Configuration and Administration Guide for Cisco Unified Customer Voice Portal

Release 8.5(1)

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Preface

Purpose

This manual describes how to configure, run, and administer Cisco Unified Customer Voice Portal (Unified CVP).

Important Assumptions

The following bullets refer to **user documentation available from the Cisco website** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html).

Understanding Unified CVP and planning your Unified CVP solution are important parts of setting up the product. Therefore, this manual assumes that you have:

- Carefully read **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html). This guide provides the following foundational material:
  - Unified CVP introductory material
  - Descriptions of Unified CVP Call Flow Models (deployment models)
  - Design and planning material
- Created the simplified Unified CVP lab setup and performed the basic exercises as described in **Getting Started with Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).
- Read **Cisco Security Agent Installation/Deployment for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_installation_guides_list.html) in its entirety. This document provides installation instructions and information about Cisco Security Agent for Unified CVP deployment.

- Examined the list of related documentation found in *Related Documentation*, below, and made additional preparations based on these documents and your deployment needs.

Audience

This document is intended for Call Center managers, Unified CVP system managers, Cisco Unified Intelligent Contact Management Enterprise (Unified ICME)/ Cisco Unified ICM Hosted (Unified ICMH) system managers, VoIP technical experts, and IVR application developers. Readers of this manual should already have a general understanding of the Unified ICME products, as discussed in the *Pre-Installation Planning Guide for Cisco Unified Intelligent Contact Management Enterprise* and the *Product Description Guide for Cisco Unified ICM Hosted*. Readers should be familiar with general Unified ICME installation and setup procedures.

A Word About Cisco Product Names

Be aware of the following Cisco product name changes:

- Effective with Release 4.0(1), Cisco Customer Voice Portal is renamed Cisco Unified Customer Voice Portal (abbreviated as Unified CVP).
- Effective with Release 7.0(1), VoiceXML Server is renamed Cisco Unified CVP VXML Server (abbreviated as VXML Server).
- Effective with Release 7.0(1), VoiceXML Studio is renamed Cisco Unified Call Studio (abbreviated as Call Studio).
- Effective with Cisco CallManager Releases 4.1(3), 4.2(1), 5.0(2), Cisco CallManager is renamed Cisco Unified Communications Manager (abbreviated as Unified CM).
- Effective with Release 7.1(1) and later, Cisco ICM Enterprise Edition is renamed Cisco Unified Intelligent Contact Management Enterprise (abbreviated as Unified ICME). Cisco ICM Hosted Edition is renamed Cisco Unified ICM Hosted (abbreviated as Unified ICMH). This manual references this product by the new name, though the new name does not appear in the Release 7.1(1) user interface.
- Effective with Release 7.1(1) and later, Cisco IPCC Enterprise Edition is renamed Cisco Unified Contact Center Enterprise (abbreviated as Unified CCE). This manual references this product by the new name, though the new name does not appear in the Release 7.1(1) user interface.

  Note: These new names are introduced in Release 7.1(1). They are referenced in opening screens and documentation, but they do not yet appear throughout the user interface. (For example, the Supervisor Desktop title bar still reads "CTIToolkit IPCC Supervisor Desktop," and there is still an "IPCC" selection on the CTI Login dialog ConnectTo pull-down menu.)

Note: For more information about Cisco product and Unified CVP component names and acronyms, refer to the Glossary in the online help.

Organization

The manual is divided into the following parts and chapters.

Part 1: Introduction to Cisco Unified Customer Voice Portal (CVP)

Chapter	Description
Chapter 1, "Configuration Overview" (page 25)	Provides background information, prerequisites, and references for understanding and planning a Unified CVP implementation. and check-off lists for the various call flow model options available for Unified CVP.
Chapter 2, "High-level Configuration Instructions for Call Flow Models" (page 25)	Provides high-level call flow model overviews, high-level sets of instructions for configuring specific call flow models, sample gateway and other configurations, and additional configuration task outlines. Provides cross-references to more detailed information.
Chapter 3, "Writing Scripts for Unified CVP" (page 141)	Discusses Unified ICME configuration and script editing. This chapter also provides a detailed description of the Unified CVP micro-applications.
Chapter 4, "Using Cisco Support Tools with Unified CVP" (page 221)	Describes the Cisco Support Tools and its supported features on Unified CVP on non-Windows boxes.
Chapter 5, "Configuring and Modifying Unified CVP Security" (page 235)	Describes Unified CVP security.

Part 2: Configuration Detail of Customer Voice Portal Components

Chapter	Description
Chapter 6, "Configuring VXML Solution" (page 259)	Describes how to configure a VXML Server in either Standalone or Comprehensive deployment models.
Chapter 7, "Configuring Unified CVP Logging and Event Notifications" (page 285)	Describes how to configure external events and how to set trace levels and log levels.
Chapter 8, "Administering the H.323 Service" (page 313)	Gives an overview of the Unified CVP H.323 Service and instructions for using the VBAdmin tool and commands.

Part 3: Configuration Detail of Non-Customer Voice Portal Components

Chapter	Description
Chapter 9, "Using Cisco Unified ICME Warm Consult Transfer/Conference to Unified CVP" (page 347)	Details how to configure the Unified ICME Warm Consult Transfer/Conference to Unified CVP feature.
Chapter 10, "Configuring Cisco Unified Communications Manager" (page 353)	Describes how to use the Unified CVP Operations Console menus to add a pre-configured Unified CM Server to the network map.
Chapter 11, "Configuring the SIP Devices" (page 355)	Provides the configuration tasks required for the SIP devices.
Chapter 12, "Transferring and Queuing Calls with Unified CVP" (page 381)	Contains additional information about transferring and queuing calls using Unified CVP.

Chapter	Description
Chapter 13, "Configuring the H.323 Devices and VoIP" (page 465)	Describes how to configure H.323 gateways, gatekeepers, Unified CM and ICM Server to perform inbound and outbound routing.
Chapter 14, "Configuring High Availability for Unified CVP" (page 489)	Provides information about how to accommodate load balancing and redundancy in Unified CVP deployments.
Chapter 15, "Configuring the Media Servers" (page 531)	Provides information about Unified CVP media file handling and details about the system media files distributed with the Unified CVP solution.

Appendices

Appendix	Description
Appendix A, "Using the Helix Server (page 553)"	Contains high-level steps for configuring a Helix server to create a broadcast stream.

Related Documentation

Note:

- Planning your Unified CVP solution is an important part of the process in setting up Unfied CVP. Cisco recommends that you read **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) guide *before* configuring your Unified CVP solution.

- The *Planning Guide for Cisco Unified Customer Voice Portal* has been incorporated into the SRND document.

Unified CVP provides the following documentation:

- *Cisco Security Agent Installation/Deployment for Cisco Unified Customer Voice Portal* provides installation instructions and information about Cisco SecurityAgent for the Unified CVP deployment. **We strongly urge you to read this document in its entirety.**

- **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) provides design considerations and guidelines for deploying contact center voice response solutions based on Cisco Unified Customer Voice Portal (CVP) releases.

- **Getting Started with Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html) provides instructions to create a simplified Unified CVP lab setup and perform basic call flow model exercises.

- **Configuration and Administration Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html)describes how to configure, run, and administer the Cisco Unified CVP product, including associated configuration.

- **Element Specifications for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_programming_reference_guides_list.html) describes the settings, element data, exit states, and configuration options for Elements.

- **Installation and Upgrade Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html) describes how to install Unified CVP software, perform initial configuration, and upgrade.

- **Operations Console Online Help for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html) describes how to use the Operations Console to configure Unified CVP solution components.

  Note: There is a printable (PDF) version of the Operations Console online help. Refer to the **Operations Console User Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html). The user guide also explains how to log into the Operations Console.

- **Port Utilization Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html) describes the ports used in a Unified CVP deployment.

- **Programming Guide for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_programming_reference_guides_list.html) describes how to build components that run on the Cisco Unified VXML Server.

- **User Guide for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html) describes the functionality of Call Studio including creating projects, using the Call Studio environment, and deploying applications to the Unified CVP VXML Server.

- **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html) describes the Reporting Server, including how to configure and manage it, and discusses the hosted database.

- **Say It Smart Specifications for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html) describes in detail the functionality and configuration options for all Say It Smart plugins included with the software.

- **Troubleshooting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_troubleshooting_guides_list.html) describes how to isolate and solve problems in the Unified CVP solution.

For additional information about Unified ICME, refer to the **Cisco web site listing Unified ICME documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html).

Conventions

This manual uses the following conventions:

Convention	Description
boldface font	Boldface font is used to indicate commands, such as user entries, keys, buttons, and folder and submenu names. For example: • Choose **Edit** > **Find**. • Click **Finish**.
italic font	Italic font is used to indicate the following: • To introduce a new term. Example: A *skill group* is a collection of agents who share similar skills. • For emphasis. Example: *Do not* use the numerical naming convention. • A syntax value that the user must replace. Example: IF (*condition, true-value, false-value*) • A book title. Example: See the*Cisco CRS Installation Guide*.
`window font`	Window font, such as Courier, is used for the following: • Text as it appears in code or that the window displays. Example: `<html><title>Cisco Systems,Inc. </title></html>`
`< >`	Angle brackets are used to indicate the following: • For arguments where the context does not allow italic, such as ASCII output. • A character string that the user enters but that does not appear on the window such as a password.

Obtaining Documentation and Submitting a Service Request

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html

Subscribe to the *What's New in Cisco Product Documentation* as a Really Simple Syndication (RSS) feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service and Cisco currently supports RSS Version 2.0.

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We appreciate your comments.

Part 1: Introduction to Cisco Unified Customer Voice Portal (Unified CVP) Configuration

This part of the manual presents the Unified CVP call flow models using a referenced set of high-level steps for each of the call flow models. The references link to detailed procedures in other parts of this guide.

It also provides an overview of the Unified CVP Operations Console, which allows you to configure devices and users, import and export system configurations, and to interact with Cisco products, such as Unified CM, Cisco Unified Intelligent Contact Management Enterprise , and Cisco Support Tools.

The final chapter in this section presents information about securing communication between the various Unified CVP components.

Chapter 1

Configuration Overview

This chapter presents general Unified CVP configuration information and prerequisites needed before you begin configuring your Unified CVP solution. At the end of the chapter is a table summarizing the Unified CVP call flow models (deployment models) and linking to the high-level configuration instructions for each call flow model, found later in the guide.

This chapter contains the following topics:

Prerequisites for Configuring Your Call Flow Model

This topic describes information you need and tasks you should perform before you select one of the call flow model high-level configuration procedures and attempt to implement it.

Prior Skills Required

The information in this chapter *assumes* that you are already familiar with:

- Configuring Cisco Gateways and Gatekeepers.
- The ICM Configuration Manager and ICM Script Editor tools for call center operations and management.
- Configuring Unified CM.

Prior Planning and Design Tasks and their Resources

Understanding Unified CVP and planning your Unified CVP solution are important parts of setting up the product. Therefore, this manual assumes that you have completed the following prerequisite reading and design work:

- **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html)
  - Study the Unified CVP overview material and the detailed descriptions of the various call flow models.
  - Also, carefully examine the design information in the SRND.
  - Based on your study, choose the appropriate callflow model for your desired Unified CVP implementation.
- **Getting Started with Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html)
  - Create the simplified all-in-one-box step-by-step call model examples.
  - Use the troubleshooting information to experiment with the working examples.

Additional Information Required

In addition to design information, you need the following configuration reference material:

- To successfully configure Unified ICME and use its features in conjunction with Unified CVP, obtain, and refer to copies of the following guides:
  - **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html)
  - **Scripting and Media Routing Guide for Cisco Unified ICM/Contact Center Enterprise & Hosted** (http://www.cisco.com/en/US/products/sw/custcosw/ps1844/products_user_guide_list.html)
  - Refer to the **Unified ICME documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html) link for detailed information about Unified ICME.
- Read **Cisco Security Agent Installation/Deployment for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_installation_guides_list.html) in its entirety. This document provides installation instructions and information about Cisco Security Agent for Unified CVP deployment.

- Examined the list of related documentation found in Related Documentation (page 4) in the Preface, for the complete list of Unified CVP solution documentation, and made additional preparations based on these documents and your deployment needs.

Information included elsewhere in this guide:

- For information on writing scripts for Unified ICME use with Unified CVP, refer to "Writing Scripts for Unified CVP (page 141)."
- Call Studio's drag-and-drop scripting interface is available to create VoiceXML scripts. Refer to "Integrating VoiceXML Scripts with Unified ICME Scripts (page 269)" for more information.

Note: For details on components required for your Unified CVP solution, but not covered or referenced in this chapter (such as the TDM side of the Voice Gateway configuration), refer to that component's documentation.

Before You Begin Configuring Your Call Flow Model Solution

The following topics relate to configuring your Unified CVP solution and should be examined before you begin following one of the specific call flow model configuration outlines in this chapter.

Developer Services

Cisco Technical Support is limited to standard Cisco product installation/configuration, and Cisco-developed applications. Questions and/or support issues related to such items as Call Studio scripting or ASR grammar are **not** covered by Cisco Technical Support.

Developers using these and similar components might be interested in joining the Cisco Developer Services Program. The Developer Services Program provides formalized services for Cisco Systems interfaces to enable developers, customers, and partners in the Cisco Technology Developer Program to accelerate their delivery of compatible solutions. The Developer Services engineers are an extension of the product technology engineering teams. They have direct access to the resources necessary to provide expert support in a timely manner

A separate service agreement and subscription fee is required to participate in the Developer Services Program. For information on how to subscribe, go to **Getting Started!** on the Developer Support Web site at **http://www.cisco.com/go/developersupport**

See Also

For additional information, refer to Frequently asked Questions about the Program and Support under **Q&A** on the Developer Support Web site at **http://www.cisco.com/go/developersupport**.

Network Information and Preparation Required

In order to use the information in this chapter, you need to know the following:

- The Unified CVP call flow model you will be implementing.

  Note: Be aware that some call flow model names have changed and new call flow models have been added since the last release. Refer to **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html).

- The network topology for your system, including addresses and names of the solution components.
- The failover strategy for Gateways, Unified CVP components, and media servers.
- The strategy for inbound call routing (that is, dial-peers versus Gatekeeper or Proxy Server).
- The naming resolution system for Gateways (DNS versus configured on the Gateway).
- Naming schemes to be used for Unified ICME PGs, Peripherals, and routing clients.
- If using a VRU other than Unified CVP, the VRU trunk group number and number of trunks.
- The locale values to be used for ASR and/or TTS.
- Whether the same set of VRUs are to be used for all cases, or whether that will be determined separately for each customer (dialed number).

  Note: If all dialed numbers will use the same VRUs, it is easiest to use a default Network VRU, rather than to configure multiple Network VRUs. For more information, refer to the section, "Common Configuration for Differentiating VRUs (Unified CVPs) Based on Dialed Number (page 138)."

Converting from H.323 to SIP

The **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) guide provides a discussion of H.323 and SIP call protocols and things to consider when deciding which protocol to use.

If you have examined the criteria in the design guide and determined that you need to convert from H.323 to SIP, complete the following steps:

1. When converting to SIP from the Queue and Transfer call flow model, first convert to the Comprehensive call flow model. In addition, adjust the VoiceXML gateway dial-peer configurations to SIP dial-peers, and specify global SIP parameters as well.

2. Decide whether you will be using the CUP Server or a non-proxy deployment (local static route configuration on the Operations Console). Also, if using redundant Proxy Servers (primary and secondary or load balancing), then decide whether to use DNS server lookups for SRV records or non-DNS based local SRV record configuration. Configure the SRV records on the DNS server or locally with an xml file (local xml configuration avoids the overhead of DNS lookups with each call). (If you decide to use a single CUP Server, then SRV record usage is not required.)

Ring No Answer (RNA) Settings with SIP

If you are using the CVP RNA settings in SIP (default 60 seconds, minimum 5 seconds, and maximum 200 seconds), make this value 2 or 3 seconds greater than the Unified ICME Agent Desk Setting RNA timeout, in order to allow time for the signaling to the agent after the ICM Router picks the agent via the link with the Peripheral Gateway.

Unified CVP makes a call to the ringtone service on the VXML gateway, followed by sending the call to the Unified CM trunk for the agent, and this allows enough time for the agent to get the delivered event after being reserved, and ensuring that Unified ICME reporting is correct in terms of handled time and RNA call disposition reporting.

Unified CVP Installation

Before using the information in this chapter, you need to do the following:

- Install the Unified CVP software.

  Note: Refer to the **Installation and Upgrade Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html) for complete installation instructions.

- Install all the solution components.

- If you are using Unified CVP as a Unified ICME queuing platform, make sure the VRU PGs use service control with Service Control reporting enabled. If you are using it strictly as a self-service platform, disable Service Control reporting. Also, take note of the VRU Connection Port used for each VRU PG peripheral (PIM).

  Note: Refer to the **Reporting Guide for Cisco IPCC Enterprise & Hosted Editions** (http://www.cisco.com/en/US/products/sw/custcosw/ps4145/products_user_guide_list.html) and the **Reporting Guide for Cisco Unified ICM Enterprise & Hosted** (http://www.cisco.com/en/US/products/sw/custcosw/ps4145/products_user_guide_list.html) for more information on IVR-related Service Control reporting and queue reporting. For Unified CVP reporting refer to **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html)

- Make sure the NIC cards, Voice GW, and network components all have the Ethernet interfaces configured with matching speed and duplex settings.

Note: Refer to the Setting Speed and Duplex for Unified CVP (page 16) for details about the required Ethernet Switch / Server NIC settings. Refer to the **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) for additional information.

Routing Calls Through the Network to the VRU

Most call flow models involve a step in which the call must be transferred to a VoiceXML gateway. Depending on the specific call flow model in use, one of two techniques is applied to direct that transfer. Both techniques involve a label or labels being provided by Unified ICME or Unified ICMH, and those labels must be configured in the solution's other call routing components to deliver the call to an appropriate VoiceXML gateway. Such labels are part of the overall dialed number plan of the contact center, and must be determined prior to configuring Unified CVP.

For call flow models that use Network VRUs of Type 3, 7, or 10:

- Determine the Network Routing Number. This number is the base for routing calls through the network to the VRU; a correlation ID is appended to this number to transfer calls to a Network VRU through the network.

For call flow models with a Customer VRU, and in Unified ICMH environments, and for NIC Type 8 call flow models:

- Determine the translation route pools to use for each VRU.
- Determine the labels to be sent to the network to connect the call to the VRU and the corresponding DNIS that will be seen by the VRU. For example, the label for the network might be 18008889999 and the DNIS received by the VRU and sent back to Unified ICME to identify the call might be 9999.

Setting Speed and Duplex Values for Unified CVP

Make sure your Ethernet Switch / Server NIC, gateways, and Call Server settings are set as follows:

Caution: The "Auto" option, is *only* applicable for matched port/NIC at Gigabit Ethemet (1000Mbps). If you are unsure of the adjacent station configuration, select 1000/Full on the Gigabit interface. You can only use the Auto option, if both stations supply Gigabit interfaces.

Ethernet Switch Speed	Server/Gateway NIC Speed	Speed/Duplex Setting for Switch Port MUST be:	Speed/Duplex Setting for Server/GW NIC MUST be:
1000 Mb	1000 Mb	1000/ Full	1000/ Full

Ethernet Switch Speed	Server/Gateway NIC Speed	Speed/Duplex Setting for Switch Port MUST be:	Speed/Duplex Setting for Server/GW NIC MUST be:
1000 Mb	1000 Mb	Auto / Auto	Auto / Auto
1000 Mb	100 Mb	100 Mb / Full	100 Mb / Full
100 Mb	100 Mb	100 Mb / Full	100 Mb / Full
100 Mb	1000 Mb	100 Mb / Full	100 Mb / Full

Applying Contact Center Gateway Debug Settings

To enable required minimum contact center debug settings, perform the following steps on the gateway.

Step 1 Log into the gateway.

Step 2 Type **`enable`** and the password to enter enable mode.

Step 3 Enter the configure terminal command to enter configuration mode.

Step 4 Type **`ivr contact-center`** to apply default debug settings.

Step 5 Configure the logging buffer size using **`set logging buffer`**. For example, **`set logging buffer 1000000`**

The recommended logging buffer size is 1000000.

Step 6 Exit configuration mode and return to the enable prompt by pressing **Ctrl-Z**.

To see the current operating configuration, including the change you just made, enter the show running-config command: **`show running-config`**

Step 7 To persist configuration changes (make changes permanent), enter the copy running-config startup-config command at the enable prompt: **`write`**

Example:

```
User Access Verification

Password:
ccbu-doc-gw4>en
Password:
ccbu-doc-gw4#config t
Enter configuration commands, one per line.  End with CNTL/Z.
ccbu-doc-gw4(config)#ivr
ccbu-doc-gw4(config)#ivr contact-center
ccbu-doc-gw4(config)#^Z
ccbu-doc-gw4#show debug
....
```

Deprecation of Queue and Transfer Model

The Queue and Transfer call flow model, which was supported in previous versions of CVP, is not supported for new customers in Unified Customer Voice Portal 4.0 and later. New customers are required to use the Comprehensive model instead.

This model, which was supported in previous versions of CVP, is not supported for new customers in Unified CVP 4.0 and later. New customers are advised to use the Comprehensive model instead.

However, existing customers who are upgrading to Unified CVP 4.0 and later while continuing to use Unified ICME 6.0 or earlier, do not always have this option.

Such customers who:

- Do have IP-originated calls, must stay with H.323 and continue terminating media on the H.323 Service (formerly called the CVP Voice Browser)
- Do not have IP-originated calls, are encouraged to move to SIP; they might choose to stay with H.323, but in that case they must remove media termination from the H.323 Service—since, within the Unified CVP *solution*, Gateways and Unified CM devices are the only components that terminate voice

Network VRU Types and Unified CVP Call Flow Models

In Unified ICME, Network VRU is a configuration database entity. It is accessed using the ICM Configuration Manager's Network VRU Explorer tool. A Network VRU entry has two pieces of information:

- Type - This is a number from 2 to 10, and corresponds to the types described above.
- Labels - This is a list of labels, which Unfied ICME can use to transfer a call to the particular Network VRU being configured. These labels are only relevant for Network VRU's of Types *3*, *7*, and *10* (that is, those that use the Correlation ID mechanism to transfer calls). They are also required, but never used in the case of Type 5. (Labels for Types *8* and *2* are defined in the ICM Configuration Manager's Translation Route Explorer tool, and invoked via a TranslationRouteToVRU node.)

  Each label is made up of two parts:

  – A digit string, which becomes a DNIS that can be understood by the gatekeeper (when using H.323), by a SIP Proxy Server or static route table (when using SIP without a Proxy Server), SIP), or by gateway dial-peers.

  – A routing client (also known as a switch leg peripheral). In other words, each peripheral device which can act as a switch leg must have its own label, even *if* the digit strings are the same in all cases.

Unified ICME Release 7.1(1) introduced Network VRU Type 10, which simplifies the configuration of Network VRU's for Unified CVP. For most call flow models, a single Type 10 Network VRU can take the place of the Type 2, 3, 5, 7, or 8 Network VRUs, which were associated with the Customer Instance and/or the Switch and VRU leg peripherals. The VRU-Only call flow models still require Type 8; however, in one specific case Type 3 or 7 is still required.

Network VRU configuration entries themselves have no value until they are associated with active calls. There are three places in Unified ICME where this association is made:

- Advanced tab for a given peripheral in the ICM Configuration Manager's PG Explorer tool
- Customer Instance configuration in the ICM Configuration Manager's ICM Instance Explorer tool
- On every VRU Script configuration in the ICM Configuration Manager's Network VRU Script List tool

Depending on the call flow model, Unified ICME looks at either the peripheral or the customer instance to determine how to transfer a call to a VRU. Generally speaking, Unified ICME examines the Network VRU which is associated with the switch leg peripheral when the call first arrives on a switch leg, and the Network VRU which is associated with the VRU leg peripheral when the call is being transferred to VRU using the Translation Route mechanism. It examines the Network VRU, which is associated with the Customer Instance or the default Network VRU from the System Information tool, when the call is being transferred to the VRU using the Correlation ID mechanism.

Unified ICME also examines the Network VRU which is associated with the VRU Script every time it encounters a RunExternalScript node in its routing script. If Unified ICME does not believe the call is currently connected to the designated Network VRU, it will not execute the VRU Script.

Note: The previously recommended VRU types still work as before, even in Unified ICME 7.1(1) and later; however, new installations must use Type 10 if possible, and existing installations can optionally switch to Type 10.

SIP DN Pattern Matching Algorithm

Throughout this document, when you are configuring static routes using dialed number (DN) patterns, keep the following configuration and application concepts in mind:

The following details refer to creating dialed number patterns:

- Wildcarded DN patterns can contain "." and "X" in any position to match a single wildcarded character.
- Any of the wildcard characters in the set ">*!T" will match multiple characters but can only be used trailing values because they will always match all remaining characters in the string.

- The highest precedence of pattern matching is an exact match, followed by the most specific wildcard match. When the number of characters are matched equally by more than one wildcarded pattern, precedence is given from top to bottom of the configured DN list.
- There is no explicit software limit on the number of items in the DN pattern list.

After You Complete the Configuration Prerequisites

Once you complete the prerequisite study and preparation described in Prerequisites for Configuring Your Call Flow Model (page 25), choose your task from one of the lists below.

- The first table enables you to choose a call flow model and go to the related high-level instructions based on the links and reference points listed in the table.
- The second table provides references to additional configuration topics.

Note:

- Configuration instructions are broken into separate instructions for the SIP and the H.323 versions of the call flow models.
- For a full description of each callflow model, refer to **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html).

Unified CVP Call Flow Models

Standalone Call Flow Models

- Unified CVP VXML Server (Standalone) Call Flow Model (page 27)

  Install the VXML Server as a standalone component, without Unified ICME, the Unified CVP Call Server, or H.323 Service components--a complete solution for rapidly creating and deploying dynamic VoiceXML applications.

- Unified CVP Standalone with ICM Lookup Call Flow Model (page 29)

  To the previous standalone solution, add the capability of invoking a routing script in Unified ICME and receiving a response.

Call Director Call Flow Models

- Unified CVP Call Director (SIP/H.323) Call Flow Model, Unified ICME (page 37)

  Unified CVP only provides Unified ICME with VoIP call switching capabilities. You provide your own Service Control VRU, if you are using Unified ICME to queue calls or you might queue calls directly on your ACD. Calls might be transferred multiple times, from Ingress,

to customer-provided VRU, to either Unified CCE or customer-provided ACD or agent, and back again.

When calls are connected to customer-provided equipment (either VoIP or TDM), their voice paths must go to an egress gateway, which is connected by TDM to that equipment. If the signaling is SIP, then this call flow model works with customer-provided SIP endpoints which have been tested and certified to interoperate with Unified CVP.

Note: The Call Director call flow model is the most inclusive call flow because it potentially incorporates all components of the Unified CVP solution.

- Unified CVP Call Director (SIP/H.323) Call Flow Model, Unified ICMH (page 39)

  Unified CVP only provides the Network Applications Manager (NAM) with VoIP call switching capabilities. You provide your own Service Control VRU, if you are using the NAM to queue calls or you might queue calls directly on your ACD. Calls might be transferred multiple times, from Ingress, to customer-provided VRU, to either the NAM or customer-provided ACD or agent, and back again.

  When calls are connected to customer-provided equipment, their voice paths must go to an egress gateway, which is connected by TDM to that equipment. If the signaling is SIP, then this call flow model works with customer-provided SIP endpoints which have been tested and certified to interoperate with Unified CVP.

  Note: In a two-tier service bureau (carrier) configuration, the *NAM* is the tier providing direct communication with the carrier PSTN. Route requests arrive at the NAM from the IXC carrier network and are forwarded, based on specific call properties, to the appropriate Customer ICM (CICM). A NAM usually contains only a small configuration that allows it to directly route a subset of calls and dispatch other calls to the appropriate CICM. The NAM receives route responses from all CICMs and forwards them to the carrier network.

Comprehensive Call Flow Models

- Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICME (page 61)

  Unified CVP acts as the Switch, transferring the call to the Network VRU and to agents. The Unified CVP IVR Service in the Operations Console is also configured to work with the VoiceXML Gateway to provide VRU treatment, which might include ASR/TTS and VXML Server applications.

- Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICMH (page 64)

  Unified CVP is deployed at the NAM where it acts as the Switch, transferring the call to the Network VRU (using the Correlation ID transfer mechanism) and to agents. The Unified CVP IVR Service in the Operations Console is also configured to work with the VoiceXML Gateway to provide VRU treatment, which might include ASR/TTS and VXML Server applications.

Type 8 Call Flow Models

- Type 8 Unified CVP VRU-Only Call Flow Model, Unified ICME (page 105)

  Unified CVP works with the Voice Gateway to act as the VRU; VRU voice treatment is provided by the Gateway and can include ASR/TTS and VXML Server applications. This model would be used when call switching is provided by some component other than Unfied CVP, such as a Unified ICME NIC.

- Type 8 Unified CVP VRU-Only Call Flow Model, Unified ICMH (page 106)

  The Unified CVP Call Server is deployed at the CICM level to act only as the VRU leg for the call. The VRU voice treatment is provided at the Voice Gateway, and might include ASR/TTS and VXML Server applications.

  This model would be used when call switching is provided by some component other than Unified CVP, such as a Unified ICME NIC.

  Note: The *CICM* is the Customer ICM. In the optional two-tier service bureau (carrier) configuration, the CICM is the tier providing the carrier customer-specific routing function. CICMs receive customer-specific call route requests from the NAM; they typically perform more elaborate scripted call routing using customer-specific advanced services or agent and skill context.

Type 3 or 7 Call Flow Models

- Type 3 or 7 Unified CVP VRU-Only Call Flow Model Network VRU, Unified ICMH (page 115)

  Unified CVP is deployed as a Network VRU at the NAM. The Unified CVP IVR Service in the Operations Console works with the Voice Gateway to act as the VRU; VRU voice treatment is provided at the Voice Gateway and can include ASR/TTS and VXML Server applications.

  This call flow model is used when Unified CVP is connected to the NAM.

  This model is also used when call switching is provided by some component other than Unified CVP, such as a Unified ICME NIC.

Configuration instructions for the following geographic and physical models are not listed in this guide. Refer to the **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) at for details about these models.

Unified CVP Geographic models:

- Centralized Single-Site
- Centralized Multi-Site
- Centralized Branch
- Standalone Branch

Unified CVP Physical models:

- Typical for Unified ICME Integrated
- Typical for Standalone
- Streamlined for SIP Call Director
- Laboratory (All-in-a-Box)

  The laboratory all-in-one-box implementation is also documented in **Getting Started with Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

Links to Additional Configuration Instructions

- Unified CVP Comprehensive Call Flows for Pre-Routed Calls (page 124)

  This class of call flows is similar to the Unified CVP Comprehensive call flow models, except that calls are first introduced into Unified ICME or Unified ICMH using some path other than through Unified CVP. A Unified ICME routing script is given the chance to "pre-route" such calls before Unified CVP ever sees them. Once the script transfers the call to Unified CVP, for either self-service or queuing, a more standard Unified CVP Comprehensive call flow model is used.

- Common Unified ICMH Configuration for Unified CVP Switch Leg (page 129)

  Describes Unified ICMH configuration instructions common to Comprehensive Unified ICMH and VRU-Only with NIC routing, with Correlation ID call routing call flow models for Unified CVP switch legs.

- Common Unified ICMH Configuration: Define Unified CVP ECC Variables (page 131)

  Provides instructions on how to set up ECC variables that Unified CVP uses to exchange information with Unified ICMH.

- Using the Metadata ECC Variable (page 137)

  Defines the values for the *user.microapp.metadata* ECC variable.

- Common Configuration for Differentiating VRUs (Unified CVPs) Based on Dialed Number (page 138)

  Provides instructions on how to configure Unified ICME to differentiate the VRUs.

- Gatekeeper Redundancy (page 138)

  Hot Standby Router Protocol (HSRP) is not the only method of Gatekeeper redundancy. Instead, use Gatekeeper Clustering and Alternate Gatekeeper configuration on Unified CVP. This is the preferred method of Gatekeeper redundancy.

- SIP Proxy Redundancy

Refer to "How to Set Up the Ingress Gateway to Use Redundant Proxy Servers (page 357)" and "How to Set Up the Unified CVP Call Server with Redundant Proxy Servers (page 358)" for details.

Cisco Unified CVP Operations Console

The Unified CVP Operations Console enables you to centrally operate, administer, maintain, and provision your Unified CVP solution. The Operations Console is a web-based tool that provides the capability to administer and monitor your Unified CVP components remotely.

For Unified CVP release 8.0(1) there is a separate guide for the Operations Console. Refer to **Cisco Unified CVP Operation Console Guide** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

Chapter 2

High-level Configuration Instructions for Call Flow Models

This chapter presents a set of high-level instructions for configuring many of the Unified CVP call flow models (deployment models).

Each set of call flow model instructions contains:

- A brief overview of that call flow model
- High-level instructions for configuring the components in that call flow model
- References to detailed instructions (elsewhere in this guide, in online help, or in other documents) for performing each high-level task

This chapter also includes information, or pointers to information, for configuring the Gateway, Gatekeeper, Unified ICME VRU handling, Unified CVP Call Server (including the SIP Service, ICM Service, and IVR Service), and the Unified CVP H.323 Service.

This chapter contains the following topics:

- Gatekeeper Redundancy, page 138
- Local SRV File Configuration Example for SIP Messaging Redundancy, page 140

Best Practices for Order of Device Operations

The following information is a guide for setting up the order of the device operations. Please apply the instructions accordingly, based on the call flow model.

Device Deployment:

- SIP Proxy Server device (optional)
- Unified CVP Call Server device
- Unified CVP VXML Server device
- Unified CVP Reporting Server device
- Other Devices (Gateways, Unified CM, etc.)

System Configuration:

- SIP Server Groups
- Dialed Number Pattern
- Locations
- Courtesy Callback

Miscellaneous:

- Transfer of licenses (required)
- Transfer of VXML applications (required)
- Bulk transfer of default Gateway files (required)

Device management order:

1. Add new CVP device.
2. Configure CVP device.
3. Save and deploy CVP device.
4. Transfer license.
5. Restart CVP device to activate license.

6. Verify CVP devices states are "Up" in OAMP Control Center.
7. Deploy system level configuration, Dialed Number Pattern, SIP Server Groups, Locations, and Courtesy Callback and confirm the status.
8. Save and deploy SNMP Configuration.

Unified CVP Standalone Call Flow Models

This section describes the following call flows and provides their high-level configuration instructions.

- Unified CVP VXML Server (Standalone) Call Flow Model (page 27)
- Unified CVP Standalone with ICM Lookup Call Flow Model (page 29)

Unified CVP VXML Server (Standalone) Call Flow Model

In this call flow model, VXML Server is a J2EE-compliant server that provides a complete solution for rapidly creating and deploying dynamic VoiceXML applications. You can install the VXML Server as a standalone component, without the Unified CVP Call Server or H.323 Service components, and with or without reporting.

The Unified CVP VXML Server (standalone) call flow model is available in the following variations:

- Standalone without reporting: Use the **VXML Server (Standalone)** option in the Operations Console. This call flow model *does not* require a Call Server and a Reporting Server.
- Standalone with reporting: Use the **VXML Server** option in the Operations Console. This call flow model *requires* a Call Server and a Reporting Server.
- Standalone, but adding reporting *after* the VXML Server (Standalone) version has already been configured: Configure the Unified CVP Call Server, delete the VXML Server (Standalone), and use the **VXML Server** option in the Operations Console to add the VXML Server.

Refer to "High-level Configuration Instructions for Unified CVP VXML Server Standalone Call Flow Models" (page 31) for configuration instructions.

In this call flow model with reporting, the Unified CVP Call Server is used to route messages between the components. Calls arrive through a VoiceXML gateway and interact directly with a VXML Server to execute VoiceXML applications. The gateway performs both ingress and VoiceXML functions. This call flow model provides a sophisticated VoiceXML-based VRU, for applications which, in many cases, do not need to interact with a Unified ICME Server.

Some of the features of this option include:

- VoiceXML features that control each of the following:
  - Audio input and output
  - Presentation logic
  - Call flow
  - Telephony connections
  - Event handling for errors
  - ASR/TTS
  - Digit detection and generation count
- Drag-and-drop Graphical User Interface for the rapid creation of voice applications.
- VXML Server provides a J2EE-compliant development framework.

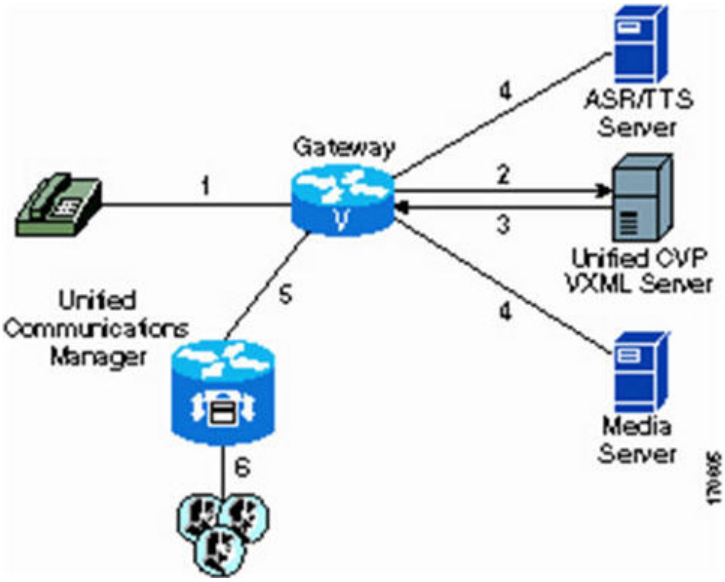
In the Unified CVP VXML Server (standalone) call flow model, *only* the VXML Server, Call Studio, and a Gateway are required, except when using reporting which requires a Call Server and a Reporting Server.

This standalone model has functions similar to the Unified CVP VRU Call Flow Models with NIC Routing (page 105).

Note: The CVP VXML standalone call flow model allows only one synchronous blind or bridged transfer. A synchronous blind transfer indicates that once the call has been transferred, a Unified CVP Standalone script has no ability to asynchronously take it back and deliver it somewhere else, whereas Unified ICME scripts, in the Unified ICME-integrated models, do have that ability.

The following figure displays the call flow for the Unified CVP VXML Server (standalone) call flow model.

Figure 1: Unified CVP VXML Server (standalone) Call Flow



The following, numbered, call flow description for the previous figure assumes:

- You installed and licensed the VXML Server.
- You created a Call Studio application and deployed it on the VXML Server.

The call flow shown in the previous figure is as follows:

1. The call arrives from the PSTN network to the Gateway.
2. The Gateway sends an HTTP URL request to the VXML Server.
3. The VXML Server returns the VoiceXML instructions to be executed on the Gateway Voice Browser.
4. The VoiceXML instructions returned to the Gateway can include references to ASR/TTS to recognize voice input and play TTS files, and references to Media Servers to play .wav files.
5. The gateway can, optionally, transfer the call to any destination that it can deliver a call to, such as Unified CM.
6. Unified CM can then send the call to an agent.

Unified CVP Standalone with ICM Lookup Call Flow Model

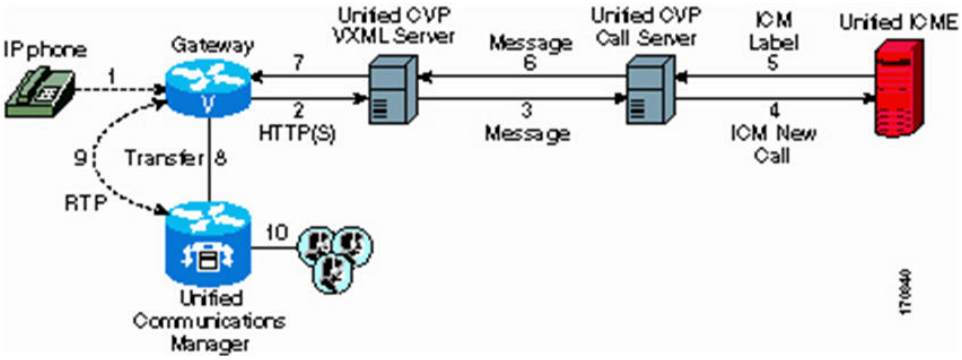
This call flow model invokes a new incoming dial number from a Unified CVP peripheral, which invokes a routing script in Unified ICME.

Note: This call flow model only returns a label and call context variables; you cannot run scripts or do queuing.

The following figure displays the call flow for the Unified CVP Standalone with ICM Lookup call flow model.

Note: In this diagram, solid lines indicate voice paths and dashed lines indicate signaling paths.

Figure 2: Unified CVP Standalone with ICM Lookup Call Flow Model



The following, numbered call flow description for the previous figure assumes:

- A Unified CVP Call Server has been defined using the Operations Console.
- A Call Studio application has been created that contains a Unified ICME request label element.
- The Unified ICME script must be set up to handle this request, so Unified ICME can interpret the label.
- The Call Studio application must be deployed on the VXML Server.

The call flow in the previous figure is as follows:

1. The call arrives from the PSTN network to the Gateway.
2. The Gateway sends a HTTP URL request to the VXML Server.
3. VXML Server sends a message to the Unified CVP Call Server.

   Note: The request in Step#2 is only sent to the Call Server when the ReqICMLabel node is used in the Call Studio script. Otherwise, the call flow is the same as in standalone call flow model; that is, Steps #3 through #6 only happen if the ReqICMLabel node is used in the script.

4. Unified CVP Call Server sends a Unified ICME new call to Unified ICME.
5. Unified ICME returns a Unified ICME label to the Unified CVP Call Server.
6. Unified CVP Call Server sends a response message to the VXML Server.

7. The VoiceXML returned to the Gateway can include references to ASR/TTS and Media Servers to recognize voice input, play TTS files, and play .wav files, respectively.

8. The gateway can, optionally, transfer the call to any destination that it can deliver a call to, such as Unified CM.

9. Unified CM can then send the call to an agent.

High-level Configuration Instructions for Unified CVP VXML Server Standalone Call Flow Models

This task provides high-level steps for configuring Unified CVP as a standaloneVXML Server. The instructions cover all three versions of Unified CVP standalone: without reporting, with reporting, and with ICM lookup and optional reporting.

There are four types of pre-instruction labels in the steps that follow. Complete the steps based on the type of standalone implementation you want. Skip steps that do not apply to your implementation, according to the following label descriptions:

- **All Versions:** Steps that apply to all standalone configurations.
- **Non-reporting:** Steps for the non-reporting version of standalone Unified CVP.
- **Reporting:** Steps for a standalone VXML Server **with** reporting. Includes steps to define a Call Server, Reporting Server, and reporting configurations.
- **ICM Lookup:** Steps, or unique portions of steps, required for the ICM Lookup version of Unified CVP standalone.

Note:

- **Converting from a Non-reporting Configuration (no Call Server defined) to a Reporting or ICM Lookup Configuration:** If you have previously configured Unified CVP for standalone **without** reporting, the version of the VXML Server you defined cannot be associated with a Call Server, which is required for reporting and for ICM LookupTherefore you must **delete** the existing VXML Server definition and begin with Step #3 to incorporate a Call Server and the remaining Reporting Server and ICM Lookup configuration steps.
- The **Getting Started with Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html) guide provides a very detailed example of creating a standalone implementation. It includes both reporting configuration and testing.

Step 1 **All Versions:** Transfer the following script, configuration, and .wav files using the Unified CVP Operations Console (**or**, using the Unified CVP product CD):

- CVPSelfService.tcl

  Note: This file contains a gateway configuration example.

- CVPSelfServiceBootstrap.vxml
- critical_error.wav

a. Select: **Bulk Administration > File Transfer >Scripts and Media**

b. From the Select device type drop-down, select **Gateway**.

c. Highlight each required file in the Available pane and click the **right arrow icon** to move the file to the Selected pane..

d. Click **Transfer**.

Step 2 **All Versions:** Configure the gateway for VXML Server (Standalone) applications:

a. Define the VXML Server applications on the gateway.

Note: Backup server is optional. For the Tomcat Application Server and the WebSphere Application Server, set the port to **7000**. The backup server cannot be the same server as the Primary Server.

b. Configure the base gateway settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Standalone Call Flow Model (page 35)."

c. Configure the service gateway settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Standalone Call Flow Model (page 35)."

d. Configure a dial-peer, which will call the service to reach the Unified CVP VXML Server.

Refer to "VoiceXML Gateway Configuration: Example of Dial-peer for Standalone Call Flow Model (page 36)."

e. Optionally, create additional dial-peers for any outgoing transfer destinations your application uses.

Review the updated gateway configuration by issuing the **show run** command to examine the running configuration.

Step 3 **All Versions, with Noted Difference for ICM Lookup:** Create an application using Call Studio and deploy it as a zip file.

Note: For ICM Lookup, Use the ReqICMLabel Element As Follows:

The application must use the ReqICMLabel element.This element has two exit states: error and done. The *done* path must connect to a transfer element to transfer the caller to ReqICMLabel as referenced by the ReqICMLabel element.

For details on the ReqICMLabel element, refer to **Element Specifications for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_programming_reference_guides_list.html).

Refer to "How to Deploy VoiceXML Scripts Using Call Studio (page 215)." For information about Call Studio, refer to **User Guide for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html).

Step 4 **Reporting and ICM Lookup:** Enable logging.

Refer to **User Guide for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html) for details on configuring loggers using Call Studio.

Step 5 **ICM Lookup:** Enable the CVPSNMPLogger for SNMP monitoring.

Note: By default, CVPSNMPLogger is enabled when a new Call Studio application is created and deployed to the VXML Server.

Step 6 **Reporting:** Configure a Call Server.

Configure a standard Call Server and *do not* enable any Unified CVP services.

In the Operations Console select **Device Management > Unified CVP Call Server**.

Click **Help > This Page** for details.

Step 7 **ICM Lookup:** Configure a Call Server and select the ICM Service.

Configure a standard Call Server and enable the ICM Service.

In the Operations Console select **Device Management > Unified CVP Call Server**.

Click **Help > This Page** for details.

Step 8 **Without Reporting:** Configure the VXML Server as a standalone server..

In the Operations Console, select**Device Management >VXML Server (standalone)** and add a new VXML Server.

For details, select **Help > This Page** and examine **Configuring a VXML Server > Adding a VXML Server**.

Step 9 **Reporting and ICM Lookup:** Configure the VXML Server.

a. In the Operations Console, select**Device Management >VXML Server** and add aVXML Server with an associated Primary Call Server.

Click **Help > This Page** to access the **Adding a VXML Server** topic for details.

b. To enable reporting for this VXML Server, in the Operations Console, select the **Configuration** tab and select **Enable Reporting for this VXML Server**.

c. Add appropriate filtering.

For help, from Configuration tab select **Help** and select **VXML Server Configuration Properties** and scroll down to the filter discussion.

Step 10 **Non-reporting:** Deploy the Call Studio Application on the VXML Server.

Note: This is the final step for a non-reporting, standalone call flow.

In the Operations Console, select **Device Management > VXML Server (standalone)**, select the VXML Server, and click **Save and Deploy**.

This **completes** the configuration for non-reporting standalone call flow model.

Step 11 **Reporting and ICM Lookup:** Deploy the Call Studio Application on the VXML Server.

In the Operations Console, select **Device Management > VXML Server**, select the VXML Server, and click **Save and Deploy**.

Step 12 **ICM Lookup:** Using the ICM Script Editor, create a Unified ICME script that returns a label.

In order to transfer information from Unified ICME to the VXML Server besides the label, use the ToExtVXML0 - 4 ECC Variables and/or Peripheral Variables 1 - 10. The format for using the ToExtVXML 0 - 4 is with name value pairs that are delimited by semi-colons.

For example:

```
ToExtVXML0 = "company=Cisco Systems;state=MA"
```

Use the Peripheral Variables 1 - 10 to pass information to the VXML Server. The values in these variables will be taken as is.

For more information about creating a Unified ICME script that returns a label in, refer to the **Unified ICME documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html).

For more information about using the ReqICMLabel element, refer to "Passing Data to Unified ICME (page 281)."

Step 13 **Reporting:** Configure the Reporting Server.

a. In the Operations Console, select **Device Management > CVP Reporting Server > General tab** and configure the Reporting Server.

b. Select a Call Server to associate with this Reporting Server.

c. Check the default values of the Reporting properties and change, if desired.

For more information, refer to **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

VoiceXML Gateway Configuration: Example Gateway Settings for Standalone Call Flow Model

The first part of the following example provides the basic configuration for setting a VoiceXML Standalone gateway:

- Applies a timestamp to debugging and log messages
- Turns on logging
- Turns off printing to the command line interface console
- Sends RTP packets
- Configures ASR/TTS Server
- Configures gateway settings

The last part of this example provides the following:

- Standalone Service settings for hello_world application on the VXML Server
- Service requirements for configuring self-service call flow models

```
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
!
service internal
logging buffered 99999999 debugging
no logging console
!
ip cef
!
voice rtp send-recv

ip host tts-en-us <IP of TTS or MRCP Server>
ip host asr-en-us <IP of ASR or MRCP Server>

voice class codec 1
   codec preference 1 g711ulaw
   codec preference 2 g729r8

voice service voip
   signaling forward unconditional
   h323
!
gateway
   timer receive-rtcp 6
```

```
!
ip rtcp report interval 3000
!
!
voice translation-profile block
translate called 1
!
!
ivr prompt memory 15000
ivr prompt streamed none
ivr asr-server rtsp://asr-en-us/recognizer
ivr tts-server rtsp://tts-en-us/synthesizer

mrcp client timeout connect 10
mrcp client timeout message 10
mrcp client rtpsetup enable
rtsp client timeout connect 10
rtsp client timeout message 10
vxml tree memory 500
http client cache memory pool 15000
http client cache memory file 500
http client connection timeout 60
http client response timeout 30
http client connection idle timeout 10

application
  service hello_world flash:CVPSelfService.tcl
    param CVPPrimaryVXMLServer <ip address>
    param CVPBackupVXMLServer <ip address>
    param CVPSelfService-port 7000
    param CVPSelfService-SSL 0
    -OR-
    param CVPSelfService-port 7443
    param CVPSelfService-SSL 1
    param CVPSelfService-app HelloWorld

  service CVPSelfService
    flash:CVPSelfServiceBootstrap.vxml
    !
```

Note:

- The optional **`param CVPSelfService-SSL 1`** line enables HTTPS.
- **Important!** Calls may be rejected with a *403 Forbidden* response if Toll Fraud security is not configured correctly. The solution is to add the IP address as a trusted endpoint, or else disable the IP address trusted list authentication altogether using this configuration entry: **`voice service voip -> "no ip address trusted authenticate"`**

VoiceXML Gateway Configuration: Example of Dial-peer for Standalone Call Flow Model

The following example provides the configuration for an incoming Pots and VoIP call for the Unified CVP VXML Server (standalone) call flow model:

Note: VXML Server (Standalone) only supports an incoming call with a TDM through a T1 port. Using an FXS port is not supported.

```
dial-peer voice 8 pots
  description Example incoming POTS dial-peer calling HelloWorld VXML
 Server app
  service hello_world
  incoming called-number <your DN pattern here>
  direct-inward-dial

dial-peer voice 800 voip
  description Example incoming VOIP dial-peer calling HelloWorld VXML
 Server app
  service hello_world
  incoming called-number 800.......
  voice-class codec 1
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
!
```

Unified CVP Call Director Call Flow Models

This section describes the call director call flow models, broken down by Unified ICME and Unified ICMH products as follows:

- Unified CVP Call Director (SIP/H.323) Call Flow Model, Unified ICME (page 37)
- Unified CVP Call Director (SIP/H.323) Call Flow Model, Unified ICMH (page 39)

This section provides high-level instruction, broken down by SIP and H.323 implementations as follows:

- Configuration Instructions for the Unified CVP Call Director Call Flow Model Using SIP, for Both Unified ICME and Unified ICMH (page 41)
- Configuration Instructions for the Unified CVP Call Director Call Flow Model Using H.323 for Both ICME and ICMH (page 50)

Unified CVP Call Director (SIP/H.323) Call Flow Model, Unified ICME

The SIP Call Director call flow model sends calls to other third-party ACDs, SIP gateways, or to a Unified CM server.

In this call flow model, Unified CVP only provides Unified ICME with VoIP call switching capabilities. You provide your own Service Control VRU, if you are using Unified ICME to queue calls or you might queue calls directly on yourACD. Calls might be transferred multiple times, from Ingress, to customer-provided VRU, to either Unified CCE or customer-provided ACD or agent, and back again. When calls are connected to customer-provided equipment (either VoIP or TDM), their voice paths must go to an egress gateway, which is connected by

TDM to that equipment. If the signaling is SIP, then this call flow model works with customer-provided SIP endpoints which have been tested and certified to interoperate with Unified CVP.

In this call flow model, Unified CVP *stays* in the signaling path after the transfer.

Note: VRU scripts and transfer to a VRU leg are not available in this call flow model.

The Unified CVP Call Director call flow model requires the following components:

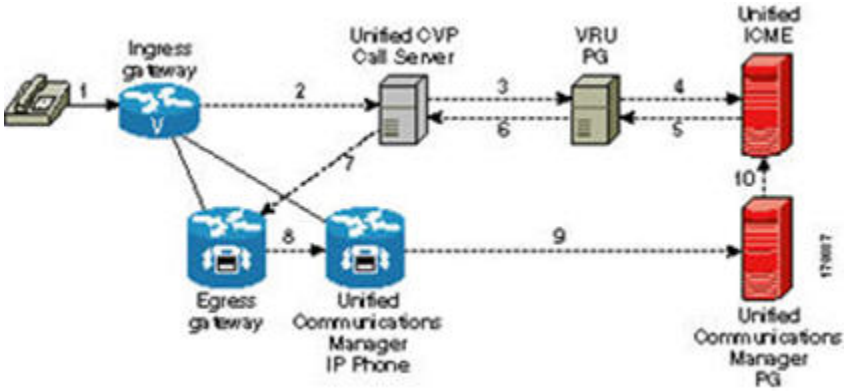
- Call Server (using either SIP or H.323 call signaling)
- Ingress Gateway
- Operations Console
- Unified ICME

Note: The Reporting Server is optional. If the Call Server is configured to use SIP signaling, a SIP Proxy Server is optional.

The following figure shows the call flow for this call flow model using SIP without a Proxy Server.

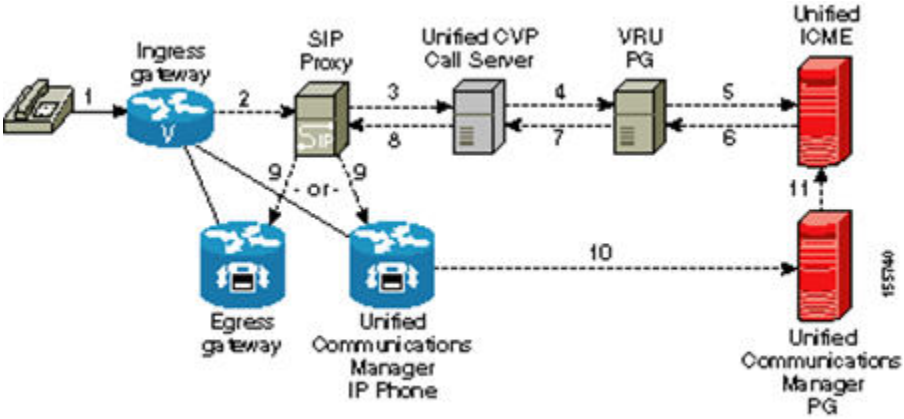
Note: In the following diagrams, solid lines indicate voice paths and dashed lines indicate signaling paths.

Figure 3: Unified CVP Call Director with SIP without Proxy Server, Unified ICME



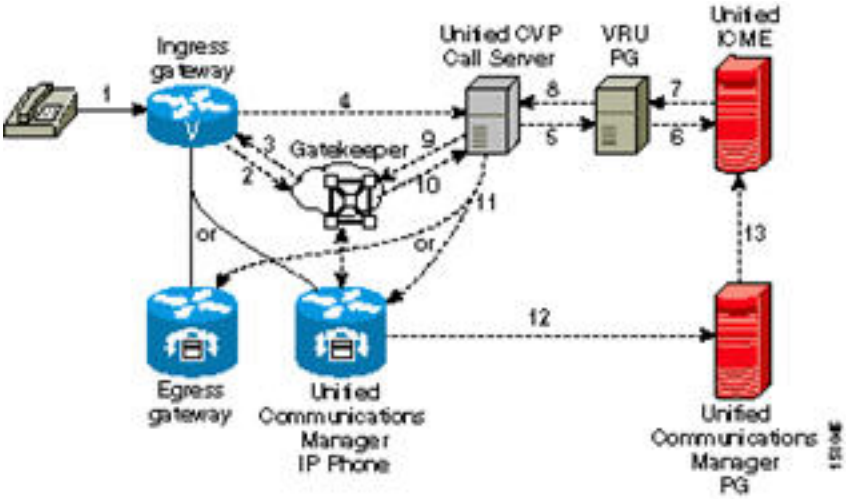
The following figure shows the call flow for this call flow model using SIP with a Proxy Server.

Figure 4: Unified CVP Call Director with SIP with Proxy Server, Unified ICME



The following figure shows the call flow for this call flow model using H.323.

Figure 5: Unified CVP Call Director with H.323, Unified ICME



Note: Refer to "REFER Transfers Using "RFXXXX" Type Unified ICME Labels (page 361)" and "Using the sendtooriginator Setting in the SIP Service (page 365)" for more information.

Unified CVP Call Director (SIP/H.323) Call Flow Model, Unified ICMH

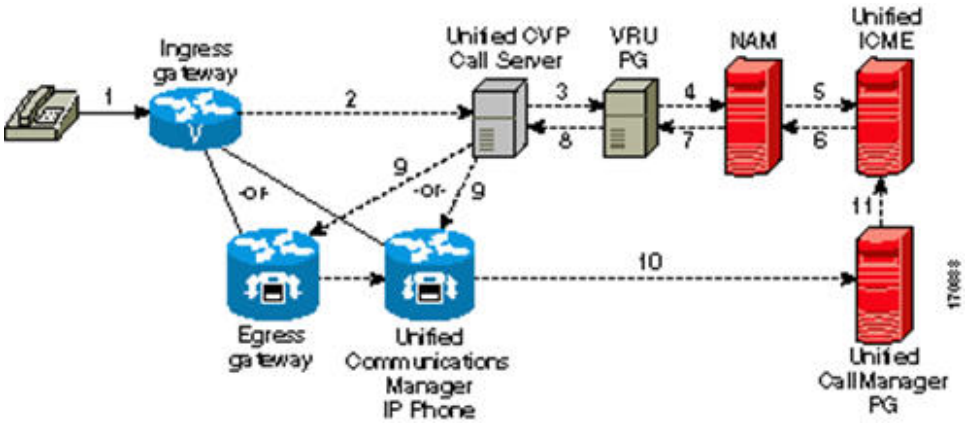
In this call flow model, Unified CVP only provides the Network Applications Manager (NAM) with VoIP call switching capabilities. You provide your own Service Control VRU, if you are using the NAM to queue calls or you might queue calls directly on your ACD. Calls might be transferred multiple times, from Ingress, to customer-provided VRU, to either the NAM or customer-provided ACD or agent, and back again. When calls are connected to customer-provided equipment, their voice paths must go to an egress gateway, which is connected by TDM to that equipment. If the signaling is SIP, then this call flow model works with customer-provided SIP endpoints which have been tested and certified to interoperate with Unified CVP.

Note: VRU scripts and transfer to a VRU leg are not available in this call flow model.

The following figure shows the call flow for this call flow model using SIP without a Proxy Server.

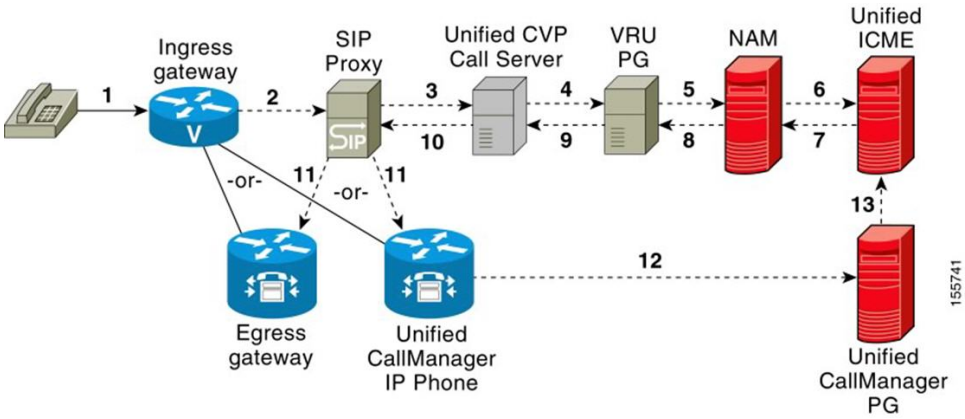
Note: In the following diagrams, solid lines indicate voice paths and dashed lines indicate signaling paths.

Figure 6: Unified CVP Call Director with SIP without Proxy Server, Unified ICMH



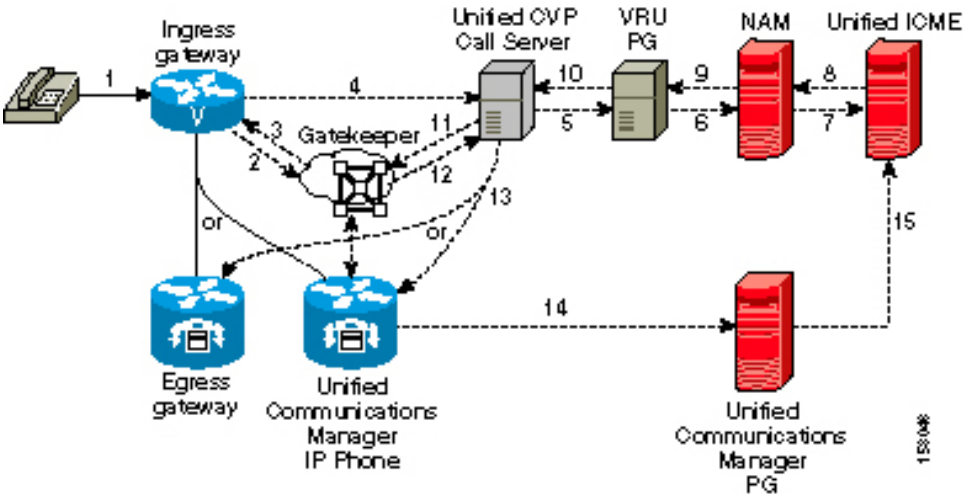
The following figure shows the call flow for this call flow model using SIP with a Proxy Server.

Figure 7: Unified CVP Call Director with SIP with Proxy Server, Unified ICMH



The following figure shows the call flow for this call flow model using H.323.

Figure 8: Unified CVP Call Director with H.323, Unified ICME



Refer to "Configuration Instructions for the Unified CVP Call Director Call Flow Model Using SIP, for Both Unified ICME and Unified ICMH (page 41)" for configuration instructions for this call flow model.

Note: Refer to "REFER Transfers Using "RFXXXX" Type Unified ICME Labels (page 361)" and "Using the sendtooriginator Setting in the SIP Service (page 365)" for more information.

Configuration Instructions for the Call Director Call Flow Model Using SIP, for Both Unified ICME and Unified ICMH

The following task contains the configuration instructions for the Call Director Call Flow Model using **SIP** for *both* Unified ICME and Unified ICMH.

Step 1 Install the IOS image on the Ingress Gateway.

For detailed information, refer to the **Cisco IOS documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html).

Step 2 Configure the Ingress Gateway:

a. Transfer the following script, configuration, and .wav files to the Ingress Gateway using the Unified CVP Operations Console (or the Unified CVP product CD):

- bootstrap.tcl
- handoff.tcl
- survivabilty.tcl
- bootstrap.vxml
- recovery.vxml
- ringtone.tcl

- cvperror.tcl
- ringback.wav
- critical_error.wav

In the Operations Console, select **Bulk Administration > File Transfer > Scripts and Media** and select **Help > This Page** for details.

b. Configure the Ingress Gateway base gateway settings.

Refer to "Ingress Gateway Configuration: Example Gateway Settings for SIP (page 56)

c. Configure the Ingress Gateway service settings.

Refer to "Ingress Gateway Configuration: Example Gateway Settings for SIP (page 56)

d. Configure the Ingress Gateway dial-peer for the Unified CVP Call Server.

Refer to "Ingress Gateway Configuration: Example of Incoming Pots Dial-peer (page 57)."

e. Configure a dial-peer for ringtone and error.

Refer to "Ingress Gateway Configuration: Example of SIP Ringtone Dial-Peer (page 58)" and "Ingress Gateway Configuration: Example of SIP Error Dial-peer (page ?)."

f. If you are using a Proxy Server, configure your session target in the outbound dial peer to point to the Proxy Server.

Refer to "Ingress Gateway Configuration: Example of Dial-peer to Reach the Unfied CVP Call Server or CUP Server (page 58)."

g. If you are using the sip-server global configuration, then configure the sip-server in the sip-ua section to be your Proxy Server and point the session target of the dial-peer to the sip-server global variable.

Refer to "Ingress Gateway Configuration: Example of Dial-peer to Reach the Unfied CVP Call Server or CUP Server (page 58)."

Note:

- Make sure your dial plan includes this information. You will need to refer to the dial plan when you configure the SIP Proxy Server for Unified CVP.
- The SIP Service voip dial peer and the destination pattern on the Ingress Gateway must match the DNIS in static routes on the SIP Proxy Server or Unified CVP Call Server.

Refer to "Configuring the SIP Devices (page 355)," and to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

Step 3 For SIP *without* a Proxy Server, complete the following steps:

a. If you are using DNS query with SRV or A types from the gateway, configure the gateway to use DNS.

Refer to the Operations Console online help for detailed instructions. If you are using DNS query with SRV or A types from the gateway, use the gateway configuration CLI as shown below:

Non-DNS Setup:

```
sip-ua
  sip-server ipv4:xx.xx.xxx.xxx:5060
!
```

DNS Setup:

```
ip domain name patz.cisco.com
ip name-server 10.10.111.16
!
sip-ua
  sip-server dns:cvp.pats.cisco.com
!
```

b. Configure the DNS zone file, for the separate DNS server, that displays how the Service (SRV) records are configured.

Note: SRV with DNS can be used in *any* of the SIP call flow models, with or without a Proxy server. Standard A type DNS queries can be used as well for the calls, without SRV, but they lose the load balancing and failover capabilities.

Refer to "DNS Zone File Configuration (page 59)" for more information.

Step 4 For SIP *with* a Proxy Server, use one of the following methods:

Note: Or, configure the Gateway statically instead of using DNS.

The following example shows how both the A and SRV type records could be configured:

```
ip host cvp4cc2.cisco.com 10.4.33.132
ip host cvp4cc3.cisco.com 10.4.33.133
ip host cvp4cc1.cisco.com 10.4.33.131
```

For **SIP/TCP**:

```
ip host _sip._tcp.cvp.cisco.com srv 50 50 5060 cvp4cc3.cisco.com
ip host _sip._tcp.cvp.cisco.com srv 50 50 5060 cvp4cc2.cisco.com
ip host _sip._tcp.cvp.cisco.com srv 50 50 5060 cvp4cc1.cisco.com
```

For **SIP/UDP**:

```
ip host _sip._udp.cvp.cisco.com srv 50 50 5060 cvp4cc3.cisco.com
ip host _sip._udp.cvp.cisco.com srv 50 50 5060 cvp4cc2.cisco.com
ip host _sip._udp.cvp.cisco.com srv 50 50 5060 cvp4cc1.cisco.com
```

Note: The DNS Server must be configured with all necessary A type or SRV type records.

Refer to "Configuring the SIP Devices (page 355).

If you are using the DNS Server, you can set your SIP Service as the Host Name (either A or SRV type).

Step 5 On the Unified CM server, CCMAdmin Publisher, complete the following **SIP-specific actions**:

a. Add a CUP Server.

To add the CUP Proxy Server, from the Unified CM Administration page, if you are using a version of CUP prior to 7.(3), select **System > Application Server > Add New:** Enter your CUP proxy in the Add New window in the Application Server area. (This step is done automatically by versions of CUP later than7.0(3))

Note: This step is not required if you are using CUSP server.

b. Create SIP trunks.

- If you are using a SIP Proxy Server, set up a SIP trunk to the SIP Proxy Server.
- Add a SIP Trunk for the Unified CVP Call Server.
- Add a SIP Trunk for each Ingress gateway that will send SIP calls to Unfied CVP that might be routed to Unified CM.

To add an SIP trunk, select **Device > Trunk > Add New** and use the following parameters:

- Trunk Type: **SIP trunk**
- Device Protocol: **SIP**
- Destination Address: IP address or host name of the SIP Proxy Server (if using a SIP Proxy Server). If not using a SIP Proxy Server, enter the IP address or host name of the Unified CVP Call Server.
- DTMF Signaling Method: **RFC 2833**
- Do **not** check the *Media Termination Point Required* checkbox.
- If you are using UDP as the outgoing transport on Unified CVP (and CUP server routes if using a proxy server), also set the outgoing transport to **UDP** on the SIP Trunk Security Profile.
- The default port to use for connection to CUP is 5060.

c. Add route patterns for outbound calls from the Unified CM devices using a SIP Trunk to the Unified CVP Call Server. Also, add a route pattern for error DN.

Select **Call Routing > Route/Hunt > Route Pattern > Add New**

Add the following:

- Route Pattern: Specify the route pattern; for example: “3XXX" for a TDM phone that dials 9+3xxx and all Unified ICME scripts are set up for 3xxx dialed numbers.
- Gateway/Route List: Select the SIP Trunk defined in the previous substep.
- Note the username and password for Unified CM. If you are using a SIP Proxy Server, you will need the Unified CM username and password when installing the CUP Server.

Note: For warm transfers, the call from Agent 1 to Agent 2 does not typically use a SIP Trunk, but you must configure the CTI Route Point for that dialed number on the Unified CM server and associate that number with your peripheral gateway user (PGUSER) for the JTAPI gateway on the Unified CM peripheral gateway. An alternative is to use the Dialed Number Plan on Unified ICME to bypass the CTI Route Point.

d. If you are sending calls to Unified CM using an SRV cluster domain name, select **Enterprise Parameters > Clusterwide Domain Configuration** and add the Cluster fully qualified domain name **FQDN**.

e. Note the username and password for Unified CM. If you are using a SIP Proxy Server, you will need the Unified CM username and password when installing the CUP Server.

For detailed instructions about using Unified CM and the CUP Server, refer to the Unified CM and **Cisco Unified Presence Server Serviceability Administration Guide** (http://www.cisco.com/en/US/products/ps6837/prod_maintenance_guides_list.html).

Step 6 (Optionally) Configure the **SIP Proxy Server**.

a. Configure the SIP static routes to the Unified CVP Call Server(s), Unified CM SIP trunks, and Gateways.

Configure the SIP static routes for intermediary transfers for ringtone, playback dialed numbers, and error playback dialed numbers.

Note: For failover and load balancing of calls to multiple destinations, configure the CUP server static route with priority and weight.

Select **Cisco Unified Presence Server > Proxy Server > Static Routes > Add New** (in CUP 7.0 **Presence > Routing > Static Routes**) and specify:

- Transport: **UDP**
- Destination Pattern: Specify the dialed number pattern for the destination.

  For example:

  777* (VRU leg label starting with pattern 777)

  9* (Errors are 9292 and ringtone is 9191)

  1* (All agent devices beginning with one, for example, 1XXX extensions)

14085551234 (Only match this specific number)

1408555.... (matches this number followed by any 4 digits in place of the .'s)

- Refer to *Valid Formats for Dialed Numbers* in the Operations Console online help.

  Also refer to "Configuring the SIP Devices (page 355)," and to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

- Next Hop: *<IP of Gateway or Unified Communications_Manager_short>*, depending on the Destination Pattern

  If the Destination Pattern is the VRU leg label, then the Next Hop would be the IP address of the VoiceXML Gateway.

  If the Destination Pattern is an agent label pattern, then the Next Hop would be the IP address of Unified CM on which the agent device is registered.

- Protocol Type: **UDP** (Best Practice) or, optionally **TCP**.

b. Configure Access Control Lists for Unified CVP calls.

Select **Proxy Settings > Incoming ACL** (in CUP 7.0, **System > Security > Incoming ACL**)

Address pattern: **all**

c. Configure the service parameters.

Select **Service Parameters**, then set the following:

- Add record route: **off**
- Maximum invite retransmission count: **2**
- Proxy Domain and Cluster Name: if using DNS SRV, set to the FQDN of your Proxy Server SRV name.

d. Write down the IP address and host name of the SIP Proxy Server. (You need this information when configuring the SIP Proxy Server in Unified CVP.)

Note: You must use the SIP Proxy components of the CUP server, not the Presence Engine component.

e. If using redundant SIP Proxy Servers (primary and secondary or load balancing), then decide whether to use DNS server lookups for SRV records or non-DNS based local SRV record configuration.

Note: If a single CUP Server is used, then SRV record usage is not required.

Configure the SRV records on the DNS server or locally on Unified CVP with a .xml file (local xml configuration avoids the overhead of DNS lookups with each call).

Note: Refer to the "Local SRV File Configuration Example for SIP Messaging Redundancy (page 140)" section for detailed information.

The Call Director call flow model with SIP calls will typically be deployed with dual CUP servers for redundancy. In some cases, you might want to purchase a second CUP server. Regardless, the default transport for deployment will be UDP, and make sure you *always* set the AddRecordRoute setting to **Off** with a CUP server.

For the required settings in the Unified CM Publisher configuration, refer to **Cisco Unified Presence Server documentation** (http://www.cisco.com/en/US/products/ps6837/tsd_products_support_series_home.html).

Step 7 Configure the PGs for the switch leg.

On Unified ICME, ICM Configuration Manager, **PG Explorer** tool:

a. Configure each peripheral gateway (PG) to be used for the **Switch** leg. In the tree view pane, select the applicable PG.

1. **Logical Controller** tab:
   - Client Type: **VRU**
   - Name: A name descriptive of this PG.

     For example: **<location>_A** for side A of a particular location
2. **Peripheral** tab:
   - Peripheral Name: A name descriptive of this Unified CVP peripheral

     For example: **<location>_<cvp1> or <dns_name>**
   - Client Type: **VRU**
   - Select the checkbox: **Enable Post-routing**
3. **Routing Client** tab:
   - Name: By convention, use the same name as the peripheral.
   - Client Type: **VRU**

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

b. Configure a peripheral for each Unified CVP Call Server to be used for a Switch leg connected to each PG.

Step 8 Configure dialed numbers.

On the Unified ICME or Unified ICMH Server, in the ICM Configuration Manager, configure the following items:

a. **Dialed Number List Tool** tab: Configure the dialed numbers.

b. **Call Type List tool** tab: Configure the call types.

c. **ICM Instance Explorer tool** tab: Configure the applicable customers.

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 9 Create a Routing Script.

On the Unified ICME or Unified ICMH Server in the ICM Script Editor tool:

Create a routing script that handles the incoming call. The routing script must execute a Label node or Select node (node that returns a label right away).

Note: *Do not* use the Queue node in the routing script.

The label must be configured in the Gatekeeper or SIP Proxy Server to the IP address of the device that the label corresponds to. The Proxy Server is optional. If you do not have one, you must configure the Gateway dial-peer to point to the Call Server (refer to the first step in this process). Also, you must configure the **destination labels** in the SIP Service for the Call Server.

For more information about creating scripts, refer to "Writing Scripts for Unified CVP (page 141)."

Refer to **Scripting and Media Routing Guide for Cisco Unified ICM/Contact Center Enterprise & Hosted** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_user_guide_list.html) for more information.

Step 10 If you are using a **SIP Proxy Server**, configure the SIP Proxy Server using the Operations Console.

Select: **Device Management > SIP Proxy Server**

For detailed information about configuring a SIP Proxy Server, refer to "Configuring a SIP Proxy Server (page 356)."

Step 11 Install and Configure Call Server(s).

In the Operations Console, install and configure the Call Server(s):

a. Enable the ICM and SIP Services on the Call Server.

In the Operations Console select **Device Management > Unified CVP Call Server**.

Click **Help > This Page** for details.

Select the check boxes: **ICM** and **SIP**

b. Configure the SIP Service:

Select **Device Management > CVP Call Server > SIP tab**.

- If you are using a SIP Proxy Server, enable the Outbound Proxy and select the SIP Proxy Server. If using a SIP Proxy Server, configure Local Static Routes on the SIP Proxy Server itself.
- If you are not using a SIP Proxy Server, configure Local Static Routes using the Dialed Number Pattern system configuration in the Operations Console. A local static route must be configured for each SIP gateway/ACD, SIP endpoint in order to receive calls
- Check the default values for the SIP Service and change, if desired.

Refer to "Configuring the SIP Devices (page 355)," and to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

c. Configure the ICM Service by setting the maximum length DNIS to the length of the Network Routing Number.

Select: **Device Management > CVP Call Server > ICM tab**

Set the Maximum Length of DNIS to length of the Network Routing Number.

Example: if the Gateway dial pattern is 1800******, the maximum DNIS length is 10.

For detailed information, refer to the Operations Console online help.

Step 12 Configure local static routes:

If an outbound proxy is enabled on the Operations Console, configure local static routes on the SIP Proxy Server.

If no outbound proxy is enabled, configure local static routes using the Operations Console Dialed Number Pattern system configuration. Refer to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

The following is an example of a local static route configuration. A local static route contains a dialed number pattern and a routing address (IP Address, Hostname, or SIP Server Group name):

- **`22291>,cvp-ringtone.cisco.com`**
- **`22292>,cvp-error.cisco.com`**
- 1>,ccm-subscribers.cisco.com
- 2>,ccm-subscribers.cisco.com

- 3>,ccm-subscribers.cisco.com

Step 13 **Optionally:** In the Operations Console, configure the **Reporting Server**. Select **Device Management > CVP Reporting Server > General tab**:

a. Configure the Reporting Server.

b. Select a Call Server to associate with this Reporting Server.

c. Check the default values of the Reporting properties and change, if desired.

For more information, refer to the **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

Configuration Instructions for the Unified CVP Call Director Call Flow Model Using H.323 for Both ICME and ICMH

The following task contains the configuration instructions for the Call Director Call Flow Model using **H.323** for *both* Unified ICME and Unified ICMH.

Note: Refer to the "Unified CVP Call Director (SIP/H.323) Call Flow Model, Unified ICMH (page 39)" section for detailed information about the Unified ICMH version of this call flow model.

Step 1 Install the IOS image on the Ingress Gateway.

For detailed information, refer to the **Cisco IOS documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html).

Step 2 Configure the Ingress Gateway:

a. Transfer the following script, configuration, and .wav files to the Ingress Gateway using the Unified CVP Operations Console:

- bootstrap.tcl
- handoff.tcl
- survivabilty.tcl
- bootstrap.vxml
- recovery.vxml
- cvperror.tcl
- critical_error.wav

In the Operations Console, select **Bulk Administration > File Transfer > Scripts and Media** and select **Help > This Page** for details.

Note: Optionally, you can transfer the files from the Unified CVP product CD.

b. Configure the Ingress Gateway base gateway settings.

Refer to "Ingress Gateway Configuration: Example Gateway Settings for Call Director Call Flow Model (page 56).

c. Configure the Ingress Gateway service settings.

Refer to "Ingress Gateway Configuration: Example Gateway Settings for Call Director Call Flow Model (page 56).

d. Configure the Ingress Gateway dial-peer for the Unified CVP Call Server.

Refer to "Ingress Gateway Configuration: Example of Incoming Pots Dial-peer for Call Director Call Flow Model (page 57)."

Refer to "Configuring the H.323 Devices and VoIP (page 465)."

Step 3 Configure the **Gatekeeper**:

Refer to "Ingress Gateway Configuration: Example of Using Gatekeeper Lookup to Reach the Call Server for Call Director Call Flow Model (page 58)."

a. Register all Gateways.

Add the boldfaced lines to the Gateway configuration. The Gateway auto-registers with the Gatekeeper once these lines are configured.

```
interface FastEthernet0/0
ip address 10.86.129.79 255.255.255.0
ip route-cache same-interface
duplex full
speed 100
no cdp enable
h323-gateway voip interface
h323-gateway voip id <zone ID specified on the GK> ipaddr <IP of GK>
1719
h323-gateway voip h323-id <IP of this GW>
h323-gateway voip tech-prefix 1#
```

For more information, refer to "Configuring the H.323 Devices and VoIP (page 465).

b. Configure the Gatekeeper to send all VRU connections to the applicable Gateway.

Include the following:

```
zone prefix  <GK> <Network Routing Number>* gw-priority 10 <H323-ID of gateway>
```

For example:

```
zone prefix gk-stooge 8001112222* gw-priority 10 vru1@provider.com vru2@provider.com
```

c. Configure the Unified ICME Label Node to the IP address of the device to which the label corresponds. Substitute your values for the placeholders in **bold**.

```
zone prefix sox-gk 1* gw-priority 10 <CCM Trunk ID>
zone prefix sox-gk 2* gw-priority 10 <CCM Trunk ID>
zone prefix sox-gk 4* gw-priority 10 <CCM Trunk ID>
zone prefix sox-gk 777* gw-priority <VXML GW1 IP> <VXML GW2 IP>
```

For example:

```
zone prefix sox-gk 1* gw-priority 10 SOX_CCM_H323_TRUNK_2 SOX_CCM_H323_TRUNK_3
zone prefix sox-gk 2* gw-priority 10 SOX_CCM_H323_TRUNK_2 SOX_CCM_H323_TRUNK_3
zone prefix sox-gk 4* gw-priority 10 SOX_CCM_H323_TRUNK_2 SOX_CCM_H323_TRUNK_3
zone prefix sox-gk 777* gw-priority 10 10.86.129.106 10.86.129.20
```

Step 4 On the Unified CM server, CCMAdmin Publisher, configure **H.323-specific actions**.

From the Unified CM Administration page:

a. Add the Gatekeeper.

Select: **Device > Gatekeeper > Add New**

Add the Host Name/IP address of Gatekeeper.

b. Add the Unified CVP H.323 Service as an H.323 gateway.

Select: **Device > Gateway > Add New**

Add the following:

- Gateway Type: **H.323 Gateway**
- Device Name: **<Unified CVP Call Server IP address>**
- Description: **<Unified CVP Call Server IP address (or other identifying text)>**
- Device Pool: **<device pool>**
- Calling Party Selection: From the Calling Party Selection list, select **Originator**
- Calling Party Presentation: **Allowed**

c. Add an H.323 Trunk for Unified CM to register with the Gatekeeper.

Select: **Device > Trunk > Add New**

Add the following:

- Trunk Type: **H.225 Trunk** (Gatekeeper controlled)

- Device Protocol: **H.225**
- Device Name: **<name for your trunk>**
- For example: **MY_CCM_TRUNK**
- Device Pool: **<device pool>**
- Gatekeeper Name: From the drop-down list, select the Gatekeeper to which this trunk will register.
- Terminal Type: **Gateway**
- Technology Prefix: **1#**

For detailed instructions about using Unified CM, refer to the Unified CM documentation.

Step 5 Configure the PGs for the switch leg.

On Unified ICME, ICM Configuration Manager, **PG Explorer** tool:

a. Configure each peripheral gateway (PG) to be used for the **Switch** leg.

In the tree view pane, select the applicable PG.

- **Logical Controller** tab:
  - Client Type: **VRU**
  - Name: A name descriptive of this PG.

    For example: **<location>_A** for side A of a particular location.
- **Peripheral** tab:
  - Peripheral Name: A name descriptive of this Unified CVP peripheral.

    For example: **<location>_<cvp1> or <dns_name>**
  - Client Type: **VRU**
  - Select the checkbox: **Enable Post-routing**
- **Advanced** tab:
  - Select the name of the Unified CVP VRU from the Network VRU field drop-down list.

    For example: **cvpVRU**
- **Routing Client** tab:
  - Name: By convention, use the same name as the peripheral.

– Client Type: **VRU**

b. Configure a peripheral for each Unified CVP Call Server to be used for a Switch leg connected to each PG.

For more information, refer to the **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 6 On the Unified ICME or Unified ICMH Server, in the ICM Configuration Manager:

a. Configure the dialed numbers, using the Dialed Number List tool.

b. Configure the call types, using the Call Type List tool.

c. Configure the applicable customers, using the ICM Instance Explorer tool.

d. Configure the Network VRU, using the Network VRU Explorer tool.

For more information, refer to the **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 7 On the Unified ICME or Unified ICMH Server in the ICM Script Editor tool, create a routing script that handles the incoming call.

The routing script must execute a Label node or Select node (node that returns a label right away).

The label must be configured in the Gatekeeper to the IP address of the device that the label corresponds to.

Note: *Do not* use the Queue node in the routing script.

For more information about creating scripts, refer to "Writing Scripts for Unified CVP (page 141)."

Refer to **Scripting and Media Routing Guide for Cisco Unified ICM/Contact Center Enterprise & Hosted** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_user_guide_list.html) for more information.

Step 8 Install and configure the **Call Server(s)**.

- In the Operations Console, select: **Device Management > CVP Call Server**
- Select the check boxes: **ICM**, **IVR**, and **H.323**

Click **Help > This Page** for details.

Step 9 Configure the **ICM Service**.

Note: For the following substeps, click **Help > This Page** for details.

- In the Operations Console, select: **Device Management > CVP Call Server > ICM tab**
- Specify the following required information:

a. VRU Connection Port

Set the VRU Connection Port to match the VRU connection Port defined in ICM Setup for the corresponding VRU peripheral gateway (PIM).

b. Maximum Length of DNIS

Set the maximum length DNIS to the length of the Network Routing Number.

Example: if the Gateway dial pattern is 1800******, the maximum DNIS length is 10.

c. Call service IDs: New Call and Pre-routed

Configure the ports for both groups according to the licenses purchased, call profiles, and capacity by completing the required fields on this tab.

d. Trunk group IDs: New Call and Pre-routed

Check the default values and change, if desired.

Refer to the Operations Console online help for information about other settings you might want to adjust from their default values.

Step 10 Configure the **IVR Service**.

a. In the Operations Console, select: **Device Management > CVP Call Server > IVR tab**

b. Check the default values and change, if desired.

Refer to the Operations Console online help for information about other settings you might want to adjust from their default values.

Step 11 Configure the **H.323 Service**.

a. In the VBAdmin tool, select: **Cisco Unified Customer Voice Portal H.323 Service**

b. Set the IP address of the Gatekeeper.

Specify: **SetGateKeeper** <*NewValue*>

Where <*NewValue*> is the IP address for the Gatekeeper serving the H.323 Service.

c. Configure the capacity.

For Unified ICMH only, configure the total number of calls and IVR ports according to the licenses purchased, call profiles, and capacity.

d. Check the default values and change, if desired.

Refer to the Operations Console online help for information about other settings you might want to adjust from their default values.

Step 12 (Optionally) Configure the **Reporting Server** .

a. In the Operations Console, select: **Device Management > CVP Reporting Server > General tab**

b. Configure the Reporting Server.

c. Select a Call Server to associate with the Reporting Server.

d. Check the default values of the Reporting properties and change, if desired.

For more information, refer to the **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

Ingress Gateway Configuration: Example Gateway Settings for Call Director Call Flow Model

The first part of the following example provides the basic configuration for setting an Ingress gateway:

- Applies a timestamp to debugging and log messages
- Turns on logging
- Turns off printing to the command line interface console
- Sends RTP packets
- Configures gateway settings

The last part of this example provides the following:

- Allows SIP to play a .wav file that enables caller to hear message from critical_error.wav
- Performs survivability
- Enables SIP to play ringtone to caller while caller is being transferred to an agent
- Logs errors on the gateway when the call fails

- Defines requirements for SIP Call Server

```
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
!
service internal
logging buffered 99999999 debugging
no logging console
!
ip cef
!
voice rtp send-recv
!
voice service voip
  signaling forward unconditional
  h323
  sip
    min-se 360
    header-passing
!
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
!
application
  service cvperror flash:cvperror.tcl
!
  service cvp-survivability flash:survivability.tcl
!
  service ringtone flash:ringtone.tcl
!
  service handoff flash:handoff.tcl
!
gateway
  timer receive-rtcp 6
!
ip rtcp report interval 3000
!
sip-ua
  retry invite 2
  timers expires 60000
  sip-server ipv4:<IP of CUP Server or Call Server>:5060
  reason-header override
!
```

Ingress Gateway Configuration: Example of Incoming Pots Dial-peer for Call Director Call Flow Model

The following example provides the configuration for an incoming Pots call for the Call Director call flow model:

```
dial-peer voice 8 pots
description Example incoming POTS dial-peer
service cvp-survivability
incoming called-number <your DN pattern here>
```

```
direct-inward-dial
!
```

Ingress Gateway Configuration: Example of SIP Ringtone Dial-peer for Call Director Call Flow Model

The following example provides the configuration for a SIP ringtone for the Call Director call flow model:

```
dial-peer voice 9191 voip
  description SIP ringtone dial-peer
  service ringtone
  voice-class codec 1
  voice-class sip rel1xx disable
  incoming called-number <your ringtone DN pattern here>
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

Ingress Gateway Configuration: Example of SIP Error Dial-peer for Call Director Call Flow Model

The following example provides the configuration for a SIP error dial-peer for the Call Director call flow model:

```
dial-peer voice 9292 voip
description SIP error dial-peer
service cvperror
  voice-class codec 1
  voice-class sip rel1xx disable
  incoming called-number <your error DN pattern here>
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

Ingress Gateway Configuration: Example of Dial-peer to Reach the Unified CVP Call Server or CUP Server for Call Director Call Flow Model

The following example provides the configuration for a dial-peer to reach the Unfied CVP Call Server or CUP Server for the Call Director call flow model:

```
dial-peer voice 800 voip
  description Example Call Server Dialpeer with CUP Server
  destination-pattern <your DN pattern here>
  voice-class codec 1
  session protocol sipv2
  session target sip-server
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

Ingress Gateway Configuration: Example of Using Gatekeeper Lookup to Reach the Call Server for Call Director Call Flow Model

The following example provides the configuration to use gatekeeper lookup for the Call Director call flow model:

```
dial-peer voice 800 voip
  description Example Call Server Dialpeer with GK
  destination-pattern <your DN pattern here>
  voice-class codec 1
  session target ras
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

DNS Zone File Configuration for Call Director Call Flow Model

DNS Zone File Linux NAMED Configuration Example

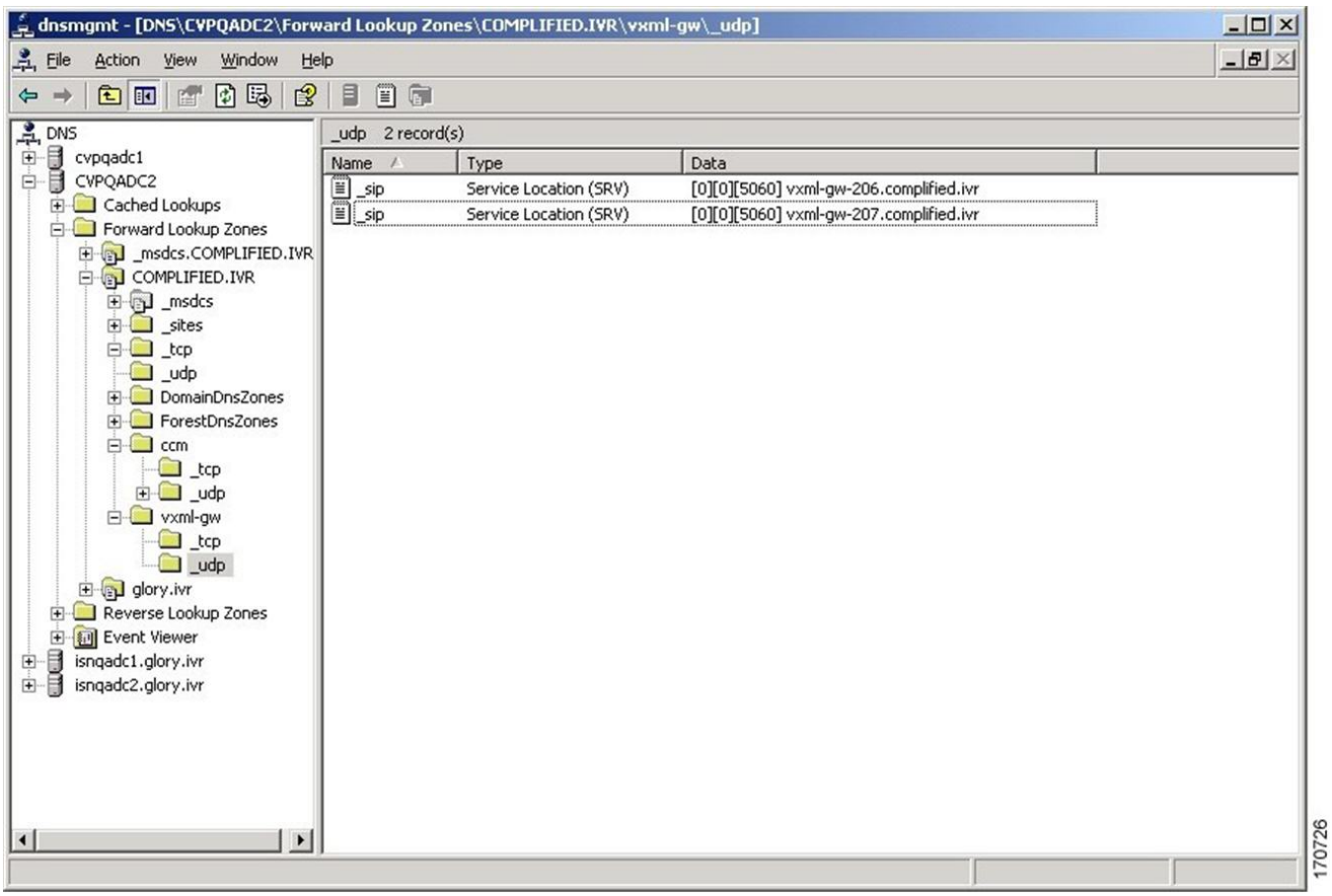
The following is an example of a DNS zone file Linux NAMED configuration.

```
ringtone-1 IN A 10.86.129.20
ringtone-2 IN A 10.86.129.229
vxml-1 IN A 10.86.129.20
vxml-2 IN A 10.86.129.229
vxml-3 IN A 161.44.81.254
cvp-1 IN A 10.86.129.211
cvp-2 IN A 10.86.129.220
cvp-3 IN A 161.44.81.254
; Priority Weight Port Target
sip._tcp.ringtone.sox.cisco.com. SRV 1 1 5060 ringtone-1.sox.cisco.com.
_
SRV 1 1 5060 ringtone-2.sox.cisco.com.
sip._udp.ringtone.sox.cisco.com. SRV 1 1 5060 ringtone-1.sox.cisco.com.
_
SRV 1 1 5060 ringtone-2.sox.cisco.com.
_sip._tcp.vxml.sox.cisco.com. SRV 1 1 5060 vxml-1.sox.cisco.com.
SRV 1 1 5060 vxml-2.sox.cisco.com.
SRV 1 1 5060 vxml-3.sox.cisco.com.
_sip._udp.vxml.sox.cisco.com. SRV 2 1 5060 vxml-1.sox.cisco.com.
SRV 2 1 5060 vxml-2.sox.cisco.com.
SRV 1 1 5060 vxml-3.sox.cisco.com.
_sip._tcp.cvp.sox.cisco.com. SRV 1 1 5060 cvp-1.sox.cisco.com.
SRV 2 1 5060 cvp-2.sox.cisco.com.
SRV 3 1 5060 cvp-3.sox.cisco.com.
_sip._udp.cvp.sox.cisco.com. SRV 1 1 5060 cvp-1.sox.cisco.com.
SRV 2 1 5060 cvp-2.sox.cisco.com.
SRV 3 1 5060 cvp-3.sox.cisco.com.
```

DNS Zone File MS DNS Configuration Example

The following is an example of a DNS zone file MS DNS configuration.

Figure 9: DNS Zone File MS DNS Configuration Example



Unified CVP Comprehensive (SIP/H.323) Call Flow Models

This section describes the comprehensive call flow models, broken down by Unified ICME and Unified ICMH products as follows:

- Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICME (page 61)
- Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICMH (page 64)

This section provides high-level instructions, broken down by SIP and H.323 implementation as follows:

- Configuration Instructions for the Unified CVP Comprehensive Call Flow Model Using SIP for Both ICME and ICMH (page 67)
- Configuration Instructions for the Unified CVP Comprehensive Call Flow Model Using H.323 for Both ICME and ICMH (page 86)

Note:

Both SIP and H.323 calls using the Unified CVP micro-applications will use the Call Server's IVR Service that has the switch leg of the call. VoiceXML fetches are always sent to the Call Server that has the switch leg of the call. The VoiceXML traffic for micro-applications must return to the same Call Server as the switch leg and no other server.

Sending VoiceXML traffic to multiple application servers by the H.323 Voice Browser (as was done in previous Unified CVP/ISN releases) is no longer supported. This is implemented in the Unified CVP 4.0(1) and later releases by extracting the Call Server's IP from the signaling SIP and H.323 messages in the bootstrap service rather than using IOS static configuration in the service parameter for the VoiceXML Gateway's bootstrap service, as was done in previous Unified CVP releases.

The service configuration parameters for the Call Server host and port are meant for the Unified CVP VRU-Only call flow model. These parameters are optional, if, for some reason, you need to override the IP address/port# of the Call Server that comes in through the SIP app-info header, or the H.323 headers.

```
application
  service vru-leg flash:bootstrap.tcl
  param cvpserverhost xxx.xxx.xxx.xxx  <IP of primary Call Server>
  param cvpserverbackup xxx.xxx.xxx.xxx  <IP of backup Call Server>
  param cvpserverport 8000   <TCP Port # of Call Server>
```

The Unified CVP Comprehensive call flow model with SIP/H.323 extracts the Call Server host from the signaling. That is, the Unified CVP SIP Service or the H.323 Service is handling the switch legs of the call. If you are performing a SIP call that does not involve the switch leg with Unified CVP, then the service parameters below will apply for the "VRU leg only" call flow. Cisco requires that Comprehensive calls always use the same Call Server for both switch leg and VRU legs. Using the same Call Server simplifies the solution and make it easier to troubleshoot and debug.

Note: The app-info header is only for SIP calls. The incoming H.323 calls use the *leg_remote_signaling_ip_address* parameter in the call. If this parameter is empty, then the primary Call Server IP, as configured on the service, will be used. If that server is "out-of-service," then it will try the backup Call Server.

Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICME

The Unified CVP Comprehensive call flow model combines the Call Director and the VRU-Only call flow models. It provides initial prompt and collect, self service IVR, queuing, and VoIP switching among Unified ICME and TDM agents.

In this call flow model, Unified CVP acts as the Switch, transferring the call to the Network VRU and to agents. The Unified CVP IVR Service in the Operations Console is also configured to work with the VoiceXML Gateway to provide VRU treatment, which might include ASR/TTS.

The Unified CVP Comprehensive call flow model requires the following components:

- Call Server (using either SIP or H.323 call signaling)
- Unified ICME

- Ingress Gateway
- VoiceXML Gateway
- Operations Console

The following optional components can be used in this call flow model:

- Reporting Server
- VXML Server
- Call Studio
- Speech Servers

Caution: The Operations Console can only manage speech servers installed on *Windows*, not on Linux. If the speech server is installed on Linux, the server cannot be managed.

- Media Servers
- DNS Server
- Content Services Switch or Application Control Engine
- Gatekeeper
- SIP Proxy Server (if the Call Server is configured to use SIP signaling, a SIP Proxy Server is optional)

In this call flow model, both the Voice Gateway and the Call Server see two call legs for the same call:

- One for the Switch leg.
- One for the VRU leg.

For the **Switch** leg, the Gateway simply provides Gateway capabilities from TDM to VoIP and call-switching capabilities. For the **VRU** leg, the Gateway provides VRU voice treatment.

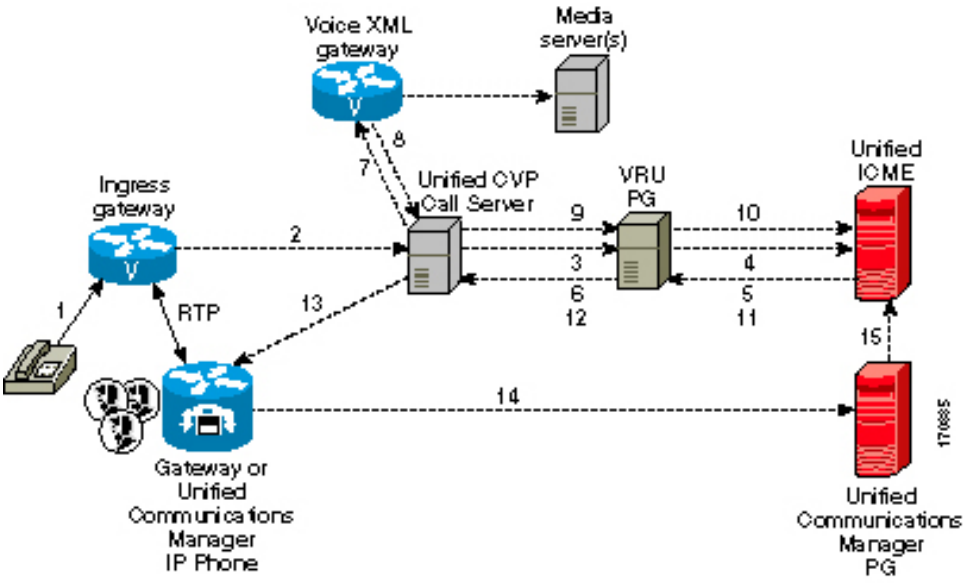
Note:

- Although the following figures show two Gateways (one where the call arrives and a separate one for the VRU leg), these could be the same physical Gateway.
- For simplicity, the figures do not illustrate a call flow model for redundancy and failover.

The following figure shows the call flow for this call flow model using SIP without a Proxy Server.

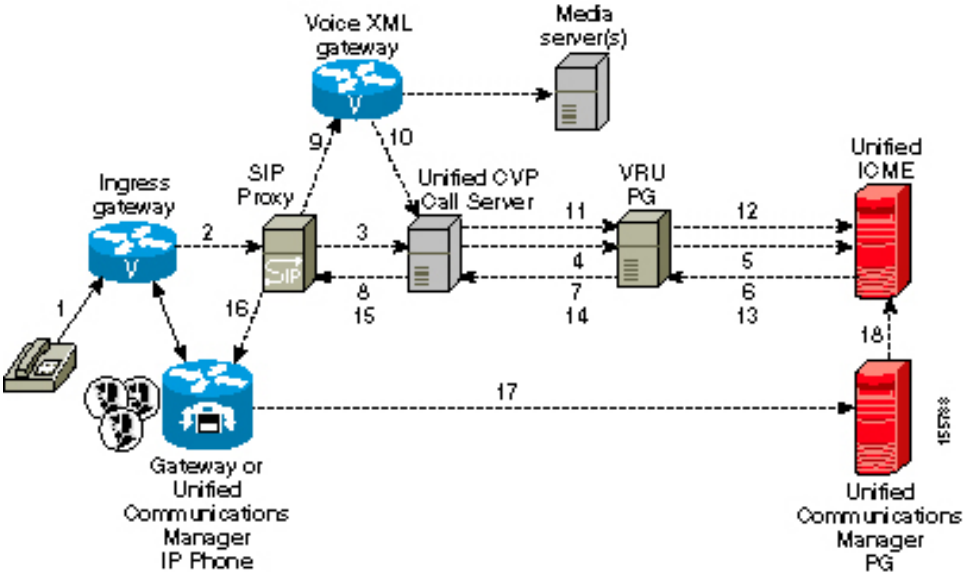
Note: In the following diagrams, solid lines indicate voice paths and dashed lines indicate signaling paths.

Figure 10: Unified CVP Comprehensive with SIP without Proxy Server, Unified ICME



The following figure shows the call flow for this call flow model using SIP with a Proxy Server.

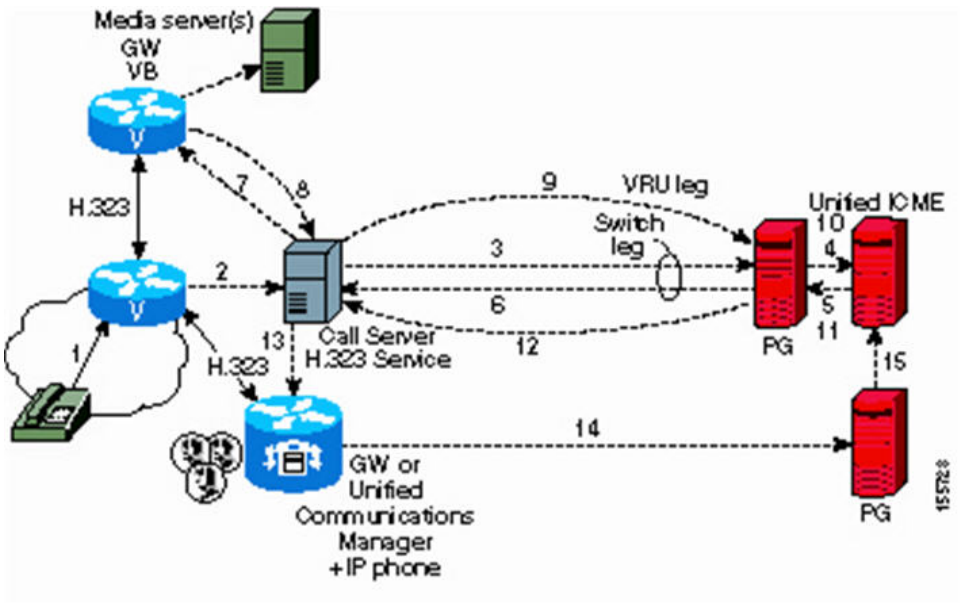
Figure 11: Unified CVP Comprehensive with SIP with Proxy Server, Unified ICME



Note: For simplicity, the Gatekeeper is not shown in this diagram.

The following figure shows the call flow for this call flow model using H.323.

Figure 12: Unified CVP Comprehensive with H.323, Unified ICME



Note: The numbers in the figure indicate call flow progression.

Configuration overview for this call flow model:

- There is one Network VRU, Type 10, which serves both the Switch and VRU legs.
- Use the ICM Script Editor's SendToVRU node to connect the call to the Network VRU.

Note: Refer to "REFER Transfers Using "RFXXXX" Type Unified ICME Labels (page 361)" and "Using the sendtooriginator Setting in the SIP Service (page 365)" for more information.

Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICMH

In this call flow model, Unified CVP is deployed at the NAM where it acts as the Switch, transferring the call to the Network VRU (using the Correlation ID transfer mechanism) and to agents. The Unified CVP IVR Service in the Operations Console is also configured to work with the VoiceXML Gateway to provide VRU treatment, which might include ASR/TTS.

Note:

- This call flow model does not support calls that originate in IP.
- Refer to the "Calls Which are Originated by Cisco Unified Communications Manager (page 126)" section for instructions on how to implement IP-originated calls in a way which is supplemental to this call flow model (Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICMH); however, this requires that an additional Unified CVP Call Server be attached to the CICM.

In this call flow model, both the VoiceXML Gateway and the Unified CVP Call Server see two calls:

- One for the Switch leg.
- One for the VRU leg.

For the Switch leg, the Gateway simply provides Gateway capabilities from TDM to VoIP and call-switching capabilities. For the VRU leg, the Gateway provides VRU voice treatment.

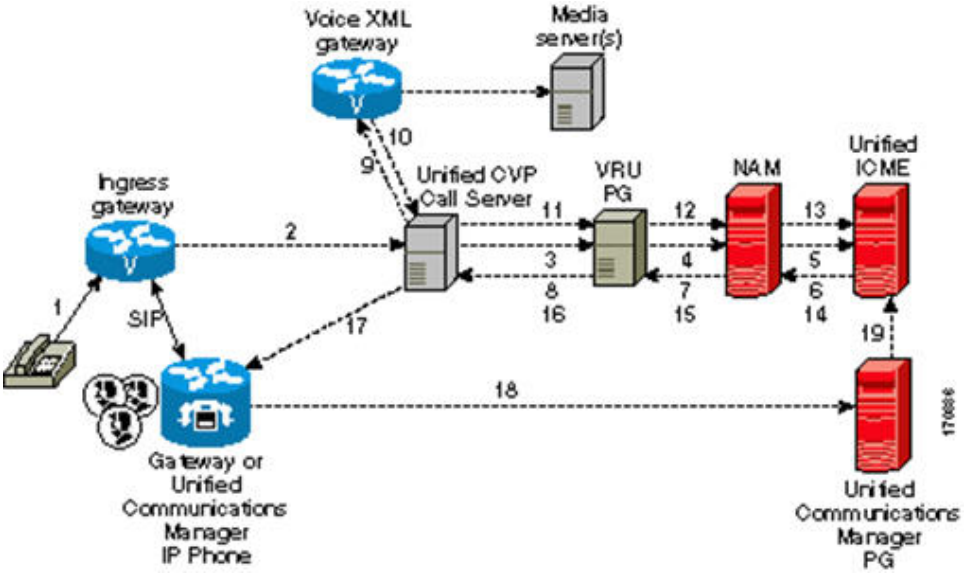
Unified ICMH sees these as a single call routed through different peripherals for different purposes.

The following figure shows the call flow for this call flow model using SIP without a Proxy Server.

Note:

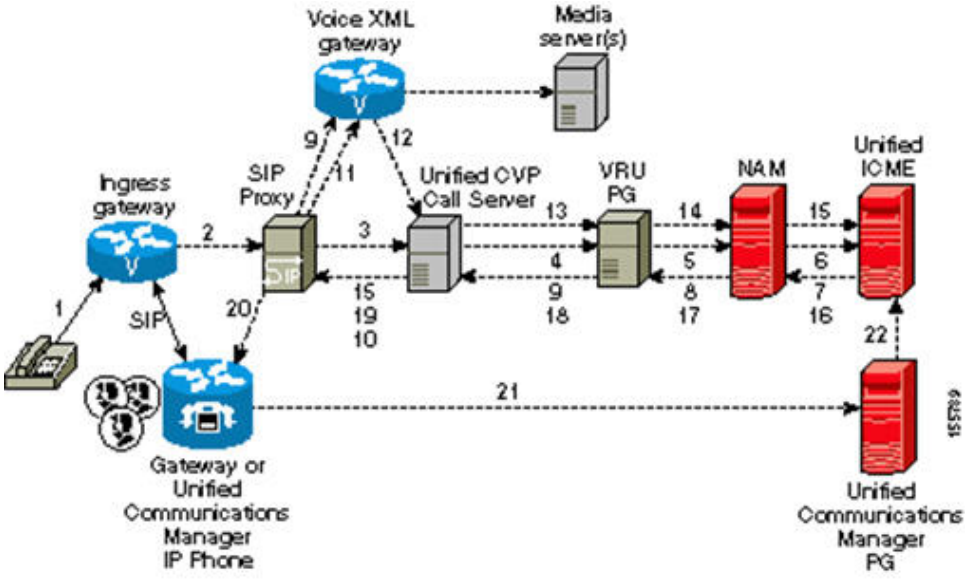
- In the following diagrams, solid lines indicate voice paths and dashed lines indicate signaling paths.
- Although the following figure shows two Gateways (one where the call arrives and a separate one for the VRU leg), these could be the same physical Gateway. Similarly, the Unified CVP IVR Service in the Operations Console and the PG could be the same physical machine.
- For simplicity, the figure does not illustrate a call flow model for redundancy and failover. Also, the Gatekeeper is not shown.

Figure 13: Unified CVP Comprehensive with SIP without Proxy Server, Unified ICMH



The following figure shows the call flow for this call flow model using SIP with a Proxy Server.

Figure 14: Unified CVP Comprehensive with SIP with Proxy Server with Proxy Server, Unified ICMH

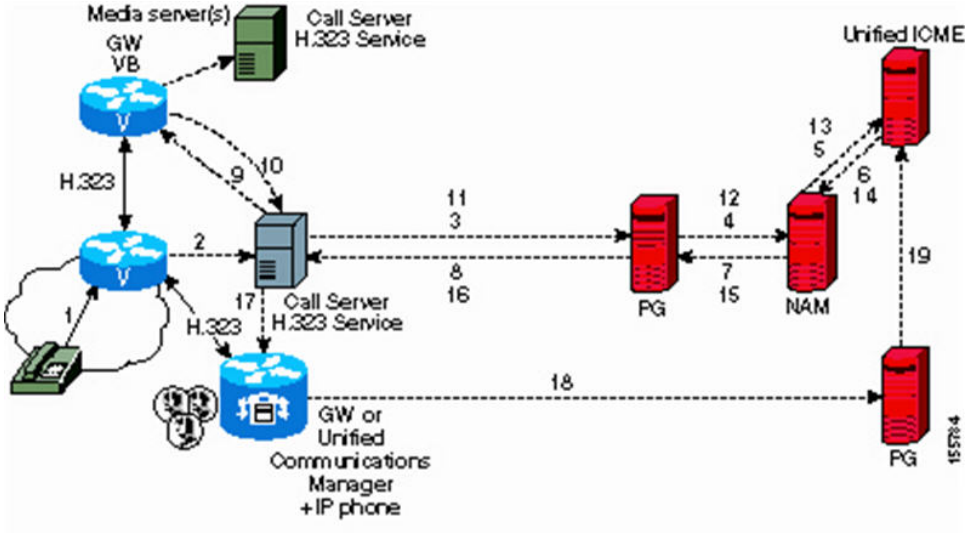


Note: The numbers in the figure indicate call flow progression.

The following figure shows the call flow for this call flow model using H.323.

Note: For simplicity, this diagram does not show interactions with the Gatekeeper.

Figure 15: Unified CVP Comprehensive with H.323, Unified ICMH



Configuration overview for this call flow model:

- There are two Network VRUs:
  - One on the NAM for the Switch leg and the VRU leg (Type 10).
  - One for the CICM for the INCRP connection.

- The Network VRU names (where applicable) and the ECC variable configurations must be identical on the NAM and CICM. All Labels must also be duplicated but their routing clients will be different.
- Use the ICM Script Editor's SendToVRU node to connect the call to the Network VRU.

Refer to "Configuration Instructions for the Unified CVP Comprehensive Call Flow Model Using SIP for Both ICME and ICMH (page 67)."

Note: Refer to "REFER Transfers Using "RFXXXX" Type Unified ICME Labels (page 361)" and "Using the sendtooriginator Setting in the SIP Service (page 365)" for more information.

Configuration Instructions for the Comprehensive Call Flow Model Using SIP for Both ICME and ICMH

The following high-level configuration steps provide configuration instructions for the Comprehensive Call Flow Model using *SIP* for *both* Unified ICME and Unified ICMH.

Note: Refer to "Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICMH (page 64)" for detailed information about the Unified ICMH version of this call flow model.

Step 1 Install the IOS image on the Ingress Gateway.

For detailed information, refer to the **Cisco IOS documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html).

Step 2 Transfer the following script, configuration, and .wav files to the Ingress gateway using the Unified CVP Operations Console (or the Unified CVP product CD):

- bootstrap.tcl
- handoff.tcl
- survivabilty.tcl
- bootstrap.vxml
- recovery.vxml
- ringtone.tcl
- cvperror.tcl
- ringback.wav
- critical_error.wav

In the Operations Console, select **Bulk Administration > File Transfer > Scripts and Media** and select **Help > This Page** for details.

Step 3 Configure the Ingress Gateway base settings.

Refer to "Ingress Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model (page 97)."

Step 4 Configure the Ingress Gateway service settings.

Refer to "Ingress Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model (page 97)."

Step 5 Configure an Ingress Gateway incoming Pots dial-peer.

Refer to "Ingress Gateway Configuration: Example of Incoming Pots Dial-peer for Comprehensive Call Flow Model (page 99)."

Step 6 (Optionally) Configure a dial-peer for ringtone and error.

Refer to "Ingress Gateway Configuration: Example of SIP Ringtone Dial-Peer for Comprehensive Call Flow Model (page 99)" and "Ingress Gateway Configuration: Example of SIP Error Dial-peer for Comprehensive Call Flow Model (page 100)."

Step 7 If you are **using a Proxy Server**, configure your session target in the outbound dial peer to point to the Proxy Server.

Refer to "Ingress Gateway Configuration: Example of Dial-peer to Reach the Unified CVP Call Server or CUP Server for Comprehensive Call Flow Model (page 100)."

Step 8 If you are using the sip-server global configuration, configure the sip-server in the sip-ua section to be your Proxy Server and point the session target of the dial-peer to the sip-server global variable.

Note:

- Make sure your dial plan includes this information. You will need to refer to the dial plan when you configure the SIP Proxy Server for Unified CVP.
- The SIP Service voip dial peer and the destination pattern on the Ingress Gateway must match the DNIS in static routes on the SIP Proxy Server or Unified CVP Call Server.

Refer to "Configuring the SIP Devices (page 355)," and to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

Step 9 For **SIP without a Proxy Server**, complete the following steps:

a. If you are using DNS query with SRV or A types from the gateway, configure the gateway to use DNS.

Also, if you are using DNS query with SRV or A types from the gateway, use CLI as shown below:

Note: Normally, non-DNS setup is: `sip-server ipv4:xx.xx.xxx.xxx:5060`.

```
ip domain name pats.cisco.com
ip name-server 10.86.129.16
sip-ua
sip-server dns:cvp.pats.cisco.com
```

OR:

```
ipv4:xx.xx.xxx.xxx:5060
```

b. Configure the DNS zone file, for the separate DNS server, that displays how the Service (SRV) records are configured.

Note: SRV with DNS can be used in *any* of the SIP call flow models, with or without a Proxy server. Standard A type DNS queries can be used as well for the calls, without SRV, but they lose the load balancing and failover capabilities.

Refer to "DNS Zone File Configuration for Comprehensive Call Flow Model (page 102)."

Step 10 For **SIP with a Proxy Server**, if you are using the DNS Server, you can set your SIP Service as the Host Name (either A or SRV type).

You can also configure the Gateway statically instead of using DNS. The following example shows how both the A and SRV type records could be configured:

```
ip host cvp4cc2.cisco.com 10.4.33.132
ip host cvp4cc3.cisco.com 10.4.33.133
ip host cvp4cc1.cisco.com 10.4.33.131
```

For SIP/TCP:

```
ip host _sip._tcp.cvp.cisco.com srv 50 50 5060 cvp4cc3.cisco.com
ip host _sip._tcp.cvp.cisco.com srv 50 50 5060 cvp4cc2.cisco.com
ip host _sip._tcp.cvp.cisco.com srv 50 50 5060 cvp4cc1.cisco.com
```

For SIP/UDP:

```
ip host _sip._udp.cvp.cisco.com srv 50 50 5060 cvp4cc3.cisco.com
ip host _sip._udp.cvp.cisco.com srv 50 50 5060 cvp4cc2.cisco.com
ip host _sip._udp.cvp.cisco.com srv 50 50 5060 cvp4cc1.cisco.com
```

Note: The DNS Server must be configured with all necessary A type or SRV type records.

Refer to "Configuring the SIP Devices (page 355)" and the Operations Console online help, **Managing devices > Configuring a SIP Proxy Server** for detailed information.

Step 11 Transfer files to the **VXML** Gateway.

Transfer the following scripts, configuration, and .wav files to the gateway, using the Operations Console (or the Unified CVP product CD):

- bootstrap.tcl

- handoff.tcl
- survivabilty.tcl
- bootstrap.vxml
- recovery.vxml
- ringtone.tcl
- cvperror.tcl
- ringback.wav
- critical_error.wav

In the Operations Console, select **Bulk Administration > File Transfer > Scripts and Media** and select **Help > This Page** for details.

Step 12 Configure the VXML Gateway base settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model (page 100)."

Step 13 Configure the VXML Gateway service settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model (page 100)."

Step 14 Configure the ICM VRU Label.

Refer to "VoiceXML Gateway Configuration: Example of ICM VRU Label for Comprehensive Call Flow Model (page 102)."

Refer to "Configuring the SIP Devices (page 355)," for detailed information.

Step 15 Optionally, enable security media fetches.

The VXML that the IVR Service returns as a response to an HTTP/HTTPS request from the VXML gateway contains URLs to media servers, so that the gateway knows where to fetch the media files from.

The URLs to the media servers in the VXML returned by the IVR Service can be controlled so that they are either HTTP or HTTPS URLs. This property is a boolean property called **Use Security for Media Fetches**. By default, it is set to "false." A value of "true" means generate HTTPS URLs to media servers and a value of "false" means generate HTTP URLs to media servers.

This property is only applicable if both of the following conditions are true:

- The client is not the H.323 Service. This is because the H.323 Service does not support HTTPS.

- In the Unified ICME script, the media server (specified in ECC variable "call.user.microapp.media_server") is not set to a URL that explicitly specifies an HTTP or HTTPS scheme.

  An example of a URL that explicitly specifies an HTTP scheme is "http://<servername>:80." One that specifies an HTTPS scheme is "https://<servername>:443." An example of a URL that does **not** specify the scheme is "<servername>."

  In the Operations Console, the user-visible text for this property is "Use Security for Media Fetches." The Call Server does not have to be re-started for this property to take effect.

Click the **Use Security for Media Fetches** checkbox on the IVR Service tab.

Refer to the Operations Console online help for detailed information about the IVR Service.

Refer to the Micro-Application ECCs table (page 132) for details about the user.microapp.media_server ECC variable.

Step 16 ASR and TTS Servers

If you will use ASR and TTS Servers, specify IP addresses for those servers for each locale using the applicable name resolution system for the Gateway (DNS or “ip host” commands).

Note: If ASR and TTS use the same server, the MRCP server might allocate one license for the ASR session and a second license for the TTS section. If you are hosting both ASR and TTS on the same speech server, you must select the **ASR/TTS use the same MRCP server** option in the IVR Service configuration tab in the Operations Console and follow the instructions in the step below.

Do **one** of the following:

- If you *are* using CSS or ACE, the server name is configured to the virtual IP (VIP) of the Call Server on CSS or ACE. Refer to "Configuring High Availability for Unified CVP (page 489)."

- If you are *not using* a CSS, primary and backup servers must be configured. If using name resolution local to the Gateway (rather than DNS) specify:

  `ip host asr-`*<locale><ASR server for locale>*

  `ip host asr-`*<locale>***`-backup`***<backup ASR server for locale>*

  `ip host tts-`*<locale><TTS server for locale>*

  `ip host tts-`*<locale>***`-backup`***<backup TTS server for locale>*

  For example: For English US, use:

```
ip host asr-en-us 10.86.129.215
```

Step 17 ASR and TTS Share Same MRCP Server

If you want the ASR and TTS to use the same MRCP server option, you must configure the gateway as follows.

a. In the IVR Service in the Operations Console, select the **ASR/TTS use the same MRCP server** option.

b. Add the following two host names to the gateway configuration:

- **`ip host asrtts-`**<*locale*> <*IP Address Of MRCP Server*>
- **`ip host asrtts-`**<*locale*> **`-backup`** <*IP Address Of MRCP Server*>

  Where the *locale* might be something like en-us or es-es, resulting in **`asrtts-en-us`** or **`asrtts-es-es`**.

c. Change the 'ivr asr-server' and 'ivr tts-server' lines as follows:

- **`ivr asr-server rtsp://asrtts-en-us/recognizer`**
- **`ivr tts-server rtsp://asrtts-en-us/synthesizer`**

Step 18 ASR and/or TTS with External Grammar

If you are using ASR and/or TTS with *external grammar* and expect the Gateway to resolve the addresses for those servers, specify the following ASR and/or TTS gateway commands.

- When using **Nuance/Scansoft 3.0** servers:

  ```
  ivr asr-server rtsp://<hostname or ip address of ASR server> /
  recognizer ivr tts-server rtsp://<hostname or ip address of TTS
  server> /synthesizer
  ```

- When using **Scansoft 2.0** servers:

  ```
  ivr asr-server rtsp://<hostname or ip address of ASR server> /media/
  speechrecognizer ivr tts-server rtsp://<hostname or ip address of
  TTS server> /media/speechsynthesizer
  ```

  Note: This setting does not accommodate multiple locales and assumes that a CSS is deployed for failover handling.

Step 19 Configure the speech servers to work with Unified CVP.

Caution: The Operations Console can only manage speech servers installed on *Windows*, not on Linux. If the speech server is installed on Linux, the server cannot be managed.

To ensure that the speech servers work with Unified CVP, you must make the following changes on each speech server as part of configuring the Unified CVP solution.

If you are using Nuance SpeechWorks MediaServer (SWMS), the configuration file is osserver.cfg. If you are using Nuance Speech Server (NSS), the configuration file is NSSserver.cfg.

Make the following changes to the Nuance configuration file:

- **Change:** server.resource.2.url VXIString media/speechrecognizer

  To: server.resource.2.url VXIString recognizer

- **Change:** server.resource.4.url VXIString media/speechsynthesizer

  To: server.resource.4.url VXIString synthesizer

- **Change:** server.mrcp1.resource.3.url VXIString media/speechrecognizer

  To: server.mrcp1.resource.3.url VXIString /recognizer

- **Change:** server.mrcp1.resource.2.url VXIString media/speechsynthesizer

  To: server.mrcp1.resource.2.url VXIString media/synthesizer

- **Change:** server.mrcp1.transport.port VXIInteger 4900

  To: server.mrcp1.transport.port VXIInteger 554

In the IBM WebSphere opt/websphere/VoiceServer/config/cwv.properties

- **Change:** com.ibm.voice.server.rtsbridge.recourl=/media/recognizer

  To: com.ibm.voice.server.rtsbridge.recourl=recognizer

- **Change:** com.ibm.voice.server.rtsbridge.syntheurl=/media/synthesizer

  To: com.ibm.voice.server.rtsbridge.syntheurl=synthesizer

If you are using Nuance Speech Server 5 and/or Nuance Vocalizer for Network 5, you will need to make changes to configuration files for each application. Make the following changes to the Nuance Speech Server 5 configuration file (NSSserver.cfg):

- **Change:** server.mrcp1.resource.3.url VXIString media/speechrecognizer

  To: server.mrcp1.resource.3.url VXIString /recognizer

- **Change:** server.mrcp1.resource.2.url VXIString media/speechsynthesizer

  To: server.mrcp1.resource.2.url VXIString /synthesizer

- **Change:** server.mrcp1.transport.port VXIInteger 4900

  To: server.mrcp1.transport.port VXIInteger 554

- **Change:** server.mrcp1.transport.dtmfPayloadType VXIInteger 96

  To: server.mrcp1.transport.dtmfPayloadType VXIInteger 101

- **Uncomment the following:** server.rtp.dtmfTriggerLeading VXIInteger 0

  If you are using the NuanceVocalizer for Network 5 TTS System, the following configuration files will need to be updated:

  <install path>\Nuance Vocalizer for Network 5.0\config\ttsrshclient.xml

- **Change:** <ssml_validation>strict</ssml_validation>

  To:<ssml_validation>warn</ssml_validation>

  <install path>\Nuance Vocalizer for Network 5.0\config\ttssapi.xml

- **Change:** <ssml_validation>strict</ssml_validation>

  To: <ssml_validation>warn</ssml_validation>

Step 20 Configure the characteristics for the VRU leg.

Specify the commands listed in the "Characteristics for the VRU Leg (page 104)" section to provide voice treatment.

Note:

- You do not need to configure all services listed. Required services depend on the call flow model you are using and whether SIP or H.323 calls are being processed.
- These characteristics are for VRU legs requiring ASR and/or TTS treatment. If you have other requirements for DTMF relay, Codecs or VAD settings, you must modify the commands accordingly.

Step 21 Configure SIP-Specific Actions.

On the Unified CM server, CCMAdmin Publisher, configure **SIP-specific actions**:

a. Add the CUP Server.

Select: **System > Application Server > Add New**

Enter your CUP server in the Add New window in the Application Server area.

Note: This step is not required if you are using CUSP server.

b. Create SIP trunks:

- If you are using a SIP Proxy Server, set up a SIP trunk to the SIP Proxy Server.
- Add a SIP Trunk for the Unified CVP Call Server.

- Add a SIP Trunk for each Ingress gateway that will send SIP calls to Unified CVP that might be routed to Unified CM.

Select **Device > Trunk > Add New** and add the following:

- Trunk Type: **SIP trunk**
- Device Protocol: **SIP**
- Destination Address: IP address or host name of the SIP Proxy Server (if using a SIP Proxy Server). If not using a SIP Proxy Server, enter the IP address or host name of the Unified CVP Call Server.
- DTMF Signaling Method: **RFC 2833**
- Do **not** check the **Media Termination Point Required** checkbox.
- If you are using UDP as the outgoing transport on Unified CVP (and CUP Server routes if using a proxy server), also set the outgoing transport to **UDP** on the SIP Trunk Security Profile.

c. Add route patterns for outbound calls from Unified CM devices using a SIP Trunk to the Unified CVP Call Server. Also, add a route pattern for error DN.

Note:

- CVP solution does not support 100rel. Under the SIP profile for the Trunk, confirm that SIP Rel1xx Options are disabled.
- For warm transfers, the call from Agent 1 to Agent 2 does not typically use a SIP Trunk, but you must configure the CTI Route Point for that dialed number on the Unified CM Server and associate that number with your peripheral gateway user (PGUSER) for the JTAPI gateway on the Unified CM peripheral gateway. An alternative is to use the Dialed Number Plan on Unified ICME to bypass the CTI Route Point.

Select: **Call Routing > Route/Hunt > Route Pattern > Add New**

- Route Pattern: Specify the route pattern; for example: “3XXX" for a TDM phone that dials 9+3xxx and all Unified ICME scripts are set up for 3xxx dialed numbers.
- Gateway/Route List: Select the SIP Trunk defined in Step 2.
- Note the username and password for Unified CM. If you are using a SIP Proxy Server, you will need the Unified CM username and password when installing the CUP Server.

d. If you are sending calls to Unified CM using an SRV cluster domain name, configure the cluster domain name.

- Select: **Enterprise Parameters > Clusterwide Domain Configuration**
- Add the Cluster fully qualified domain name: **FQDN**

e. Note the username and password for Unified CM. If you are using a SIP Proxy Server, you will need the Unified CM username and password when installing the CUP Server.

For detailed instructions about using Unified CM and the CUP Server, refer to the Unified CM and **Cisco Unified Presence Server documentation** (http://www.cisco.com/en/US/products/ps6837/tsd_products_support_series_home.html).

Step 22 (Optionally) Configure the **SIP Proxy Server**.

From the CUP Server Administration web page (**http://<CUP server>/ccmadmin**):

a. Configure the SIP static routes to the Unfied CVP Call Server(s), Unified CM SIP trunks, and Gateways.

Configure the SIP static routes for intermediary transfers for ringtone, playback dialed numbers, and error playback dialed numbers.

Note: For failover and load balancing of calls to multiple destinations, configure the CUP Server static route with priority and weight.

Select **Cisco Unified Presence Server > Proxy Server > Static Routes > Add New** (in CUP 7.0 **Presence > Routing > Static Routes**) and specify:

- Transport: **UDP**
- Destination Pattern: Specify the dialed number pattern for the destination.

  For example:

  777* (VRU leg label starting with pattern 777)

  9* (Errors are 9292 and ringtone is 9191)

  1* (All agent devices beginning with one, for example, 1XXX extensions)

  14085551234 (Only match this specific number)

  1408555.... (matches this number followed by any 4 digits in place of the .'s)

  Refer to *Valid Formats for Dialed Numbers* in the Operations Console online help for format and precedence information.

- Next Hop: *<IP of Gateway or Unified Communications_Manager_short>*, depending on the Destination Pattern.

  If the Destination Pattern is the VRU leg label, then the Next Hop would be the IP address of the VoiceXML Gateway.

  If the Destination Pattern is an agent label pattern, then the Next Hop would be the IP address of Unified CM on which the agent device is registered.

- Protocol Type: **UDP** (Best Practice) or, optionally **TCP**

Refer to "Configuring the SIP Devices (page 355)," and to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

b. Configure Access Control Lists for Unified CVP calls.

- Select **Proxy Settings > IncomingACL** (in CUP 7.0, **System > Security > IncomingACL**)
- Set address pattern: **all**

c. Configure the service parameters.

Select **Service Parameters**, and set the following:

- Add record route: **off**
- Maximum invite retransmission count: **2**
- Proxy Domain and Cluster Name: if using DNS SRV, set to the FQDN of your Proxy Server SRV name.

d. Write down the IP address and host name of the SIP Proxy Server. You will need this information when configuring the SIP Proxy Server in Unified CVP.

Note: You must use the SIP Proxy components of CUP Server, not the Presence Engine component.

e. If using redundant SIP Proxy Servers (primary and secondary or load balancing), then decide whether to use DNS server lookups for SRV records or non-DNS based local SRV record configuration.

Note: If a single CUP Server is used, then SRV record usage is not required.

The Comprehensive call flow model with SIP calls will typically be deployed with dual CUP Servers for redundancy. In some cases, you might want to purchase a second CUP Server. Regardless, the default transport for deployment will be UDP, and make sure you *always* set the AddRecordRoute setting to **Off** with CUP Servers.

Configure the SRV records on the DNS server or locally on Unified CVP with a .xml file (local xml configuration avoids the overhead of DNS lookups with each call).

For the required settings in the Unified CM Publisher configuration, refer to **Cisco Unified Presence Server documentation** (http://www.cisco.com/en/US/products/ps6837/tsd_products_support_series_home.html).

Step 23 Define Network VRUs.

a. On Unified ICME or the NAM, ICM Configuration Manager, **Network VRU Explorer** tool, define a Network VRU for the VRU leg and labels for each Unified CVP Call Server.

b. On the *CICM only*, ICM Configuration Manager, Network VRU Explorer tool, define a Network VRU for the VRU leg and labels for reaching the NAM.

For each of the two previous substeps, specify the following:

- Type: **10**
- Name: *<Network VRU Name>*

  For example: **cvp**
- Define a label for each Unified CVP Call Server that is handling the Switch leg:
  - Label: *<Network Routing Number>*
  - Type: **Normal**
  - Routing client for Unified ICME or the NAM: Select the routing client configured for that Unified CVP Call Server peripheral from the drop-down list.
  - Routing client for *CICM only*: Select the INCRP routing client from the drop-down list.

Note: The Network VRU label in the NAM and CICM must be identical. The Network VRU Names on the NAM and CICM must also be identical to avoid confusion.

Step 24 Define network VRUs and PGs for the switch leg in the ICM Configuration Manager.

On Unified ICMH, on the NAM and CICMs, Network VRU Explorer tool, define one label per Unified CVP Call Server or NIC routing client.

Note: Use the same Type 10 Network VRU that you defined in the previous steps for the VRU leg.

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 25 Set the client type for the INCRP NIC.

On the **CICM**, ICM Configuration Manager, NIC Explorer tool, set the client type for the INCRP NIC.

- Client Type: **VRU**

Step 26 Define a VRU that uses INCRP.

On the **CICM**, ICM Configuration Manager, Network VRU Explorer tool:

a. Define a Network VRU with a label that uses INCRP as its routing client.

Specify the following:

- Type: **10**
- Name: *<name of Unified CVP VRU>*

  For example: **cvpVRU**

b. Define one label for the NAM routing client.

Specify the following:

- Type: **Normal**
- Label: *<Network Routing Number>*
- Routing client: **INCRP NIC**

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 27 Configure Peripheral Gateways (PGs).

On the NAM, ICM Configuration Manager, **PG Explorer** tool, configure a peripheral gateway (PG) to be used for Unified CVP. Configure a PG for each Unified CVP Call Server as follows:

In the tree view pane, select the applicable PG.

Logical Controller tab:

- Client Type: **VRU**
- Name: A name descriptive of this PG.

  For example: **<location>_A** for side A of a particular location.

Peripheral tab:

- Peripheral Name: Descriptive name of this Unified CVP peripheral

  For example: **<location>_<cvp1> or <dns_name>**
- Client Type: **VRU**
- Select: **Enable Post-routing**

Advanced tab:

- Select the name of the Unified CVP VRU from the Network VRU field drop-down list.

For example: **cvpVRU**

Routing Client tab:

- Name: By convention, use the same name as the peripheral.
- Client Type: **VRU**
- If you are in a Unified ICMH environment and configuring the CICM, then do the following:
  - *Do not* select the **Network Transfer Preferred** checkbox.
  - Routing client: **INCRP NIC**

Step 28 Define a default network VRU.

On Unified ICME or the NAM, in the ICM Configuration Manager, the **System Information** tool:

a. For Unified ICME or on the **CICM only**, define a default Network VRU.

- Define the Default Network VRU: *<Network VRU Name>*

  For example: **cvpVRU**

b. If there are Routing Scripts on the **NAM**, define a default Network VRU.

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 29 Configure dialed numbers, call types, and customers.

On the Unified ICME or Unified ICMH Server in the ICM Configuration Manager:

a. **Dialed Number List Tool tab:** Configure the dialed numbers.

b. **Call Type List tool tab:** Configure the call types.

c. **ICM Instance Explorer tool tab:** Configure the applicable customers.

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 30 Configure ECC variables.

On Unified ICME, ICM Configuration Manager, configure ECC variables.

For more information, refer to "Common Unified ICMH Configuration: Define Unified CVP ECC Variables (page 131)."

Step 31 Create a routing script that handles the incoming calls.

On the Unified ICME or Unified ICMH Server in the ICM Script Editor tool, use the Send To VRU node to connect the call to the Network VRU.

For more information about creating scripts, refer to "Writing Scripts for Unified CVP (page 141)."

Refer to **Scripting and Media Routing Guide for Cisco Unified ICM/Contact Center Enterprise & Hosted** (http://www.cisco.com/en/US/products/sw/custcosw/ps1844/products_user_guide_list.html) for more information.

Step 32 Optionally, configure the **SIP Proxy**.

If using a SIP Proxy Server, configure it in the Unified CVP Operations Console.

Select: **Device Management > SIP Proxy Server**

For detailed information about configuring a SIP Proxy Server, refer to "Configuring a SIP Proxy Server (page 356)."

Step 33 Install and configure the **Call Server(s)**.

In the Operations Console:

a. Enable the ICM and SIP Services on the Call Server.

- In the Operations Console select **Device Management > Unified CVP Call Server**.

  Click **Help > This Page** for details.

- Select the check boxes: **ICM** and **SIP**

b. In the Operations Console select **Device Management > Unified CVP Call Server** > **SIP**. Configure the SIP Service:

- If you are using a SIP Proxy Server, enable the Outbound Proxy and select the SIP Proxy Server.

  Select the **SIP tab** and configure the following:

  – Enable Outbound Proxy: **Yes**

  – Outbound Proxy Host: Select from drop-down list.

  – Configure Local Static Routes on the SIP Proxy Server itself.

- If you are **not** using a SIP Proxy Server, configure Local Static Routes using the Dialed Number Pattern system configuration on the Operations Console. A Local Static Route must be configured for each SIP gateway/ACD, SIP endpoint in order to receive calls.

  Local Static Routes, Dialed Number (DN): Specify the dialed number pattern for the destination.

  Valid number patterns include the following characters:

  – Use the period (**.**) or **X** character for single-digit wildcard matching in any position.

  – Use the greater than (**>**), asterisk (*****), or exclamation (**!**) characters as a wildcard for 0 or more digits at the end of the DN.

  – Do *not* use the **T** character for wildcard matching.

  – Dialed numbers must not be longer than 24 characters.

  – Refer to *Valid Formats for Dialed Numbers* in the Operations Console online help for format and precedence information.

  Example: **9>** (Errors are 9292 and ringtone is 9191)

  Refer to "Configuring the SIP Devices (page 355)," and to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

  The following static route configuration is *not* correct because the least explicit routes must appear at the end. Load balancing and/or failover of calls will require DNS SRV domain names, not multiple routes with the same DN Pattern, but a single route to an SRV domain name.

  Incorrect Example:

```
1>,10.2.6.1
2>,10.2.6.2
3>,10.2.6.20
2229191>,10.2.6.241
2229292>,10.2.6.241
2229191>,10.2.6.242
2229292>,10.2.6.242
2>,ccm-subscribers.cisco.com
3>,ccm-subscribers.cisco.com
```

  Correct static route configuration example:

```
22291>,cvp-ringtone.cisco.com
22292>,cvp-error.cisco.com
1>,ccm-subscribers.cisco.com
2>,ccm-subscribers.cisco.com
3>,ccm-subscribers.cisco.com
```

  Note: "91919191>" pattern will not match an exact DN of "91919191."

- Check the default values for the SIP Service and change, if desired.

c. Configure the ICM Service by setting the maximum length DNIS to the length of the Network Routing Number.

Select **Device Management > CVP Call Server > ICM tab**: Maximum Length of DNIS: length of the Network Routing Number.

Example: if the Gateway dial pattern is 1800******, the maximum DNIS length is **10**.

For detailed information, click **Help > This Page**.

Step 34 Configure Local Static Routes:

If an outbound proxy is enabled on the Operations Console, configure local static routes on the SIP Proxy Server.

If no outbound proxy is enabled, configure local static routes using the Operations Console Dialed Number Pattern system configuration. Refer to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

The following is an example of a local static route configuration. A local static route contains a dialed number pattern and a routing address (IP Address, Hostname, or SIP Server Group name):

- **`22291>,cvp-ringtone.cisco.com`**
- **`22292>,cvp-error.cisco.com`**
- 1>,ccm-subscribers.cisco.com
- 2>,ccm-subscribers.cisco.com
- 3>,ccm-subscribers.cisco.com

Step 35 Configure custom ringtone patterns.

In the Operations Console, configure the custom ringtone patterns to play different ring tones based on the dialed number destination.

Select **> System > Dialed Number Pattern** and complete the following steps:

a. Click **Add New**. The Add New Dialed Number Pattern configuration page displays.

b. Fill in the appropriate configuration settings:

Table 1: Dialed Number Pattern Configuration Settings

Property	Description	Default	Value	
General Configuration				

Property	Description	Default	Value
Dialed Number Pattern	The actual Dialed Number Pattern.	None	Must be unique Maximum length of 24 characters Can contain alphanumeric characters, wildcard characters such as exclamation point (!) or asterisk (*),single digit matches such as the letter X or period (.) Can end with an optional greater than (>) wildcard character
Description	Information about the Dialed Number Pattern.	None	Maximum length of 1024 characters
Enable Local Static Route	Enable local static routes on this Dialed Number Pattern. If Local Static Routes are enabled: • **Route to Device -** Select the device from the drop down list which contains a list of configured, supported devices. Once a selection is made, the IP Address/Hostname/Server Group Name field is automatically updated with the IP Address of the selected device.	Disabled	Maximum length of 128 characters Must be a valid IP address, hostname, or fully qualified

Property	Description	Default	Value
	• **Route to SIP Server Group -** Select the device from the drop down list which contains a list of configured, support devices. Once a selection is made, the IP Address/Hostname/Server Group Name field is automatically updated with the IP Address of the selected device. • **IP Address/Hostname/Server Group Name -** If you have not selected a **Route to Device** or **Route to SIP Server Group**, enter the IP address, hostname, or the server group name of the route.		domain name
Enable Send Calls to Originator	Enables calls to be sent to originator.	Disabled	n/a
Enable RNA Timeout for Outbound Calls	Enables Ring No Answer (RNA) timer for outbound calls. • **Timeout -** Enter the timeout value in seconds.	Disabled none	n/a Valid integer in the inclusive range from 5 to 200.
Enable Custom Ringtone	Enables customized ring tone. • **Ringtone media filename -** Enter the name of the file that contains the ringtone.	Disabled none	Maximum length of 256 characters Cannot contain whitespace characters
Enable Post Call Survey for Incoming Calls	Enables post call survey for incoming calls. • **Survey Dialed Number Pattern -** Enter the survey dialed number pattern.	Disabled none	n/a Maximum length of 24 characters Can contain alphanumeric characters, wildcard characters such as

Property	Description	Default	Value
			exclamation point (!) or asterisk (*), single digit matches such as period (.) or X, and can end with an optional greater than (>)wildcard character

c. Click **Save** to save the Dialed Number Pattern.

You are returned to the **Dialed Number Pattern** page. To deploy the Dialed Number Pattern configuration, click **Deploy** to deploy the configuration to all Unified CVP Call Server devices.

Step 36 (Optionally) Configure the Reporting Server and associate it with a Call Server.

In the Operations Console, select **Device Management > CVP Reporting Server > General tab** and complete the following steps:

a. Configure the Reporting Server.

b. Select a Call Server to associate with this Reporting Server.

c. Check the default values of the Reporting properties and change, if desired.

For more information, refer to **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

Configuration Instructions for the Comprehensive Call Flow Model Using H.323 for Both ICME and ICMH

The following task contains the configuration instructions for the Comprehensive Call Flow Model using *H.323* for both Unified ICME and Unified ICMH.

Note: Refer to "Unified CVP Comprehensive (SIP/H.323) Call Flow Model, Unified ICMH (page 64)" for detailed information about the Unified ICMH version of this call flow model.

High-level Configuration Steps, Unified CVP Comprehensive Call Flow Model, Unified ICME and Unified ICMH Using H.323

Step 1 Install the IOS image on the Ingress Gateway.

For detailed information, refer to the **Cisco IOS documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html).

Step 2 Transfer the following script, configuration, and .wav files to the Ingress gateway using the Unified CVP Operations Console (or the CVP product CD):

- bootstrap.tcl
- handoff.tcl
- survivabilty.tcl
- bootstrap.vxml
- recovery.vxml
- cvperror.tcl
- critical_error.wav

In the Operations Console, select **Bulk Administration > File Transfer > Scripts and Media** and select **Help > This Page** for details.

Step 3 Configure the Ingress Gateway base settings.

Refer to "Ingress Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model (page 97)."

Step 4 Configure the Ingress Gateway service settings.

Refer to "Ingress Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model (page 97)."

Step 5 Configure an Ingress Gateway incoming Pots dial-peer.

Refer to "Ingress Gateway Configuration: Example of Incoming Pots Dial-peer for Comprehensive Call Flow Model (page 99)."

Step 6 Configure the gatekeeper.

Refer to "Ingress Gateway Configuration: Example of Using Gatekeeper Lookup to Reach the Call Server for Comprehensive Call Flow Model (page 100).

Refer to "Configuring the H.323 Devices and VoIP (page 465)," for detailed information.

Step 7 Transfer files to the **VXML** Gateway.

Transfer the following scripts, configuration, and .wav files to the gateway.

- bootstrap.tcl
- handoff.tcl
- survivabilty.tcl
- bootstrap.vxml
- recovery.vxml
- cvperror.tcl
- critical_error.wav

In the Operations Console, select **Bulk Administration > File Transfer > Scripts and Media** and select **Help > This Page** for details.

Note:

- If you have installed a Unified CVP Call Server, you can also copy gateway scripts from the **<cvp root>\Cisco\CVP\OpsConsoleServer \GWDownloads** folder on the Call Server.
- Optionally, you can transfer the files from the Unified CVP product CD.

Step 8 Configure the VXML Gateway base settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Setting for Comprehensive Call Flow Model (page 100)."

Step 9 Configure the VXML Gateway service settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model (page 100)."

Step 10 Configure the ICM VRU Label.

Refer to "VoiceXML Gateway Configuration: Example of ICM VRU Label for Comprehensive Call Flow Model (page 102)."

Refer to "Configuring the H.323 Devices and VoIP (page 465)" for detailed information.

Step 11 Configure the **Gatekeeper**.

a. Register all Gateways. Add the **boldfaced** lines to the Gateway configuration. The Gateway auto-registers with the Gatekeeper once these lines are configured.

```
interface FastEthernet0/0
ip address 10.86.129.79 255.255.255.0
```

```
ip route-cache
same-interface
duplex full
speed 100
no cdp enable
h323-gateway voip interface
h323-gateway voip id <zone ID specified on the GK> ipaddr <IP of GK>
1719
h323-gateway voip h323-id <IP of this GW>
h323-gateway voip tech-prefix 1#
```

b. Configure the Gatekeeper to send all VRU connections to the applicable Gateway. Include the following:

```
zone prefix <GK> <Network Routing Number>* gw-priority 10
 <H323-ID of gateway>
```

For example:

```
zone prefix gk-stooge 8001112222* gw-priority 10 vru1@provider.com
vru2@provider.com
```

c. Configure the Gatekeeper to send all agent device target labels to the IP address of the applicable Unified CM trunks.

```
zone prefix sox-gk 1* gw-priority 10 <CCM Trunk ID>
zone prefix sox-gk 2* gw-priority 10 <CCM Trunk ID>
zone prefix sox-gk 4* gw-priority 10 <CCM Trunk ID>
zone prefix sox-gk 777* gw-priority <VXML GW1 IP><VXML GW2 IP>
```

For example:

```
zone prefix sox-gk 1* gw-priority 10 SOX_CCM_H323_TRUNK_2 SOX_CCM_H323_TRUNK_3
zone prefix sox-gk 2* gw-priority 10 SOX_CCM_H323_TRUNK_2 SOX_CCM_H323_TRUNK_3
zone prefix sox-gk 4* gw-priority 10 SOX_CCM_H323_TRUNK_2 SOX_CCM_H323_TRUNK_3
zone prefix sox-gk 777* gw-priority 10 10.86.129.106 10.86.129.20
```

Refer to "Configuring the H.323 Devices and VoIP (page 465)," for detailed information.

Step 12 Configure H.323-specific actions.

On the Unified CM server, CCMAdmin Publisher, configure **H.323-specific actions**.

From the Unified CM Administration page:

a. Add the Gatekeeper.

Select **Device > Gatekeeper > Add New** and add the Host Name/IP address of Gatekeeper.

b. Add the Unified CVP H.323 Service as an H.323 gateway.

Select **Device > Gateway > Add New** and add the following:

- Gateway Type: **H.323 Gateway**

- Device Name: **<Unified CVP Call Server IP address>**
- Description: **<Unified CVP Call Server IP address (or other identifying text)>**
- Device Pool: **<device pool>**
- Calling Party Selection.

  From the Calling Party Selection list, select: **Originator**
- Calling Party Presentation: **Allowed**

c. Add an H.323 Trunk for Unified CM to register with the Gatekeeper.

Select **Device > Trunk > Add New** and add the following:

- Trunk Type: **H.225 Trunk** (Gatekeeper controlled)
- Device Protocol: **H.225**
- Device Name: **<name for your trunk>**

  For example: **MY_CCM_TRUNK**
- Device Pool: **<device pool>**
- Gatekeeper Name.

  From the drop-down list, select the Gatekeeper to which this trunk will register.
- Terminal Type: **Gateway**
- Technology Prefix: **1#**

For detailed instructions about using Unified CM, refer to the Unified CM documentation.

Step 13 Define VRUs.

- On Unified ICME or the NAM, ICM Configuration Manager, **Network VRU Explorer** tool, define a Network VRU for the VRU leg and labels for each Unified CVP Call Server.
- On the *CICM only*, ICM Configuration Manager, Network VRU Explorer tool, define a Network VRU for the VRU leg and labels for reaching the NAM.

For each of the preceding bullet items, specify the following:

- Type: **10**
- Name: *<Network VRU Name>*.

  For example: **cvp**

- Define a label for each Unified CVP Call Server that is handling the Switch leg:
  - Label: *<Network Routing Number>*
  - Type: **Normal**
  - Routing client for Unified ICME or the NAM: Select the routing client configured for that Unified CVP Call Server peripheral from the drop-down list.
  - Routing client for *CICM only*: Select the INCRP routing client from the drop-down list.

Note: The Network VRU label in the NAM and CICM must be identical. The Network VRU Names on the NAM and CICM must also be identical to avoid confusion.

Step 14 Configure the Peripheral Gateways (PGs).

Note: If you are using the same Unified CVP Call Server for the Switch and VRU legs, skip this step because this configuration is already completed. If you are using the same PG but different Call Servers, you need to perform only the second part of the step.

- Configure each peripheral gateway (PG) to be used for the VRU Client leg.
- Configure a peripheral for each Unified CVP Call Server to be used for aVRU leg connected to each PG.

For each of the preceding bullet items, configure the following parameters for the PGs:

On Unified ICME or the NAM, ICM Configuration Manager, **PG Explorer** tool, in the tree view, select the applicable PG:

- **Logical Controller** tab:
  - Client Type: **VRU**
  - Name: A name descriptive of this PG

    For example: **<location>_A** for side A of a particular location
- **Peripheral** tab:
  - Peripheral Name: A name descriptive of this Unified CVP peripheral

    For example: **<location>_<cvp1> or <dns_name>**
  - Client Type: **VRU**
  - Select the checkbox: **Enable Post-routing**
- **Advanced** tab:
  - Select the name of the Unified CVP VRU from the Network VRU field drop-down list.

For example: **cvpVRU**

- **Routing Client** tab:
  - Name: By convention, the same name as the peripheral
  - Client Type: **VRU**

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 15 Configure the **Network VRUs and PGs for the switch leg**.

On Unified ICMH, on the NAM and CICMs, configure the Network VRUs and PGs for the switch leg.

In the ICM Configuration Manager, Network VRU Explorer tool, define one label per Unified CVP Call Server or NIC routing client.

Note: Use the same Type 10 Network VRU that you defined in the previous step.

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 16 Set the client type for the INCRP NIC.

On the **CICM**, ICM Configuration Manager, NIC Explorer tool, set the Client Type to: **VRU**.

Step 17 Define a network VRU.

a. Define a Network VRU with a label that uses INCRP as its routing client.

Specify the following:

- Type: **10**
- Name: *<name of Unified CVP VRU>*

  For example: **cvpVRU**

b. Define one label for the NAM routing client.

Specify the following:

- Type: **Normal**
- Label: *<Network Routing Number>*
- Routing client: **INCRP NIC**

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 18 Configure Peripheral Gateways (PGs) on the NAM for each ICM Service.

a. Configure each PG to be used for the **Switch** leg.

b. Configure a peripheral for each ICM Service to be used for a Switch leg connected to each PG.

For each of the above substeps, on the **NAM**, ICM Configuration Manager, **PG Explorer** tool, in the tree view pane, select the applicable PG and configure the following parameters:

Logical Controller tab:

- Client Type: **VRU**
- Name: A name descriptive of this PG.

  For example: **`<location>_A`**, for side A of a particular location.

Peripheral tab:

- Peripheral Name: A name descriptive of this Unified CVP peripheral.

  For example: **`<location>_<cvp1>`** or **`<dns_name>`**
- Client Type: **VRU**
- Select the checkbox: **Enable Post-routing**

Advanced tab:

- Select the name of the Unified CVP VRU from the Network VRU field drop-down list.

  For example: **cvpVRU**

Routing Client tab:

- Name: By convention, use the same name as the peripheral.
- Client Type: **VRU**
- *Do not* select the **Network Transfer Preferred** checkbox.
- Routing client: **INCRP NIC**

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 19 Configure the Peripheral Gateways (PGs) for each CVP Call Server.

a. Configure each peripheral gateway (PG) to be used for the **Switch** leg.

b. Configure a peripheral for each Unified CVP Call Server to be used for a Switch leg connected to each PG.

Note: If you are using the same Unified CVP Call Server for the Switch and VRU legs, skip this step as this configuration is already completed. If you are using the same PG but different Call Servers, you need to perform only the second part of the step.

For the previous substeps, on Unified ICME, ICM Configuration Manager, **PG Explorer** tool, In the tree view pane, select the applicable PG and configure:

- **Logical Controller** tab:
  - Client Type: **VRU**
  - Name: A name descriptive of this PG.

    For example: **<location>_A** for side A of a particular location.
- **Peripheral** tab:
  - Peripheral Name: A name descriptive of this Unified CVP peripheral

    For example: **<location>_<cvp1> or <dns_name>**
  - Client Type: **VRU**
  - Select the checkbox: **Enable Post-routing**
- **Advanced** tab:
  - Select the name of the Unified CVP VRU from the Network VRU field drop-down list.

    For example: **cvpVRU**.
- **Routing Client** tab:
  - Name: By convention, use the same name as the peripheral.
  - Client Type: **VRU**

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 20 Define a network VRU.

a. For Unified ICME or on the **CICM only**, define a default Network VRU.

b. If there are Routing Scripts on the **NAM**, define a default Network VRU.

On Unified ICME or the **NAM**, ICM Configuration Manager, **System Information** tool, **General** section:

- Define the Default Network VRU: *<Network VRU Name>*

  For example: **cvpVRU**

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 21 Configure dialed numbers, call types, and customers.

On the Unified ICME or Unified ICMH Server in the ICM Configuration Manager:

a. **Dialed Number List Tool tab:** Configure the dialed numbers.

b. **Call Type List Tool tab:** Configure the call types.

c. **ICM Instance Explorer Tool tab:** Configure the applicable customers.

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 22 On Unified ICME, ICM Configuration Manager, configure the ECC variables.

For more information, refer to "Common Unified ICMH Configuration: Define Unified CVP ECC Variables (page 131)."

Step 23 Create a routing script that handles the incoming call.

On the Unified ICME or Unified ICMH Server in the ICM Script Editor tool, use the Send To VRU node to connect the call to the Network VRU.

For more information about creating scripts, refer to "Writing Scripts for Unified CVP (page 141)."

Refer to *ICM Scripting and Media Routing Guide for Cisco ICM/IPCC Enterprise & Hosted Editions* for more information.

Step 24 Enable the ICM, IVR, and H.323 Services on the Call Server.

In the Operations Console:

- Select: **Device Management > CVP Call Server**
- Select the check boxes: **ICM**, **IVR**, and **H.323**

Step 25 Configure the **ICM Service**.

In the Operations Console, select **Device Management > CVP Call Server > ICM tab** and specify the information for the following components:

a. VRU Connection Port.

Set the VRU Connection Port to match the VRU connection Port defined in ICM Setup for the corresponding VRU peripheral gateway (PIM). Configure the ports for both groups according to the licenses purchased, call profiles, and capacity by completing the required fields on this tab.

b. Maximum Length of DNIS.

Set the maximum length DNIS to the length of the Network Routing Number.

For example: if the Gateway dial pattern is 1800******, the maximum DNIS length is **10**.

c. Call service IDs: New Call and Pre-routed.

Check the default values and change, if desired.

d. Trunk group IDs: New Call and Pre-routed.

Check the default values and change, if desired.

Refer to the Operations Console online help for information about other settings you might want to adjust from their default values.

Step 26 Configure the **IVR Service**.

In the Operations Console, select **Device Management > CVP Call Server > IVR tab** and check the default values. Change, if desired.

Refer to the Operations Console online help for information about other settings you might want to adjust from their default values.

Step 27 Configure the **H.323 Service**.

In the VBAdmin tool, select **Cisco Unified Customer Voice Portal H.323 Service** and configure the following items:

a. Set the IP address of the Gatekeeper.

Specify: **SetGateKeeper**<*GK IP Address*>

Where <*GK IP Address*> is the IP address for the Gatekeeper serving the H.323 Service.

b. Configure the capacity.

For Unified ICMH only, configure the total number of calls and IVR ports according to the licenses purchased, call profiles, and capacity.

Refer to the Operations Console online help for information about other settings you might want to adjust from their default values.

c. Restart the H.323 Service.

1. Select: **Start > Programs > Cisco Unified Customer Voice Portal > H.323 Service**
2. Click **Shutdown**.
3. Click **Start**.

Verify that the H.323 Service can communicate with the Gatekeeper.

Step 28 (Optionally) Configure the **Reporting Server**.

In the Operations Console, select **Device Management > CVP Reporting Server > General tab**:

a. Configure the Reporting Server.

b. Select a Call Server to associate with this Reporting Server.

c. Check the default values of the Reporting properties and change, if desired.

For more information, refer to **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

Ingress Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model

The first part of the following example provides the basic configuration for setting an Ingress gateway:

- Applies a timestamp to debugging and log messages

- Turns on logging
- Turns off printing to the command line interface console
- Sends RTP packets
- Configures gateway settings

The last part of this example provides the following:

- Allows SIP to play a .wav file that enables caller to hear message from critical_error.wav
- Performs survivability
- Enables SIP to play ringtone to caller while caller is being transferred to an agent
- Logs errors on the gateway when the call fails
- Defines requirements for SIP Call Server

Note: CVP solution does not support 100rel. It can be disabled on the dial peer level or on a global level under the voice service VoIP section.

```
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
!
service internal
logging buffered 99999999 debugging
no logging console
!
ip cef
!
voice rtp send-recv
!
voice service voip
  signaling forward unconditional
  h323
  sip
    min-se 360
    header-passing
!
voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8
!
application
  service cvperror flash:cvperror.tcl
!
  service cvp-survivability flash:survivability.tcl
!
  service ringtone flash:ringtone.tcl
!
  service handoff flash:handoff.tcl
```

```
!
gateway
  timer receive-rtcp 4
!
ip rtcp report interval 2000
!
sip-ua
  retry invite 2
  timers expires 60000
sip-server ipv4:<IP of CUP server or Call Server>:5060
reason-header override
!
```

Ingress Gateway Configuration: Example of Incoming Pots Dial-peer for Comprehensive Call Flow Model

The following example provides the configuration for an incoming Pots call for the Comprehensive call flow model:

```
dial-peer voice 8 pots
  description Example incoming POTS dial-peer
  service cvp-survivability
  incoming called-number <your DN pattern here>
  direct-inward-dial
!
```

Ingress Gateway Configuration: Example of SIP Ringtone Dial-peer for Comprehensive Call Flow Model

The following example provides the configuration for a SIP ringtone for the Comprehensive call flow model:

```
dial-peer voice 9191 voip
  description SIP ringtone dial-peer
  service ringtone
  voice-class codec 1
  voice-class sip rel1xx disable
  incoming called-number <your ringtone DN pattern here>
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

The following example provides the gateway configuration for setting ringtone time-out to the caller if the network connection between ingress gateway and CVP Call Server fails. The default value for zombie-timeout for ringback tone to the caller is 120 seconds. The following configuration sets the ringtone timeout to 60 seconds.

```
service ringtone flash:ringtone.tcl
    param zombie-timeout 60

ip rtcp report interval 2000

gateway
    time receive-rtcp 4
```

Ingress Gateway Configuration: Example of SIP Error Dial-peer for Comprehensive Call Flow Model

The following example provides the configuration for a SIP error dial-peer for the Comprehensive call flow model:

```
dial-peer voice 9292 voip
  description SIP error dial-peer
  service cvperror
  voice-class codec 1
  voice-class sip rel1xx disable
  incoming called-number <your error DN pattern here>
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

Ingress Gateway Configuration: Example of Dial-peer to Reach the Unified CVP Call Server or CUP Server for Comprehensive Call Flow Model

The following example provides the configuration for a dial-peer to reach the Unfied CVP Call Server or CUP Server for the Comprehensive call flow model:

```
dial-peer voice 800 voip
  description Example Call Server Dialpeer with CUP Server
  destination-pattern <your DN pattern here>
  voice-class codec 1
  session protocol sipv2
  session target sip-server
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

Ingress Gateway Configuration: Example of Using Gatekeeper Lookup to Reach the Call Server for Comprehensive Call Flow Model

The following example provides the configuration to use gatekeeper lookup for the Comprehensive call flow model:

```
dial-peer voice 800 voip
  description Example Call Server Dialpeer with GK
  destination-pattern <your DN pattern here>
  voice-class codec 1
  session target ras
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

VoiceXML Gateway Configuration: Example Gateway Settings for Comprehensive Call Flow Model with Gatekeeper

The first part of the following example provides the basic configuration for setting a VoiceXML gateway:

- Applies a timestamp to debugging and log messages

- Turns on logging
- Turns off printing to the command line interface console
- Sends RTP packets
- Configures ASR/TTS Server
- Configures gateway settings

The last part of this example provides the following:

- Initiates the VoiceXML leg
- Initiates the switch leg of the call
- Plays a .wav file that enables caller to hear message from critical_error.wav
- Logs errors on the gateway when the call fails

```
service timestamps debug datetime msec
service timestamps log datetime msec

service internal

logging buffered 99999999 debugging
no logging console
ip cef
no ip domain lookup
ip host tts-en-us <IP of TTS or MRCP Server>
ip host asr-en-us <IP of ASR or MRCP Server>

voice rtp send-recv
!

voice service voip
  signaling forward unconditional
  h323
  sip
    min-se 360
    header-passing

  voice class codec 1
    codec preference 1 g711ulaw
      codec preference 2 g729r8
!
!
voice translation-profile block
translate called 1
!
!
ivr prompt memory 15000
ivr prompt streamed none
ivr asr-server rtsp://asr-en-us/recognizer
```

```
ivr tts-server rtsp://tts-en-us/synthesizer

mrcp client timeout connect 10
mrcp client timeout message 10
mrcp client rtpsetup enable
rtsp client timeout connect 10
rtsp client timeout message 10
vxml tree memory 500
http client cache memory pool 15000
http client cache memory file 500
http client connection timeout 60
http client response timeout 30
http client connection idle timeout 10

gateway
  timer receive-rtcp 6
!
ip rtcp report interval 3000
application
  service new-call flash:bootstrap.vxml

  service cvperror flash:cvperror.tcl

  service handoff flash:handoff.tcl

  service bootstrap flash:bootstrap.tcl
    param cvpserverss1 1
    !
```

Note: The optional **`param cvpserverss1 1`** line enables HTTPS.

VoiceXML Gateway Configuration: Example of Dial-peer for ICM VRU Label for Comprehensive Call Flow Model

The following example provides the configuration for an ICM VRU label dial-peer for the Comprehensive call flow model:

```
dial-peer voice 777 voip
  description ICM VRU label
  service bootstrap
  codec g711ulaw
  incoming called-number <your sendtovru label pattern here>
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

DNS Zone File Configuration for Comprehensive Call Flow Model

DNS Zone File Linux NAMED Configuration Example

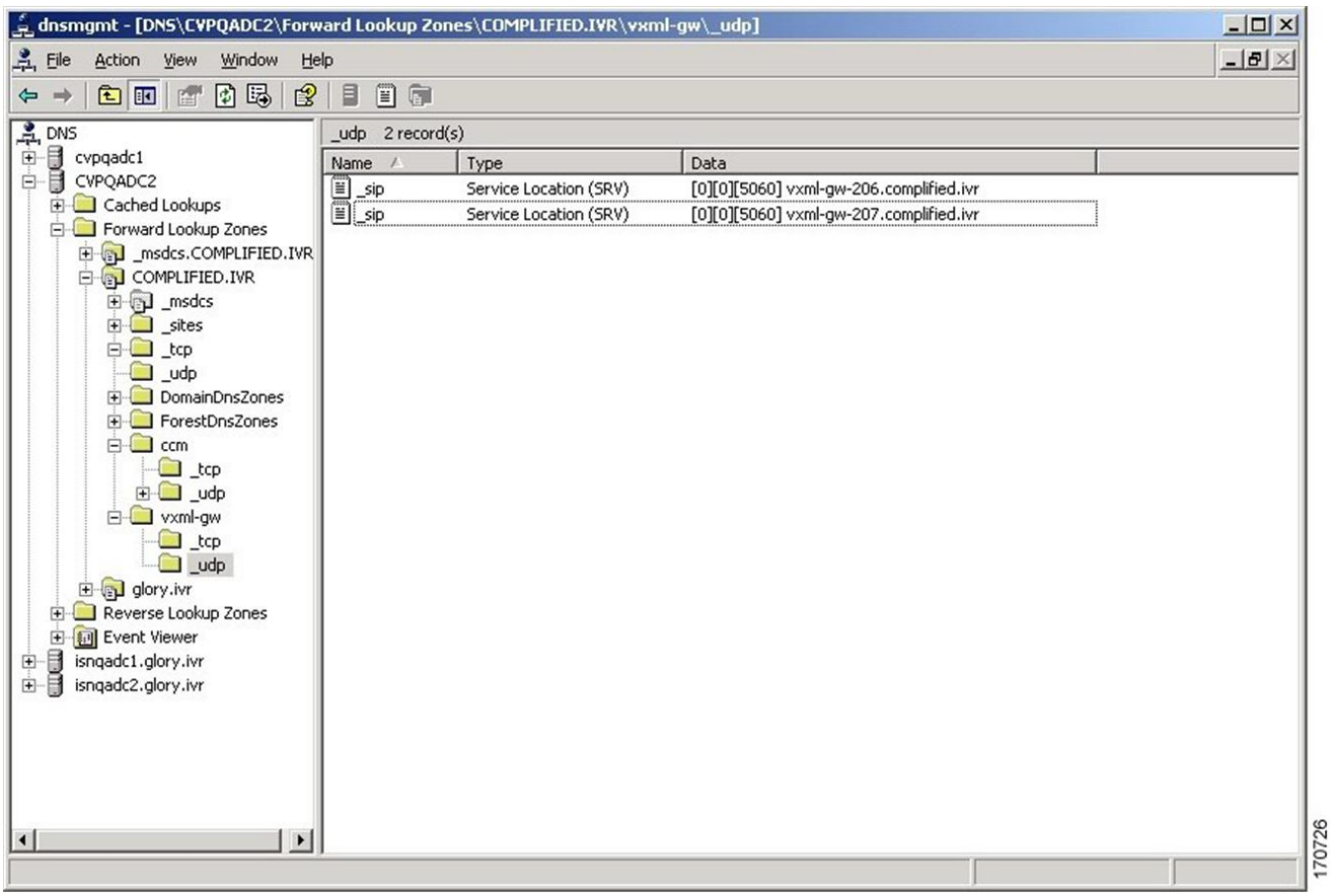
The following is an example of a DNS zone file Linux NAMED configuration.

```
ringtone-1 IN A 10.86.129.20
ringtone-2 IN A 10.86.129.229
vxml-1 IN A 10.86.129.20
vxml-2 IN A 10.86.129.229
vxml-3 IN A 161.44.81.254
cvp-1 IN A 10.86.129.211
cvp-2 IN A 10.86.129.220
cvp-3 IN A 161.44.81.254
; Priority Weight Port Target
sip._tcp.ringtone.sox.cisco.com. SRV 1 1 5060 ringtone-1.sox.cisco.com.
_
SRV 1 1 5060 ringtone-2.sox.cisco.com.
sip._udp.ringtone.sox.cisco.com. SRV 1 1 5060 ringtone-1.sox.cisco.com.
_
SRV 1 1 5060 ringtone-2.sox.cisco.com.
_sip._tcp.vxml.sox.cisco.com. SRV 1 1 5060 vxml-1.sox.cisco.com.
SRV 1 1 5060 vxml-2.sox.cisco.com.
SRV 1 1 5060 vxml-3.sox.cisco.com.
_sip._udp.vxml.sox.cisco.com. SRV 2 1 5060 vxml-1.sox.cisco.com.
SRV 2 1 5060 vxml-2.sox.cisco.com.
SRV 1 1 5060 vxml-3.sox.cisco.com.
_sip._tcp.cvp.sox.cisco.com. SRV 1 1 5060 cvp-1.sox.cisco.com.
SRV 2 1 5060 cvp-2.sox.cisco.com.
SRV 3 1 5060 cvp-3.sox.cisco.com.
_sip._udp.cvp.sox.cisco.com. SRV 1 1 5060 cvp-1.sox.cisco.com.
SRV 2 1 5060 cvp-2.sox.cisco.com.
SRV 3 1 5060 cvp-3.sox.cisco.com.
```

DNS Zone File MS DNS Configuration Example

The following is an example of a DNS zone file MS DNS configuration.

Figure 16: DNS Zone File MS DNS Configuration Example



Characteristics for the VRU Leg for Comprehensive Call Flow Model

Use the following commands to provide voice treatment:

Note: `new-call` is a required name.

Continue with the VRU Leg Example.

```
service vru-leg flash:bootstrap.tcl
  !
service new-call flash:bootstrap.vxml
  !
service handoff flash:handoff.tcl
  !
service ringtone flash:ringtone.tcl
  !
service cvperror flash:cvperror.tcl
  !
service cvp-survivability flash:survivability.tcl
  !
```

Unified CVP VRU Call Flow Models with NIC Routing

This section describes the call flows and provides configuration instructions for the following Unified CVP call flow models:

- Type 8 Unified CVP VRU-Only Call Flow Model, Unified ICME (page 105)
- Type 8 Unified CVP VRU-Only Call Flow Model, Unified ICMH (page 106)
- Type 3 or 7 Unified CVP VRU-Only Call Flow Model Network VRU, Unified ICMH (page 115)

Note: Be aware that in VRU-Only mode, Unified CVP by itself does not provide queuing capability; however, it can hold calls being queued when used with Unified ICME/Unified CCE with appropriate Unified ICME network interface controllers.

Type 8 Unified CVP VRU-Only Call Flow Model, Unified ICME

In this call flow model, Unified CVP works with the Voice Gateway to act as the VRU; VRU voice treatment is provided by the Gateway and can include ASR/TTS.

When deployed with a NIC being used to queue and transfer calls (VRU Type 8), the NIC interfaces to the TDM switch or to the PSTN to transfer the call to an agent.

Note: Neither the Unified CVP H.323 Service nor the Unified CVP SIP Service are part of this call flow model.

The Unified CVP VRU-Only call flow model requires the following components:

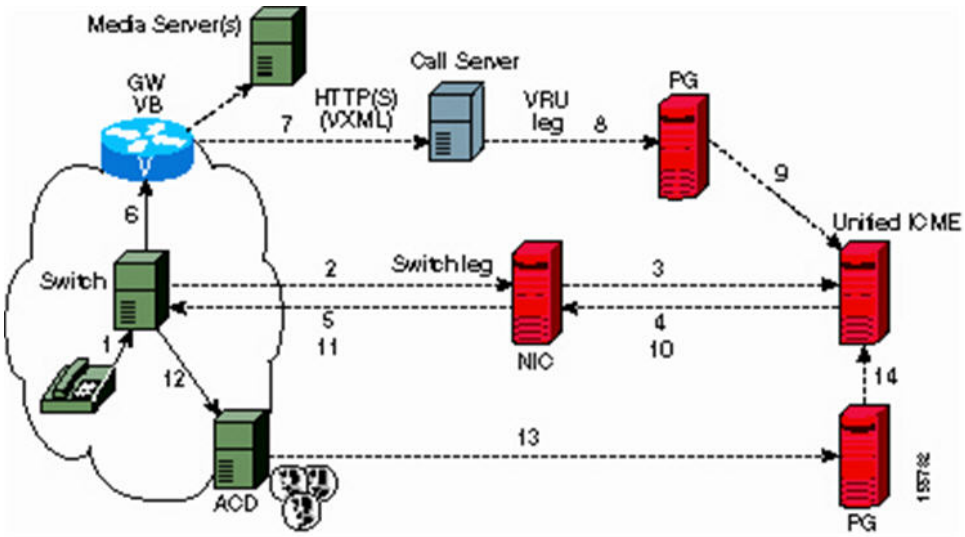
- Call Server with IVR and ICM Services enabled
- VoiceXML Gateway
- Operations Console
- Unified ICME

Note: The VXML Server and the Reporting Server are optional.

The following figure shows the call flow for the Type 8 call flow model; in this figure, the NIC transfers the call.

Note: In the following diagram, solid lines indicate voice paths and dashed lines indicate signaling paths.

Figure 17: Type 8 Unified CVP with Type 8 VRU-Only, Unified ICME



Note: The numbers in the figure represent call flow progression.

Configuration overview for this call flow model:

- There is one Network VRU: a Type 8 when NIC is queuing and transferring calls.
- Define a Translation Route and labels for the VRU Peripheral (Network VRU labels do not need to be configured.)
- Use the ICM Script Editor's TranslationRouteToVRU node to connect the call to the Network VRU.

Type 8 Unified CVP VRU-Only Call Flow Model, Unified ICMH

In this call flow model, the Unified CVP Call Server is deployed at the CICM level to act only as the VRU leg for the call. The VRU voice treatment is provided at the Voice Gateway, and might include ASR/TTS.

Note: This call flow model is used when Unified CVP is connected to the CICM. The routing client in this call flow model is connected to the NAM.

When deployed with a NIC being used to queue and transfer calls (VRU Type 8), the NIC interfaces to the TDM switch to transfer the call to an agent.

Note: Neither the Unified CVP H.323 Service nor the Unified CVP SIP Service are part of this call flow model.

The Unified CVP VRU-Only call flow model requires the following components:

- Call Server with IVR and ICM Services enabled
- VoiceXML Gateway

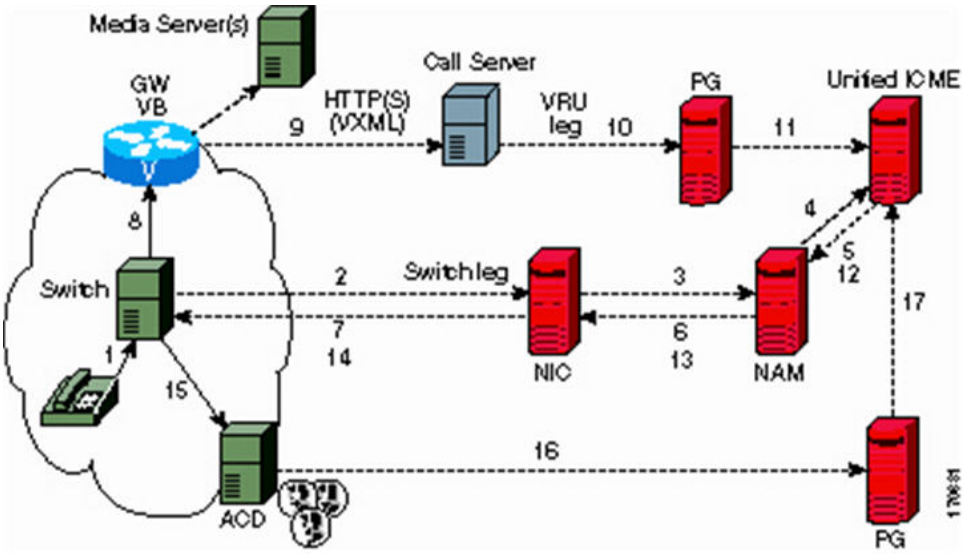
- Operations Console
- Unified ICME

Note: The VXML Server and the Reporting Server are optional.

The following figure shows the call flow for this call flow model.

Note: In the following diagram, solid lines indicate voice paths and dashed lines indicate signaling paths.

Figure 18: Type 8 Unified CVP with Type 8 VRU-Only, Unified ICMH



Note: For simplicity, the figure does not illustrate a callflow model for redundancy and failover.

Configuration overview for this call flow model:

- There are two Network VRUs configured:
  - One on the NAM (Type 8).
  - One on the CICM for the INCRP connection (Type 8).
- Use the ICM Script Editor's TranslationRouteToVRU node to connect the call to the Network VRU.

High-level Configuration for Type 8, Unified CVP VRU-Only, Unified ICME and Unified ICMH

The following task contains the configuration instructions for this call flow model for **both** ICME and ICMH.

Step 1 Using the Unified CVP Operations Console (or the Unified CVP product CD), transfer the following script, configuration, and .wav files to the **VoiceXML Gateway** used for the VRU leg.

In the Operations Console, select **Bulk Administration > File Transfer > Scripts and Media** and select **Help > This Page** for details.

Transfer the following files:

- bootstrap.tcl
- handoff.tcl
- survivabilty.tcl
- bootstrap.vxml
- recovery.vxml
- ringtone.tcl
- cvperror.tcl
- ringback.wav
- critical_error.wav

Step 2 Configure the VXML gateway base settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Type 8 Call Flow Model (page 114)."

Step 3 Configure the VXML gateway service settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Type 8 Call Flow Model for Type 8 Call Flow Model (page 114)."

Step 4 Configure the ICM VRU Label.

Refer to "VoiceXML Gateway Configuration: Example of ICM VRU Label for Type 8 Call Flow Model (page 115)."

Step 5 Define a Network VRU on Unified ICME or (for Unified ICMH) on the NAM and each CICM.

Using the ICM Configuration Manager, the **Network VRU Explorer** tool, specify the following:

- Type: **8**
- Name: **cvpVRU**

Note: Although any name will work, **cvpVRU** is used by convention, and is the example name referenced elsewhere in this document.

Step 6 Configure the Peripheral Gates (PGs) on Unified ICME or (for Unified ICMH) on each CICM.

a. Configure each PG.

b. Configure a peripheral for each Unified CVP ICM Service connected to each PG.

Use the ICM Configuration Manager, the **PG Explorer** tool. For each Unified CVP ICM Service connected to this PG, in the tree view pane, select the applicable PG and configure the following items:

Logical Controller tab:

- Client Type: **VRU**
- Name: A name descriptive of this PG

  Example: **`<location>_A`** for side A of a particular location.

Peripheral tab:

- Peripheral Name: A name descriptive of this Unified CVP peripheral

  Examples: **`<location>_<cvp1>`** or **`<dns_name>`**
- Client Type: **VRU**
- Select the checkbox: **Enable Post-routing**

Advanced tab:

- From the Network VRU field drop-down list, select the name: **cvpVRU**

Routing Client tab:

- Name: By convention, use the same name as the peripheral.
- Client Type: **VRU**

Step 7 Configure a Service and Route for each VRU on Unified ICME or (for Unified ICMH) on each CICM.

Note: You can also use service arrays. Refer to the Unified ICME documentation set for more information.

Using the ICM Configuration Manager, the **Service Explorer** tool, specify the following:

- Service Name: **cvpVRU**
- Route Name: **PeripheralName_cvpVRU**
- Peripheral Number: **2**

  Must match the "Pre-routed Call Service ID" in the Call Server configuration on the ICM tab in the Operations Console.

- Select the checkbox: **Enable Post-routing**

Step 8 Define trunk groups.

Note: You must configure one Network Transfer Group and one associated Trunk Group for each VRU leg Unified CVP ICM Service.

Define and configure the network trunk group on Unified ICME or (for Unified ICMH) on each CICM.

Using the ICM Configuration Manager, the Network **Trunk Group Explorer** tool, specify:

a. Identify the network trunk group.

- Network Trunk Group Name: A name descriptive of this trunk group.

b. For each Unified CVP ICM Service for theVRU leg, configure an associated trunk group.

- Peripheral Name: A name descriptive of this trunk group.
- Peripheral Number: **200**

  Must match the "Pre-routed Call Trunk Group ID" in the Call Server configuration on the ICM tab in the Operations Console.

- Trunk Count: Select **Use Trunk Data** from the drop-down list.
- *Do not* configure any trunks.

Step 9 Define translation route(s).

Define and configure a Translation Route for each VRU Peripheral on Unified ICME or (for Unified ICMH) on each CICM.

On Unified ICME, ICM Configuration Manager, **Translation Route Explorer** tool:

a. Define a Translation Route for each VRU Peripheral. Specify the following:

Translation Route tab:

- Set the **Name** field to the name of the target VRU peripheral. (This is by convention; this value must be unique in the enterprise.)

- Set the **Type** field to **DNIS** and select the Service defined in the previous step.

b. Configure translation route and label information for each VRU peripheral. Complete the following:

Route tab:

- Set the **Name**. By convention, this is the name of the target VRU peripheral, followed by the DNIS that this route will use, for example, MyVRU_2000.

  This value must be unique in the enterprise.

- Service Name drop-down, select: **PeripheralName.cvpVRU**

Peripheral Target tab:

- Enter the first DNIS that will be seen by the VRU that you will be using for this translation route.

  Note: The DNIS pool used for each VRU peripheral must be unique.

- From the drop-down list, select a **Network Trunk Group** which belongs to the target VRU.

Label tab:

- Enter the translation route label (which might or might not be the same DNIS you entered on the Peripheral Target tab).
- Type: **Normal**
- Routing Client: Select the NIC Routing Client.

You must create an additional label for each NIC routing client.

Note: Repeat the Route and corresponding Peripheral Target and Label information for each DNIS in the pool.

Step 10 Create VRU and routing scripts.

Create VRU scripts and routing scripts for IVR treatment and agent transfer on Unified ICME or (for Unified ICMH) on each CICM .

Using the ICM **Script Editor** tool, create the VRU scripts and routing scripts to be used for IVR treatment and agent transfer, as described in other sections of this manual and in the ICM manuals.

The VRU scripts are associated with the applicable Network VRU.

For example, **cvpVRU**

Use the ICM Script Editor's TranslationRouteToVRU node to connect the call to the Network VRU.

Step 11 Configure the ECC variables on Unified ICME or (for Unified ICMH) on the NAM and each CICM.

Using the ICM Configuration Manager, create the ECC variables.

For more information, refer to "Common Unified ICMH Configuration: Define Unified CVP ECC Variables (page 131)."

Step 12 Configure dialed numbers and call types on Unified ICME or (for Unified ICMH) on the NAM and each CICM.

On Unified ICME, using the ICM Configuration Manager, configure dialed numbers and call types.

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 13 On Unified CM configure Unified CM.

For more information, refer to the Unified CM user documentation.

Step 14 Install and configure the Call Server(s).

Using the Operations Console, select **Device Management > CVP Call Server** and install and configure the **Call Server(s)**.

Select the check boxes: **ICM** and **IVR**

For detailed information, refer to the Operations Console online help.

Step 15 Configure the ICM service.

Using the Operations Console, select **Device Management > CVP Call Server > ICM tab**. On **each** Unified CVP Call Server, configure the **ICM Service** by specifying the following required information:

a. VRU connection port number.

Set the VRU Connection Port to match the VRU connection Port defined in ICM Setup for the corresponding VRU peripheral gateway (PIM).

b. Maximum Length of DNIS.

Set the maximum length DNIS to a number which is at least the length of the translation route DNIS numbers.

Example: if the Gateway dial pattern is 1800******, the maximum DNIS length is 10.

c. Call service IDs: New Call and Pre-routed.

Enter the new and pre-routed call service IDs. Configure the ports for both groups according to the licenses purchased, call profiles, and capacity by completing the required fields on this tab

d. Trunk group IDs: New Call and Pre-routed.

- Enter the new and pre-routed call trunk group IDs.
- Configure the group number for the Pre-routed Call Trunk group. The group number must match the trunk group number in the Network Trunk group used for the translation route.
- Configure the number of ports according to the licenses purchased and capacity.
- Configure each of the numbers used for translation routes. (The "New Call" group is not used since the calls are being sent to the VRU (Unified CVP) after some initial processing by the NIC/Unified ICME.)

e. Dialed numbers used in the translation route.

Add the dialed numbers in the DNIS field.

f. Check the default values of the other settings and change, if desired.

Step 16 Configure the **IVR Service**.

In the Operations Console, select **Device Management > CVP Call Server > IVR tab**.

Check the default values and change, if desired.

Refer to the Operations Console online help for information about other settings you might want to adjust from their default values.

Step 17 (Optionally) Configure the Reporting Server.

In the Operations Console, select **Device Management > CVP Reporting Server > General tab**:

1. Configure the Reporting Server.
2. Select a Call Server to associate with this Reporting Server.
3. Check the default values of the Reporting properties and change, if desired.

For more information, refer to **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

VoiceXML Gateway Configuration: Example Gateway Settings for Type 8 Call Flow Model

The first part of the following example provides the basic configuration for setting a VoiceXML gateway:

- Applies a timestamp to debugging and log messages
- Turns on logging
- Turns off printing to the command line interface console
- Sends RTP packets
- Configures ASR/TTS Server
- Configures gateway settings

The last part of this example provides the following:

- Initiates the VoiceXML leg
- Plays a .wav file that enables caller to hear message from critical_error.wav
- Logs errors on the gateway when the call fails

```
service timestamps debug datetime msec
service timestamps log datetime msec

service internal

logging buffered 99999999 debugging
no logging console
ip cef
no ip domain lookup
ip host tts-en-us <IP of TTS or MRCP Server>
ip host asr-en-us <IP of ASR or MRCP Server>
voice rtp send-recv
!

voice service voip
  allow-connections h323 to h323
  signaling forward unconditional
  h323
  sip
    min-se 360
    header-passing
voice class codec 1
  codec preference 1 g711ulaw
    codec preference 2 g729r8
!
!
voice translation-profile block
translate called 1
```

```
!
!
ivr prompt memory 15000
ivr prompt streamed none
ivr asr-server rtsp://asr-en-us/recognizer
ivr tts-server rtsp://tts-en-us/synthesizer

mrcp client timeout connect 10
mrcp client timeout message 10
mrcp client rtpsetup enable
rtsp client timeout connect 10
rtsp client timeout message 10
vxml tree memory 500
http client cache memory file 500
http client connection timeout 60
http client response timeout 30
http client connection idle timeout 10

gateway
  timer receive-rtcp 6
!
ip rtcp report interval 3000
application
  service new-call flash:bootstrap.vxml

  service cvperror flash:cvperror.tcl

  service handoff flash:handoff.tcl

  service bootstrap flash:bootstrap.tcl
    !
```

VoiceXML Gateway Configuration: Example of Dial-peer for ICM VRU Label for Type 8 Call Flow Model

The following example provides the configuration for an ICM VRU label dial-peer for the Type 8 Unified CVP VRU-Only call flow model:

```
dial-peer voice 777 voip
  description ICM VRU label
  service bootstrap
  voice-class codec 1
  incoming called-number <your sendtovru label pattern here>
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

Type 3 or 7 Unified CVP VRU-Only Call Flow Model Network VRU, Unified ICMH

In this call flow model, Unified CVP is deployed as a Network VRU at the NAM. The Unified CVP IVR Service in the Operations Console works with the Voice Gateway to act as the VRU; VRU voice treatment is provided at the Voice Gateway and can include ASR/TTS. (This call flow model is used when Unified CVP is connected to the NAM.)

The NIC interfaces to the TDM switch to transfer calls to Unified CVP for VRU treatment and to queue and transfer calls using a VRU Type 3 or 7 call flow.

Note:

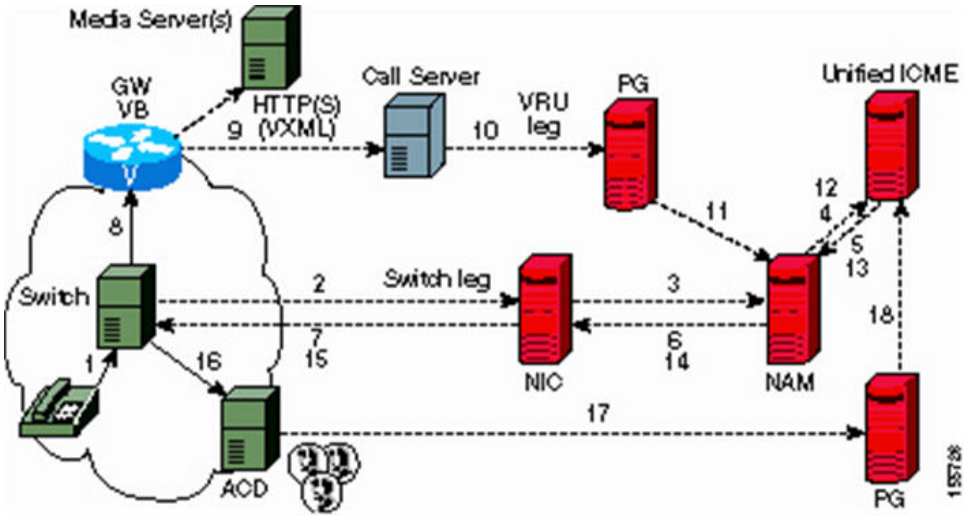
- Use this call flow model only if the PSTN to which the NIC is connected can transport a Correlation ID when it transfers a call. If this is not the setup you are using, then use the VRU-Only Type 8 Unified ICMH call flow model (page 106) instead.
- Neither the Unified CVP H.323 Service nor the Unified CVP SIP Service are part of this call flow model.

The following figure shows the call flow for this call flow model.

Note:

- In the following diagram, solid lines indicate voice paths and dashed lines indicate signaling paths.
- For simplicity, the figure does not illustrate a call flow model for redundancy and failover.

Figure 19: Unified CVP with Type 3 or 7 VRU-Only, Unified ICMH



Note: The numbers in the figure indicate call flow progression.

Configuration overview for this call flow model:

- Set the Network VRU Type to 3 or 7. There is no difference between these two types except that Type 7 causes Unified ICME to explicitly inform Unified CVP when it is about to transfer the call away from Unified CVP. (Most customers use Type 7.)
- The Network VRU names (where applicable), correlation IDs, and the ECC variable configurations must be identical on the NAM and CICM. All Labels must also be duplicated, but their routing clients will be different.
- Use the CICM Script Editor's SendToVRU node to connect the call to the Network VRU.

High-level Configuration Instructions Type 3 or 7 VRU-Only Call Flow Model Network VRU, Unified ICMH

The following table contains the high-level configuration instructions for this call flow.

Step 1 Transfer the following script, configuration, and .wav files to the **VoiceXML Gateway** used for the VRU leg, using the Unified CVP Operations Console:

In the Operations Console, select **Bulk Administration > File Transfer > Scripts and Media** and select **Help > This Page** for details.

- bootstrap.tcl
- handoff.tcl
- survivabilty.tcl
- bootstrap.vxml
- recovery.vxml
- ringtone.tcl
- cvperror.tcl
- ringback.wav
- critical_error.wav

Note: Optionally, you can transfer the files from the Unified CVP product CD.

Step 2 Configure the VoiceXML gateway base settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Type 3 or Type 7 (page 122)."

Step 3 Configure the VoiceXML gateway service settings.

Refer to "VoiceXML Gateway Configuration: Example Gateway Settings for Type 3 or Type 7 (page 122)."

Step 4 Configure the ICM VRU Label.

Refer to "VoiceXML Gateway Configuration: Example of ICM VRU Label for Type 3 or Type 7 (page 123)."

Step 5 Configure each PG.

On the **NAM**, ICM Configuration Manager, **PG Explorer** tool:

a. Configure each PG to be used for the **VRU Client** leg.

b. Configure a peripheral for each Unified CVP ICM Service to be used as a VRU leg connected to each PG.

For each Unified CVP ICM Service connected to this PG, in the tree view pane, select the applicable PG.

Logical Controller tab, configure:

- Client Type: **VRU**
- Name: A name descriptive of this PG

  For example: **`<location>_A`** for side A of a particular location

Peripheral tab, configure:

- Peripheral Name: A name descriptive of this VRU peripheral.

  For example: **`<location>_<cvp1>`** or **`<dns_name>`**
- Client Type: **VRU**
- Select the checkbox: **Enable Post-routing**

Routing Client tab:

- Name: By convention, use the same name as the peripheral.
- Client Type: **VRU**

Step 6 Define a Network VRU and labels.

On the **CICM**, ICM Configuration Manager, **Network VRU Explorer** tool, define a Network VRU for the VRU leg and labels for reaching the NAM.

Specify the following:

- Type: **3** or **7**
- Name: **cvpVRU**

  Note: This name is used by convention. Although any name will do, since it is referenced elsewhere in this document, **cvpVRU** is assumed.
- Define a **Label** for the NAM.
  - Label: Network routing number
  - Type: **Normal**

– Routing client: Select the INCRP Routing Client from the drop-down list.

Step 7 Define a Network VRU and a label for each NIC.

On the **NAM**, ICM Configuration Manager, **Network VRU Explorer** tool, define a Network VRU and a label for each NIC that is using this VRU.

Specify the following:

- Type: **3** or **7**
- Name: **cvpVRU**

  Note: This name is used by convention. Although any name will work, since it is referenced elsewhere in this document, **cvpVRU** is assumed.

- Define a **Label** for each NIC that is using this VRU:
  - Label: Network routing number
  - Type: **Normal**
  - Routing client: Select the Routing Client for that NIC from the drop-down list.

Note: Make sure the Network VRU label is identical in the NAM and CICM. The Network VRU Name must be identical as well to avoid confusion.

Step 8 If there will be Routing Scripts on the NAM, define a default Network VRU.

On the **NAM**, ICM Configuration Manager, **System Information** tool, in the General section:

- Define the Default Network VRU: **cvpVRU**

Step 9 Define a default VRU.

On the **CICM**, ICM Configuration Manager, **System Information** tool, in the General section:

- Define a default Network VRU: **cvpVRU**

Step 10 Create the VRU and routing scripts.

On the **CICM**, ICM **Script Editor** tool:

Create the VRU scripts and routing scripts to be used for IVR treatment and agent transfer, as described in other sections of this manual and in the Unified ICME manuals. The VRU scripts are associated with the applicable Network VRU, that is, **cvpVRU**.

Use the ICM Script Editor's SendToVRU node to connect the call to the Network VRU.

Note: A RunVRU Script or Queue node is an “implicit” SendToVRU node, although error handling will be easier if the explicit “SendToVRU” node is used.

Step 11 Configure the ECC variables.

On the **NAM** and **CICM**, ICM Configuration Manager, configure the ECC variables.

For more information, refer to "Common Unified ICMH Configuration: Define Unified CVP ECC Variables (page 131)."

Step 12 Configure dialed numbers and call types.

On the **NAM** and **CICM**, ICM Configuration Manager, configure dialed numbers and call types.

For more information, refer to **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html).

Step 13 Define customers.

On the **NAM** and **CICM**, ICM Configuration Manager:

a. If necessary, differentiate VRUs (Unified CVPs) based on dialed number.

b. Define customers and their Network VRU.

For more information, refer to "Common Configuration for Differentiating VRUs (Unified CVPs) Based on Dialed Number (page 138)."

Step 14 On Cisco Unified CM, configure Unified CM.

For more information, refer to the Unified CM user documentation.

Step 15 Install and configure the Call Server(s).

In the Operations Console, select **Device Management > CVP Call Server**.

Click **Help > This Page** for details.

a. Install and configure the **Call Server(s)**.

b. To enable the ICM and IVR Services on the Call Server, select the check boxes: **ICM** and **IVR**

Click **Help > This Page** for details.

Step 16 Configure the ICM Service for each Call Server.

In the Operations Console, select **Device Management > CVP Call Server > ICM tab**. For each Unified CVP Call Server, configure the **ICM Service** by specifying the following required information:

a. VRU connection port number.

Set the VRU Connection Port to match the VRU connection Port defined in ICM Setup for the corresponding VRU peripheral gateway (PIM).

b. Set the maximum length DNIS to the length of the Network Routing Number.

Example: if the Gateway dial pattern is 1800******, the maximum DNIS length is 10.

c. Call service IDs: New Call and Pre-routed.

Enter the new and pre-routed call service IDs. Configure the ports for both groups according to the licenses purchased, call profiles, and capacity by completing the required fields on this tab

d. Trunk group IDs: New Call and Pre-routed.

Enter the new and pre-routed call trunk group IDs. Configure the group number for the Pre-routed Call Trunk group. The group number must match the trunk group number in the Network Trunk group used for the translation route.

Configure the number of ports according to the licenses purchased and capacity. Configure each of the numbers used for translation routes. (The "New Call" group is not used since the calls are being sent to the VRU (Unified CVP) after some initial processing by the NIC/Unified ICME.)

e. Check the default values of other settings and change, if desired.

Step 17 Configure the IVR service.

In the Operations Console, select **Device Management > CVP Call Server > IVR tab** and configure the **IVR Service**.

Check the default values and change, if desired.

Refer to the Operations Console online help for information about other settings you might want to adjust from their default values.

Step 18 (Optionally) Configure the Reporting Server.

In the Operations Console, select **Device Management > CVP Reporting Server > General tab** and configure the Reporting Server.

a. Configure the Reporting Server.

b. Select a Call Server to associate with this Reporting Server.

c. Check the default values of the Reporting properties and change, if desired.

For more information, refer to **Reporting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_installation_and_configuration_guides_list.html).

VoiceXML Gateway Configuration: Example Gateway Settings for Type 3 or Type 7

The first part of the following example provides the basic configuration for setting a VoiceXML gateway:

- Applies a timestamp to debugging and log messages
- Turns on logging
- Turns off printing to the command line interface console
- Sends RTP packets
- Configures ASR/TTS Server
- Configures gateway settings

The last part of this example provides the following:

- Initiates the VoiceXML leg
- Plays a .wav file that enables caller to hear message from critical_error.wav
- Logs errors on the gateway when the call fails

```
service timestamps debug datetime msec
service timestamps log datetime msec

service internal

logging buffered 99999999 debugging
no logging console

ip cef
no ip domain lookup
ip host tts-en-us <IP of TTS or MRCP Server>
ip host asr-en-us <IP of ASR or MRCP Server>

voice rtp send-recv
!

voice service voip
  allow-connections h323 to h323
  signaling forward unconditional
  h323
  sip
    min-se 360
    header-passing
```

```
voice class codec 1
  codec preference 1 g711ulaw
    codec preference 2 g729r8
!
!
voice translation-profile block
translate called 1
!
!
ivr prompt memory 15000
ivr prompt streamed none
ivr asr-server rtsp://asr-en-us/recognizer
ivr tts-server rtsp://tts-en-us/synthesizer

mrcp client timeout connect 10
mrcp client timeout message 10
mrcp client rtpsetup enable
rtsp client timeout connect 10
rtsp client timeout message 10
vxml tree memory 500
http client cache memory pool 15000
http client cache memory file 500
http client connection timeout 60
http client response timeout 30
http client connection idle timeout 10

gateway
  timer receive-rtcp 6
!
ip rtcp report interval 3000
application
  service new-call flash:bootstrap.vxml

  service cvperror flash:cvperror.tcl

  service handoff flash:handoff.tcl

  service bootstrap flash:bootstrap.tcl
    !
```

VoiceXML Gateway Configuration: Example of Dial-peer for ICM VRU Label for Type 3 or Type 7

The following example provides the configuration for an ICM VRU label dial-peer for the Type 3 or 7 Unified CVP VRU-Only call flow model:

```
dial-peer voice 777 voip
  description ICM VRU label
  service bootstrap
  voice-class codec 1
  incoming called-number <your sendtovru label pattern here>
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  no vad
  !
```

Unified CVP Comprehensive Call Flows for Pre-Routed Calls

This class of call flows is similar to the Unified CVP Comprehensive call flow models, except that calls are first introduced into Unified ICME or Unified ICMH using some path other than through Unified CVP. A Unified ICME routing script is given the chance to "pre-route" such calls before Unified CVP ever sees them. Once the script transfers the call to Unified CVP, for either self-service or queuing, a more standard Unified CVP Comprehensive call flow model is used.

In this section we will discuss the following call flows:

- Calls which arrive at Unified ICME through a pre-route-only NIC (page 125)
- Calls which are pre-routed through a GKTMP NIC (not applicable for SIP deployments) (page 126)
- Calls which are originated by Cisco Unified Communications Manager (page 126)
- Calls which are originated by an ACD or Call Routing Interface (CRI) based VRU using a Peripheral Gateway (PG) (page 128). A VXML Server running as a Standalone with ICME Lookup call flow model also falls into this category, if the goal of the ICM Lookup is to transfer the call into a Unified CVP Comprehensive call flow model deployment.

All these call flows share the characteristic that the original routing client (a NIC, Unified CM, an ACD, or a VRU) is capable of a single route request only. By definition, the routing client makes a single request to Unified ICME, and Unified ICME returns a destination label and the routing client affects the transfer. At that point the route request dialog is ended, and Unified ICME has no ability to send a subsequent label or conduct any other form of third-party call control.

The routing script might continue, however. This would be the case if the returned label was a translation route to VRU label, or if it was a correlation ID label resulting from a SendToVRU node. In those cases the call is transferred to Unified CVP, and the routing script continues executing once Unified CVP successfully receives the call. The script then invokes micro-application requests as part of its queuing or self service treatment. If the call will then be transferred to an agent or skill group, that label must go to Unified CVP rather than to the original routing client. If the call will later be blind-transferred to another agent or skill group, or back into Unified CVP for additional queuing or self service, that label too must go to Unified CVP rather than to the original routing client.

When the call is delivered to Unified CVP, in order for micro-applications to be supported, it must establish both the Switch and the VRU leg. In short, it must enter a normal Unified CVP Comprehensive call flow model. The only difference between the pre-routed case and a normal Unified CVP Comprehensive call flow model case is in how the call first arrives at Unified CVP. In the pre-routed case, it arrives using either a translation route or correlation-id transfer, whereas in the more typical case it arrives as a new call from the PSTN. In either case, a subsequent transfer to Unified CVP's VRU leg is required.

The following sections describe the important configuration points for each of the above call flows.

Calls Which Arrive at Unified ICME Through a Pre-Route-Only NIC

The following Unified ICME NICs fall into this category: ATT, GKTMP, MCI, Sprint, Stentor. The GKTMP NIC is a special case of this category and will be discussed in the next section. This call flow applies to both Unified ICME and Unified ICMH implementations. In the latter case, both Unified CVP and the NIC would be deployed at the NAM.

For Unified ICME Release 7.1 and later, do the following:

1. Configure a single Type 10 Network VRU and associate it with all Unified CVP peripherals in the PG Explorer configuration tool, and also define it as the default system Network VRU in the System Information tool.

2. In order to support the initial call transfer to Unified CVP from the pre-route routing client, configure Translation Route labels to target the Unified CVP peripherals.

3. In order to support the transfer to VRU leg, configure the Type 10 Network VRU that you defined above in Step 1 with Network Routing Number labels for each Unified CVP peripheral routing client.

4. Associate all micro-application VRU scripts with that same Type 10 Network VRU. When the routing script transfers the call into Unified CVP, it must use a TranslationRouteToVRU node. No subsequent node is necessary in order to perform the transfer to Unified CVP's VRU leg; this will take place automatically.

   Note: Non-prerouted calls can also share the same Network VRU and the same Unified CVP Call Servers.

For Unified ICME Release 7.0, do the following:

1. Configure two Network VRUs: one Type 2 and one Type 7.

2. In the PG Explorer tool, assign all Unified CVP Call Servers to the Type 2 Network VRU.

3. Configure one set of Translation Route labels to target the Type 2 Call Servers; these will be used to transfer the call from the original routing client to the Unified CVP Switch leg.

4. Assign a label to the Type 7 Network VRU for each Unified CVP Call Server routing client, whose label string is set to the Network Routing Number.

5. Configure the Type 7 Network VRU as the system default Network VRU in the System Information configuration tool.

6. Associate all micro-application VRU scripts with the Type 7 Network VRU.

7. When the routing script transfers the call into Unified CVP, it must use *two* nodes in succession: first, a TranslationRouteToVRU, and then an explicit SendToVRU node. The first node transfers the call from the initial routing client to one of the Type 2 Call Servers

(Unified CVP Switch leg); the second one transfers the call from the Type 2 Call Server to the Unified CVP VRU leg. (The VRU leg will usually end up running through the same Unified CVP Call Server as the Switch leg.)

Note: Non-prerouted calls can also share the same Type 2 Call Servers and Type 2 and Type 7 Network VRUs; however, scripts which handle non-prerouted calls must also use an explicit SendToVRU node before they can execute any micro-applications.

Calls Which are Pre-Routed Through a GKTMP NIC (Not Applicable for SIP Deployments)

This call flow is actually a special case of those calls which arrive at Unified ICME through a pre-route-only NIC, since GKTMP does in fact fall into that category. However, it differs in that when the call arrives there, it is already in the H.323 realm. In fact, it is an H.323 Gatekeeper which issues the GKTMP request.

The GKTMP routing script returns not only a translation route to VRU label, but also the IP address of the specific Unified CVP Call Server which represents the Unified CVP Switch leg for that call. (The IP address is returned in an ECC variable.) Therefore, the GKTMP pre-route can be used to perform a kind of custom-scripted load balancing across Unified CVP Call Servers.

Note: This call flow applies to both Unified ICME and Unified ICMH implementations. In the latter case, both Unified CVP and the NIC would be deployed at the NAM.

Calls Which are Originated by Unified CM

This category includes all of the following types of calls:

- "Internal Help Desk" calls, in which a Unified CM phone user calls a CTI Route Point, which starts a routing script that can optionally deliver the call to Unified CVP for queuing or self-service.
- Unified ICME Outbound Option calls, in which a dialer makes outbound calls and then transfers them to a CTI Route Point; this starts a routing script that can optionally deliver the call to Unified CVP for queuing or self-service.
- Consultative Warm Transfer, in which a Unified CM agent places the caller on hold and dials in to Unified ICME to reach a second agent; this starts a routing script that can optionally deliver the call to Unified CVP for queuing or self-service.

  Note: Refer to "Using Cisco Unified ICME Warm Consult Transfer/Conference to Unified CVP (page 347)" for additional information about Consultative Warm Transfer.

Note: If these call flows are used in a Cisco Unified Contact Center Management Portal environment, the target Unified CVP Call Servers are required to be connected to the same CICM as the Unified CM from which the call originates. Just as multiple CICMs will require multiple Unified CMs, so will they require multiple Unified CVP Call Servers.

Further configuration points differ depending on whether Unified CVP is being deployed with Unified ICME Release 7.0 or 7.1.

For Unified ICME Release 7.1 and later, do the following:

1. Configure a single Type 10 Network VRU and defined as the default system Network VRU in the System Information tool.

2. Configure the Type 10 Network VRU with two sets of labels. Associate the first set with the Unified CM routing client, which contains a label that Unfied CM uses to transfer the call to Unified CVP. Configure Unified CM with a series of route patterns, which include that label followed by one to five arbitrary digits. For example, if the selected label is 1111, then the following route pattern is needed: 1111!. The second set of Network VRU labels must contain the usual Comprehensive Model "Network Routing Number," which must be associated with each Unified CVP Call Server routing client.

3. Associate all micro-application VRU scripts with that same Type 10 Network VRU.

4. When the routing script transfers the call into Unified CVP, it should use a single SendToVRU node. No subsequent node is necessary in order to perform the transfer to Unified CVP's VRU leg; this will take place automatically. (The SendToVRU node can be omitted since any micro-application script node will invoke the same functionality automatically; however, you can include this node explicitly in the script for troubleshooting purposes.)

   Note: Non-prerouted calls can also share the same Network VRU and the same Unified CVP Call Servers as those calls which are transferred from Unified CM; however, note that the scripts which handle non-prerouted calls must also use an explicit SendToVRU node before they can execute any micro-applications.

For Unified ICME Release 7.0, do the following:

1. Configure two Network VRUs: one Type 2 and one Type 7.

2. In the PG Explorer tool, assign the Unfied CVP Call Servers to the Type 2 Network VRU.

3. Configure one set of Translation Route labels to target the Type 2 Call Servers; these will be used to transfer the call from the original routing client to the Unfied CVP Switch leg.

4. Assign a label to the Type 7 Network VRU for each Unified CVP Call Server routing client, whose label string is set to the Network Routing Number.

5. Configure the Type 7 Network VRU as the system default Network VRU in the System Information configuration tool.

6. Associate all micro-application VRU scripts with the Type 7 Network VRU.

7. When the routing script to transfers the call into Unified CVP, it should use *two* nodes in succession: first, a TranslationRouteToVRU, and then an explicit SendToVRU node (which contrary to the Unified ICME 7.1 case, is *not* optional). The first node transfers the call from the initial routing client to one of theType 2 Call Servers (Unified CVP Switch leg);

the second one transfers the call from the Type 2 Call Server to the Unified CVP VRU leg. (The VRU leg will usually end up running through the same Unfied CVP Call Server as the Switch leg.)

Note: Non-prerouted calls can also share the same Type 2 Call Servers and Type 2 and Type 7 Network VRUs.

Calls Which are Originated By an ACD or Call Routing Interface (CRI) Based VRU Using a Peripheral Gateway (PG)

These calls are very similar to those which arrive from a preroute-only NIC, except that the routing client is connected to Unified ICME using a PG rather than using a NIC. Therefore, if this call flow is used in a Unified ICMH environment, the target Unified CVP Call Servers are required to be connected to the same CICM as the ACD or CRI-based VRU from which the call originates. Just as multiple CICMs will require multiple ACD or VRU peripherals, so will they require multiple Unified CVP Call Servers.

Further configuration points differ depending on whether Unified CVP is being deployed with Unified ICME Release 7.0 or 7.1.

For Unified ICME Release 7.1 and later, do the following:

1. Configure a single Type 10 Network VRU and associate it with all Unfied CVP peripherals in the PG Explorer configuration tool, and also define it as the default system Network VRU in the System Information tool.

2. In order to support the initial call transfer to Unfied CVP from the pre-route routing client, configure Translation Route labels to target the Unified CVP peripherals.

3. In order to support the transfer to VRU leg, configure the Type 10 Network VRU with Network Routing Number labels for each Unified CVP peripheral routing client.

4. Associate all micro-application VRU scripts with that same Type 10 Network VRU.

5. When the routing script transfers the call into Unified CVP, it must use a TranslationRouteToVRU node. No subsequent node is necessary in order to perform the transfer to Unified CVP's VRU leg; this will take place automatically.

   Note: Non-prerouted calls can also share the same Network VRU and the same Unified CVP Call Servers.

For Unified ICME Release 7.0, do the following:

1. Configure two Network VRUs: one Type 2 and one Type 7.

2. In the PG Explorer tool, assign all Unfied CVP Call Servers to the Type 2 Network VRU.

3. Configure one set of Translation Route labels to target the Type 2 Call Servers; these will be used to transfer the call from the original routing client to the Unfied CVP Switch leg.

4. Assign a label to the Type 7 Network VRU for each Unified CVP Call Server routing client, whose label string is set to the Network Routing Number.

5. Configure the Type 7 Network VRU as the system default Network VRU in the System Information configuration tool.

6. Associate all micro-application VRU scripts with the Type 2 Network VRU.

7. When the routing script transfers the call into Unified CVP, it should use *two* nodes in succession: first, a TranslationRouteToVRU, and then an explicit SendToVRU node. The first node transfers the call from the initial routing client to one of the Type 2 Call Servers (Unified CVP Switch leg); the second one transfers the call from the Type 2 Call Server to the Unified CVP VRU leg. (The VRU leg will usually end up running through the same Unified CVP Call Server as the Switch leg.)

   Note: Non-prerouted calls can also share the same Type 2 Call Servers and Type 2 and Type 7 Network VRUs.

Common Unified ICMH Configuration for Unified CVP Switch Leg

The steps in the following table describe Unified ICMH configuration instructions common to Comprehensive Unified ICMH and VRU-Only with NIC routing, with Correlation ID call routing call flow models for Unified CVP switch legs.

High-level Configuration Steps for Common Unified ICMH Unified CVP Switch Leg

Step 1 On the **NAM**, ICM Configuration Manager, **Network VRU Explorer** tool:

a. Define a Network VRU for Unified CVP.

- Type: **10**
- Name: **cvpVRU**

  Note: This name is used by convention. Any name will do; since it is referenced elsewhere in this document, **cvpVRU** will be assumed.

b. Assign labels.

Define one **Label** per Unified CVP or NIC routing client.

- Type: **Normal**
- Label: Network Routing Number

Step 2 Set the client type.

On the **CICM**, using the ICM Configuration Manager, **NIC Explorer** tool:

- Select the **Routing Client** tab for the INCRP NIC.
- Specify the Client Type: **VRU**

Step 3 Define a Network VRU.

On the **CICM**, using the ICM Configuration Manager, **Network VRU Explorer** tool, define a Network VRU with a label that uses INCRP as its routing client.

Specify the following:

- Type: **10**
- Name: **cvpVRU**

  Note: This name is used by convention. Although any name will work; since it is referenced elsewhere in this document, **cvpVRU** is assumed.

- Define one **Label** for the NAM routing client:
  - Label: Network Routing Number.
  - Type: **Normal**.
  - Routing client: **INCRP NIC**

Step 4 Define the Peripheral Gates (PGs).

On the **NAM**, using the ICM Configuration Manager, **PG Explorer** tool, configure a peripheral gate for each ICM Service to be used for a Switch leg connected to each PG.

For each Unified CVP ICM Service connected to this PG, in the tree view pane, select the applicable PG.

Logical Controller tab:

- Client Type: **VRU**
- Name: A name descriptive of this PG

  For example: **`<location>_A`**, for side A of a particular location.

Peripheral tab:

- Peripheral Name: A name descriptive of this Unified CVP peripheral, for example, **`<location>_<cvp1>`** or **`<dns_name>`**
- Client Type: **VRU**

- Select the checkbox: **Enable Post-routing**

  On the **Advanced** tab, select the name **cvpVRU** from the Network VRU field drop-down list.

Routing Client tab:

- Name: By convention, use the same name as the peripheral.
- Client Type: **VRU**
- *Do not* select the **Network Transfer Preferred** checkbox.

Common Unified ICME/ICMH Configuration: Define Unified CVP ECC Variables

You need to set up ECC variables that Unified CVP uses to exchange information with Unified ICME/ICMH.

Note: Information about how Unified CVP uses these ECC variables can be found throughout the manual.

Step 1 Within the ICM Configuration Manager, select **Tools > Miscellaneous Tools > System Information** and select the **Enable expanded call context** checkbox.

Step 2 Within the ICM Configuration Manager, select **Tools > List Tools > Expanded Call Variable List**.

Step 3 In the Expanded Call Variable List window, enable the **Add** button by clicking **Retrieve**.

Step 4 Click **Add**. The Attributes property tab is enabled.

Step 5 Create each of the variables in the following table, clicking **Save** after defining each variable.

Note: Any time you change the configuration of any ECC variable with the Expanded Call Variable List tool, you must stop and restart the Unified CVP Call Server.

Caution: It is important that you enter the ECC's Name values listed in following table exactly as specified. If you do not, the Unified ICME/ICMH software will not be able to successfully communicate with the micro-applications on the ICM Service.

Length values are more flexible. Unless the values listed in following table are *specifically* noted as "required," the value in the Length column is the *maximum* that Unified ICMH can handle for that ECC; you can specify a value between 1 and the maximum length.

Note:

- In a Unified ICME/ICMH configuration, the ECC variable configuration, including the length, defined in the NAM must be defined *exactly* the same in the CICM.

- If you change the length of the ECC variables while the Unified CVP ICM Service is running, you need to restart the Unified CVP ICM Service for it to work properly.

Step 6 When finished, click **Save** to apply your changes.

Table 2: Micro-Application ECCs

Name	Length	Definition
user.CourtesyCallbackEnabled	**Required** for using Courtesy Callback. Length:1	Used to determine if Courtesy Callback should be offered to a caller. Valid values are: "1" = Yes "0" = No
user.cvp_server_info	Length: 15	Used by Unified CVP to send the IP address of the Call Server sending the request to Unified ICME. Example: An IPv4 address like 192.168.150.181
user.media.id	**Required** value: 36	A number identifying a call to the ICM Service and, optionally, the H.323 Service: • For a call arriving from the network, consists of an H.323 Conference ID. • For a call arriving from a non-H.323 Service client, consists of a random number.
user.microapp.currency	value: 6	Currency type.
user.microapp.error_code	value: 2	Return status error code to be returned from the Unified CVP to Unified ICME/ICMH upon a False return code in the Run Script Result.
user.microapp.fetchaudio	Recommended length: 20; but length depends on the file name.	Filename for audio to be played by the VXML gateway while the gateway loads and processes the requested resource when there is signficant network latency. Default: none Example: "flash:holdmusic.wav"
user.microapp.fetchdelay	Length: 1	The length of time (in seconds) to wait at the start of the fetch delay before playing the audio specified by *user.microapp.fetchaudio*. This setting only takes effect if the value of fetchaudio is not empty.

Name	Length	Definition
		Default: 2 seconds; used to avoid a "blip" sound heard in a normal network scenario Setting this value to zero will play hold music immediately, for a minimum of five seconds. Values: 1 through 9
user.microapp.fetchminimum	Length: 1	The minimum length of time to play audio specified by *user.microapp.fetchaudio*, even if the requested resource arrives in the meantime. This setting only takes effect if value of fetchaudio is not empty. Default: 5 seconds Values; 1 through 9
user.microapp.isPostCallSurvey	Length: 1	Used to determine if post call survey should be offered to a caller after the agent hangs up. Valid values: "y" or "Y" is "Yes" "n" or "N" is "No" Default value is "Yes"
user.microapp.locale	value: 5	Locale, a combination of language and country which defines the grammar and prompt set to use.
user.microapp.media_server	**Required** for any IVR scripting. Maximum length: 210 characters Recommended length: 30	Root of the URL for all media files and external grammar files used in the script. HTTP and HTTPS schemes can be specified as: • HTTP scheme is specified as "http://<servername>" • HTTPS scheme is specified as "https://<servername>"
user.microapp.play_data	40	Default storage area for data for Play Data micro-application.
user.microapp.sys_media_lib	10	Directory for all system media files, such as individual digits, months, default error messages, and so forth.
user.microapp.app_media_lib	Maximum length: 210 characters Recommended length: 10	Directory for all application-specific media files and grammar files. You can also set this value to "..", (literally two periods in quotes) which bypasses the

Name	Length	Definition
		user.microapp.app_media_lib and user.microapp.locale ECC Variables when writing a URL path. For example, if you set the user.microapp.app_media_lib to "..", the path: **`http://server/locale/../hello.wav`** would really be: **`http://server/hello.wav`**
Note: The system and application media libraries need message and prompt files created/recorded for each locale that will be referenced. For more information, refer to "Configuring the Media Servers (page 531)."		
user.microapp.grammar_choices	Configurable on Unified ICME. Maximum length: 210 characters.	Specifies the ASR choices that a caller can input for the Get Speech micro-application. Each option in the list of choices is delimited by a forward slash (/). **Note:** If text is placed in this variable that is longer than the variable is configured to handle, only the first 210 characters are sent.
user.microapp.inline_tts	Configurable on the ICM. Maximum length: 210 characters.	Specifies the text for inline Text To Speech (TTS). **Note:** If text is placed in this variable that is longer than the variable is configured to handle, only the first 210 characters are sent.
user.microapp.input_type	value: 1	Specifies the type of input that is allowed. Valid contents are: • **D** - DTMF • **B** - (Both, the default) DTMF and Voice If you are not using an ASR, you need to set this variable to D. If you are using anASR, you can set the variable to either D or B. **Note:** With input_mode set to "B" (both), either DTMF or speech will be accepted, but mixed mode input is not. Once you begin entering with one mode, input using the other mode is ignored and has no effect.

Name	Length	Definition
user.microapp.caller_input	Configurable on Unified ICME/ICMH. Maximum length: 210 characters.	Storage area for any ASR input that is collected from Get Speech. **Note:** Get Speech text results will be written to this ECC variable. Results from Get Digits or Menu micro-applications will be written to the CED.
user.microapp.pd_tts	value: 1	Specifies whether Unified CVP's Text To Speech (TTS) or media files must be played to the caller. Valid contents are: • **Y** - Yes, use TTS capabilities • **N** - No, do not use TTS capabilities; play media files instead. **Note:** Used only with Play Data micro-application.
user.microapp.UseVXMLParams	value: 1	This parameter specifies the manner in which you pass information to the external VoiceXML. Set this parameter to either "Y" (for yes) or "N" (for no). Y uses the values in the user.microapp.ToExtVXML variable array. N appends the name/value pairs in user.microapp.ToExtVXML to the URL of the external VoiceXML. Default: "N" Refer to "Writing Scripts for Unified CVP (page 141)," for more information.
user.microapp.ToExtVXML	210	This variable array sends information to the external VoiceXML file. Must be configured as Array variables, not Scalar variables, even if the array length is set to 1. Refer to "Passing Information to the External VoiceXML (page 200)" for more information.
user.microapp.FromExtVXML	210	This variable array returns information from the external VoiceXML file. Must be configured

Name	Length	Definition
		as Array variables, not Scalar variables, even if the array length is set to 1. Refer to "Passing Data Back to Unified ICME with External VoiceXML (page 203)" for more information.
user.microapp.override_cli	Configurable on Unified ICME/ICMH. Maximum length: 200 characters.	Used by system to override the CLI field on outgoing transfers.
user.microapp.metadata	The variable length would normally be configured as 62 bytes, but if ECC space is restricted, you can configure it as 21 bytes.	Following the Menu (M), Get Data (GD) and Get Speech (GS) micro-applications, Unified CVP returns information about the execution of that micro-application. The user.microapp.metadata ECC variable is structured as follows: **`m|con|tr|to|iv|duratn|vruscriptname`**
user.microapp.uui	Configurable on Unified ICME/ICMH. Maximum length: 131 characters.	Used to pass user-to-user information back to Unified CVP from Unified ICME/ICMH.
user.sip.refertransfer	Optional Maximum length: 1 character.	SIP Service will use REFERs when transferring to the agents: • y – Use REFER when transferring • n – Do not use REFER when transferring
user.suppress.sendtovru	optional Length: 1	Suppress the Temporary Connect message generated by SendToVRU node (explicitly or implicitly, for example by a Translation Route to VRU node). Used in call flows where the Temporary Connect is generated right before the Connect message (that is, no Run Script messages expected) to avoid the extra overhead of setting up a VRU leg temporarily before the Connect arrives. Valid values are : "y" or "Y" (yes, suppress the message)

Using the Metadata ECC Variable

Metadata ECC Variable

Following the Menu (M), Get Data (GD) and Get Speech (GS) micro-applications, Unified CVP returns information about the execution of that micro-application. The information is returned in the ECC variable **user.microapp.metadata**. Its format is defined in terms of a number of subfields, each separated by a vertical bar character ('|'). Also, the subfields are of fixed length in order to facilitate extraction either at reporting time or within the ICM routing script itself.

The **user.microapp.metadata** ECC variable is structured as follows:

m|con|tr|to|iv|duratn|vruscriptname

The following table shows the values for this variable.

Table 3: Metadata ECC Variable Values

Metadata	ECC Variable Value
m	D, V or N – Indicates whether the user responded withVoice (V), DTMF (D), or not at all (N). (Note that for the Menu micro-application, any successful single digit entry will result in m being set to V or D, even if the entry was an invalid menu selection.)
con	000 to 100 – Indicates the ASR percent confidence level at which the voice input was finally recognized. This field is only valid if m is Voice (V).
tr	00 to 99 – Indicates how many tries were required. 01 means user responded successfully after the first prompt.
to	00 to 99 – Indicates how many timeouts occurred. Does not include interdigit timeouts.
iv	00 to 99 – Indicates how many invalid entries were received, including interdigit timeouts.
duratn	000000 to 999999 – Indicates the micro-application duration in milliseconds. Duration is defined as the elapsed time between entering and exiting the RunExternalScript node, as measured in the IVR Service.
vru script name	Full name of theVRU script which was executed. This is the only variable length field.

This ECC variable is optional. If used, you must define it in the Unified ICME Expanded Call Context Variables configuration tool. The variable length would normally be configured as 62 bytes, but if ECC space is restricted, you can configure it as 21 bytes. This will drop the vruscriptname subfield. If you do define this variable, its contents get written to the Unified

ICME database with every termination record, and can be used to provide a record of meta-information about the execution of each input micro-application.

Common Configuration for Differentiating VRUs (Unified CVPs) Based on Dialed Number

The Network VRU configuration instructions in this guide assume that all callers will be routed to the same VRUs (Unified CVPs) for VRU treatment purposes. Under this assumption, it is always simplest to rely on the system default Network VRU. However, it is sometimes necessary to differentiate the VRUs (Unified CVPs) based on dialed number.

Note: This section is only applicable to call flow models which use the SendToVRU node to transfer the call to Unified CVP's VRU leg (it does not apply to Translation Route transfers).

For example, some calls:

- Require Comprehensive treatment, while others only use the H.323 with MediaTermination call flow model which does not support ASR/TTS.
- Need to assign different customers or applications to their own Unified CVP machines.

To configure Unified ICME to differentiate the VRUs, do the following:

- Configure more than one Network VRU.
- On Unified ICME, ICM Configuration Manager, **ICM Instance Explorer** tool:
  - Configure the customer(s).
  - Configure the Network VRU for each customer if that customer will use a Network VRU other than the default.
- Associate the dialed number(s) to the customer in the Dialed Number List tool.
- Since each configured VRU script is specific to one specified Network VRU, you need to create a distinct set of VRU scripts for each Network VRU. Also, be sure that the ICM routing script calls the correct set of VRU scripts.

Gatekeeper Redundancy

Hot Standby Router Protocol (HSRP) is not the only method of Gatekeeper redundancy. Instead, use Gatekeeper Clustering and Alternate Gatekeeper configuration on Unified CVP. This is the preferred method of Gatekeeper redundancy.

Gatekeeper HSRP Configuration

On the primary gatekeeper, enter these commands:

```
interface ethernet 0
  ip address 10.0.1.98 255.255.255.0
  ! Unique IP address for this GK
  standby 1 ip 10.0.1.100
  ! Member of standby group 1, sharing virtual address 10.0.1.100
  standby 1 preempt
  ! Claim active role when it has higher priority.
  standby 1 priority 110
  ! Priority is 110
  !
```

On the backup gatekeeper, enter these commands:

```
interface ethernet 0
  ip address 10.0.1.99 255.255.255.0
  standby 1 ip 10.0.1.100
  standby 1 preempt
  standby 1 priority 100
  !
```

On both Gatekeepers, configure identical gatekeeper configurations, for example:

```
gatekeeper
  ! Enter gatekeeper configuration mode
  zone local gk-sj cisco.com 10.0.1.100
! Define local zone using HSRP virtual address as gatekeeper RAS address
!
```

Alternate Gatekeeper Configuration

Configure the following examples using the Operations Console H.323 configuration tab for the Call Server:

Examples:

- Example 1: **`set GK "10.0.1.100, 10.0.2.100, 10.0.3.100"`**

  This example sets up three gatekeepers, which could be registered to the H.323 Service. In each case, the H.323 Service registers to the first local zone that is configured in that gatekeeper. It also uses the default RAS port 1719.

- Example 2: **`setGK "10.0.1.100:zone1:1718, 10.0.2.100"`**

  This example causes the H.323 Service tofirst attempt to register to gatekeeper 10.86.129.33 on port 1718 with local zone "zone1". If that gatekeeper fails, the H.323 Service subsequently attempts to register to 10.86.129.34 on port 1719 with the first local zone defined on that gatekeeper.

See Also

Refer to the Operations Console online help for more information.

Local SRV File Configuration Example for SIP Messaging Redundancy

Below is an example local SRV configuration file. It must be named in a text file named srv.xml and manually placed in the c:\cisco\cvp\conf directory of the Call Server where the SIP Service is running.

```
<?xml version="1.0" encoding="UTF-8" ?>
<locater>
<host name="cups.proxy.cisco.com">
 <record weight="30" priority="1" destination="10.86.129.23" port="5060"/>
 <record weight="30" priority="2" destination="10.86.129.109" port="5060"/>
</host>
<host name="ringtone.services.cisco.com">
 <record weight="30" priority="1" destination="10.86.129.23" port="5060"/>
 <record weight="30" priority="1" destination="10.86.129.109" port="5060"/>
</host>
</locater>
```

Chapter 3

Writing Scripts for Unified CVP

This chapter discusses using Unified ICME configuration and script editing to access the Unfied CVP solution.

It includes information about how to:

- Set up Unified ICME to interact with Unified CVP
- Write applications for Unified CVP

Note: This chapter contains important information for IVR application developers. It also may be of interest to Call Center Managers, Unified CVP System Managers, and Unified ICME System Managers.

This chapter contains the following topics:

Before You Begin

This chapter makes the following assumptions:

- The information in this chapter assumes that you are already familiar with using Unified ICME software's ICM Configuration Manager and Script Editor tools for call center operations and management.

Note: You should have a copy of the following Unified ICME documentation available in addition to this manual in order to successfully configure Unified ICME and use its features in conjunction with Unified CVP: **ICM Configuration Guide for Cisco ICM Enterprise Edition** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/products_installation_and_configuration_guides_list.html) and *ICM Scripting and Media Routing Guide for Cisco ICM/IPCC Enterprise & Hosted Editions*.

- When creating applications that interact with Unified CVP, only use alphanumeric characters for application, element, and field names; *do not* use special characters such as periods, asterisks or brackets. Following this practice will avoid potential issues with data transfer between different systems.

Making Unified ICME Work with Unified CVP

Unified ICME determines where to route calls—whether to ACDs, specific agents, or to VRUs. However, the routing services themselves must be provided by an external routing client.

Traditionally, Unified ICME's routing clients were PSTN network switches, or in some cases, customer provided ACDs. Unified CVP provides VoIP routing capability for the Unified ICME and Unified CCE products. Unified CVP makes it possible for Unified ICME to use VoIP gateways as routing clients, as well traditional routing services.

Unified ICME and Unified CVP work together to perform such tasks as:

- Playing media, such as a recording stating office hours, to a caller.
- Playing streaming audio, such as a radio broadcast, to a caller.
- Retrieving caller-entered data, DTMF, or speech.
- Playing back different types of data, such as an account number or balance, to a caller.
- Moving calls to other destinations. For example, forwarding calls to an agent.

Unified ICME uses IVR messaging technology to direct Unified CVP and to receive the responses from Unified CVP.

Using Scripts to Access Unified CVP from Unified ICME

Both Unified ICME and Unified CVP use scripts to invoke their features. In fact, Unified ICME references Unified CVP scripts from *within* its own scripts . This method of invoking Unified CVP from within Unified ICME enables Unified ICME to take advantage of the features of Unified CVP.

The two products (Unified ICME and Unified CVP) provide two service creation (scripting) environments. Each environment is used for different purposes:

- **ICM Script Editor.** Use this scripting tool to develop agent routing scripts and to invoke the Unified CVP **micro-applications**: Play Media, Get Speech, Get Digits, Menu, Play Data, and Capture. These applications are the basic building blocks of a voice interaction design.
- **Call Studio.** Use Call Studio to develop sophisticated IVR applications.

  Note: For more information, refer to "Scripting for Unified CVP with Call Studio (page208)."

Unified ICME Scripting

The Unified ICME Script Editor is used to develop agent routing scripts, and to invoke Unified CVP micro-applications—basic building blocks of a voice interaction design. The Unified CVP micro-applications are: Play Media, Get Speech, Get Digits, Menu, Play Data, and Capture. These applications are combined and customized in the Unified ICME routing script to produce a viable voice interaction with the caller.

While it is possible to develop full scale IVR applications using micro-applications, it is not recommended. Micro-application-based scripts are primarily used for initial prompt and collection operations, as well as for directing the playing of .wav files while calls are in queue. Instead, use the IVR scripts developed using Call Studio to create the IVR applications.

In an environment where Unified ICME script works with VXML script (the 2-script implementation for Unified ICME-integrated models described here), the Unified ICME script remains in control (and receives control back), even while it *delegates* the more complex self service activity to the VXML Server script. Data can be passed from one script to the other and back through ECC variables.

Note: The capability of using Unified ICME scripting for anything other than simple functions has been kept in support of legacy deployments. New customers are strongly advised to use the VoiceXML scripting environment of Unified CVP for creating IVR applications.

Unified CVP VoiceXML Scripting

Sophisticated IVR applications can be developed using Call Studio which is an Eclipse-based service creation environment whose output is an intermediary file describing the application flow. That file gets loaded onto the VXML Server machines for execution. To invoke a VXML Server application, the script writer includes a special micro-application in his Unified ICME routing script. This micro-application instructs the VoiceXML Gateway to interact with the VXML Server directly to execute the application. The final results are passed back to Unified ICME.

Some of the VoiceXML scripting environment features include:

- A drag-and-drop interface with a palette of IVR functions
- The ability to do database queries
- Extensibility with Java code written to perform any task a Java application can perform

Note:

- Unified CVP does not support using the *MicroApp* nodes that are available in the ICM Script Editor. All MicroApp implementation must be done using the *Run External Script* node in ICM Script Editor. Refer to *ICM Scripting and Media Routing Guide for Cisco ICM/IPCC Enterprise & Hosted Editions* for detailed information about the Run External Script node. Refer to Writing Unified ICME Applications for Unified CVP (page 154) for detailed information about setting Unified CVP-specific parameters in this node for each Unified CVP micro-application.

- For more information about creating scripts, refer to "Scripting for Unified CVP with Unified ICME (page 144)."

Micro-application Use Versus VXML Server Use

The same special micro-application that is used to invoke VXML Server applications can also be used to invoke arbitrary "External VXML" pages from a Media Server or other customer-provided source. However, only use this capability for very simple VoiceXML needs, because Cisco has no way to verify that *customer-provided* VoiceXML documents are compatible with the IOS Voice Browser. (As opposed to VoiceXML documents that are generated by VXML Server, which *are* guaranteed by Cisco to be compatible with the IOS Voice Browser.) **Although the capability to use the micro-application has not been removed from the Unified CVP 4.0 and later offerings, customers are discouraged from using it directly.**

Additionally, all the VoiceXML Gateway sizing metrics that Cisco provides are based on the specific VoiceXML documents that are generated using either micro-applications or VXML Server applications. Using VoiceXML from another source will require you to perform your own empirical performance and capacity testing in order to determine how to size the VoiceXML Gateways.

Scripting for Unified CVP with Unified ICME

The sections that follows include:

- A discussion of micro-applications.
- A sample Unified ICME script.
- A discussion of how Unified ICME and Unified CVP exchange information.

What are Micro-Applications?

Micro-applications are a set of specific IVR functions that can be invoked by Unified ICME, enabling communication with the caller.

There are six Unified CVP micro-applications:

- **Play Media**. Plays a message to the caller.

- **Play Data**. Retrieves data from a storage area and plays it to the caller in a specific format called a data play back type.
- **Get Digits**. Plays a TTS or media file and retrieves digits from the caller.
- **Menu**. Plays a TTS or media menu file and retrieves a single telephone keypad entry from the caller.
- **Get Speech**. Collects ASR or DTMF input after prompting a caller.
- **Capture**. The Capture (CAP) micro-application enables you to trigger the storage of current call data at multiple points in the Unified ICME routing script.

Micro-applications are interpreted by the IVR Service, which resides on the Call Server. The IVR Service sends VoiceXML code to the VoiceXML Gateway Voice Browser.

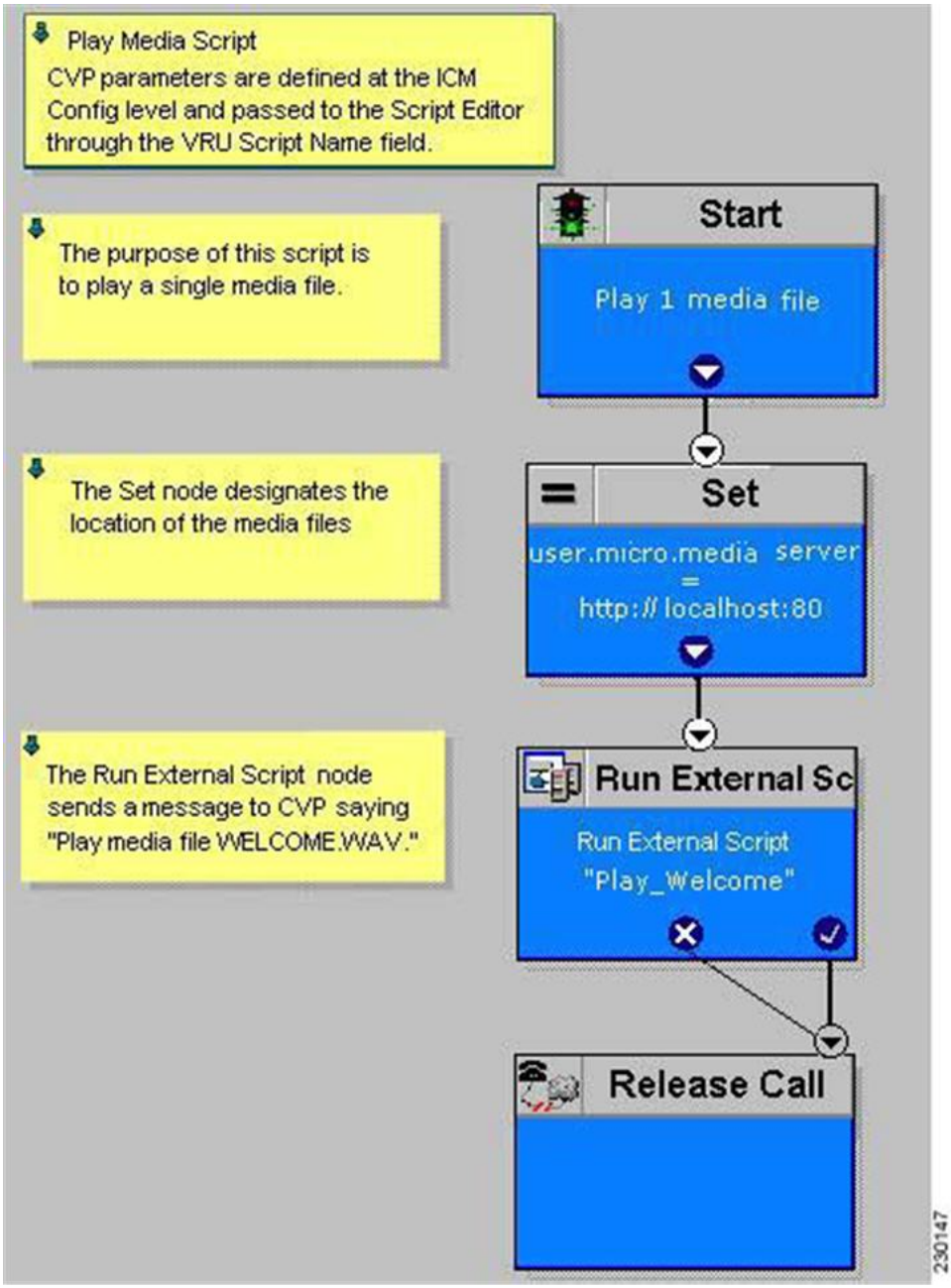
The IVR Service also accepts HTTP requests from the VoiceXML Gateway's Voice Browser, and communicates those requests to Unified ICME's Service Control Interface using the ICM Service.

Note: In H.323 implementations only, the IVR Service also transmits call control instructions (new call, transfer, release) between the Unified ICME and the H.323 Service.

Simple Example Script: “Welcome to XYZ Corporation”

Suppose you want to create a Unified ICME script that simply plays a message, “Welcome to XYZ Corporation.” From the Unified ICME’s perspective, there is no difference between a script written for a standard “black box” IVR or the Unified CVP, so you can create a script such as the one shown in the following figure.

Figure 20: Example Play Media Script



This simple script performs three functions:

- Sends the Run External Script request to Unified CVP.
- Indicates the location of the "Welcome" media file.
- Releases the call.

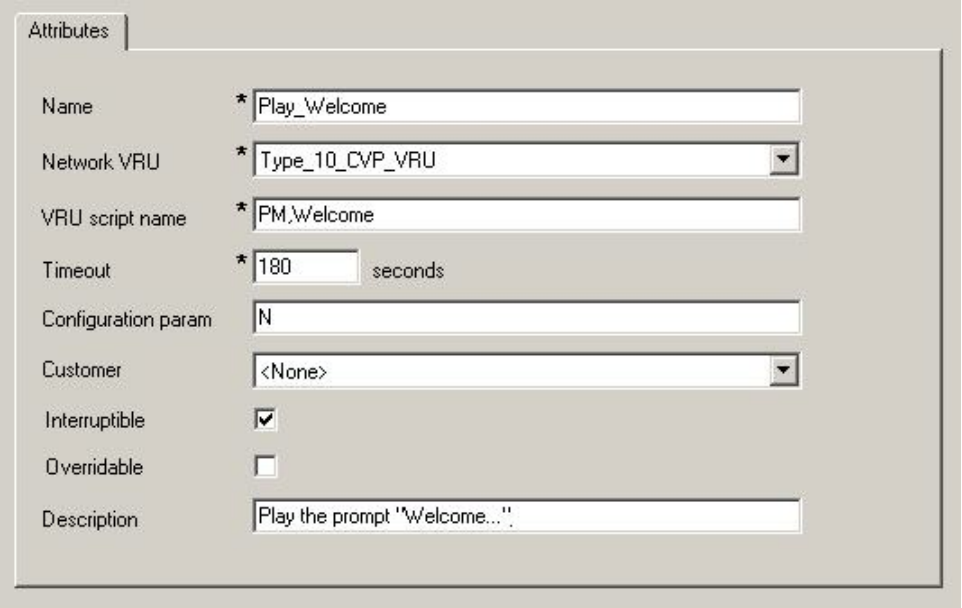
Note: In a “real life” application, any Unified ICME script you create would include error checking to ensure that micro-applications instructions are properly executed.

Cisco Unified Intelligent Contact Management Enterprise /Unified CVP Micro-App Connection

Before the Unified CVP IVR solution can be accessible through the Script Editor's Run External Script node, you must first set up Unified ICME with special Unified CVP parameters using the ICM Configuration Manager tool.

Begin by using the ICM Configuration Manager's Network VRU Script window to define Unified CVP parameters.

Figure 21: Define ICM Script Parameters for Unified CVP



In the figure above:

- **PM,Welcome**. (VRU Script Name field.) This means: "Use the instructions in the Play Media micro-application to play the Welcome.wav media file."
- **N**. (Configuration Param field.) This means: "Do not allow barge-in." (Barge-in is when the caller can interrupt message play by entering a digit, causing the script to move to the next prompt.)
- You *must* check the **Interruptible** checkbox as shown in the figure above. This specification allows the script to be interrupted by the Unified CVP script functions.

Note: As shown in the two columns of the following table, certain entries for the VRU Script Name and Configuration Param fields are case-sensitive.

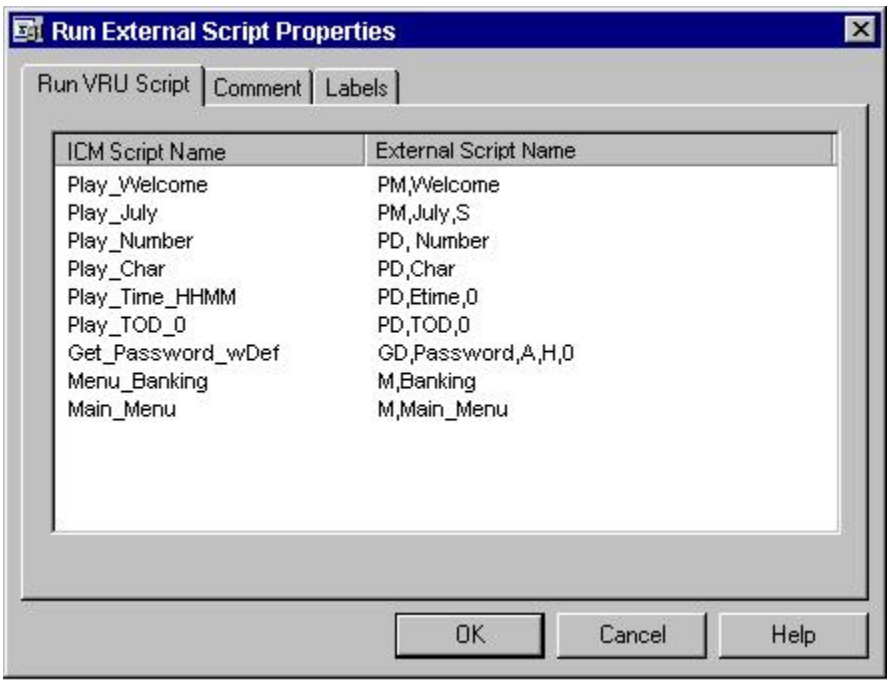
Network VRU Script Field Attributes That *Are* Case-Sensitive	**Network VRU Script Field Attributes That *Are Not* Case-Sensitive**
Applies to: All micro-applications.	**Applies to:** All micro-applications.

Network VRU Script Field Attributes That *Are* Case-Sensitive	Network VRU Script Field Attributes That *Are Not* Case-Sensitive
Attribute: Media File Name (for example, media or .vxml)	**Attribute:** VRU Script Name (for example, PM, GD).
Applies to: Get Speech (GS). **Attribute:** External Grammar File name.	**Applies to:** All micro-applications. **Attribute:** Media Library Type (A, S, V)
	Applies to: All micro-applications. **Attribute:** Barge-in Allowed (Y/N),
	Applies to: PlayData (PD). **Attribute:** Data playback type (for example, Number, Char).
	Applies to: PlayData (PD). **Attribute:** Time Format (HHMM, HHMMSS, HHMMAP),
	Applies to: Get Digits (GD), Get Speech (GS), Menu (M). **Attribute:** Timeout Message Override (Y/N)
	Applies to: Get Digits (GD), Get Speech (GS), Menu (M). **Attribute:** Invalid Entry Message Override (Y/N).
	Applies to: All micro-applications. **Attribute:** DTMF Termination Key (only N)
	Applies to: Get Speech (GS). **Attribute:** Type of Data to Collect (for example, boolean, date).

Once the ICM Configuration Manager's settings have been saved, the information is available to the Script Editor. When you place a Run External Script node in the Script Editor workspace and open the Properties dialog box, it displays all the script names defined in the system.

The Run External Script node below shows that the ICM Script Name Play_Welcome was selected.

Figure 22: Run External Script Node Properties



Information Exchange Between Unified ICME and Unified CVP

When Unified ICME processes a Run External Script node, parameters are sent to Unified CVP.

These parameters contain instructions about how to interact with a caller, such as:

- What micro-application to use.
- The location of the media files to be played to the caller.
- Timeout settings to be used during caller digit entry.

Some IVR parameters are passed to Unified CVP through Expanded Call Context (ECC) variables and/or Call.Peripheral variables. Other parameters are sent in the normal VRU messaging interface (Unified ICME/IVR Service Control Interface).

Unified ICME Data Handling

In defining scripts, you might specify strings, numbers, or formulas to be sent to Unified CVP. When passing numbers to Unified CVP, always enclose them in quotes so that they will be processed as a string.

This is especially important if:

- Leading 0's are significant to the data type (times, character), enter the number as a quoted string (example: "031524").

- Trailing 0's after a decimal point are significant to the data type (number, character, currency), enter the number as a quoted string (examples: "42.00" or "42.10").
- The number is very large (example: a number normally expressed through exponential notation).

Unified CVP Script Error Checking

Unified CVP uses the **user.microapp.error_code** ECC variable to return information regarding problems encountered while running a script.

Unified CVP software tests for the following conditions when processing Unified ICME scripts:

- **ASR Error**. Failure of an Advanced Speech Recognition component.
- **General error**. General error occurred.
- **Invalid Configuration Param**. Data passed from Unified ICME to the IVR Service is not consistent with what the micro-application requires for processing.
- **Invalid variable data**. The variable data passed was not valid for the script type being processed.
- **Invalid VRU Script Name format**. VRU Script Name data passed from Unified ICME to the IVR Service does not contain the expected components (micro-application name, media file name, media file type, uniqueness value).
- **Locale**. Locale was not supported. (Only applies to Play Data micro-applications that use .wav files. Does not apply to Play Data micro-applications that use TTS, or to Play Media, Get Digits, Menu, Get Speech, or Capture micro-applications.)
- **Misconfigured ECC variable**. An ECC variable was set to a value the IVR Service did not recognize. ECC variable definitions must be the same in Unified ICME and Unified CVP.
- **Network Error**. Failure of an IP network connection.
- **Reached Maximum Invalid Tries**. Caller was unsuccessful in entering digits during each of the tries allowed by the micro-application. (Only applies to Get Digits, Menu, and Get Speech micro-applications.)
- **Reached Maximum Number Entry Tries**. Caller did not enter digits in response to the prompt for each of the tries allowed by the micro-application. (Only applies to Get Digits and Get Speech micro-applications.)
- **Semantic-Runtime**. Semantic error occurred while running a micro-application.
- **System Error**. Unexpected failure of a Unified CVP component.
- **Timed Out**. Caller did not enter digits in response to the prompt in the time allowed by the micro-application.

- **TTS Error**. Failure of a Text-to-Speech component.
- **Unavailable Media file**. Media file name passed from Unified ICME to the IVR Service did not exist on the Media Server.
- **Unknown micro-application**. Micro-application name passed from Unified ICME to the IVR Service did not exist on the IVR Service.
- **Unsupported locale**. The VoiceXML Interpreter (that is, Gateway or H.323 Service) did not recognize the locale passed from the IVR Service.
- **Unsupported VoiceXML element**. The VoiceXML Interpreter (that is, Gateway or H.323 Service) did not recognize aVoiceXML element passed from the IVR Service,VXML Server, or media server (using External VoiceXML).
- **Unsupported VoiceXML format**. The VoiceXML Interpreter (that is, Gateway or H.323 Service) did not recognize aVoiceXML format passed from the IVR Service,VXML Server, or media server (using External VoiceXML).

Each Unified CVP micro-application has individualized settings for **user.microapp.error_code** (non-video and video), as shown in the following table.

Table 4: Possible user.microapp.error_code ECC Variable Settings for Non-Video

Error Code	Play Media	Play Data	Get Digits	Menu	Get Speech	Capture
0	No error	No error	No error	No error	No error	No error
1	Caller Hangup	Caller Hangup	Caller Hangup	Caller Hangup	Caller Hangup	N/A
2	Network Error	Network Error	Network Error	Network Error	Network Error	N/A
3	System Error	System Error	System Error	System Error	System Error	System Error
5	Unknown micro-application	Unknown micro-application	Unknown micro-application	Unknown micro-application	Unknown micro-application	Unknown micro-application
6	Invalid VRU Script Name format	Invalid VRU Script Name format	Invalid VRU Script Name format	Invalid VRU Script Name format	Invalid VRU Script Name format	N/A
7	Invalid Configuration Param	Invalid Configuration Param	Invalid Configuration Param	Invalid Configuration Param	Invalid Configuration Param	N/A
8	Misconfigured ECC variable	Misconfigured ECC variable	Misconfigured ECC variable	Misconfigured ECC variable	Misconfigured ECC variable	N/A
9	One of the following: • Media or external VXML file does not exist	One of the following: • Media file does not exist	One of the following: • Media file does not exist	One of the following: • Media file does not exist	One of the following: • Media or external VXML file does not exist	N/A

Error Code	Play Media	Play Data	Get Digits	Menu	Get Speech	Capture
	• Invalid URL for Media or external VXML file • External VXML is in an invalid format	• Invalid URL for Media L file	• Invalid URL for Media L file	• Invalid URL for Media L file	• Invalid URL for Media or external VXML file • External VXML is in an invalid format	
10	Semantic-Runtime Error	Semantic-Runtime Error	Semantic-Runtime Error	Semantic-Runtime Error	Semantic-Runtime Error	N/A
11	Unsupported VoiceXML format	Unsupported VoiceXML format	Unsupported VoiceXML format	Unsupported VoiceXML format	Unsupported VoiceXML format	N/A
12	Unsupported VoiceXML element	Unsupported VoiceXML element	Unsupported VoiceXML element	Unsupported VoiceXML element	Unsupported VoiceXML element	N/A
13	N/A	Variable data is invalid	N/A	N/A	N/A	N/A
14	N/A	Location of variable data is empty	N/A	N/A	N/A	N/A
15	N/A	Time format is invalid	N/A	N/A	N/A	N/A
16	N/A	N/A	Reached Maximum Invalid Tries	Reached Maximum Invalid Tries	Reached Maximum Invalid Tries	N/A
17	N/A	N/A	Reached Maximum No Entry Tries	Reached Maximum No Entry Tries	Reached Maximum No Entry Tries	N/A
20	N/A	Data value out of range	N/A	N/A	N/A	N/A
23	No answer	No answer	No answer	No answer	No answer	N/A
24	Busy	Busy	Busy	Busy	Busy	N/A
25	General transfer error	General transfer error	General transfer error	General transfer error	General transfer error	N/A
26	Invalid extension	Invalid extension	Invalid extension	Invalid extension	Invalid extension	N/A
27	Called party hung up	Called party hung up	Called party hung up	Called party hung up	Called party hung up	N/A
28	Error after transfer established	Error after transfer established	Error after transfer established	Error after transfer established	Error after transfer established	N/A

Error Code	Play Media	Play Data	Get Digits	Menu	Get Speech	Capture
30	Unsupported locale	Unsupported locale	Unsupported locale	Unsupported locale	Unsupported locale	N/A
31	ASR error	ASR error	ASR error	ASR error	ASR error	N/A
32	TTS error	TTS error	TTS error	TTS error	TTS error	N/A
33	General ASR/TTS error	General ASR/TTS error	General ASR/TTS error	General ASR/TTS error	General ASR/TTS error	N/A
34	Unknown error	Unknown error	Unknown error	Unknown error	Unknown error	N/A
40	VXML Server system unavailable	N/A	N/A	N/A	VXML Server system unavailable	N/A
41	VXML Server application error	N/A	N/A	N/A	VXML Server application error	N/A
42	VXML Server application used hangup element instead of subdialog return element	N/A	N/A	N/A	VXML Server application used hangup element instead of subdialog return element	N/A
43	VXML Server application is suspended	N/A	N/A	N/A	VXML Server application is suspended	N/A
44	VXML Server session error (for example, application has not yet been loaded)	N/A	N/A	N/A	VXML Server session error (for example, application has not yet been loaded)	N/A
45	VXML Server encounters a bad fetch error (for example, media or grammar file not found)	N/A	N/A	N/A	VXML Server encounters a bad fetch error (for example, media or grammar file not found)	N/A
46	Audio streaming error	N/A	N/A	N/A	N/A	N/A

Note:

- **user.microapp.error_code** is always zero, indicating success, if control proceeds out the Checkmark (success) branch of the Run External Script node. Usually, if control proceeds out the X (failure) branch, Unified CVP sets this variable to one of the codes listed here. (Set up your routing script to always test the error code after an X branch is taken.)

- However, if a configuration error, or a network or component failure of some sort, prevents the micro-application from being executed at all, then Unified CVP does not get a chance to

set this variable at all. Such cases can be identified by using a Set node to pre-set **user.microapp.error_code** to some known invalid value such as -1, and then to test for that value using an If node, following the X branch of the Run External Script node.

- Two classes of problems can prevent the micro-application from being executed at all: (1) inability to route the call to an appropriate VoiceXML-enabled gateway and IVR Service (VRU-Only call flow model only); (2) mismatch between Network VRU associated with the configured VRU script and Network VRU associated with Unified CVP that is handling the call. The second case can only be caused by an ICM configuration error, but the first case may also be caused by a temporary network outage or other component failure.

Unified ICME Setup

Before you can use Unified ICME features to access the Unfied CVP solution, you must perform some initial setup tasks to enable communication between Unified ICME and Unified CVP. These setup tasks are determined by Unified CVP call flow model; refer to "High-level Configuration Instructions for Call Flow Models" (page 25), for complete setup instructions for each model.

Note: For more information about the supported Unified CVP call flow models, refer to *Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)*.

Writing Unified ICME Applications for Unified CVP

Once Unified ICME-to-Unified CVP initial setup is complete, you can create Unified ICME applications to access Unified CVP micro-applications.

You do this using two Unified ICME software tools:

- Configuration Manager
- Script Editor

The sections that follow give a brief overview of how to use these tools to access Unified CVP functionality.

How to Configure a Unified CVP Network VRU Script

Step 1 Within the ICM Configuration Manager, select **Tools > List Tools > Network VRU Script List**.

Step 2 In the Network VRU Script List window, enable the **Add** button by clicking **Retrieve**.

Step 3 Click **Add**.

The Attributes property tab is enabled.

Step 4 Complete the Attributes tab as described below.

Warning: The format of the strings for the VRU Script Name and Configuration Param fields are *very specific* and vary for different micro-applications (Play Media, Play Data, Get Digits, Menu, and Get Speech).

- **Network VRU**. (Drop-down list.) The name of the Network VRU to be associated with the Network VRU script.
- **VRU Script Name**. A 39-character, comma-delimited string used by Unified CVP to pass parameters to the IVR Service. The content of string depends on the micro-application to be accessed. For more information on what to specify in thisfield, refer to the following sections:
  - "Play Media (PM) Micro-Application" (page 162)
  - "Play Data (PD) Micro-Application" (page 169)
  - "Get Digits (GD) Micro-Application" (page 178)
  - "Menu (M) Micro-Application" (page 185)
  - "Get Speech (GS) Micro-Application" (page 191)
- **Name**. A unique name for the VRU script. Unified ICME generates a name based on the Network VRU and script names.
- **Timeout**. The number of seconds Unified ICME is to wait for a response from theVRU after invoking the script before assuming that the Unified CVP script has failed.

Warning: This setting is designed to detect VRU failures only; attempting to use it as a technique for interrupting script processing can lead to unexpected results. Use the 180-second default or lengthen the setting to a duration that is longer than the longest time the script is expected to take.

- **Configuration Param**. A string used by Unified CVP to pass additional parameters to the IVR Service. The content of string depends on the micro-application to be accessed. For more information on what to specify in this field, refer to the following sections:
  - "Play Media (PM) Micro-Application" (page 162)
  - "Play Data (PD) Micro-Application" (page 169)
  - "Get Digits (GD) Micro-Application" (page 178)
  - "Menu (M) Micro-Application" (page 185)
  - "Get Speech (GS) Micro-Application" (page 191)
- **Description**. Any additional information about the script.

- **Customer**. (Optional.) A customer associated with the script. For Service Provider solutions, this field is mandatory, due to multiple tenancy solutions (customer-specific data needs to be separated).
- **Interruptible**. (Checkbox.) Whether Unified ICME can interrupt the script (for example, if a routing target becomes available).
- **Overridable**. (Checkbox.) Indicates whether the script can override its own Interruptible attribute. Options: This setting does not apply to Unified CVP micro-applications.

Step 5 When finished, click **Save** to apply your changes.

How to Specify a Run External Script Node that Accesses a Unified CVP Micro-Application

Step 1 Within Script Editor, place the Run External Script object in the workspace, right-click, and open the Properties dialog box.

The Run External Script Properties dialog box lists all Network VRU scripts currently configured

Note: The ICM Script Name column reflects the values defined through the Name field in ICM Configuration Manager's Network VRU Script List tool.

Step 2 Select the **ICM Script/VRU Script Name** you want to execute.

Step 3 Optionally, modify the Comments tab

Step 4 Optionally, modify the Labels tab.

Step 5 When finished, click **OK** to submit the changes and close the dialog box.

Using Unified CVP Micro-Applications

Note: Not all third-party ASR servers use Unified CVP micro-application parameters in the same manner. This affects how third-party ASR servers interact with the Unified CVP micro-applications. For example, although Unified CVP allows timeout parameters to be set to a value in the range of 1 to 99 seconds, a particular ASR server may only support a range of 1 to 32 seconds. Another ASR server requires a "#" to indicate that digits are to be collected before the inter-digit timeout is reached. **Be sure to follow the instructions provided by your third-party vendor.** Also, be sure to test all of your micro-applications before deploying them.

The sections that follow describe the parameters that can be defined through ICM Configuration Manager for each of the six Unified CVP micro-applications.

Keep the following in mind as you configure each Network VRU Script to be used with Unified CVP:

- Each micro-application *parameter* in fields of the Network VRU Script List's Attributes tab must be separated by a comma.
- If a parameter value is not specified, the micro-application uses its default.

Each section concludes with sample ICM Configuration Manager and Script Editor screen captures for the micro-application.

Note: For detailed examples of Unified CVP IVR scripts, refer to "Transferring and Queuing Calls with Unified CVP (page 381)."

How Micro-Applications Use Automatic Speech Recognition (ASR) and Text-to-Speech (TTS)

Unified CVP micro-applications can use ASR in two ways:

- In Get Digits and Menu micro-applications, to recognize data for built-in data types, such as numbers, dates or currency, using digits and/or voice. The **user.microapp.input_type** ECC variable specifies the collection type. The script writer uses this variable in a Script Editor Set node to allow the caller to input DTMF only (**D**) or both DTMF and Voice (**B**, the default). If you are not using an ASR, you need to set this variable to **D**. If you are using an ASR, you can set the variable to either **D** or **B**. Regardless of the value of **user.microapp.input_type**, the recognized digit(s) are always returned to ICM in the CED variable.

  Note: With input_mode set to "B" (both), either DTMF or speech will be accepted, but mixed mode input is not. Once you begin entering with one mode, input via the other mode is ignored and has no effect.

- In Get Speech micro-applications, to collect voice input according to a specified grammar. The grammar to be used is specified either as inline grammar (through the setting in the **user.microapp.grammar_choices** ECC variable) or as an external grammar file (through a text file, the name of which is given in the Network VRU Script's Configuration Param field). The recognized result is returned to ICM in the **user.microapp.caller_input** ECC variable.

Unified CVP micro-applications can use TTS for two purposes:

- As an alternative for playing recorded announcement prompts with the Play Media, Get Digits, Menu, and Get Speech micro-applications, using either the contents of the **user.microapp.inline_tts** or an external .vxml file. (For more information, refer to "How Micro-Applications Use External VoiceXML (page 158).") The ECC variable is useful if the amount of text is relatively short and simple. The external .vxml file is useful for more lengthy text or text that needs to be changed frequently using tools other than the ICM Script Editor
- As a method of playing data using the Play Data micro-application. If the **user.microapp.pd_tts** ECC variable contains **Y**, Unified CVP will use TTS to speak the data (depending on the TTS locale support and capabilities); if **N**, Unified CVP will use the system recorded announcements to speak the data (depending on IVR Service locale support and capabilities).

Note: These ECC variables must be set in the Unified ICME script prior to executing the micro-application that they modify.

How Micro-Applications Use External VoiceXML

The Play Media and Get Speech micro-applications can be used to render external .vxml; that is, text Voice-XML files. To access the external file, the Media File Component of the Network VRU Script's VRU Script Name field must point to a .vxml file and specify **V** as the Media Library Type parameter.

The external VoiceXML file must contain particular call control catch blocks and must not execute call control, as Unified CVP and Unified ICME must be responsible for all call control. (For more information, refer to "External VoiceXML File Contents (page 206).")

Dynamic Audio File Support for Micro-Applications

In ISN 2.0 (an earlier release of Unified CVP), all audio files needed to be specified in the VRU Script Name of the Play Media, Menu, Get Digits and Get Speech micro-applications. Unified CVP gives you the capability to use a single micro-application and specify the prompt using call variables and the ICM formula editor.

To provide dynamic audio file capability, you set the second VRU script parameter to a numeric value, 1-10, prefixed by a dash. You then set the Media Library to either "A", "S", or "V". Unified CVP looks in the corresponding Call.PeripheralVariable for the name of the audio file to play.

When you set the Media Library to "A" or "S", Unified CVP plays the audio file specified by the Call Variable after the "-(number)". For example, if the second VRU Script Parameter is set to "-4", it plays the audio file specified in Call.PeripheralVariable4. This functionality is added for Play Media, Menu, Get Digits, and Get Speech micro-applications.

If you set the Media Library to "V", Unified CVP calls the external VoiceXML file specified by the Call Variable after the "-(number)". If the Script Parameter is set to "-7", for example, it calls the external VoiceXML file specified in Call.PeripheralVariable7.

Note: The "V" option is only supported for the Play Media and Get Speech micro-applications.

Second VRU Script Parameter	Corresponding Call Variable
-1 to -10	Call.PeripheralVariable (1 to 10)

For an example of how to use a dynamic audio file, refer to the following table.

VRU Script Parameter Example	Definition
PM, -3,V	**PM** - Utilizes the Play Media micro-application. **-3** - Plays the file specified in Call.PeripheralVariable3. **V** - Acquires the file from the external VoiceXML Media Library.

Example of Using the Dynamic Prompt

To use the dynamic prompt, do the following:

- In the Set node in a Unified ICME script, set the value of ToExtVXML[0] to:

```
prompt=http://152.217.34.252/en-us/app/Welcome.wav
```

- In the external VoiceXML file specify the following configuration:

```
<?xml version="1.0"?>
<vxml version="2.0">
<form id="BilingualMenu" scope="dialog">
<var name="prompt"/>
<field name="caller_input">
<prompt bargein="true" timeout="3s">
<audio expr="prompt"/>
</prompt>
```

Note: A specific hostname, wav filename, form ID, et cetera, was used for the purpose of this example. Replace these elements with your own configuration settings.

Notes

- If you do not specify a file extension for the file name in the Call.PeripheralVariable, the default media file extension is applied.
- If you set the second VRU script parameter to a value prefixed with a dash and don't specify a file name in the corresponding Call.PeripheralVariable, the IVR Service creates a VoiceXML that does not contain a media prompt.
- If you set the second VRU Script Parameter to a value prefixed with a dash and set the "App Media Library" to **V**, signifying external VoiceXML, you must specify a VoiceXML file in the corresponding Call.PeripheralVariable. If you do not, an "Invalid VRU Script Name" error is returned to ICM. If the specified VoiceXML filename does not contain an extension, and **user.microapp.UseVXMLParams** is not set to **N**, the default extension of .vxml is automatically added.
- You can only specify the name of a single file in the Peripheral Variable. You cannot set this value to a name/value pair.

For more information, refer to the sections on individual micro-applications in this chapter.

Default Media Server for Micro-Applications

In prior releases (before Unified CVP 8.5), the only way to specify a media server for a micro-application was to use the ECC variable **`user.microapp.media_server`**. You may now use the Operations Console to designate a default media server for the entire deployment.

The global default media server may be specified in the Operations Console by selecting **Device Management --> Media Server**. The default media server will be used by the micro-applications if the ECC variable **`user.microapp.media_server`** is missing or empty in the Unfied ICM script.

The following list specifies the order in which the micro-application tries to resolve which media server to use:

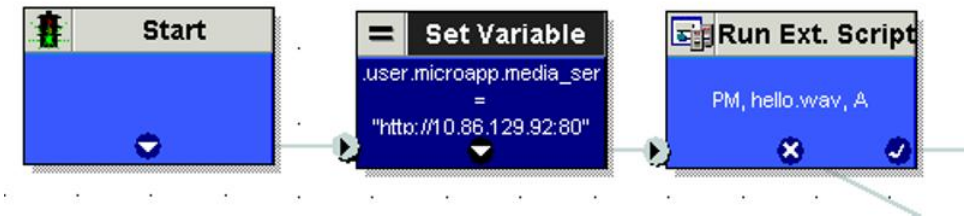
1. Media server is specified by the ECC variable: **`user.microapp.media_server`**
2. Global default media server is specified

The first non-empty media server value encountered in the above order will be used by the micro-application. This applies to all micro-applications including:

- Play Media (PM)
- Play Data (PD)
- Get Digits (PD)
- Menu (M)
- Get Speech (GS) only if the Media Library Type in the VRU Script is set to A (Application) or **`S`** (System) but not **`V`** (ExternalVXML)

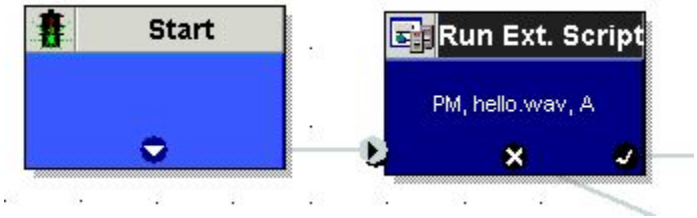
The following screen shot shows the Unified ICM script where Play Media micro-application is used to play a media file using the ECC variable **`user.microapp.media_server`**.

Figure 23: Media Server ECC



The following screen shot shows the Unified ICM script where Play Media micro-application is used to play a media file using a default media server configured in the Operations Console.

Figure 24: Media Server Ops Console



Capture (CAP) Micro-Application

The Capture (CAP) micro-application allows you to trigger the storage of current call data at multiple points in the ICM routing script. The CAP micro-application must be configured as a VRU script, and it is executed using a RunExternalScript node, just as with any other Unified CVP micro-application. The VRU Script Name value is "CAP" or "CAP,xxx," where "xxx" is any arbitrary string to be used if necessary for uniqueness purposes. There is no VRU Script Config string.

Executing a Capture micro-application causes the ICM PG to produce an intermediate termination record. Specifically, it writes a record in the Termination_Call_Detail (TCD) table which includes all current call variables (but not the VRUProgress variable), router call keys, date and time, caller entered digits, et cetera. Together with the TCD record, the Capture micro-application writes a set of records to the Termination_Call_Variable (TCV) table which includes the current values of all ECC variables.

Unified ICME provides no standard reporting templates for TCD and TCV records. These tables are very large and minimally indexed, and are optimized for writing rather than querying, in order to minimally impact call handling throughput. If you plan to report on this data, create off-hours extract processes which copy rows in their raw format into a database which is external to ICM. From there you can organize the tables in the way that best supports your querying requirements.

Some information you need concerning these records:

- TCD records for a given call may be identified because they contain the same RouterCallKeyDay and RouterCallKey. Successive TCD records are ordered by incrementing RouterCallKeySequenceNumber.
- Intermediate TCD records may be identified because they contain a CallDisposition of 53, “PartialCall”. Only the last TCD record for the call contains the actual disposition.
- TCV records corresponding to a particular TCD record may be obtained by joining on TCV.TCDRecoveryKey. This key matches the RecoveryKey value in the TCD record.
- As of Unified ICME 6.0(0), the TCD record’s CallTypeId is populated even for VRU peripherals. This means you can determine the call’s current CallType at the time of each Capture micro-application invocation, as well as at the end of the call.
- In Unified CVP Comprehensive call flow models, these records will be associated with the VRU leg peripheral. If you are doing VRU application reporting, you may want to filter for TCD records which contain the PeripheralID of the ISN VRU leg.

If using the Capture micro-application, keep in mind that it places a heavy demand on ICM resources. Each time you use it, ICM writes one TCD record and multiple TCV records. Though it can conveniently capture the information you need, it is also likely to capture a great deal of extra information which you do not require. If you overuse this micro-application, you may end up placing a heavy load on ICM both in terms of processing time and disk space, which despite the minimal indexing, may nevertheless impact ICM’s ability to handle the expected call load.

Choose carefully where in your scripts you really need to capture information, and that you spread data items into as many different call variables as possible in order to maximize the usefulness of each invocation.

Play Media (PM) Micro-Application

The Play Media (PM) micro-application can be configured to play a message that is contained in a media file or streaming audio file.

How to Configure Network VRU Script for Play Media

Use the ICM Configuration Manager's Network VRU Script List tool's Attributes tab to specify parameters.

Note: DTMF digit type-ahead is not supported by Play Media and Play Data micro-apps when executing in Comprehensive mode (Type 7). However, this feature is supported for Type 5 calls.

Step 1 Configure VRU Script field parameters:

- **Micro-application type**. For Play Media, valid options are: **PM** or **pm**.
- **Media File Name**. Name of the media file to be played (that is, the prompt file) or the name of the external VoiceXML file.

  The valid options are:

  – A file name (for instance, a .wav file).

    Note: Media file names are case-sensitive.

  – **null** - (default) If this field is empty, Unified CVP examines the contents of the **user.microapp.inline_tts** ECC variable. If this ECC variable contains a value, Unified CVP prompts using TTS. If the ECC is empty, no prompt is played.

  – **-(number 1-10)** - Unified CVP plays the file in the corresponding Call.PeripheralVariable file. For example, a value of 2 instructs Unified CVP to look at Call.PeripheralVariable2.

    Note: If you use the - (number 1-10) option and set the Media Library Type to "V," Unified CVP plays the external VoiceXML file specified in the corresponding Call.PeripheralVariable. If you set the value to - (no value) and set the Media Library Type to "A" or "S", the IVR Service creates VoiceXML without a media prompt.

  – **-a** - Unified CVP will automatically generate the media file name for agent greeting when this option is specified. The file name is based on GED-125 parameters received from Unified ICM. This option is only valid if the Media Library Type is not set to **V**.

- **Media Library Type**. Flag indicating the location of the media files to be played.

  The valid options are:

– **A** - (default) Application

– **S** - System

– **V** - External VoiceXML

- **Uniqueness value**. Optional. A string identifying a VRU Script Name as unique.

Step 2 Configure the Configuration Param field parameters:

- **Barge-in Allowed**. Specifies whether barge-in (digit entry to interrupt media playback) is allowed.

  The valid options are:

  – **Y** - (default) barge-in allowed

  – **N** - barge-in not allowed

  Note:

  - Voice barge-in is not supported by Play Media and Play Data micro-applications. However, DTMF barge-in is supported for these micro-applications.

  - Unified CVP handles barge-in as follows: If barge-in *is not* allowed, the SIP/H.323 Service/Gateway continues prompt play when a caller starts entering digits and the entered digits are discarded. If barge-in *is* allowed, the H.323 Service/Gateway discontinues prompt play when the caller starts entering digits. (For more information, refer to "Get Speech and External VoiceXML (page 199).")

- **RTSP Timeout**. Specifies the Real-time Streaming Protocol (RTSP) timeout—in seconds—when RTSP is used.

  The valid range is 0 - 43200 seconds (default is 10 seconds). If the value is set to 0 or a timeout value is not provided, the stream will not end.

  Refer to How to Configure the Play Media Micro-Application to Use StreamingAudio (page 164) below for more details.

- **Type-ahead Buffer Flush**. The Cisco VoiceXML implementation includes a type-ahead buffer that holds DTMF digits collected from the caller. When the VoiceXML form interpretation algorithm collects user DTMF input, it uses the digits from this buffer before waiting for further input. This parameter controls whether the type-ahead buffer is flushed after the prompt plays out. A false value (default) means that the type-ahead buffer is not flushed after the prompt plays out. If the prompt allows barge-in, the digit that barges in is not flushed.

  The valid options are:

  – **Y** - flush the type-ahead buffer

– **N** - (default) do not flush the type-ahead buffer

Note: This parameter is only applicable when using the Cisco IOS gateway with DTMF barge-in. This parameter is not applicable when using external VXML. This parameter is normally used when 2 or more PM and/or PD microapps are used in a loop in the ICM script (such as while in queue for an agent). If the PM and/or PD microapps are enabled for barge-in, one would set this parameter to **Y** to prevent an uncontrolled looping in the ICM script when the user barges in.

How to Configure the Play Media Micro-Application to Use Streaming Audio

Use the ICM Script Editor to configure Play Media (PM) micro-application to play .wav files from a streaming audio server.

Cisco does not sell, OEM, or support any Media Servers. At present, the IOS gateway only supports µ-law wav files in 8 bit format. Media Servers such as RealNetwork's Helix™ Server will serve RTSP broadcast audio streams in the µ-Law format. Refer to Appendix B: Using the Helix Server (page 553) for instructions on how to configure Helix Server for use with CVP.

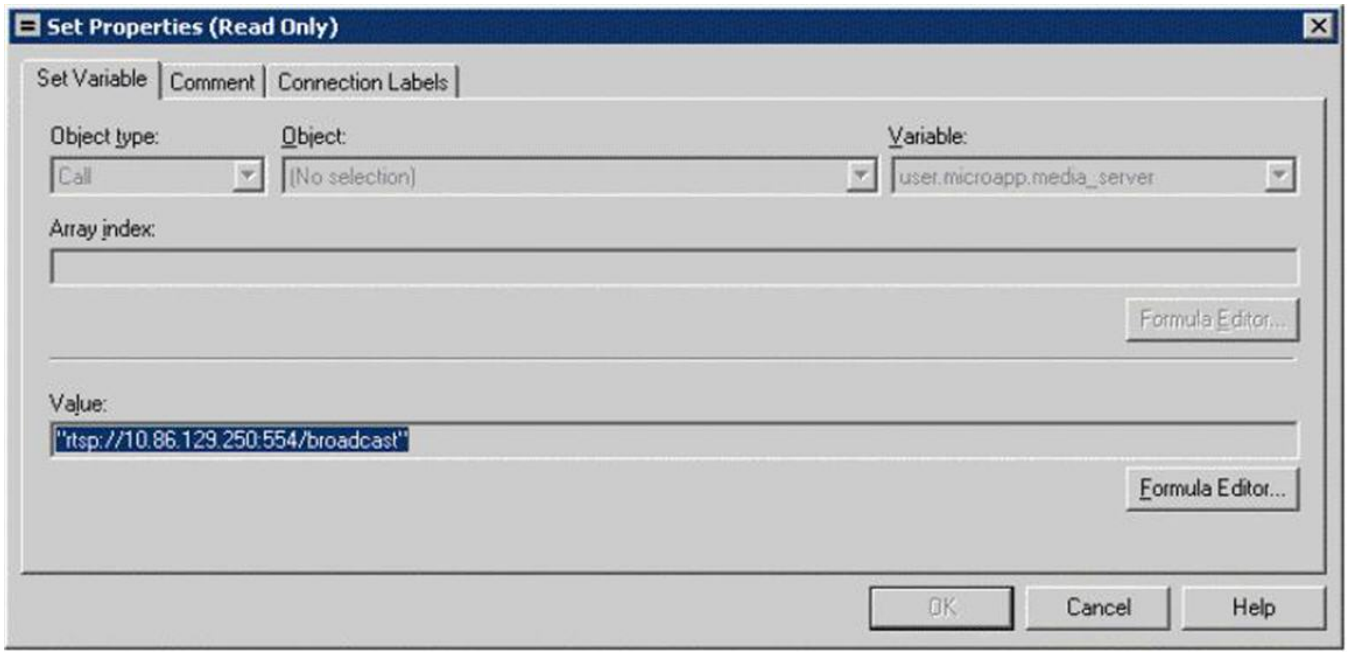
Note:

- At present, the IOS gateway only supports µ-law wav files in 8 bit format.
- You must enclose the stream URL and stream name values in quotation marks.

Step 1 Add a Set Node in the script to configure the media_server ECC variable.

- On the Set Variable tab of the Set Properties dialog box, select **Call** from the Object Type drop down and then set the Variable to **user.microapp.media.server**.

Figure 25: Set Node Dialog: rtsp value



- In the Value field, specify the URL up to, but not including, the stream name.

  Note: The URL must begin with an *rtsp://* prefix (Real-time Streaming Protocol) to stream audio over the network. A trailing forward slash is not permitted in the URL.

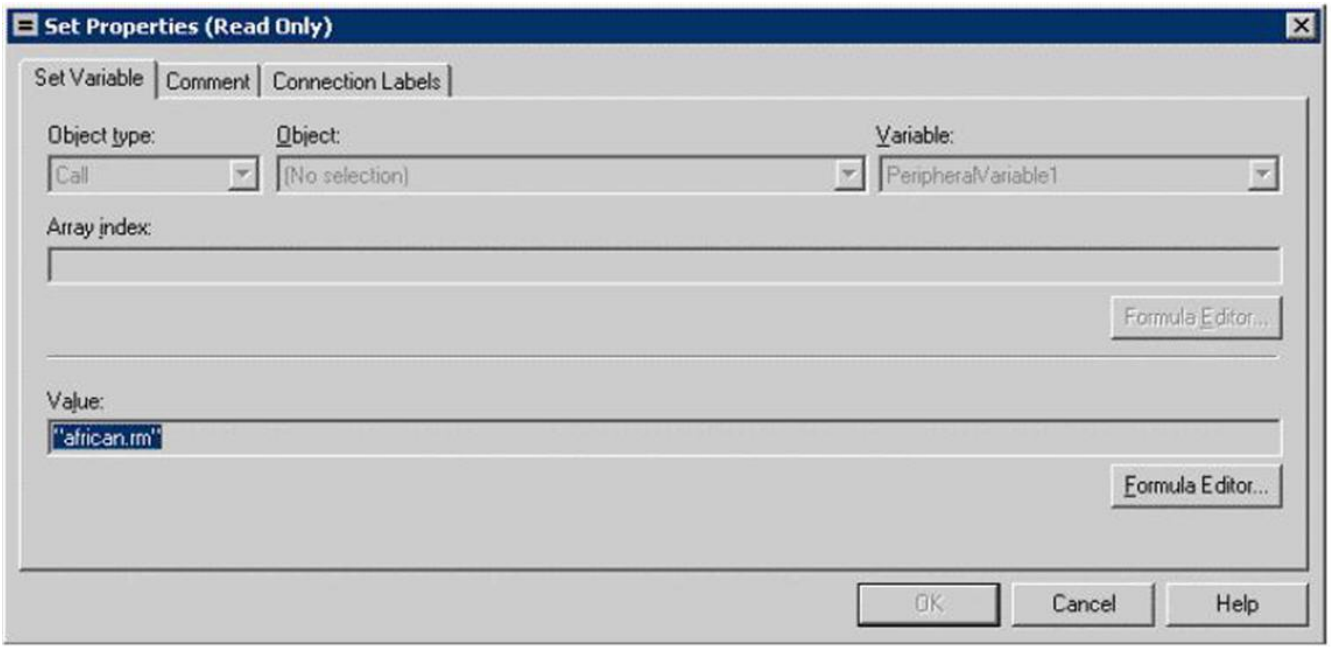
- Click **OK**.

Step 2 Add another Set Node in the script to configure the stream name.

- On the Set Variable tab of the Set Properties dialog box, select Call from the Object Type drop down and set the Variable to **PeripheralVariable<1>**.

  The range for standard ICM Peripheral Variables is PeripheralVariable1 through PeripheralVariables10.

Figure 26: Set Node Dialog: stream name



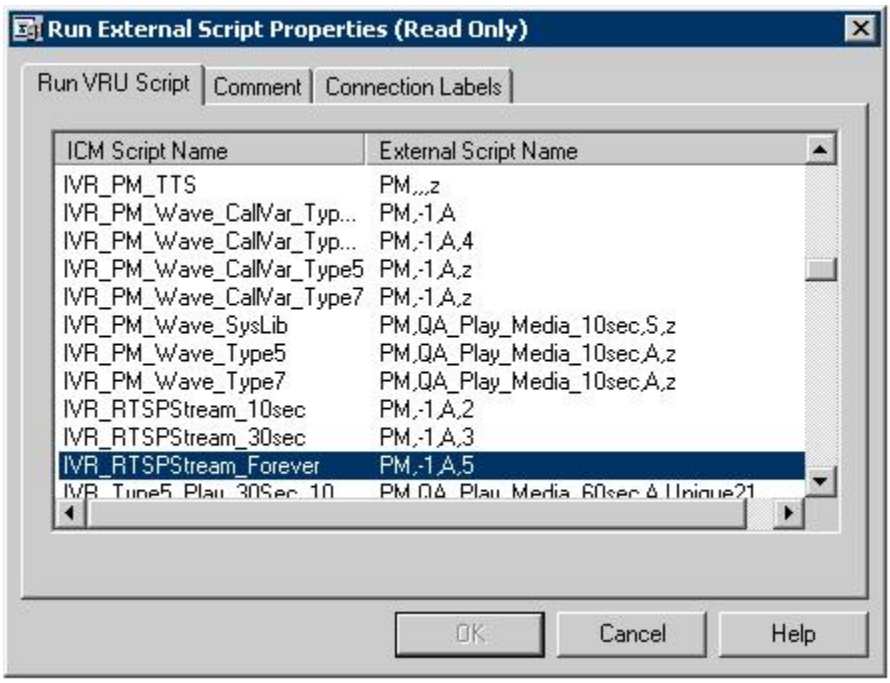
- In the Value field, specify the stream name and click **OK**.

  Note: Stream names are case-sensitive.

Step 3 Add a Run External Script node to the workspace and double-click Run External Script.

The Run External Script Properties dialog box lists all of the Network VRU scripts that are currently configured.

Figure 27: Run External Script Properties Dialog



Note:

- In the example above, the IVR_RTSPStream_Forever script's external script name contains four parameters: PM, -1, A, 5. The second parameter, **-1**, instructs CVP to play the stream name declared in **PeripheralVariable1** (shown in Step 2). It is recommended that you configure streaming audio following the steps outlined so that you may easily change the stream name within the Script Editor, if necessary.

- You can also use the Run External Script node in the ICM Script Editor to configure ICM to failover to a new streaming server. For example, if you want to point to an alternate streaming server (IP address), you would use the X-path out of the Run External Script node to redefine the media_server ECC variable. In a failover situation, the script executes and the stream plays from the targeted streaming server and proceeds normally.

Step 4 From the Run VRU Script tab, select the ICM Script Name desired and click **OK**.

Step 5 Optionally, you can use the ICM Configuration Manager's Network VRU Script List tool's Attributes tab to configure the timeout value for the stream.

Configure the Configuration Param field parameter:

- In the RTSP Timeout field, enter a timeout value (in seconds).
  - The valid range is 0 - 43200 seconds.
  - If the value is set to 0 or a timeout value is not provided the stream will not end.

Step 6 Access the IOS device in global configuration mode and use the **rtsp client timeout connect** command to set the number of seconds the router waits before it reports an error to the Real-time Streaming Protocol (RTSP) server.

The range is 1 to 20. The recommended value is 10 seconds.

If the H.323 Gateway/SIP Call with IVR Service is Terminated with **Reason Code: Q.850;Cause=38** then be sure that the network interface configuration is as follows:

```
ip route-cache same-interface
ip route-cache cef
ip route-cache
ip mroute-cache
no cdp enable
```

If specified, remove the following line from the network interface:

```
keepalive 1800
```

This issue arises if the Unified CVP loses network connectivity, then the VXML Server Gateway is not able to get information from the IVR Service, and as a result a code 38 rejection is generated in the Gateway logs.

See Also

Configuring Custom Streaming Ringtones (page 363)

Appendix B: Using the Helix Server (page 553)

Play Media Examples: Play Welcome Message

The following table shows some Network VRU Script configuration examples for Play Media.

Example	Field Name	Field Contents	Tells Unified CVP...
1	VRU Script Name	**PM,Welcome**	To use the Play Media (PM) micro-application to play the "Welcome.wav" Media file and accept the defaults for remaining settings. **Note:** If no file extension is specified, .wav is assumed.
	Configuration Param	**N**	That Barge-in *is not* allowed.
2	VRU Script Name	**pm,July,S**	To use the Play Media (PM) micro-application to play the "July.wav" Media file, using the System (S) Media library
	Configuration Param	**Null** (Accept default.)	That Barge-in *is* allowed.
3	VRU Script Name	**PM,WebSite,,0**	To use the Play Media (PM) micro-application to play the "Website.wav" Media file, using the default Media Type (Application library), and setting **`0`** as the Uniqueness value. **Note:** A **,** (comma) indicates a skipped parameter. When a parameter is skipped, Unified CVP applies its default.
	Configuration Param	**Null** (Accept default.)	That Barge-in *is* allowed.
4	VRU Script Name	**PM,WebSite,,1**	To use the Play Media (PM) micro-application to play the "Website.wav" Media file, using the default Media Type (Application library), and setting **`1`** as the Uniqueness value.
	Configuration Param	**N**	That Barge-in *is not* allowed.
5	VRU Script Name	**PM,customer.vxml,V,1**	To use the Play Media (PM) micro-application to the external VoiceXML file "customer.vxml", using the VoiceXML Media library, and setting **`1`** as the Uniqueness value.
	Configuration Param	**Note:** Any barge-in setting is ignored when using external VoiceXML.	

Example	Field Name	Field Contents	Tells Unified CVP...
6	VRU Script Name	**PM**	To use the Play Media (PM) micro-application and accept the defaults for remaining settings. **Note:** If the **user.microapp.inline_tts** ECC contains a value, the PM micro-application will play its contents (for example, “Hello world”).
	Configuration Param	**N**	That Barge-in *is not* allowed.
7	VRU Script Name	**PM, -3, A**	To use the Play Media (PM) micro-application, using the file listed in Call.PeripheralVariable3, acquiring the file from the Application (A) media library.
	Configuration Param	**N**	That Barge-in *is not* allowed.
8	VRU Script Name	**PM, -7, V**	To use the Play Media (PM) micro-application, Calls the external VoiceXML listed in Call.PeripheralVariable7, acquiring external VoiceXML (V) media library.
	Configuration Param	**Note:** Any barge-in setting is ignored when using external VoiceXML.	
9	VRU Script Name	**PM, stream.rm**	To use the Play Media (PM) micro-application to play “stream.rm" from a streaming audio server and accept the defaults for remaining settings.
	Configuration Param	**N, 30**	That Barge-in *is not* allowed, and the stream is configured to stop playing in 30 seconds.

Note: Play Media sets the ECC variable **user.microapp.error_code** to zero, indicating success, if control proceeds out the Checkmark (success) branch of the Run External Script node. If control proceeds out the X (failure) branch, Play Media typically sets this variable to one of the codes listed in Unified CVP Script Error Checking (page 150).

Play Data (PD) Micro-Application

The Play Data micro-application retrieves data from a storage area and plays it to the caller in a specific format, called a *data play back type*.

Some possible sources of the data to be played back:

- Information retrieved from a database look-up.
- Information entered by the caller.

Play Data and Data Storage

Before this micro-application can be called, you must specify the location of the play back data. You do this with a Script Editor Set node that points to one of the following storage areas:

- One of the standard ICM Peripheral Variables (PeripheralVariable1 through PeripheralVariables10).
- The **user.microapp.play_data** elements.

How to Configure Network VRU Script Settings for the Play Data Micro-application

Use the ICM Configuration Manager's Network VRU Script List tool's Attributes tab to specify parameters.

Note:

- DTMF digit type-ahead is not supported by Play Media and Play Data micro-apps when executing in Comprehensive mode (Type 7). However, this feature is supported for Type 5 calls.
- Voice barge-in is not supported by Play Media and Play Data micro-applications. However, DTMF barge-in is supported for these micro-applications.
- If you are using integers that are larger than 9 digits, enclose the value in quotation marks, so it will be treated as a string.

Step 1 Configure VRU Script field parameters:

- **Micro-application type**. For Play Data, valid options are: **PD** or **pd**.
- **Data Playback Type**. The kind of the data to be returned ("played") to the caller. The valid options are:
  - **Number**
  - **Char** (character)
  - **Date**
  - **Etime** (elapsed time)
  - **TOD** (Time of Day)
  - **24TOD** (24-hour Time of Day)
  - **DOW** (Day of Week)
  - **Currency**

Note:

- 24TOD and DOW data play back types are not supported when using TTS. Also, currency other than US dollar (USD) is not supported.
- For more information about each of these playback types, including input format and output examples, refer to "Play Back Types for Voice Data (page 172)."

- **Uniqueness value**. Optional. A string identifying a VRU Script Name as unique.

Step 2 Configure the Configuration Param field parameters:

- **Location of the data to be played**. The valid options are:
  - *null* (default) - If you leave this option empty, uses the ECC variable **user.microapp.play_data**.
  - A **number** representing a Call Peripheral Variable number (for example, a 1 to represent Call.PeripheralVariable1).

  Note: For more information on data location, refer to "Play Data and Data Storage (page 169)."

- **Barge-in Allowed**. Specifies whether barge-in (digit entry to interrupt media playback) is allowed.

  The valid options are:

  - **Y** - (default) barge-in allowed
  - **N** - barge-in not allowed

  Note:

  - Voice barge-in is not supported by Play Media and Play Data micro-applications. However, DTMF barge-in is supported for these micro-applications.
  - Unified CVP deals with barge-in as follows: If barge-in *is not* allowed, the SIP/H.323 Service/Gateway continues prompt play when a caller starts entering digits and the entered digits are discarded. If barge-in *is* allowed, the H.323 Service/Gateway discontinues prompt play when the caller starts entering digits. (For more information, refer to "Get Speech and External VoiceXML (page 199).")

  Barge-in works the same for ASR as DTMF. If the caller speaks during prompt play, the prompt play stops. Unlike DTMF input, ASR caller input is checked against the grammar that is defined. If a match is not found, an Invalid Entry error is generated and the caller input is deleted. Voice barge-in is not supported during a Play Media or Play Data script because there is not a grammar specified for these micro-applications.

- **Time Format**

Valid only for the time Data Playback types (Etime, TOD, 24TOD).

The available formats are:

– *null* - leave this option empty for non-time formats

– **HHMM** - default for time formats

– **HHMMSS**

– **HHMMAP** - includes am or pm; valid only for TOD

- **Type-ahead Buffer Flush**. The Cisco VoiceXML implementation includes a type-ahead buffer that holds DTMF digits collected from the caller. When the VoiceXML form interpretation algorithm collects user DTMF input, it uses the digits from this buffer before waiting for further input. This parameter controls whether the type-ahead buffer is flushed after the prompt plays out. A false value (default) means that the type-ahead buffer is not flushed after the prompt plays out. If the prompt allows barge-in, the digit that barges in is not flushed.

  The valid options are:

  – **Y** - flush the type-ahead buffer

  – **N** - (default) do not flush the type-ahead buffer

  Note: This parameter is only applicable when using the Cisco IOS gateway with DTMF barge-in. This parameter is not applicable when using external VXML. This parameter is normally used when 2 or more PM and/or PD microapps are used in a loop in the ICM script (such as while in queue for an agent). If the PM and/or PD microapps are enabled for barge-in, one would set this parameter to **Y** to prevent an uncontrolled looping in the ICM script when the user barges in.

Play Back Types for Voice Data

Configuring how voice data is presented to a caller is an important part of setting up your Unified CVP IVR. The "Data Play Back Types" table below describes each type, along with sample valid values and formats for the supported locales when **not** using TTS:

- **en-us**. English (United States)
- **en-gb**. English (Great Britain)
- **es-mx**. Spanish (Mexico)
- **es-es**. Spanish (Spain)

Locale is selected by setting the **user.microapp.locale** variable.

Note: For information about locale support when usingTTS, check with your third-party vendor.

Keep in mind that any string of characters typically used in the language may need to be spoken back character by character (this includes special keyboard symbols and numbers). If a particular symbol is not used by a particular language, there is still the possibility that a string containing that symbol will be spelled out with a Play Data with Char data type.

For example, assume that an IVR application in the US (a locale of **en-us**) queries a database for an account owner's name and spells the name back to the caller. If the name pulled from the database was "Hänschen Walther," the media files that would need to be pulled from the Media Server would have been derived from a URL including the **en-us** locale. The symbol **ä** has a decimal value of 228, which is different than the symbol a which has a value of 97. It is the translator's task to record the proper word(s) for each symbol to be supported. For detailed information on character translation, refer to "System Media Files (page 536)."

Note: When using TTS, some output format may vary between Unified CVP playback types and TTS playback types. For example, TTS may pronounce a Play Data number "1234" as "twelve thirty four". When not using TTS, the output is "one thousand, two hundred, thirty four". Please check with your third-party vendor on TTS outputs for playback types.

Table 5: Data Play Back Types

Data Play Back Type	Description	Input Format	Output Examples (When Not Using TTS)
Number	Play the stored data as a number.	-###############.###### The leading minus (-) is optional and is played as "minus." The whole number portion of the string can contain a maximum of 15 digits (for a maximum value of 999 trillion, 999 billion, et cetera). The decimal point is represented as a period (.) and played as "point." It is optional if there is no floating portion. The floating point portion of the number is optional and can contain a maximum of 6 digits. Trailing zeros are played.	**en-us** and **en-gb** typical spoken form: • -123 = "minus one hundred twenty three" • 35.67 = "thirty five point six seven" • 1234.0 = "one thousand, two hundred, thirty four point zero" **es-mx** and **es-es** typical spoken form: • -120 = "menos ciento veinte" • 10.60 = "diez coma seis cero" • 1,100 = "mil cien"
Char	Play the stored data as individual characters.	All printable American National Standards Institute (ANSI) characters are supported. **Note:** Code Page 1252 is ANSI standard. It contains ASCII	**en-us** and **en-gb** typical spoken form: • abc123= "A, B, C, one, two, three" **es-mx** and **es-es** typical spoken form: • abc123 = "A, B, C, uno, dos, tres"

Data Play Back Type	Description	Input Format	Output Examples (When Not Using TTS)
		(characters 0-127) and extended characters from 128 to 255.	
Date	Play the stored data as a date.	YYYYMMDD, regardless of locale. **YYYY** options: the range of 1800 through 9999. **MM** options: the range of 01 through 12. **DD** options: the range of 01 through 31. **Note:** The software does not validate the date (for example, 20000231 is valid and played accordingly). However, a failure occurs if any bounds are broken (for example, 34 for month).	**en-us** typical spoken form: • MMDDYYYY format: 20000114 = “January fourteenth, two thousand” **en-gb** typical spoken form: • DDMMYYYY format: 20000114 = “Fourteenth of January, two thousand” **es-mx** and **es-es** typical spoken form: • DDMMYYYY format: 20001012 = "doce octubre dos mil” **Note:** All spoken forms use the proper grammar for the locale.
Etime (elapsed time)	Play the stored data as an amount of elapsed time.	HHMM or HHMMSS Maximum 99 hours, 59 minutes, 59 seconds Leading zeros are ignored.	**en-us** and **en-gb** typical spoken form: • HHMM format: 0830= “eight hours thirty minutes” • HHMMSS format: 083020= “eight hours, thirty minutes, twenty seconds” **es-mx** and **es-es** typical spoken form: • HHMM format: 0205 = “dos horas cinco minutos” • HHMMSSS format: 020101 = “dos horas un minuto un segundo”
TOD (Time of Day)	Play the stored data as a time of day.	HHMM or HHMMSS 24 hour time **HH** options: 00 - 24 **MM** options: 00 - 59 **SS** options: 00 - 59	**en-us** and **en-gb** typical spoken form: • HHMM format: 0800 = “eight o’clock” 0830 = “eight thirty” 1430 = “two thirty” • HHMMSS format: 083020 = “eight thirty and twenty seconds”

Data Play Back Type	Description	Input Format	Output Examples (When Not Using TTS)
			• HHMMAP format: 1430 = “two thirty p.m.” **es-mx** and **es-es** typical spoken form: • HHMM format: 0100 = “una a.m.” • HHMMAP format: 1203 = “doce y tres p.m.” • HHMMSS format: 242124 = “doce veintiuno a.m.”
24TOD (24-hour Time of Day)	Play the stored data as military time.	HHMM or HHMMSS 24 hour time. **HH** options: 00 - 24 **Note:** 24-hour time and military time may have a discrepancy as to valid hours. Unified CVP plays back the value 00 or 24 “as is.” The application developer is free to make alterations to the data passed to the micro-application, if so desired. **MM** options: 00 - 59 **SS** options: 00 - 59 Unified CVP validates the ranges as stated above. For example, if a time ends in 00 minutes (that is, 2300), one would say “hundred hours” (that is, “twenty-three hundred hours”). The range is 0000 (12 a.m.) through 2459 (after midnight) or 0059, if you prefer 1300 equals 1 o’clock in the afternoon. **Note:** The 24TOD play back type is not supported when using TTS.	**en-us** and **en-gb** typical spoken form: • HHMM format: 0815 = “eight fifteen” 2330 = “twenty three thirty” 2300 = “twenty three hundred hours” • HHMMSS format: 233029 =“twenty three thirty and twenty nine seconds” **es-mx** and **es-es** typical spoken form: • HHMM format: 2121 = “veintiuno veintiuno” 2100 = “veintiún horas” **Note:** In Spanish, when a time ends in 00 minutes the spoken form is “hours,” not “hundred hours.” • HHMMSS format: 050505 = “cinco y cinco y cinco segundos”

Data Play Back Type	Description	Input Format	Output Examples (When Not Using TTS)
DOW (Day of Week)	Play the stored data as a day of week.	An integer from 1 through 7 (1 = Sunday, 2 = Monday, et cetera). **Note:** The DOW data play back type is not supported when using TTS.	**en-us** and **en-gb** typical spoken form: • 7 = "Saturday" **es-mx** and **es-es** typical spoken form: • 7 = "Sabado"
Currency	Play the stored data as currency.	Format is [-]15(X)[.2(Y)] where the minus sign is optional as well as the decimal point and the 2 digits after the decimal point. The whole number portion of the string can contain a maximum of 15 digits (for a maximum value of 999 trillion, 999 billion). **Note:** No comma delimiters or currency symbols are recognized. Leading and trailing zeros are played. If a number does not have a decimal point, the"cents" portion of the amount will not be spoken. For example, the spoken form for the input 100 is "one hundred dollars." The grammar rules apply to the currency, not the locale. **Note:** The **user.microapp.currency** ECC variable contains the currency indicator (USD, CAD, EUR, et cetera).	**USD** (US dollar) typical spoken form: • 15.05 = "fifteen dollars and five cents" • 3.00 = "three dollars and zero cents" **Note:** Unified CVP uses the USD_dollar.wav and USD_dollars.wav media files; the dollar.wav and dollars.wav used by ISNVersion 1.0 are no longer installed. **CAD** (Canadian dollar) typical spoken form: • 15.05 = "fifteen dollars and five cents" • 3.00 = "three dollars and zero cents" **EUR** (Euro dollar) typical spoken form: • 1.10 = "one point one zero euro" **GBP** (Great Britain pound) typical spoken form: • 1.10 = "one pound ten pence" **MXN** (Mexican pesos) typical spoken form: • 1.10 = "one peso and ten centavos" **Note:** The default spoken form for a negative amount (for all currency types) is "minus <amount>."

Note: 24TOD and DOW data play back types are not supported when using TTS. Also, currency other than US dollar (USD) is not supported.

Play Data Configuration Examples

The following table shows several configuration examples for Play Data.

Table 6: Play Data Configuration Examples

If the VRU Script Name field setting is…	It means…	If the Configuration Param field is…	It means…
PD,Number **Note:** If you are using integers that are larger than 9 digits, enclose the value in quotation marks, so it will be treated as a string.	**PD** – Use the Play Data micro-app. **Number** – Play back the data as a number.	empty	Play the data in the default ECC, **user.microapp.play_data**, as a number.
PD,Char	**pd** – Use the Play Data micro-app. **Char** –Play back the data as individual characters.	1	**1** – Play the data in Call PeripheralVariable 1 as a character.
PD,Etime,0 **Note:** If you are using integers that are larger than 9 digits, enclose the value in quotation marks, so it will be treated as a string.	**PD** – Use the Play Data micro-app. **Etime** – Play back the data as a Time.	1,,HHMM	**1** – Play the data in Call PeripheralVariable 1 as an elapsed time. **,** – (Skipped parameter) Accept default setting (Y) **HHMM** – Play the time in HHMM format (for example, 8 hours, 30 minutes).
PD,Date	**PD** – Use the Play Data micro-app. **Date** – Play back the data as a Date.	1,N	**1** – Play the data in Call Variable 1 as a date. **N** – No barge-in allowed.
PD,Currency	**PD** – Use the Play Data micro-app. **Currency** – Play back the data as a Currency.	4,N	**4** – Play the data in Call Variable 4 s currency. **N** – No barge-in allowed.

Note:

- Play Data sets the ECC variable **user.microapp.error_code** to zero, indicating success, if control proceeds out the Checkmark (success) branch of the Run External Script node. If

control proceeds out the X (failure) branch, Play Data typically sets this variable to one of the codes listed in Unified CVP Script Error Checking (page 150).

- To enable Text-to-Speech (TTS) as the data play back source, you would need to need to insert a Set node in the script prior to the Run External Script node setting **user.microapp.pd_tts** to **"Y"**.

Get Digits (GD) Micro-Application

The Get Digits (GD) micro-application plays a mediafile and retrieves digits. For example, you could use Get Digits in an application that prompts a caller to enter a password.

Unified CVP passes the retrieved digits back to Unified ICME for further processing using the Caller-Entered Digits (CED) field in the ICM/IVR Messaging interface. (This is available in the Unified ICME script through the variable Call.CallerEnteredDigits.)

How to Configure Network VRU Script Settings for the Get Digits Micro-Application

Use the ICM Configuration Manager's Network VRU Script List tool's Attribute tab to specify parameters.

Step 1 Configure VRU Script field parameters:

- **Micro-application type**. For Get Digits, valid options are: **GD** or **gd**.
- **Media File Name**. Name of the media file or external VoiceXML to be played (that is, the prompt file). The valid options are:
  - A file name (for instance, a .wav file).

    Note: The file name is case-sensitive.
  - **null** - (default) If this field is empty, Unified CVP examines the contents of the **user.microapp.inline_tts** ECC variable. If this ECC variable contains a value, Unified CVP prompts using TTS. If the ECC is empty, no prompt is played.
  - **-(number 1-10)** - Unified CVP plays the file in the corresponding Call.PeripheralVariable file. For example, entering -2 causes Unified CVP to look at Call.PeripheralVariable2.
- **Media Library Type**. Flag indicating the location of the media files to be played. The valid options are:
  - **A** - (default) Application
  - **S** - System

  Note: This value is ignored if using TTS.
- **Uniqueness value**. Optional. A string identifying a VRU Script Name as unique.

Step 2 Configure the Configuration Param field parameters:

- **Minimum Field Length**. Minimum number of digits expected from the caller. The valid options are: **1-32** (the default is **1**)
- **Maximum Field Length**. Maximum number of digits expected from the caller. The valid options are: **1-32** (the default is **1**).

  Note: For information about Maximum Field Length and the DTMF Termination Key, refer to the Note in "Get Digits and Digit Entry Completion (page 183)."
- **Barge-in Allowed**. Specifies whether barge-in (digit entry to interrupt media playback) is allowed.

  The valid options are:

  – **Y** - (default) barge-in allowed

  – **N** - barge-in not allowed

  Note: Unified CVP deals with barge-in as follows: If barge-in *is* not allowed, the SIP/H.323 Service/Gateway continues prompt play when a caller starts entering digits. If barge-in *is* allowed, the H.323 Service/Gateway discontinues prompt play when the caller starts entering digits. (For more information, refer to "Get Speech and External VoiceXML (page 199).")
- **Inter-digit Timeout**. The number of seconds the caller is allowed between entering digits. If exceeded, the system times-out. The valid options are: **1-99** (the default is **3**).

  Note: This value is ignored if using ASR.
- **No Entry Timeout**. The number of seconds a caller is allowed to begin entering digits. If exceeded, the system times-out. The valid options are: **0-99** (the default is **5**).
- **Number of No Entry Tries**. Unified CVP repeats the “Get Digits” cycle when the caller does not enter any data after the prompt has been played. (Total includes the first cycle.) The valid options are: **1-9** (the default is **3**).
- **Number of Invalid Tries**. Unified CVP repeats the “Get digits” cycle when the caller enters invalid data. (Total includes the first cycle.) The valid options are: **1-9** (the default is **3**).
- **Timeout Message Override**. The valid options are:

  – **Y** - override the system default with a pre-recorded Application Media Library file

  – **N** - (default) do not override the system default

  Note: This value is ignored if using TTS.
- **Invalid Entry Message Override**. The valid options are:

- **Y** - override the system default with a pre-recorded Application Media Library file.
- **N** - (default) do not override the system default

Note:

- This value is ignored if using TTS.
- For more information about Timeout and Invalid Entry Messages, refer to "System Media Files (page 536)."

- **DTMF Termination Key**. A single character that, when entered by the caller, indicates digit entry is complete. The valid options are:
  - **0-9**
  - * (asterisk)
  - # (pound sign, the default)
  - **N** (No termination key)

Note:

- For information about Maximum Field Length and the DTMF Termination Key, refer to the Note in "Get Digits and Digit Entry Completion (page 183)."
- This value is ignored if using ASR.

- **Incomplete Timeout**. The amount of time after a caller stops speaking to generate an invalid entry error because the caller input does not match the defined grammar. The valid options are: **0-99** (the default is **3**).

  Note: This value is ignored when not using ASR. If the value is set to 0, the IVR Service treats the NoEntry Timeout as NoError.

Get Digits Configuration Examples

The following table shows several configuration examples for Get Digits for an application that prompts using .wav files and retrieves input through DTMF.

Table 7: Get Digits Configuration Examples

If the VRU Script Name field setting is…	It means…	If the Configuration Param field setting is…	It means…
GD,Password,A,0	**GD** –Use the Get Digits micro-app.	6,12	**6** – Minimum field length. **12** – Maximum field length.

If the VRU Script Name field setting is…	It means…	If the Configuration Param field setting is…	It means…
	Password – Play the Media file named "Password.wav." **A** – Application Media Library. **0** – Uniqueness value.		Accept defaults for all other settings.
GD,Password,A,1	**gd** –Use the Get Digits micro-app. **Password** - Play the Media file named "Password.wav." **A** - Application Media Library. **1** – Uniqueness value.	6,12,N,3,5,2,2,N,Y,#	**6** – Minimum field length **12** – Maximum field length **N** – No barge-in allowed. **3** – Inter-digit Timeout (seconds) **5** – No Entry Timeout (seconds) **2** – Number of no entry tries **2** – Number of invalid tries **N** – Timeout Msg Override **Y**– Invalid Entry Msg Override **#** – DTMF Termination key
Note: The two examples above both play the Password.wav file ("Please enter your password followed by the pound sign.") and collect digits. They differ in that the first example accepts most of the default settings available through the Configuration Param field; the second field does not.			
GD,ssn	**GD** – Use the Get Digits micro-app. **ssn** – Play the Media file named "ssn.wav."	9,9,	**9** – Minimum field length. **9** – Maximum field length. Accept defaults for all other settings.
GD	**GD** –Use the Get Digits micro-app. Since no Media field settings appear after **GD**, Unified CVP examines the contents of the **user.microapp.inline_tts** ECC variable. If this ECC variable contains a value—for example, "What is your account number?"	6,12,N	**6** – Minimum field length. **12** – Maximum field length. **N** – No barge-in allowed. Accept defaults for all other settings.

If the VRU Script Name field setting is…	It means…	If the Configuration Param field setting is…	It means…
	— Unified CVP prompts using TTS. **Note:** If the **user.microapp.inline_tts** is empty, no prompt is played. In turn, if the **user.microapp.input_type** ECC variable is D, Unified CVP will be set to process any DTMF input the customer supplies.		
Note: Type-ahead can only be used with the Get Digits micro-application when **user.microapp.input_type** is set to **D**. For more information, refer to "Get Speech and External VoiceXML (page 199)."			
GD, -4, S	**gd** –Use the Get Digits micro-app **-4** – Calls the file specified in Call.PeripheralVariable4. **S** – Acquires the file from the System media library.	6,12,	**6** – Minimum field length. **12** – Maximum field length. Accept defaults for all other settings.

The following table shows several configuration examples for Get Digits for an ASR/TTS application.

Table 8: Get Digits Configuration Examples

If …	It means …	If the Configuration Param field setting is…	It means …
The **user.microapp.inline_tts** ECC variable contains "What is your account number?" *and* **user.microapp.input_type** contains: **D** (DTMF) *and* The VRU Script Name field contains: **GD**	Use the Get Digits micro-app to play the contents of the ECC variable and collect DTMF input.	6, 12, N,3,5,2,2,N,Y,#	**6** – Minimum field length **12** – Maximum field length **N** – No barge-in allowed. **3** – Inter-digit Timeout (seconds) **5** – No Entry Timeout (seconds) **2** – Number of no entry tries **2** – Number of invalid tries **N** – Timeout Msg Override **Y**– Invalid Entry Msg Override **#** – DTMF Termination key

If ...	It means ...	If the Configuration Param field setting is...	It means ...
The **user.microapp.inline_tts** ECC variable contains "What is your account number?" *and* **user.microapp.input_type** contains: **B** (the default, both DTMF and voice) *and* The VRU Script Name field contains: **GD**	Use the Get Digits micro-app to play the contents of the ECC variable and collect either voice or DTMF input.	6,12,N,,,,4	**6** – Minimum field length **12** – Maximum field length **N** – No barge-in allowed. **,,,,** – Accept defaults for Inter-digit Timeout (seconds), No Entry Timeout (seconds), Number of no entry tries, Number of invalid tries, Timeout Msg Override, Invalid Entry Msg Override, DTMF Termination key **4** – Incomplete timeout
The **user.microapp.inline_tts** ECC variable contains "What is your account number?" *and* **user.microapp.input_type** contains: **B** (the default, both DTMF and voice) *and* The VRU Script Name field contains: **GD**	Use the Get Digits micro-app to play the contents of the ECC variable and collect DTMF or voice input.	6,12,N	**6** – Minimum field length **12** – Maximum field length **N** – No barge-in allowed Accept defaults for all other settings.
Note: Type-ahead can only be used with the Get Digits micro-application when **user.microapp.input_type** is set to **D**. For more information, refer to "Get Speech and External VoiceXML (page 199)."			

Note: Get Digits sets the ECC variable **user.microapp.error_code** to zero, indicating success, if control proceeds out the Checkmark (success) branch of the Run External Script node. If control proceeds out the X (failure) branch, Get Digits typically sets this variable to one of the codes listed in Unified CVP Script Error Checking (page 150).

Get Digits and Digit Entry Completion

Unified CVP tests GD digit entry input against several conditions to determine whether digit entry is complete.

Unified CVP considers digit entry to be complete if the caller enters any of the following:

- The maximum allowable number of digits (when terminator key is not used).

- The maximum number of digits, excluding a terminator key (when terminator key is used).
- Less than the maximum number of digits, followed by the terminator key.
- Less than the maximum number of digits and exceeding the inter-digit timeout.
- Nothing and reaching the no entry timeout.

Caution: It is important that you set up your Unified ICME script to test for all the scenarios mentioned below.

If Digit-Entry Input is Complete

After digit-entry input is complete, Unified CVP validates the digit string to determine if it is >= (greater than or equal to) the minimum length and <= (less than or equal to) the maximum length.

In variable-length data entry, the Maximum Field Length value does not accommodate the termination key. For example, if a GD micro-application is configured to accept a password that is between 6 and 12 digits long and digit-entry completion is indicated through a termination key (or a timeout), the Minimum Field Length setting would be **6**, the Maximum Field Length setting would be **12**, and the DTMF Termination Key would be defined as a single character.

Before passing the result back to the IVR Service, the H.323 Service or SIP Service would discard the termination key (that is, only the password digits will be included in the CED returned to Unified ICME).

Note: In this example, if the 13th digit is entered without reaching the interdigit timeout and the 13th digit is not the terminator key, the extra digits are buffered by the gateway VXML browser and will be consumed by the next digit collecting node (for example: GD or Menu micro-app).

This type-ahead behavior is described online in the Type-ahead Support section of the **Cisco VoiceXML Programmer's Guide** (http://www.cisco.com/en/US/docs/ios/voice/vxml/developer/guide/refgde1.html#wp1049198).

After validating the digit string, Unified CVP does the following:

- If the string is valid, Unified CVP stores the digit string (not including the terminator key) in the Call.CallerEnteredDigits variable, exits the node through the Checkmark (success) branch, and returns control to Unified ICME software.
- If the string is not valid, Unified CVP considers it an invalid entry and does the following:
  - If the Number of Invalid Entry Tries value has not been reached, Unified CVP plays an error message and re-plays the original prompt.
  - If the Number of Invalid Entry Tries value has been reached, Unified CVP stores the last-entered digit string in the Call.CallerEnteredDigits variable, exits the node through the X (failure) branch, sets the **user.microapp.error_code** ECC variable to **16** (Reached Maximum Invalid Tries), and returns control to Unified ICME.

If No Entry Timeout Occurs

If the caller does not enter input and No EntryTimeout period is exceeded, the following happens:

- If the Number of No Entry Tries value has not been reached, Unified CVP plays the "no entry" error message and re-plays the original prompt.
- If the Number of No EntryTries value has been reached, Unified CVP exits the node through the X (failure) branch, sets the Call.CallerEnteredDigits variable to NULL, the **user.microapp.error_code** ECC variable to **17** (Reached Maximum No Entry Tries), and returns control to Unified ICME.

Menu (M) Micro-Application

This micro-application plays a menu media file and retrieves a defined digit. (Menu is similar to the Get Digit micro-application except that it only accepts one digit, which it checks for validity.)

Unified CVP passes the retrieved digit back to Unified ICME for further processing using the Caller-Entered Digits (CED) field in the ICM/IVR Messaging interface.

How to Configure Network VRU Script Settings for the Menu Micro-Application

Use the ICM Configuration Manager's Network VRU Script List tool's Attribute tab to specify parameters.

Step 1 Configure VRU Script field parameters:

- **Micro-application type**. For Menu, valid options are: **M** or **m**.
- **Media File Name**. Name of the media file or external VoiceXML to be played (that is, the prompt file). The valid options are:
  - A file name (for instance, a .wav file)

    Note: The file name is case-sensitive.
  - **null** - (default) If this field is empty, Unified CVP examines the contents of the **user.microapp.inline_tts** ECC variable. If this ECC variable contains a value, Unified CVP prompts using TTS. If the ECC is empty, no prompt is played.
  - **-(number 1-10)** - Unified CVP plays the file in the corresponding Call.PeripheralVariable file. For example, entering -2 causes Unified CVP to look at Call.PeripheralVariable2.
- **Media Library Type**. Flag indicating the location of the media files to be played. The valid options are:

  - **A** - (default) Application
  - **S** - System

  Note: This value is ignored if using TTS.

- **Uniqueness value**. Optional. A string identifying a VRU Script Name as unique.

Step 2 Configure the Configuration Param field parameters:

- A list of **menu choices**. The valid options are:
  - **0-9**
  - * (asterisk)
  - # (pound sign)

  Formats allowed include:

  - Individual options delimited by a / (forward slash)
  - Ranges delimited by a - (hyphen) with no space

- **Barge-in Allowed**. Specifies whether barge-in (digit entry to interrupt media playback) is allowed.

  The valid options are:

  - **Y** - (default) barge-in allowed
  - **N** - barge-in not allowed

  Note: Unified CVP deals with barge-in as follows: If barge-in *is not* allowed, the SIP/H.323 Service/Gateway continues prompt play when a caller starts entering digits. If barge-in *is* allowed, the H.323 Service/Gateway discontinues prompt play when the caller starts entering digits. (For more information, refer to "Get Speech and External VoiceXML (page 199).")

- **No Entry Timeout**. The number of seconds a caller is allowed to begin entering digits. If exceeded, the system times-out. The valid options are: **0-99** (the default is **5**).

- **Number of No Entry Tries**. Unified CVP repeats the "Menu" cycle when the caller does not enter any data after the prompt has been played. (Total includes the first cycle.) The valid options are: **1-9** (the default is **3**).

- **Number of Invalid Tries**. Unified CVP repeats the prompt cycle when the caller enters invalid data. (Total includes the first cycle.) The valid options are: **1-9** (the default is **3**).

- **Timeout Message Override**. The valid options are:

- **Y** - override the system default with a pre-recorded Application Media Library file
- **N** - (default) do not override the system default

- **Invalid Entry Message Override**. The valid options are:
  - **Y** - override the system default with a pre-recorded Application Media Library file.
  - **N** - (default) do not override the system default

Note: For more information about Timeout and Invalid Entry Messages, refer to "System Media Files (page 536)."

Menu Configuration Examples

The following table shows several configuration examples for Menu for use in an ASR/TTS application:

Table 9: Menu Configuration Examples…ASR/TTS Application

If...	It means...	And, if the Configuration Param field contains...	It means...
The **user.microapp.inline_tts** ECC variable contains "Press 1 for Sales and 2 for Support." and **user.microapp.input_type** contains: **D** (DTMF) and The VRU Script Name field contains: **M**	Use the Menu micro-app to play the contents of the **user.microapp.inline_tts** ECC variable and collect DTMF input.	1-2,Y,4,3	**1-2** – Accept the DTMF digits 1 and 2. **Y** – Barge-in allowed. **4** – No Entry Timeout value (in seconds). **3**– Number of no entry tries allowed.
The **user.microapp.input_type** ECC variable contains: **D** (DTMF) and The VRU Script Name field contains: **M,SalesService,A**	Use the Menu micro-app to play the media file named "SalesService.wav" (which is located in the Application Media library) and collect DTMF input.	1-2,N,4,3,2,Y,Y	**1-2** – Accept the numbers 1 and 2. **N** – No barge-in allowed. **4** – No Entry Timeout value (in seconds). **3**– Number of no entry tries allowed.

If...	It means...	And, if the Configuration Param field contains...	It means...
			2 – Number of invalid tries allowed. **Y**– Allow Timeout Msg Override. **Y** – Allow Invalid Entry Msg Override).
The **user.microapp.inline_tts** ECC variable contains "Press or Say 1 for Sales and 2 for Support." and **user.microapp.input_type** contains: **B** (the default, both DTMF and voice) and The VRU Script Name field contains: **M**	Use the Menu micro-app to play the contents of the **user.microapp.inline_tts** ECC variable and collect either DTMF or voice input.	1-2,Y,4,3	**1-2** – Accept the DTMF digits 1 and 2. **Y** – Barge-in allowed. **4** – No Entry Timeout value (in seconds). **3**– Number of no entry tries allowed.
The **user.microapp.inline_tts** ECC variable contains "Press 1 for Sales and 2 for Support." and **user.microapp.input_type** contains: **B** (the default, both DTMF and voice) and The VRU Script Name field contains: **M**	Use the Menu micro-app to play the contents of the **user.microapp.inline_tts** ECC variable and collect DTMF or voice input.	The Configuration Param field contains: **1-2,Y,4,3**	**1-2** – Accept the DTMF digits 1 and 2. **Y** – Barge-in allowed. **4** – No Entry Timeout value (in seconds). **3**– Number of no entry tries allowed.
Note: Type-ahead can *only* be used with the Menu micro-application when **user.microapp.input_type** is set to **D**. For more information, refer to "Get Speech and External VoiceXML (page 199).			

The following table shows several configuration examples for Menu for use in an application where input type is DTMF.

Table 10: Menu Configuration Example…DTMF Application

If the VRU Script Name field setting is…	It means…	If the Config Param setting is…	It means…
M,Banking	**M** – Use the Menu micro-app. **Banking** – Play the Media file named "Banking.wav." **Note:** This file might contain a message such as: "For Checking, press 1. For Savings, press 2. For Money Market, press 3."	1-3	**1-3** – Accept numbers 1, 2, 3. Accept all other defaults (No Entry Timeout, Number of no entry tries, Number of invalid tries, Timeout Msg Override, Invalid Entry Msg Override).
M,Main_Menu	**M** – Use the Menu micro-app. **Main_Menu** – Play the Media file called "Main_Menu.wav." **Note:** This file might contain a message such as: "For information or transactions on checking, press 1. For savings or club accounts, press 2. For other information, press 0. If you know your party's extension, press 9."	0-2/9,,4,2,2	**0-2/9** – Accept numbers 0, 1, 2, and 9. **,** (Skipped parameter) – Accept the default barge-in setting (Y). **4** – No Entry Timeout value (in seconds). **2** – Number of no entry tries allowed. **2** – Number of invalid tries allowed. Accept all other defaults (Timeout Msg Override, Invalid Entry Msg Override)
M,-2,S	**M** – Use the Menu micro-app. **-2** – Plays the file specified in Call.PeripheralVariable2. **S** – Acquires the file from the System media library.	1-3	**1-3** – Accept numbers 1, 2, 3. Accept all other defaults (No Entry Timeout, Number of no entry tries, Number of invalid tries, Timeout Msg Override, Invalid Entry Msg Override).

Note: Menu sets the ECC variable **user.microapp.error_code** to zero, indicating success, if control proceeds out the Checkmark (success) branch of the Run External Script node. If control proceeds out the X (failure) branch, Menu typically sets this variable to one of the codes listed in Unified CVP Script Error Checking (page 150).

Menu and Digit Entry Completion

Unified CVP tests Menu digit entry input against two conditions to determine whether digit entry is complete:

- If a caller enters a digit, Unified CVP checks whether the digit is within the set of valid digits for this menu.
- If a caller *does not* enter a digit, Unified CVP checks whether the No Entry Timeout value has been reached.

Caution: It is important that you set up your Unified ICME script to test for all the scenarios mentioned below.

If Digit-Entry is Complete

After a caller enters a digit, Unified CVP validates the digit against the list of valid menu options that were defined through ICM Configuration Manager. Then Unified CVP does the following:

- If the digit is valid, Unified CVP stores the digit in the Call.CallerEnteredDigits variable, exits the node through the Checkmark (success) branch, and returns control to Unified ICME.
- If the digit is not valid, Unified CVP considers it an invalid entry and does the following:
  - If the Number of Invalid Entry Tries value *has not* been reached, Unified CVP plays the "invalid message" file and re-plays the menu prompt.
  - If the Number of Invalid Entry Tries value has been reached, Unified CVP stores the last-entered invalid digit in the **user.microapp.caller_input** variable, exits the node through the X (failure) branch, sets the **user.microapp.error_code** ECC variable to **16** (Reached Maximum Invalid Tries), and returns control to Unified ICME.

If No Entry Timeout Occurs

If the caller does not enter a digit within the No Entry Timeout period:

- If the Number of No Entry Tries value *has not* been reached, Unified CVP plays the "no entry" error message and re-plays the menu prompt
- If the Number of No EntryTries value has been reached, Unified CVP exits the node through the X (failure) branch, sets the Call.CallerEnteredDigits variable to NULL, the **user.microapp.error_code** ECC variable to **17** (Reached Maximum No Entry Tries), and returns control to Unified ICME.

Get Speech (GS) Micro-Application

The Get Speech (GS) micro-application collects input that can be DTMF-only, Speech, or both input modes, after prompting a caller. The prompt can be generated by a media file or a TTS source.

Note: The Get Speech (GS) micro-application collects voice and DTMF input from the caller. Get Speech supports SRGS and built-in grammars with the exception of the "Digit" grammar which is handled by GetDigit. Use the ICM Configuration Manager's Network VRU Script List tool's Attribute tab to specify parameters. The prompt can be generated by a media file or a TTS source.

Unified CVP passes the input back to Unified ICME for further processing using the **user.microapp.caller_input** ECC variable.

Get Speech and Grammar Specification

There are three ways to specify a grammar in the Get Speech micro-application:

- Include a **Type of Data to Collect** setting in the Get Speech Configuration Param field for built-in grammars such as dates and numbers. If the "Type of Data to Collect" setting is specified, the other grammar options will not be used by the IVR Service. Conversely, if you do not specify a "Type of Data to Collect" setting, then you must include either an inline or external grammar.
- Include an external grammar file name in the Get Speech Configuration Param field's "External Grammar File Name" setting.
- Include a list of inline grammar choices in the **user.microapp.grammar_choices** ECC variable. These grammar choices will only be used if a "Type of Data to Collect" or "External Grammar File Name" setting is not specified.

  Note:

  - One of these grammar options must be used for each micro-application. If no grammar option is specified, an Invalid Config Param error will be sent back to Unified ICME.
  - If you are using an external grammar, be sure to follow the instructions provided by your third-party vendor.

For details on writing an external grammar file, refer to "External VoiceXML File Contents (page 206)."

Note: For the following table, the Configuration Param field is not used if you are using external VoiceXML.

How to Configure Network VRU Script Settings for the Get Speech Micro-Application

Use the ICM Configuration Manager's Network VRU Script List tool's Attribute tab to specify parameters.

Step 1 Configure VRU Script field parameters:

- **Micro-application type**. For Get Speech, valid options are: **GS** or **gs**.
- **Media File Name**. Name of the media file or external VoiceXML to be played (that is, the prompt file). The valid options are:
  - A file name (for instance, a .wav file)

    Note: The file name is case-sensitive.
  - **null** - (default) If this field is empty, Unified CVP examines the contents of the **user.microapp.inline_tts** ECC variable. If this ECC variable contains a value, Unified CVP prompts using TTS. If the ECC is empty, no prompt is played.
  - **-(number 1-10)** - Unified CVP plays the file in the corresponding Call.PeripheralVariable file. For example, entering -2 causes Unified CVP to look at Call.PeripheralVariable2.

  Note: If you use the -(number 1-10) option and set the Media Library Type to "V," Unified CVP plays the external VoiceXML file specified in the corresponding Call.PeripheralVariable. If you set the value to - (no value) and set the Media Library Type to "A" or "S", the IVR Service creates VoiceXML without a media prompt.
- **Media Library Type**. Flag indicating the location of the media files to be played. The valid options are:
  - **A** - (default) Application
  - **S** - System
  - **V** - External VoiceXML. Refer to "Get Speech and External VoiceXML (page 199)."

  Note: This value is ignored if using TTS.
- **Uniqueness value**. Optional. A string identifying a VRU Script Name as unique.

Step 2 Configure the Configuration Param field parameters:

Note: This field does not apply if you are using external VoiceXML. For example, if you are using external VoiceXML, all Configuration Param settings (such as barge-in) will be allowed whether they're set to Y or N.

- **Type of Data to Collect**. A flag indicating the location of the media files to be played. The valid options are:
  - **null** - (default) Leave this option empty if you will be specifying an External Grammar File Name setting.
  - **boolean** - Affirmative and negative phrases appropriate to the current locale.
  - **date** - Phrases that specify a date, including a month, days and year.
  - **currency** - Phrases that specify a currency amount.

    Note: Nuance 8.5 ASR does not support negative currencies in its built-in grammar of datatype "currency."
  - **number** - Phrases that specify numbers. (For example, "one hundred twenty-three.")
  - **time** - Phrases that specify a time, including hours and minutes.

    Note: For information about the format of the currency data returned to Unified ICME in the **user.microapp.caller_input** ECC variable, refer to "Get Speech Data Format (page 197)."
- **External Grammar File Name**. The name of the grammar file that holds the grammar definition for the ASR. The valid options are:
  - **null** - (default) Leaving this option empty implies that an inline grammar, as given in the Type of Data to Collect setting, will be used.
  - A grammar file name. The Gateway retrieves the grammar file from a Web Server using HTTP.

    Note: The file name is case-sensitive.

  Note: For more information about the "Type of Data to Collect" and "External Grammar" settings, refer to "Get Speech and Grammar Specification (page 191)."
- **Barge-in Allowed**. Specifies whether barge-in (digit entry to interrupt media playback) is allowed.

  The valid options are:
  - **Y** - (default) barge-in allowed
  - **N** - barge-in not allowed

  Note: Unified CVP deals with barge-in as follows: If barge-in *is not* allowed, the SIP/H.323 Service/Gateway continues prompt play when a caller starts entering input. If barge-in *is* allowed, the H.323 Service/Gateway discontinues prompt play when the caller starts entering input. (For more information, refer to "Get Speech and External VoiceXML (page 199).")

- **No Entry Timeout**. The number of seconds a caller is allowed to begin entering digits. If exceeded, the system times-out. The valid options are: **0-99** (the default is **5**).

- **Number of No Entry Tries** Unified CVP repeats the "Get Digits" cycle when the caller does not enter any data after the prompt has been played. (Total includes the first cycle.) The valid options are: **1-9** (the default is **3**).

- **Number of Invalid Tries** Unified CVP repeats the prompt cycle when the caller enters invalid data. (Total includes the first cycle.) The valid options are: **1-9** (the default is **3**).

- **Timeout Message Override**. The valid options are:

  – **Y** - override the system default with a pre-recorded Application Media Library file

  – **N** - (default) do not override the system default

  Note: This value is ignored if using TTS.

- **Invalid Entry Message Override**. The valid options are:

  – **Y** - override the system default with a pre-recorded Application Media Library file.

  – **N** - (default) do not override the system default

  Note:

  - This value is ignored if using TTS.

  - For more information about Timeout and Invalid Entry Messages, refer to "System Media Files (page 536)."

- **Incomplete Timeout**. The amount of time after a caller stops speaking to generate an invalid entry error because the caller input does not match the defined grammar. The valid options are: **0-99** (the default is 3).

  Note: This value is ignored when not using ASR. If the value is set to 0, the IVR Service treats the NoEntry Timeout as NoError

- **Inter-digit Timeout**. The number of seconds the caller is allowed between entering DTMF key presses. If exceeded, the system times-out. The valid options are: **1-99** (the default is 3).

  Note: This value is ignored if using ASR.

- **Pass FTP Information** Specifies whether to pass FTP server information to the VXML Server. This option is only useful if the VXML Server application uses the FTP_Client Element and the FTP server information is already configured using the Operations Console. Valid options are:

  – **Y** - Pass FTP server information to the VXML Server as VXML Server session variables.

  – **N** - (default) Do not pass FTP server information.

If the **Pass FTP Information** parameter is set, the following information is passed:

– **ftpServer** - A space separated string of FTP servers. For example, **`ftp_host1|21|username|password ftp_host2`**. Everything is optional except the host name. Refer to FTP_Client Element settings located in the *Elements Specifications for Cisco Unified CVP VXML Server and Cisco Unified Call Studio* guide for more information.

– **ftpPath** - path on the FTP server. By default, this path is formed from the content of the ECC variable **`user.microapp.locale`** concatenated with path separator (**`/`**) and the content of the ECC variable **`user.microapp.app_media_lib`**. One exception is if the value of user.microapp.app_media_lib is **`..`**, then **`app`** is used instead. An example of a path is: **`en-us/app`**

Get Speech Configuration Examples

The following table shows several configuration examples for Get Speech.

Table 11: Get Speech Configuration Examples

If...	It means...	And, if...	It means...
The **user.microapp. inline_tts** ECC variable contains "What is your account value" and **user.microapp. input_type** contains: **B** (the default, both DTMF and voice) and The VRU Script Name field contains: **GS**	Use the Get Speech micro-app to play the contents of the **user.microapp.inline_tts** ECC variable and collect account balance in voice or DTMF input, which it passes in the **user.microapp.caller_input** ECC variable.	The Configuration Param field contains: **Currency,,N,5,2,1**	**Currency** – Collect a string of data in currency format , – Accept the default External Grammar File Name setting (empty). **Note:** You accept the default because you are specifying a Type of Data to Collect parameter (Currency). **N** – No barge-in allowed. **5** – No Entry Timeout (seconds) **2** – Number of no entry tries **1** – Number of invalid tries
The **user.microapp. inline_tts** ECC variable contains "What department do you wish to speak to" and	Use the Get Speech micro-app play the contents of the **user.microapp.inline_tts** ECC variable and collect the answer, which will be passed in either voice or DTMF format in the	The Configuration Param field contains: **,,N**	, – Accept the default Type of Data to Collect parameter (empty) **Note:** You accept the default Type of Data to Collect parameter because you are specifying a value in the External Grammar File Name parameter.

If...	It means...	And, if...	It means...
user.microapp. input_type contains: **B** (the default, both DTMF and voice) and **user.microapp. grammar_choices** contains **Sales/ Customer Service/ Help Desk** and The VRU Script Name field contains: **GS,Department,A**	**user.microapp.caller_input** ECC variable		, – Accept the default External Grammar File Name setting (empty) and use the grammar in the **user.microapp. grammar_choices** ECC variable **Note:** This is an inline grammar example. Each option in the list of choices specified in the **user.grammar_choices** ECC must be delimited by a forward slash (/). **N** – No barge-in allowed.
The **user.microapp. inline_tts** ECC variable contains "What is your name" and **user.microapp. input_type** contains: **B** (the default, both DTMF and voice) and **user.microapp.media_server** contains `http://grammars.com` and **user.microapp.app_media_lib** contains **Boston** and The VRU Script Name field contains: **GS,YourName,A**	Use the Get Speech micro-app play the contents of the **user.microapp.inline_tts** ECC variable and collect the answer, which will be passed in either voice or DTMF format in the **user.microapp.caller_input** ECC variable.	The Configuration Param field contains: **,customers.grxml,N**	, – Accept the default Type of Data to Collect parameter (empty) **Note:** You accept the default Type of Data to Collect parameter because you are specifying a value in the External Grammar File Name setting parameter. **customers.grxml** – Use the grammar in this file **Note:** This is an external grammar file example. For details on writing an external grammar file, refer to "External VoiceXML File Contents." (page 206) **N** – No barge-in allowed.

Note: Get Speech sets the ECC variable **user.microapp.error_code** to zero, indicating success, if control proceeds out the Checkmark (success) branch of the Run External Script node. If control proceeds out the X (failure) branch, Get Speech typically sets this variable to one of the codes listed in Unified CVP Script Error Checking (page 150).

Get Speech and DTMF Input Collection

Contrary to its name, the Get Speech micro-application can also be used to collect DTMF input. For certain grammars, the caller could type a number, time, or currency rather than saying it.

Although the Get Digits micro-application is capable of providing the same type of functionality, it does not allow for validation at collection time. If a caller inputs 2 5 0 0 in response to a Get Speech prompt prompting the caller to enter a time, the Get Speech micro-application would detect that "twenty-five hundred hours" is an invalid entry. With the Get Digits micro-application, this kind of validation would need to be done using additional Script Editor nodes.

Note: The caller cannot mix DTMF and speech in a single input, even if both are enabled. Once he starts talking, he cannot key-in characters, and vice versa.

The following table lists the rules associated with using DTMF collection in the Get Speech micro-application.

Table 12: DTMF Rules for Get Speech

Type of Data To Collect (as specified in the Config Params	Allows DTMF Input?	DTMF Rules[1]
boolean	Yes	Valid DTMF inputs are: **1** (Yes) and **2** (No).
date	Yes	Valid DTMF inputs are: four digits for the year, followed by two digits for the month, and two digits for the day.*
currency	Yes	For DTMF input, the * (asterisk) key represents the decimal point.*
number	Yes	Valid DTMF input includes positive numbers entered using digits and the * (asterisk) key to represent a decimal point.
time	Yes	Since is no DTMF convention for specifying AM/PM, in the case of DTMF input, the result will always end with **h** or **?**
External Grammars	No	None.
Inline Grammars	No	None.

Note: Regardless of the collection type (that is, "voice" or "DTMF"), caller input from Get Speech is always written to the **user.microapp.caller_input** ECC variable.

Get Speech Data Format

The data type determines the format of the information returned to Unified ICME in the user.microapp.caller_input ECC variable:

- **Boolean**. Returned to Unified ICME as "true" or "false."
- **Date**. Returned to Unified ICME as a fixed-length date string with the format *yyyymmdd* where *yyyy* is the year, *mm* is the month, and *dd* is the day.
- **Currency**. Returned to Unified ICME as a string with the format *UUUmmm.mm* or *mmm.mm*, where *UUU* is the three-character currency indicator (for example, USD), and *mmm.mm* is the currency amount with a decimal point.

1) Source: Voice Extensible Markup Language (VoiceXML) Version 2.0, W3C Working Draft, 23 October 2001, This version: http://www.w3.org/TR/2001/WD-voicexml20-2001 1023

Note: Whether *UUU* is used depends both on theASR capabilities and on whether the caller said it unambiguously (for example, "dollar" and "dinar" are ambiguous, and so the UUU segment will not be included in the return value).

- **Number**. Returned to Unified ICME as a string of digits from 0 to 9 which can optionally include a decimal point and/or a plus or minus sign as a prefix to indicate that the numbers is a positive or negative number.
- **Time**. Returned to Unified ICME as a five-character string in the format hhmmx, where hh is hours, mm is minutes and x is one of the following:

  a - AM

  p - PM

  ? - unknown/ambiguous

Get Speech and Entry Completion

The ASR Engine tests Get Speech input entry against two conditions to determine whether entry is complete:

- If a caller enters input, the ASR Engine checks whether the input is within the set of valid grammar for this script.
- If a caller *does not* enter input, the ASR Engine checks whether the No Entry Timeout value has been reached.

Caution: It is important that you set up your Unified ICME script to test for all the scenarios mentioned below.

If Input Entry is Complete

After a caller enters input, the ASR Engine validates the input against the valid grammar set that was defined using the Set node to define the **user.microapp.grammar_choices** ECC variable.

Then Unified CVP does the following:

- If the input is valid, Unified CVP stores the input in the **user.microapp.caller_input** ECC variable, exits the node through the Checkmark (success) branch, and returns control to Unified ICME.
- If the input is not valid, Unified CVP considers it an invalid entry and does the following:
  - If the Number of Invalid Entry Tries value *has not* been reached, Unified CVP plays the "invalid message" file and re-plays the menu prompt.

– If the Number of Invalid Entry Tries value *has* been reached, Unified CVP stores the last-entered invalid digit in the **user.microapp.caller_input** variable, exits the node through the X (failure) branch, sets the **user.microapp.error_code** ECC variable to **16** (Reached Maximum Invalid Tries), and returns control to Unified ICME.

Note: For more information, refer to "Unified CVP Script Error Checking (page 150)."

If No Entry Timeout Occurs

If the caller does not enter input within the No Entry Timeout period:

- If the Number of No Entry Tries value has not been reached, Unified CVP plays the "no entry" error message and re-plays the prompt.
- If the Number of No EntryTries value *has* been reached, Unified CVP exits the node through the X (failure) branch, sets the **user.microapp.caller_input** ECC variable to NULL, the **user.microapp.error_code** ECC variable to **17** (Reached Maximum No Entry Tries), and returns control to Unified ICME.

Get Speech and External VoiceXML

You can use the Get Speech micro-application to pass information to and from an external VoiceXML file. The following table describes how to set the Get Speech script to utilize external VoiceXML.

To set up the Get Speech micro-application to utilize external VoiceXML, set the Media Library Type to "V". The IVR Service creates VoiceXML that calls the external VoiceXML that is specified in the external VoiceXML file name. The URL to the external VoiceXML is formed from a combination of the media_server, locale, App_Media_Lib and external VoiceXML file name. If the VoiceXML file name does not contain a file extension, the default "*.VoiceXML" is used.

If the external VoiceXML is used, the only GetSpeech VRU Script parameters that are used are the:

- "Number of Invalid Entry" errors, and
- "Number of No Entry" errors.

The IVR Service "NoEntry" and "InvalidEntry" retry logic are utilized if the external VoiceXML returns a <noinput> or <nomatch> event, respectively.

Error Handling

The error handling for an external VoiceXML called from the Get Speech micro-application includes the following:

- If you set the "Media Library Type" to "V" and you do not set an "External VoiceXML Name" parameter, an "Invalid VRU Script Name" error is returned to Unified ICME.

Passing Information to the External VoiceXML

There are two methods of passing information to the external VoiceXML, either by <param> elements or URL elements. You can pass up to 1050 characters to the external VoiceXML by using an ECC Variable array.

Table 13: To External VoiceXML ECC Variable Array

ECC Variable Name	Type	Max. Number of Elements	Max. Size of Each Element
user.microapp.ToExtVXML	Array	5	210

This variable array contains a list of semicolon delimited name/value pairs. The following is an example of the syntax:

Table 14: Sample Array Definition

Variable Name	Values
user.microapp.ToExtVXML[0]	"Company=Cisco;Job=technical writer"
user.microapp.ToExtVXML[1]	"Location=Boxborough;Street=Main"
user.microapp.ToExtVXML[2]	"FirstName=Gerrard;LastName=Thock"
user.microapp.ToExtVXML[3]	"Commute=1hour;Car=Isuzu"
user.microapp.ToExtVXML[4]	"BadgeID=2121212"

Unified CVP links all five elements of the "ToExtVXML" array, parses the contents, and then puts each of the name/value pairs in the VoiceXML that it creates.

You define an ECC variable to determine which method to use when passing information to the external VoiceXML.

Table 15: Use External VoiceXML ECC Variable

ECC Variable Name	Type	Max. # of Elements	Possible Values
user.microapp.UseVXMLParams	Scalar	1	• **Y** - (Yes) Use the values in the user.microapp.ToExtVXML variable array elements. • **N** - (No) Append the name/value pairs in user.microapp.ToExtVXML to the URL of the external VXML. Default: "N"

- If **user.microapp.UseVXMLParams** is set to a value other than "Y" or "N," the IVR Service sends a Misconfigured ECC Variable error message to Unified ICME.

- If the **user.microapp.UseVXMLParams** variable is not set, the default method is "VXML Parameters."
- If the **user.microapp.UseVXMLParams** is set to its default value "N," the .vxml file extension is not used.

Using <Param> Elements

All name/value pairs declared in the **user.microapp.ToExtVXML** variable array are added to the VoiceXML file that the IVR Service creates in <param> elements.These <param> elements are how the information is passed to the external VoiceXML. Unlike the URL parameters, these VoiceXML parameters are a complete, 100% VoiceXML solution that does not require media server side scripting.

You must declare all names specified in the name/value pairs as variables in the external VoiceXML; otherwise, the VoiceXML Interpreter produces a error.semantic error that is returned to Unified ICME as error code "10." Using the example above in the Sample Array Definition (page 200) table, you would need to define the following form level declarations in the external VoiceXML.

```
<var name="Company"/>
<var name="Job"/>
<var name="Location"/>
<var name="Street"/>
<var name="FirstName"/>
<var name="LastName"/>
<var name="Commute"/>
<var name="Car"/>
<var name="BadgeID"/>
```

This is the reason that Unified CVP does not allow the Script Writer to specify a name without a value. Specifying only a name and not a value in a ToExtVXML parameter is useless because the external VoiceXML already has the variable defined.

URL Parameter Element

When you use the URL parameter element option, the name/value pairs that you set in the **user.microapp.ToExtVXML** parameter are appended to the URL to the external VoiceXML. The media server side scripting logic parses the URL and passes the parameters to the external VoiceXML document. Unlike the “VXML Parameters,” this is not a 100% VoiceXML solution and requires media-server side scripting.

Using the examples above in the Sample Array Definition (page 200) table, the URL to the external VoiceXML would be in the following form:

```
http://server/en-us/app/MyVXML?Company=Cisco&Job=technical+writer&
Location=Boxborough&Street=Main&FirstName=Gerrard&LastName=Thock&
Commute=1+hour&Car=Isuzu&BadgeID=2121212
```

ECC Variable Array Formula

The equation for figuring out how many bytes you are sending to and from the external VoiceXML is below. You must be careful to keep this calculation in mind so that you do not overload the ECC Variable with too many bytes.

5 + (1 + Maximum_Length) * (Maximum_Array_Size)

For example, if you are sending 3 array elements to the external VoiceXML of the maximum 210 bytes, the equation would look like this:

```
5 + (1 + 210) * 3
5 + (211 * 3)
5 + 633
638
```

Although the maximum number of bytes you can set each variable to is 210, as you can see from the formula above, maxing out each variable would go over the 1050 character limit. Thus, you need to make sure you use this formula to keep your character limit under the 1050 maximum.

You must use a single 210-byte array element in each direction.

Note: Remember that when you are returning a call from the external VoiceXML, the **user.microapp.caller_input** variable is automatically returned.

Notes

- If the **user.microapp.ToExtVXML** array is either not defined on Unified ICME or empty, the IVR Service does not pass any parameters to the external VoiceXML.
- You do not need to define the **user.microapp.ToExtVXML** array to contain 5 elements. However, it must be defined as an ARRAY variable, not a SCALAR variable, even if you are using only one element. The IVR Service can handle any number of array elements up to a maximum of 5.
- If the **user.microapp.ToExtVXML** is either undefined or partially defined, a warning message appears on the VRU PIM console window indicating that the variable is undefined or partially defined.
- Although the array elements are linked together, you can't span a name/value pair to multiple array elements. This is because before you parse for the name/value pairs, the IVR Service inserts a semicolon between two array elements if there is not one.
- The IVR Service produces a "Misconfigured ECC Variable" error if there is not a "=" symbol between two semicolons.
- The IVR Service produces a "Misconfigured ECC Variable" error if the "=" symbol is the first or last character between two semicolons (for example, if there is not a name or a value).

- The IVR Service produces a “Misconfigured ECC Variable” error if the “name” part of the name/value pair contains a space.
- The IVR Service treats each of the name/value parameters as strings. The IVR Service does not check to see if the value parameter is an integer.

Passing Data Back to Unified ICME with External VoiceXML

Unified CVP can return 1050 characters for external VoiceXML.

Note: All other Get Speech nodes are limited to the 210 characters returned in **user.microapp.caller_input**.

The following ECC Variable array has been added:

Table 16: From External VoiceXML ECC Variable Array

ECC Variable Name	Type	Max. Number of Elements	Max. Size of Each Element
user.microapp.FromExtVXML	Array	4	210

The Get Speech micro-app returns up to 1050 characters by populating the **user.microapp.caller_input** variable and each element of the**user.microapp.FromExtVXML** array.

Note: Remember to use the ECC Variable array formula listed above when defining the FromExtVXML variable. You do not need to define the user.microapp.FromExtVXML array to contain 5 elements. However, it must be defined as an ARRAY variable, not a SCALAR variable, even if you are using only one element.

VoiceXML Requirements

External VoiceXML is called via the <subdialog> VoiceXML element. Because of the design of the VoiceXML itself, the variable names in the customer defined VoiceXML must be coordinated with Unified CVP; otherwise, it won’t be possible to pass external VoiceXML data back to Unified CVP. The following table lists the VoiceXML variables and the ECC variables they correspond to.

Table 17: From External VoiceXML ECC Variable Definition

External VoiceXML Variable Name	Unified ICME ECC Variable	Max. Variable Size
caller_input	user.microapp.caller_input	210
FromExtVXML0	user.microapp.FromExtVXML[0]	210
FromExtVXML1	user.microapp.FromExtVXML[1]	210
FromExtVXML2	user.microapp.FromExtVXML[2]	210
FromExtVXML3	user.microapp.FromExtVXML[3]	210

You need to define the following form level declarations in the external VoiceXML.

```
<var name="caller_input"/>
```

When passing the information back to Unified ICME, you need to use the following syntax for caller input:

```
<assign name="caller_input" expr="input$.utterance"/>
```

If the external VoiceXML sets “input” to “sales” and “FromExtVXML2” to “stocks”, for example, the user.microapp.caller_input would be set to “sales” and user.microapp.FromExtVXML[2] would be set to “stocks”.

Notes

- The variables declared in the customer VoiceXML must be named exactly as specified above. VoiceXML is case-sensitive.
- The “caller-input” variable must be declared and used in the VoiceXML. The only way that it is acceptable to not populate this variable is if the external VoiceXML is returning an error event. Examples of error events include <badfetch>, <noinput>, and <nomatch>. If the “caller_input” variable is not set to a value and an error event is not generated, the Unified CVP assumes that a “No Entry” error occurred.
- The FromExtVXML variables are optional. You can use these variables if the “caller_input” variable is not sufficient. If you are not using this additional data then the ECC Variable array does not need to be defined.
- If the user.microapp.FromExtVXML ECC Variable array is undefined or partially defined, a warning message appears when the IVR Service starts up on the VRU PIM console window.
- If you set an “FromExtVXML” variable in the external VoiceXML, the user.microapp.FromExtVXML must be defined on Unified ICME. If it is not, the Unified CVP does not attempt to set the value and an error appears on the VRU PIM console window. However, no error appears in the Unified CVP log files.
- If you need more than 210 characters but less than the full 1050 characters, you can declare the user.microapp.FromExtVXML array to be less than four elements long. If you do this, you only use the corresponding number of External VoiceXML Variables. For example, if you configure a 2 element “FromExtVXML” ECC Variable array, you can utilize the “caller_input”, “FromExtVXML0” and “FromExtVXML1” VoiceXML variables.
- If the external VoiceXML sets a FromExtVXML variable to a value that is longer than the maximum ECC Variable length, an error message appears in the VRU PIM console window and the value is not set. No error appears in the Unified CVP log files.

Sample External VoiceXML Code

This section provides sample external VoiceXML code.

```
<?xml version=”1.0”>
<vxml version=”2.0”>
<var name="caller_input"/>
  <form id=”getcredit”>
<field name=”input”>
```

```
  <prompt>
  What is your credit card type?
  </prompt>
  <help>
  I am trying to collect your credit card type.
  <reprompt/>
  </help>
  <nomatch>
    <return event=”nomatch”/>
  </nomatch>
  <grammar src=”cctype.grxml” type=”application/srgs+xml”/>
  </field>
<field name=”FromExtVXML0”>
  <prompt>
  What is your credit card number?
  </prompt>
  <help>
    I am trying to collect your credit card information.
      <reprompt/>
  </help>
  <nomatch>
    <return event=”nomatch”/>
  </nomatch>
  <grammar src=”ccn.grxml” type=”application/srgs+xml”/>
</field>
<field name=”FromExtVXML1”>
  <grammar type=”application/srgs+xml” src=”/grammars/date.grxml”/>
  </prompt>
  <help>
    I am trying to collect the expiration date of the credit card number you provided.
      <reprompt/>
      </help>
      <nomatch>
        <return event=”nomatch”/>
        </nomatch>
</field>
<block>
  <assign name="caller_input" expr="input$.utterance"/>
    <return namelist=”caller_input FromExtVXML0 FromExtVXML1”/>
</block>
<catch event=”telephone.disconnect.hangup”>
  <return event=”telephone.disconnect.hangup”/>
</catch>
<catch event=”error.badfetch”>
  <return event=”error.badfetch”/>
</catch>
<catch event=”error.semantic”>
  <return event=”error.semantic”/>
</catch>
<catch event = “error.unsupported.format”>
  <return event=”error.unsupported.format”/>
</catch>
<catch event = “error.unsupported.element”>
  <return event=”error.unsupported.element”/>
</catch>
<catch event=”error.unsupported.language”>
  <return event=”error.unsupported.language”/>
</catch>
<catch event = “error.com.cisco.media.resource.unavailable.asr”>
  <return event=” error.com.cisco.media.resource.unavaliable.asr”/>
</catch>
<catch event = “error.com.cisco.media.resource.unavalable.tts”>
  <return event=” error.com.cisco.media.resource.unavailable.tts”/>
</catch>
<catch event = “error.com.cisco.media.resource.failure.asr”>
  <return event=” error.com.cisco.media.resource.failure.asr”/>
</catch>
<catch event = “error.com.cisco.media.resource.failure.tts”>
  <return event=” error.com.cisco.media.resource.failure.tts”/>
</catch>
```

```
<catch event = “error.com.cisco.media.resource”>
  <return event=” error.com.cisco.media.resource”/>
</catch>
<catch event = “error”>
  <return event=”error”/>
</catch>

<form>
</vxml>
```

External VoiceXML File Contents

An external VoiceXML file must obey the following rules:

- It must not use the <transfer>, <exit> or <disconnect> elements. However, it can use the <Goto> and <submit> elements.
- It must have <return> elements at all exit points in the document.
- It must check for all error events and the “telephone.disconnect.hangup” event. Each event handler must have a <return> element that includes the “event” attribute.
- It must contain <catch> event handlers for all events thrown by the Gateway. These catch handlers can have their own customer-defined logic, but they *must* include the statements that are listed in the sample VoiceXML provided in the previous section.

The External VoiceXML document example that follows illustrates the contents of aVoiceXML document that follows these rules.

Note: This example assumes that the VRU Script Name value is **PM,CustomerVXML,V**.

External VoiceXML Document Example

```
<?xml version="2.0" encoding="iso-8859-1"?>
<vxml version="2.0">

<form id="CustomerVXML" scope="dialog">

<block>
<prompt bargein="true">
<audio src="http://webserver.com/myapp/Hello_World.wav"/>
</prompt>

</block>
<catch event="error.badfetch">
<return event="error.badfetch"/>
</catch>

<catch event="error.semantic">
<return event="error.semantic"/>
</catch>

<catch event ="error.unsupported.format">
<return event="error.unsupported.format"/>
```

```
</catch>

<catch event ="error.unsupported.element">
<return event="error.unsupported.element"/>
</catch>

<catch event="telephone.disconnect.hangup">
<return event="telephone.disconnect.hangup"/>
</catch>

<catch event="error">
<return event="error"/>
</catch>

<block>
<return/>
</block>
</form>
</vxml>
```

Note: For a complete explanation of VoiceXML file grammar format, refer to **http://www.w3.org/TR/speech-grammar/**. Also, consult the user documentation for yourASR Server for a list of supported grammar elements.

The example that follows illustrates another external grammar file that might be used to prompt callers for the state that they live in.

External Grammar file <?xml version = 1.0?>

```
<grammar version= 1.0 root= action xml:lang= en-us >
<rule id= action scope= public >
<one-of>
<item> California </item>
<item> Arizona </item>
<item> Connecticut </item>
</one-of>
</rule>
</grammar>
```

After a caller responds with the name of the state she lives in, the ASR Engine determines if the caller said **California**, **Arizona**, or **Connecticut**. If the caller said the name of one of these states, the text listed in the **<item>** element will be passed to the IVR Service—and, in turn—**ICM** software. If a caller responds with a name not included in this list, an invalid entry error is returned to the IVR Service.

Type-Ahead Support for ASR

Type-ahead support for ASR is only supported for DTMF under the following conditions:

- When Unified CVP H.323 Service is the client.
- When the Gateway is the client and input type is set to **D**.

Scripting for Unified CVP with Call Studio

You can use Call Studio to build sophisticated IVR applications which can then be loaded onto a VXML Server machine for execution.

To invoke a VXML Server application, you would create a Unified ICME routing script that:

- Includes a user.microapp.ToExtVXML[0] ECC variable instructing the VoiceXML Gateway to interact with the VXML Server directly to execute the application
- Instructs the application to pass back results to Unified ICME.

This section describes:

- Call Studio and how to use it to pass data to Unified ICME.
- How to integrate Call Studio scripts with Unified ICME scripts.
- How to deploy Call Studio Scripts in Unified CVP.

About Call Studio

Call Studio is an Eclipse-based service creation environment whose output is an intermediary file which describes the application flow.

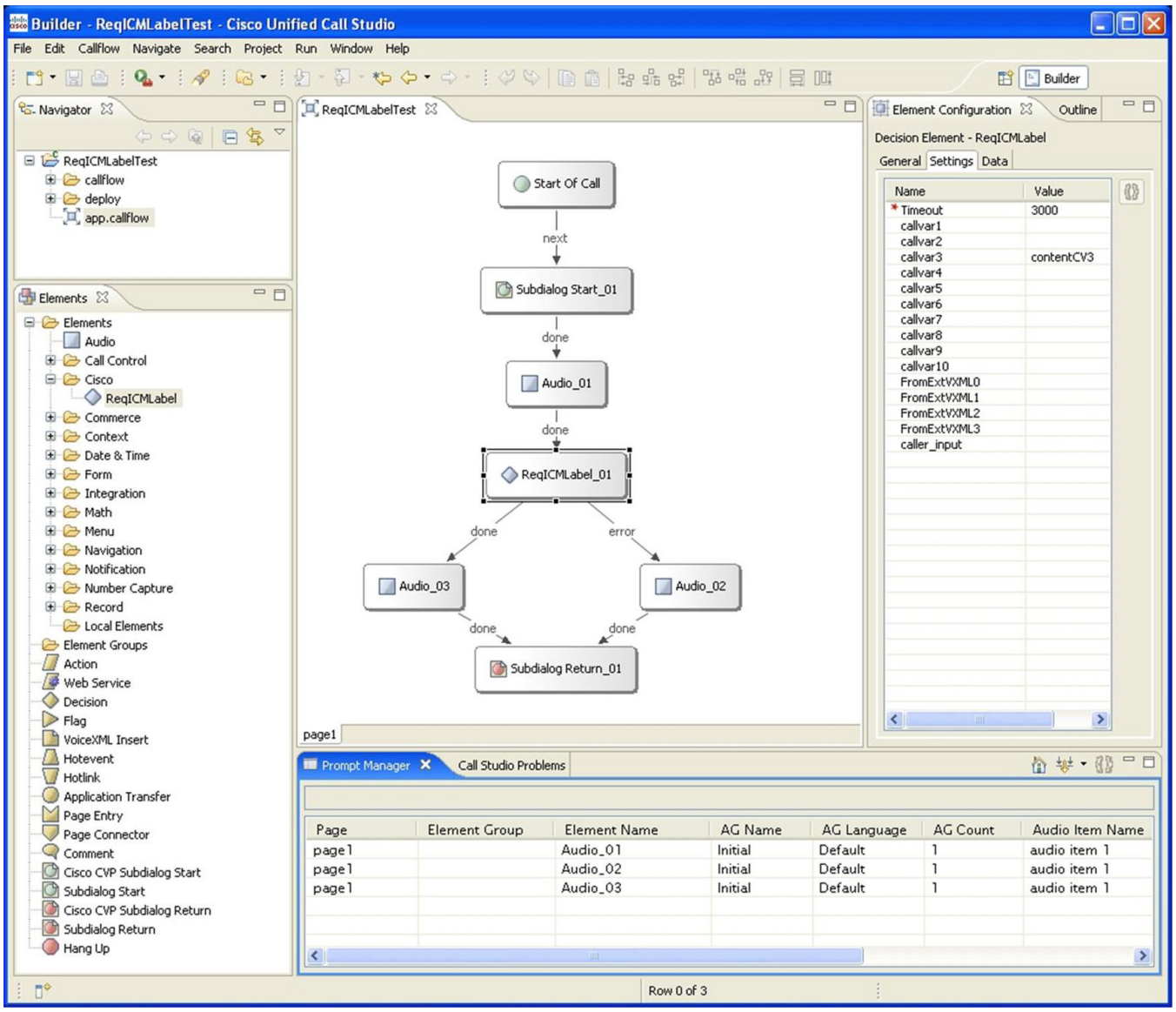
Among its many features, the Call Studio scripting environment :

- Has a drag-and-drop interface with a palette of IVR functions.
- Can perform database queries.
- Can be extended with Java code written to perform any task a Java application can perform.

The following figure shows a Call Studio application that can be used with the Unified CVP Standalone with ICM Lookup call flow model.

Note: For more information about this callflow model, refer to "Configuration Overview (page 25)."

Figure 28: VXML Server (Standalone) with ICM Lookup



Using Call Studio's ReqICMLabel Element to Pass Data

The ReqICMLabel element allows a Call Studio script to pass caller input, Call Peripheral variables, and External Call Context (ECC) variables to a Unified ICME script. The ReqICMLabel must be inserted into a Call Studio script as a decision element. In Call Studio, the returned Unified ICME label contains a result, which can be used by other elements in the same application, such as theTransfer or Audio element. The Transfer element sends instructions to the IOS Voice Browser to transfer the caller to the desired location.

After the ReqICMLabel exits its done path, you can retrieve the values set by the Unified ICME script by selecting the Element Data tab for the ReqICMLabel element.The element data value is {Data.Element.ReqICMLabelElement.result}. ReqICMLabelElement is the name of the ReqICMLabel element in the Call Studio script. The default name for this element is ReqICMLabel_<n>. For example, if you changed ReqICMLabel to GetICMLabel, the value

returned from Unified ICME would be {Data.Element.GetICMLabel.result}, where *result* is the variable of the ReqICMLabel element that contains the Unified ICME label.

Table 18: Settings

Name (Label)	Type	Required	Single Setting Value	Substitution Allowed	Default	Notes
Call Peripheral Variables 1 – 10 (callvar1 – callvar10)	String	No	Yes	Yes		Call Peripheral variables passed by the Call Studio script to the Unified ICME Server. This setting can be a maximum length of 40 characters. The Unified ICME Server returns a name-value pair for up to 10 Call Peripheral Variables in a result. Any value that is placed in callvar<n> from a Call Studio script is returned unchanged, if the Unified ICME Script does not change it.
Call Peripheral Variables Return 1 – 10 (callvarReturn1 – callvarReturn10)	String	No	Yes	Yes		Call Peripheral variables created upon the return of the Unified ICME Label request, regardless of whether or not these variables are filled by the Unified ICME Script. You need two sets of these variables to keep reporting the To ICM Call Peripheral Variables separate from what is returned from Unified ICME.
FromExtVXML0 - 3 (External VXML 0 – External VXML 3)	String Array	No	Yes	Yes		External Call Context (ECC) variables passed by the Call Studio script to the Unified ICME Server. Each variable is a string of name-value pairs, separated by semicolons, for up to 4 external VoiceXML variables. This setting can be a maximum length of 210 characters.
ToExtVXML0 - 4 (External VXML 0 – External VXML 4)	String Array	No	Yes	Yes		External Call Context (ECC) variables received from the Unified ICME script. The Unified ICME Server returns a string of name-value pairs, separated by semicolons, for up to 5 external VoiceXML variables.
Timeout	Integer	Yes	Yes	Yes	3000 (ms)	The number of milliseconds the transfer request waits for a response from the Unified ICME Server before timing out.

Name (Label)	Type	Required	Single Setting Value	Substitution Allowed	Default	Notes
						Note: This value can only be increased or decreased by increments of 500 ms.
caller_input (Caller Input)	String	No	Yes	Yes		This setting can be a maximum length of 210 characters. The caller_input is only passed to Unified ICME from Call Studio.

Table 19: Element Data

Name	Type	Notes
result	String	Unified ICME Label returned from a Unified ICME server. You can use this result as input to other Call Studio elements, such as Transfer or Audio. The element data value is {Data.Element.ReqICMLabelElement.result}.
callvar<*n*	String	Call Peripheral variables that the Call Studio scripts passes to the Unified ICME Server. Valid Call Peripheral Variables are callvar1 – callvar10.
callvarReturn<*n*	String	Call Peripheral variables that the Unified ICME script returns to the VXML Server. Valid Call Peripheral Variables are callvarReturn1 – callvarReturn10. For example, if an Unified ICME script contains Call Peripheral variable 3 with the string value "CompanyName=Cisco Systems, Inc", you can access the value of CompanyName that is returned by the Unified ICME script by using: **`Data.Element.ReqICMLabelElement.callvarReturn3.`** The returned value is "Cisco Systems, Inc."

Table 20: Session Data

Name	Type	Notes
name	String	Value for a name-value pair contained in a ToExtVXML variable returned in the Unified ICME label. You must know which name-value pairs are set in the Unified ICME script to retrieve the correct value from the Call Studio script. For example, if a Unified ICME script contains a user.microapp.ToExtVXML0 variable with the string value "CustomerName=Mantle", specify Data.Session.CustomerName. If the same Unified ICME script contains a user.microapp.ToExtVXML0 variable with the string value "BusinessType=Manufacturing", you can access the customer business type returned by the Unified ICME script by using Data.Session.BusinessType.

Table 21: Exit States

Name	Notes
done	The element execution is complete and the value was successfully retrieved.
error	The element failed to retrieve the value.

Studio Element Folder is "Cisco."

Integrating Call Studio Scripts with Unified ICME Scripts - Traditional Method

This section describes how to integrate the VXML Server into the Unified CVP solution in the traditional way. This process involves:

- Creating a Unified ICME script with ECC variables configured for VXML Server.
- Creating a VRU Script to run in the Unified ICME script.

Note: There are two ways of integrating VXML Server into the Unified CVP solution. The first method is the same as in previous releases where ECC variables are used for specification of the parameters needed for the integration. This method is named as the 'traditional' method in the description below. The second method is new for Unified CVP Release 8.5 and is an alternative method that may be used if the Call Server and the VXML Server are co-located. Unified CVP Release 8.5 continues to support the first method also. The second method is described in the next section (page 214).

The following steps describe how to call the VXML Server from an Unified ICME script in the traditional way.

Step 1 Specify the URL and port number of the VXML Server that you want to reach, for example:

```
http://12.34.567.890:7000/CVP/Server?application=HelloWorld
```

In the example above, **`12.34.567.890`** is the URL and **`7000`** port number; the values are delimited by a colon (:).

Note: 7000 is the default port number for a VXML Server.

Step 2 In the Unified ICME script, first set the media_server ECC variable to:

```
http://12.34.567.890:7000/CVP
```

Step 3 Set the app_media_lib ECC Variable to "..", (literally two periods in quotes).

Step 4 Set the user.microapp.ToExtVXML[0] ECC variable to **`application=HelloWorld`**.

Note: This example indicates that the VXML Server will execute the "HelloWorld" application. To execute a different application, change the value of user.microapp.ToExtVXML[0] accordingly.

Step 5 Set the UseVXMLParams ECC Variable to "**N**."

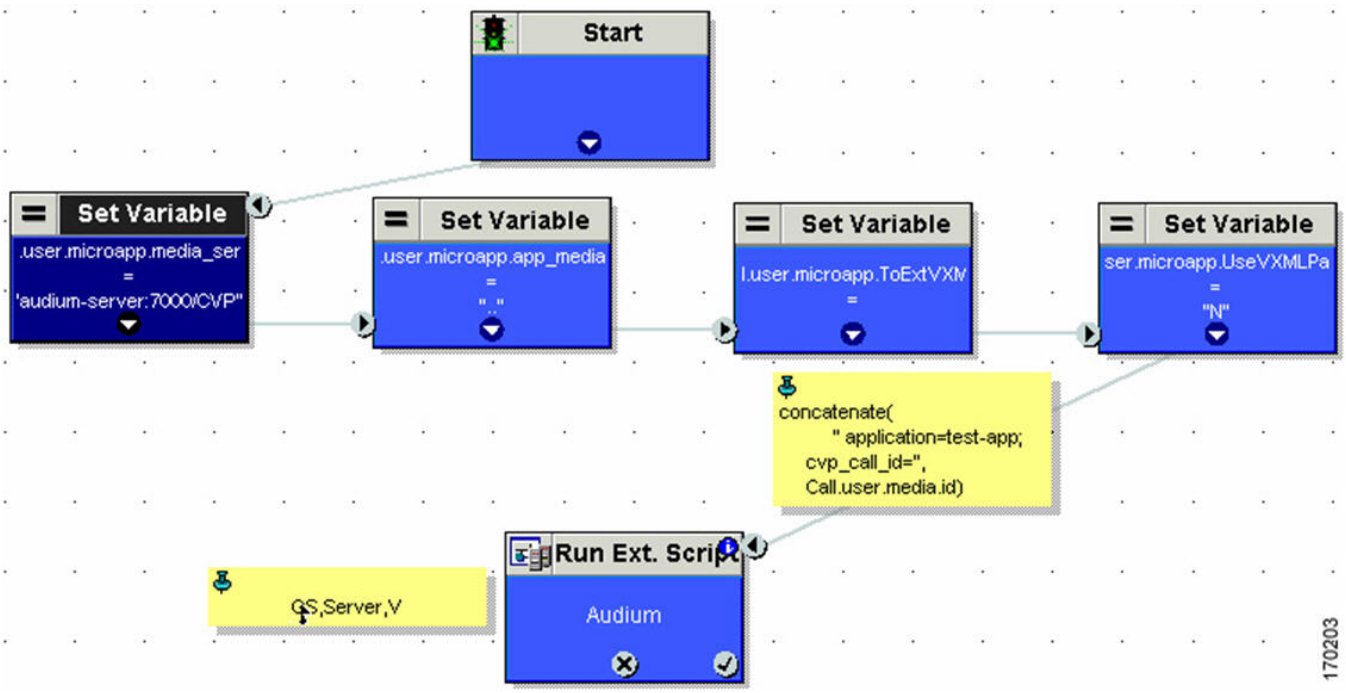
Step 6 Set the concatenate element by following the instructions in "Correlating Unified CVP/Unified ICME Logs with VXML Server Logs (page 271)."

Step 7 Create a Run External Script node within the Unified ICME script with a VRU Script Name value of **GS,Server,V**.

Note: Remember to link this node to the nodes configured in previous steps.

- Configure the timeout setting in the Network VRU Script to a value substantially greater than the length of the timeout in theVXML Server application. (This timeout would only be used for recovery from a failed VXML Server.)
- Always leave the **Interruptible** checkbox in the Network VRU Script Attributes checked; otherwise, calls queued to aVXML Server application might stay in the queue when an agent becomes available.

Figure 29: Sample Unified ICME Script for VXML Server - Traditional



After you configure the Unified ICME script, you need to configure a corresponding VXML Server script with Call Studio. The VXML Server script must:

- Begin with a Unified CVP Subdialog_Start element (immediately after the Call Start element).
- Contain a Unified CVP Subdialog_Return element on all return points (script must end with a Subdialog_Return element).
- The Unified CVP Subdialog_Return element must include a value for the call input.

- To enable reporting, you must add Data Feed/SNMP loggers.

Integrating Call Studio Scripts with Unified ICME Scripts - Simplified Method for Co-located Call Server and VXML Server

This method is available with Release 8.5. It is applicable only if the CVP Call Server and the VXML Server are co-located. This method simplifies the Unified ICME script configuration and reduces the number of script nodes that need to be configured. The GS micro-application will assume that it will invoke the VXML Server that is co-located with the Call Server if the following conditions are met:

- 1. The GetSpeech (GS) micro-application used to invoke a VXML Server application has the following:
  - The "Media File Name" of the GS is set to "Server" e.g., "GS,Server,V". (“Server” is case-sensitive).
  - - The "Media Library Type" of the GS is set to "V" (External VoiceXML).
- 2. The ECC variable user.microapp.media_server has not been encountered in the UCCE script when the GS RunScript node is run.

With this method, the following ECC variables are no longer needed:

- user.microapp.media_server
- user.microapp.UseVXMLParams
- user.microapp.app_media_lib

The following steps describe how to call the VXML Server from an Unified ICME script using the simplified method.

Step 1 Set the user.microapp.ToExtVXML[0] ECC variable to **`application=HelloWorld`**.

Note: This example indicates that the VXML Server will execute the “HelloWorld” application. To execute a different application, change the value of user.microapp.ToExtVXML[0] accordingly.

Step 2 Create a Run External Script node within the Unified ICME script with a VRU Script Name value of **`GS,Server,V`**.

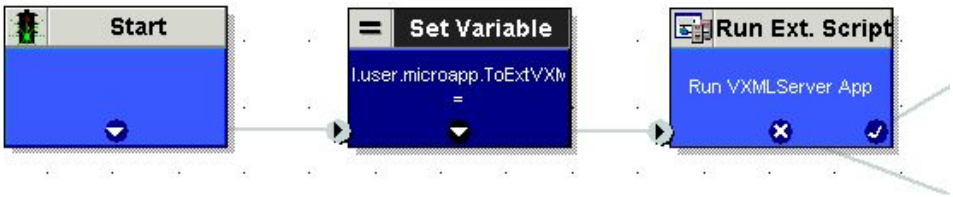
- Configure the timeout setting in the Network VRU Script to a value substantially greater than the length of the timeout in the VXML Server application. (This timeout would only be used for recovery from a failed VXML Server.)

- Always leave the **Interruptible** checkbox in the Network VRU Script Attributes checked; otherwise, calls queued to aVXML Server application might stay in the queue when an agent becomes available.

Step 3 After you configure the Unified ICME script, you need to configure a corresponding VXML Server script with Call Studio. The VXML Server script must:

- Begin with a Unified CVP Subdialog_Start element (immediately after the Call Start element).
- Contain a Unified CVP Subdialog_Return element on all return points (script must end with a Subdialog_Return element).
- The Unified CVP Subdialog_Return element must include a value for the call input.
- To enable reporting, you must add Data Feed/SNMP loggers.

Figure 30: Sample Unified ICME Script for VXML Server - Simplified



- Begin with a Unified CVP Subdialog_Start element (immediately after the Call Start element).
- Contain a Unified CVP Subdialog_Return element on all return points (script must end with a Subdialog_Return element).
- The Unified CVP Subdialog_Return element must include a value for the call input.
- To enable reporting, you must add Data Feed/SNMP loggers.

Deploying Call Studio Scripts in Unified CVP

Call Studio scripts can be deployed in one of the following ways:

- In Call Studio, create and deploy the Call Studio scripts to the local machine using the **Archive** option.
- In the Operations Console, upload the archived Call Studio script file from the local machine to the Operations Server and deploy to other VXML Server machines.

How to Deploy Call Studio Scripts Using Call Studio

Step 1 Create or modify one or more VoiceXML application scripts.

Step 2 Deploy one or more VoiceXML application scripts to the local machine using archive option. The archived scripts are saved as a zipped file under a user-specified directory, for example:

```
C:\Program Files\Cisco\CallStudio
```

Note: The sample folder is C:\Cisco\CallStudio, which is the default folder.

Step 3 Use Call Studio to set up the loggers using the ActivityLogger, ErrorLogger, and Admin Logger tools. Set up the Unified CVP Datafeed logger for each application.

Note: Call Studio also includes CVPDatafeedLogger and CVPSNMPLogger, as well. Call Studio also allows you to change other parameters for these loggers, such as log file size, log lever, et cetera.

See Also

Refer to the Call Studio documentation for more information.

How to Deploy Call Studio Scripts Using the Operations Console

---

Step 1 From the web browser, enter the following URL:

```
https://ServerIP:9443/oamp or http://ServerIP:9000/oamp
```

Step 2 Enter your user ID in the User Name field.

Note: The first time you log in after installing Unfied CVP, enter **`Administrator`**, the default user account.

Step 3 In the Password field, enter your password.

Observe the following:

- If you are logging in to the default Administrator account, enter the password that was set for this account during installation.
- If the user ID or password is invalid, the Operations server displays the message, "Invalid Username or password." Click the link, enter your user ID and password again, and click **OK**.

The main Operations Console Welcome window displays.

Step 4 Select **Bulk Administration > File Transfer >Scripts and Media**.

Step 5 From the Device Association drop down, select **Gateway**.

Step 6 In the Available pane, select one or more archived script files to deploy.

Step 7 Click the **arrow icon** to move the file from *Available* to **Selected**.

Step 8 Click **Transfer** to transfer the selected archived scripts file(s) to the selected device.

Chapter 4

Using Cisco Serviceability Tools

This chapter presents an overview of Cisco Support Tools, including the tools available with Unified CVP solutions running on Windows environments. It also presents the newer serviceability tools that use the Web Services Manager (WSM.

This chapter contains the following topics:

About Cisco Support Tools and Unified CVP

The ST Server communicates to a Unified CVP client node using the *Support Tools Protocol Adapter* (STPA).

STPA provides a subset of the standard Support Tools utilities:

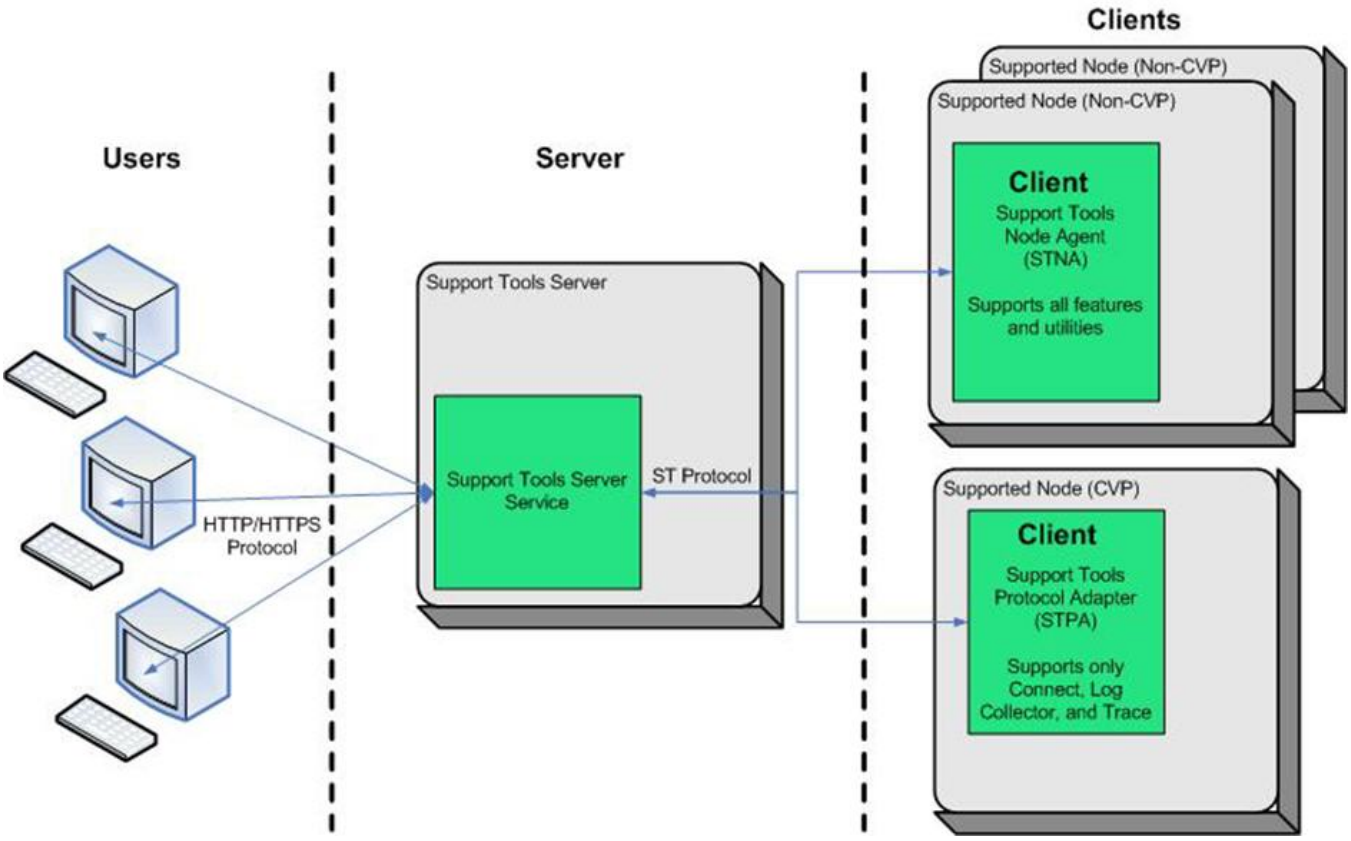
- **Connect**. This tool permits the ST Server to determine whether a node is reachable. It also supports Support Tool's Automatic Node Detection feature, which will add any new server nodes defined by Unified CVP Operations Console to the list of supported nodes.

- **Log Collector**. This tool accepts requests for logs within a certain time window. The tool extracts the desired records, places the results in a ZIP file, and sends them to the server.

- **Trace**. This tool accepts requests to change the trace level of a selected software component.

The following figure illustrates how the ST Server interacts with client nodes.

Figure 31: The Support Server and Clients



In addition to the STPA module that is now part of Unified CVP, the Unified CVP installer, when run on a Windows platform, automatically installs the Windows version of the Support Tools Node Agent (called STNA) on the Windows server. If the STPA receives any requests for commands that are *not* handled by the STPA, the STPA will attempt to forward the requests to the local STNA. This will enable other Support Tools commands to work when Unified CVP is installed on Windows servers.

Cisco Unified Contact Center Security Wizard

The Cisco Unified Contact Center Security Wizard is a new security deployment tool for Unified ICME and Unified ICME, introduced after the publication of Unified ICME 7.2, that simplifies security configuration through its step-by-step wizard based approach.

The Security Wizard is a new graphical user interface to configure security by means of Unified ICME and Unified ICME security command-line utilities:

- Security Hardening Utility
- Windows Firewall Utility
- Network Isolation Utility

The Security Hardening and Windows Firewall utilities are two command-line security utilities that have existed since the 7.0 release. The Network Isolation Utility was introduced after the Unified ICME 7.2 release. The Security Wizard installer installs both the Security Wizard and the Network Isolation Utility.

The Cisco Unified Contact Center Security Wizard works with Unified ICME 7,0, 7.1, and 7.2. That is, all three security utilities within the wizard (the Security Hardening Utility, the Windows Firewall Utility, and the Network Isolation Utility) can be used in Unified ICME 7.0, 7.1, and 7.2.

See Also

Refer to **Security Best Practices Guide for Cisco Unified ICM/Contact Center Enterprise & Hosted** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/prod_technical_reference_list.html) for a complete description of the Security Wizard and the Cisco Network Isolation Utility and for how to use them.

Using Cisco Support Tools with Unified CVP

Cisco Support Tools is Cisco's Unified Contact Center serviceability application. It provides a common, web-based UI to a number of serviceability tools and works across all Cisco Unified Contact Center products.

Support Tools uses a client/server model:

- The Support Tools (ST) Server provides a web-based front-end for users to logon to and make requests for serviceability tasks, such as log collection and trace setting, from other Cisco Unified Contact Center products.

Caution: You must install ST Server on a separate machine from the Unified CVP software. If you try to install ST Server on a machine on which Unified CVP resides, or try to install Unified CVP on a machine on which ST Server resides, the installation process aborts.

- The ST Server connects to Support Tools Node Agents (STNA) installed on each supported computer running a Cisco Unified Contact Center product. These agents listen for serviceability requests, fulfill them, and send back the results to the ST Server.

  Note: The installation of the ST Server and STNA can automatically configure IPSEC when run on Windows servers; you must take care to use the same preshared key during each Unified CVP installation. For more information on IPSEC, refer to Configuring and Modifying Unified CVP Security (page 235).

For additional information about Cisco Support Tools, refer to **Cisco Support Tools** (http://www.cisco.com/en/US/products/ps5905/tsd_products_support_series_home.html).

Accessing Cisco Support Tools from the Operations Console

You launch the Cisco Support Tools application from the Operations Console.

Step 1 Select **Tools > Support Tools**.

Note: The Support Tools options will only work if it has been configured. You must first specify the URL of the Support Tools webpage to launch; select **Tools > Configure** option. (For more information, refer to the Operations Console online help.)

A web browser window opens displaying the Cisco Support Tools Dashboard.

Step 2 Select the Support Tools utility you want to work with. Refer to **Cisco Support Tools User Guide for Cisco Unified Software** (http://www.cisco.com/en/US/products/ps5905/products_user_guide_list.html) for a list of available Support Tools utilities.

See Also

For additional information about using Cisco Support Tools, refer to **Cisco Support Tools** (http://www.cisco.com/en/US/products/ps5905/tsd_products_support_series_home.html).

Web Services Manager (WSM)

Unified CVP supports a new service layer referred to as the Web Services Manager (WSM). WSM interacts with various subsystems and infrastructure handlers, consolidates the responses, and publishes an xml result. The Web Services Manager supports https requests and sends a predefined XML response. WSM is installed on every Unified CVP device and runs automatically as a Windows service. For a device to be managed by WSM, the device must be deployed from the Operations Console.

Note: System CLI uses WSM to collect and present the data available to WSM from the various Unified CVP components.

Creating a WSM User

When Unified CVP is installed, a new user called **wsmadmin** is created with the same password as the Operations Console user. You can create and manage additional WSM users using the Operations Console.

Once you have devices deployed in the Operations Console, you can log on to any server where WSM is installed and access the System CLI. Refer to Unified System CLI (page 223).

To create an additional WSM user:

Step 1 Log into the Unified CVP Operations Console and select **User Management > Users**.

Step 2 Click **Add New**.

Step 3 Provide a **Username** and **Password**.

Step 4 Click the **User Groups** tab.

Step 5 In the **Available** panel, highlight *ServiceabilityAdministrationUserGroup*, then click the right arrow to move that group to the **Selected** panel.

Step 6 Click **Save**.

Unified System CLI

Beginning with Release 8.0(1) Unified CVP supports a new serviceability Command Line Interface (CLI) called Unified System CLI (System CLI). The System CLI enables you to collect diagnostic information (health and status) on Unified CVP servers and to collect device-specific information from each supported node connected to the Unified CVP server from which you are using System CLI. The System CLI accesses a new web services layer in Unified CVP called the Web Services Manager. System CLI commands can be run on a local server, or commands can be executed on a remote server. You can also obtain information from **all** the devices in your CVP system by switching to *system* mode. (Devices must first be configured and deployed in the Operations Console.)

Note:

- To quickly access and use the System CLI, refer to Quick Start Exercise: Accessing and Using the Unified System CLI and its Help (page 224)
- In addition to the System CLI, which is automatically installed with the Unified CVP installation, you can also obtain a GUI-based client, the Analysis Manager (part of the Unified Communication Manager product). For more information on the Analysis Manager, refer to **Cisco Unified Communications Analysis Manager User Guide** (http://www.cisco.com/en/US/docs/voice_ip_comm/uc_system/UC_Analysis_Manager801/Analysis_Manager_User_Guide801.pdf). For instructions specific to configuring Analysis Manager with Unified CVP, refer to section Integrate Analysis Manager with Unified CVP (page 557)

The System CLI is installed on all CVP servers. You can leverage the WSM and CLI functionality to collect diagnostic details like server map, version information, licenses, configuration, components, sessions, logs, traces, performance factors, and platform information for each Unified CVP Device, on a component and sub-component level. You can also set or reset debug levels using CLI on a component and sub-component level.

The System CLI provides both a local mode and a system mode:

- The local mode accesses data about the devices associated with the server you are logged into. Local mode is the default mode accessed automatically when you log into the Unified CLI.
- The system mode, accessed by typing the **`system`** command at the CLI prompt, provides access to all the devices in your Unified CVP deployment solution. In system mode, the System CLI automatically detects the Operations Console and extracts solution topology based on the devices configured in the Operations Console. Based on options you enter for a given command, System mode can be limited to a certain device group or list of servers.

Note: Before you can use the system CLI to obtain information about a device, that device must be listed in and deployed by the Operations Console.

The System CLI commands (for example **`show all`** and **`show component`**) enable you to view and zip all necessary logs or configurations on a specific server, servers, or groups of servers, and store that data on a local disk.

Quick Start Exercise: Accessing and Using the Unified System CLI and its Help

The Unified System CLI is installed on all Unified CVP Servers. Using the CLI client on a Unified CVP device enables you to connect to the local server, a remote server, or, when using *system* mode, to servers defined and deployed in the Operations Console.

Note:

- To be able to log in to the System CLI on Unified CVP, the WSM service must be up and running. By default WSM service is always running.
- The System CLI only provides information on devices that have been configured, saved, and deployed in the Operations Console. Also, if you change the configuration of a device, you must save and deploy the revised configuration before it is available to the System CLI.

Complete the following example session to quickly learn how to use the Unified System CLI.

Step 1 **CLI Launch and Login.** To launch Unified System CLI on **any** CVP server, log into a Unified CVP server through windows (you can use tools like VNC or Remote Desktop console).

Select **Start > Programs > Cisco Unified Customer Voice Portal > Unified System CLI**.

A CMD window displays with an *Enter Username* prompt.

Step 2 Log into the Unified System CLI by entering the default username, **wsmadmin** (added during Unified CVP installation), or a username and password that you created. See Creating a System CLI User (page 222).

After logging, in you should see the following message and prompt:

```
Welcome to the Platform Command Line Interface
admin:
```

Step 3 **CLI Usage Example.** You can now receive Web Services Manager data from the local machine using the system CLI commands. The following example shows a user issuing the **show tech-support** command.

```
Enter username[wsmadmin]:wsmadmin
Enter password:

   Welcome to the Platform Command Line Interface

admin:show tech-support
Warning: Because running this command can affect system performance,
Cisco recommends that you run the command during off-peak hours.
```

```
Do you want to continue? [y/n]: y

Retrieving [version] data from device [localhost] ProductType [cvp]
...
Retrieving [component] data from device [localhost] ProductType [cvp]
 ...
Retrieving [log] data from device [localhost] ProductType [cvp] ...
Default time range is last 24 hours.
...
Output is saved to "C:\Cisco\CVP\wsm\CLI\download\clioutput0.zip"
```

Note:

- The **show tech-support** command creates a single zip file in the directory *%CVP_HOME%\wsm\CLI\download*.
- By default, this command collects the traces for the last 24 hours. Use the **reltime** parameter to change that period. For example, if you want to pull traces for the last 3 days, use the following command: **show tech-support reltime 3 days**.

Step 4 **CLI Command Options.** Each System CLI command has a set of **command options**. Each option consists of a keyword and set of values. The following is an example of how to use a command with options.

In this example, the keyword *component* has a value *cvp:CallServer* and the keyword *subcomponent* has a value *cvp:ICM*. The following command tells the System CLI to get the configuration data for component *CallServer* and subcomponent *ICM* in the Unified CVP deployment.

```
admin:show config component cvp:CallServer subcomponent cvp:ICM
Downloading Configuration file: [ICM: icm.properties] ...
ICM.icmGarbageCollectorInterval : 120
ICM.locationDelimeter : --
ICM.icmHeartbeatInterval : 5000
...
ICM.preRoutedCallServiceID : 2
ICM.icmVxmlIdleTimeout : 30
```

Step 5 **Redirecting Output.** Almost all commands have a *redirect* option. This option tells the System CLI to redirect the command output to a directory (or a file). If the output is saved in a directory, it is saved in a zip file in that specific directory. The following example saves the zip file to **c:\temp\clioutput.zip**:

```
admin:show version redirect dir c:\temp
```

An advantage to saving to a directory is that the output zip file provides a consistent directory structure. When unzipped, the directory structure enables you to find the required diagnostic data quickly.

If you redirect the output to a file, the information will be stored in the form a of "flat" file similar to what you would see for the screen output. An example of redirecting to a file would be:

```
admin:show version redirect file <filename>
```

Step 6 **System Mode.** If you wish to obtain information from all the Web Services Managers that are part of your Unified CVP environment, you can change to *system* mode to issue system-wide commands.

To enable system mode, type *system* at the prompt.

Note: Also refer to Accessing Help for the System CLI (page 223).

Unified System CLI: System Mode

Enter the CLI's system mode by typing **system** at the command prompt and then execute the commands exactly like the local version of the CLI for interactive mode.

In system mode, the Unified System CLI automatically detects the Operations Console, which acts as a seed device, and extracts the solution topology automatically based on devices configured in the Operations Console. System mode enables the System CLI to iteratively go to each supported box in the background and run the command that was executed by you in system mode.

Note: System CLI initialization for the very first system mode command execution, or for the **system init** command, may take a few minutes to complete, especially when there are a lot of devices in a Unified CVP solution, or if devices are unreachable, or network timeouts are involved.

Optionally, you can limit the system command to execute only on a certain device group or list of servers.

Device group is automatically populated based on:

- Device type (CVP, ICM, IOS, CUP, UCM as an example)
- Device IP/hostname wildcard (LOC-1*, 10.86.129.* as an example for branch office deployments)
- Device pool (defined within the Operations Console)

When the System CLI executes the *system* command, the System CLI queries each device in the list and caches the responses locally during its first time initialization process, or when **system init** is executed. The cache enables the system command to be executed quickly for subsequent sessions.

If an error is reported due to an unreachable destination or incorrect credentials for a specific device during the execution of a system command , then the device will be marked *OFFLINE* in the System CLI cache. Use *system init* to retry this device.

The most common command to receive all information from all the components in a Unified CVP solution deployment is:

```
system show tech-support dtcomponent "ucm:Cisco CallManager|cup:Cisco
 UP SIP Proxy"
```

This command collects everything from all device types except the devices *ucm* and *cup*. For ucm, it applies the device type filter *Cisco CallManager* and for the device *cup* it applies the device type filter *Cisco UP SIP Proxy*.

System CLI Automated Execution

To automatically execute Unified System CLI commands, create a plain text file with the .bat extension as shown in the example below and replace the <password> as highlighted with the actual Operations Console password.

```
REM TECH-SUPPORT-COLLECTION
echo show tech-support > clicmds.txt
echo exit >> clicmds.txt
type clicmds.txt | wsccli.bat inplace nointeractive novalidation
user:wsmadmin passwd:<password>
```

In a batch file, the system command can also be executed by prefixing *system* on any regular command. For example, **system show tech-support**. The entire example would be:

```
REM SYSTEM-TECH-SUPPORT-COLLECTION
echo system show tech-support > clicmds.txt
echo exit >> clicmds.txt
type clicmds.txt | wsccli.bat inplace nointeractive novalidation
user:wsmadmin passwd:<password>
```

The most commonly used command in a Unified CVP solution deployment, to get everything from all solution components, is given below:

```
REM SYSTEM-TECH-SUPPORT-COLLECTION
echo system show tech-support dtcomponent "ucm:Cisco
CallManager\|cup:Cisco UP SIP Proxy" > clicmds.txt
echo exit >> clicmds.txt
type clicmds.txt \| wsccli.bat inplace nointeractive novalidation
user:wsmadmin passwd:<password>
```

This command collects everything from all device types except for the device *ucm* and the device *cup*. For the ucm device, it applies the device type filter *Cisco CallManager* and for the device cup, it applies the device type filter *Cisco UP SIP Proxy*.

Note: You can run a Windows scheduled job and collect traces periodically from one or multiple servers using a schedule. Refer to **How to Schedule Tasks in Windows XP** (http://support.microsoft.com/kb/308569) for information about how to schedule a Windows job.

System CLI Remote Execution

To launch System CLI remotely from your laptop, (or any Windows box), to connect to any Unified CVP server or other solution component box (for example, Unified CM, ICM, IOS, CUP Server, etc.), complete the following steps:

1. Install Unified CVP Remote Operations, if no other CVP software is installed.

   For information about only installing the Remote Operations feature of Unified CVP, refer to the **Installation and Upgrade Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html).

2. In the Operations Console, complete the following steps:

   – Select **System -> Web Services -> Remote Operations Deployment (tab)**.

   – Enter the IP address, hostname, and description information for the remote device.

   – Click **Add** to add the device and click **Save & Deploy** to made this device available for remote operations.

3. Start the System CLI on the remote system.

Note: You can also run a Windows scheduled job and collect traces periodically from one or multiple servers, using a schedule from a remote box.

Accessing Help for the System CLI

The following types of help are available:

- To see an overview of the CLI system help, type help at the admin prompt and press enter.

  admin:**help<enter>**

- To view the list of main commands, enter "?" at the admin prompt.

  admin:**?<enter>**

- To obtain the syntax of the command, that is, to see what additional options are available at any given point in the command's structure, type a "?" at the end of the current syntax. For example:

  admin:**capture start ?**

  The preceding entry then informs you that there are two options at this point in the syntax: **duration** and **<cr>**

- To obtain **detailed** help, enter a portion of the command preceded by the word help. For Example:

  admin:**help show version**

  That command provides detailed information about additional options for both local mode and for system modes.

The following entries show an example of "drilling down" to obtain more detailed help. At the option level, placing *help* in front of the command, provides detailed information, as in the example. Because this command can be used in "local" mode and in "system" mode, the detailed help also gives information about how to limit the command when using system mode.

```
admin:show version ?
Options: redirect
         <cr>

show version redirect ?
Options: dir
         file
```

Example of help for product-specific parameters such as *component*. The help system displays the actual components for your Cisco Unified product as shown in the following example for Unified CVP using system mode of the System CLI.

```
admin(system):show log component ?
Options:cvp:CallServer
        cvp:OAMP
        cvp:ORM
        cvp:Reporting
        cvp:VXMLServer
```

System CLI Troubleshooting

The System CLI may show two types of errors:

- Errors from servers (displayed unchanged)
- Errors from the System CLI itself, for which you need to check the log files in the System CLI directory

Sometimes it is necessary to change the System CLI debug level to "debug" to collect more data about a System CLI error. To do this:

1. Open CLI\conf\cli_log4j.xml and change the word "info" to "debug".
2. Restart the System CLI to reproduce the error.

See Doc Wiki troubleshooting page for more CLI troubleshooting information: **System CLI troubleshooting tips** (http://docwiki-dev.cisco.com/wiki/Category:Unified_CVP%2C_Release_8.0).

System CLI Commands and Parameters

The Unified System CLI is designed to work across multiple Cisco Unified products. For this reason, the meanings of some of the parameters for commands such as*show* vary from product to product. An example is the *components* parameter which varies in meaning because the components of systems such as Unified CVP and Unified ICM are different.

The CLI online help provides product-specific information for each command's parameters. Simply type the command and its parameter, followed by the "?" symbol. An example of this mechanism is given in Accessing Help for the System CLI (page 228).

The following table provides basic information about the Unified System CLI commands.

Command or Parameter	Description
capture	Sets up, starts, and stops, a network packet capture. Capturing network packets with the System CLI can be performed in either local mode (to run on a single machine), or system mode (to run across several machines simultaneously). The **capture start** command has an optional duration parameter, while the **capture stop** command has no parameters. The *duration* parameter indicates when the capture should stop. If no duration is provided, the capture process stops after 1 day. The capture command starts the packet capture on a single Unified CVP device or multiple devices. The capture operation defaults to the interface card that is receiving packets and is used for the Unified CVP server socket and IP address binding. The default setting of the capture command will capture the network packets and save the capture information in the Unified CVP logs folder which can be retrieved using the regular CLI *trace* command.
debug	Modifies the current debug level. **Note:** Debug levels reset to the default level for a system component when that system component is restarted, with the exception of the H.323 Voice Browser. The debug level of the Voice Browser remains persistent when the voice browser process is restarted.
help	Access the online help system overview. Use the "?" character to access command-specific and parameter-specific help. Refer to Accessing Help for the System CLI (page 228).
exit	Exit the CLI.
show	Main command for accessing, displaying, and saving (to a file) data about system configuration and operation. Refer to the next table for descriptions of the sub-level *show* commands.
system	Enter *system* mode, which provides access to all the devices in your Unified CVP deployment solution. Use the *exit* command to return to local mode.

The following table describes the variations of the *show* command.

Show Commands	Descriptions
show all	Shows all the available information in all of the sub-categories listed in this table. However, you can still enter qualifying parameters to restrict the information retrieved.
show component	Shows component-specific information. Available components are based on the Cisco Unfied product type and can be listed by entering the command: `show component ?`
show config	Displays the application configuration.
show debug	Shows the current debug level.
show devices	Shows a list of the devices in your deployment (system mode only).
show license	Shows license information.
show log	Shows log contents. You can narrow this command to specific logs.

show perf	Show system performance statistics.
show platform	Shows platform information.
show sessions	Shows the current active sessions or calls.
show tech-support	Shows information to help tech support in solving issues. This command is equivalent to *show all*. **Note:** When you issue the *show tech-support* command, the System CLI issues the other *show* commands needed to provide all of the system information. During the process of executing the other *show* commands, the System CLI passes component and sub-component parameters to the *show trace* command, but does not include component and sub-component parameters when it executes commands like *show configuration* and *show log*.
show trace	Shows trace file information.
show version	Shows Unified CVP component version information.

Details for Specific Options

This section provides detailed information of certain options that require additional explanations.

Results of the *match* Option when Information is Sent to a Directory Instead of a File

You can use the *match* option with the *show trace* and the *show log* commands to send selected output to a directory. This option enables you to specify text that the CLI should match when examining data in text files. The output is then limited to log information that matches the specified criteria. However, because the system cannot perform a text match to include or exclude information in binary files such as .zip files, these files are *included* in your *show* command execution, since they may contain pertinent information.

If you send the output of the *match* selection process to a file instead of a directory, the output only includes the actual text the CLI command would return to the screen.

Comparison of the *component* Option with the *dtcomponent* Option

The *component* option is used to limit the output of a command to the results from specific system components.

Using the command **show trace component ucm:CallManager** and the system components shown in the following list, you would receive information as indicated in the list.

- Unified CVP Call Server: **No Information Returned**
- Unified Call Manager: **Information Returned**
- Unified CM Tomcat Server: **No Information Returned**

The command **show trace component ucm:CallManager** would only return information from the Call Manager.

The option *dtcomponent* (device type component option) only restricts the information from the given device type, while allowing data to be returned from other device types.

Using the above system and the command **show trace dtcomponent ucm:CallManager** would restrict its *ucm* component output to just the Call Manager, but would return information for other device types. In this example, it would return information for the Unified CVP Call Server and the Call Manager but not the Unified CM Tomcat Server.

Information returned

- A Unified CVP Call Server: **Information Returned**
- A Unified Call Manager: **Information Returned**
- A Unified CM Tomcat Server: **No Information Returned**

Integrate Analysis Manager with Unified CVP

Follow these steps to integrate Analysis Manager with Unified CVP configuration:

Step 1 From the Unified Analysis Manager menu, choose **Inventory > Node**.

The Node window appears.

Step 2 Click Add to add a node or select a node from the list. Click Edit to edit an existing configuration.

The Add or Edit Node window appears.

Note: Fields on this window that are marked with an asterisk (*) are required fields.

Step 3 From the Product Type drop-down list box, select **CVP** as the product.

Step 4 In the IP/Hostname field, enter the hostname or the IP address of the Unified CVP OAMP box.

Step 5 In the Transport Protocol field, select the protocol you want to use. Use the default value (HTTPS).

Step 6 In the Port Number field, enter the port number of the node you will be using. Use the default value (8111).

Step 7 In the User Name and Password fields, enter the user name as **wsmadmin** and for password use the OAMP password. Reenter the password in the Confirm Password field.

Step 8 In the Description field, provide a brief description of the node you are adding.

Step 9 In the Associated Call Record Server and Associated Trace File Server fields, use the drop-down list to select the respective servers you want to use for the node.

Step 10 To add a node to an existing group, check the Associated Group check box.

Step 11 If you have a NAT or Terminal Server configuration, click **Advanced** to display the Add Node-Advanced window. Enter the appropriate information in the Alternate IP/Hostname and Alternate Port fields.

Step 12 Click **Save** to add and save the node to the list. Click **Cancel** to end the operation without adding the node.

Mapping System CLI Commands to IOS CLI Commands

The following table maps the System CLI commands to their corresponding IOS CLI commands.

System CLI Commands	IOS CLI Commands
show config	show running-config
show version	show version show clock
show license	show license
show perf	show call resource voice stat show memory statistics show processes cpu history show processes memory sorted show voice dsp group all show voice dsp voice
show debug	show debug
show log	N/A
show sessions	show call active voice compact show voice call status | inc calls show voip rtp connections | inc connect sh sip-ua calls | inc calls
show tech-support	show tech-support <Everything else given above>
show trace	show logging
show platform	show diag sh inventory

	sh int | inc media type|Ethernet|address sh controllers T1 | inc T1 sh controllers E1 | inc E1
debug	0 no debug all 1 - deb ccsip err deb cch323 err deb voip app vxml err deb http client err deb mrcp err deb rtsp err deb h225 asn1 err deb h245 asn1 err 2 - debug isdn q931 debug h225 events debug h245 events debug voip ccapi inout debug vtsp events 3 - debug ccsip messages debug h225 q931 debug h225 asn1 debug h245 asn1

Chapter 5

Configuring and Modifying Unified CVP Security

This chapter discusses security considerations for Unified CVP deployments.

Note: For detailed information about security issues in Unified ICME, refer to *Security Best Practices Guide for ICM and IPCC Enterprise Hosted Editions*.

This chapter contains the following topics:

Unified CVP Security and Cisco Support Tools

Cisco Support Tools is Cisco Unified Contact Center's serviceability application. It provides a common, web-based UI to a number of serviceability tools and works across all Cisco Unified Contact Center products. Unified CVP solutions can use the Support Tools as a mechanism to collect logs and trace levels, as well as provide other supportability functions.

Note: For more information, refer to "Using Cisco Support Tools with Unified CVP (page 221)." For detailed information about security issues when using Cisco Support Tools, refer to *Security Best Practices Guide for ICM and IPCC Enterprise Hosted Editions*.

About Cisco Support Tools and Secure Sockets Layer (SSL)

Cisco Support Tools is deployed with SSL enabled by default. This ensures all Support Tools data exchanged between the client and server, including user logins, are passed with 128-bit encryption.

About Cisco Support Tools and Unified CVP on a Windows System

When Unified CVP is installed on a Windows system, the Support Tools Protocol Adapter will attempt to become enabled using an IPSEC preshared key.

IPSEC is a set of protocols that supports secure exchange of packets at the IP Layer. IPSEC requires an authentication mechanism between participants in a data exchange. Unified CVP uses a preshared key as the authentication method; IPSEC will only let messages pass between a ST Server and node agent when each side has been configured with the same key.

Note: When specifying a preshared key, the value must be entered *exactly* the same on each server or node agent to enable them to communicate with each other.

You specify the preshared key during Unified CVP installation on the Support Tools screen.

Note: For instructions, refer to"Installing Cisco Unified CVP Software on Windows 2003 Systems," in the **Installation and Upgrade Guide for Cisco Unified Customer Voice Portal guide** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html).

Securing Communications Between Unified CVP Components

During configuration of a Unified CVP device–such as the Call Server, Reporting Server, or VXML Server–the Operations Console Server uses Java Management Extensions (JMX) to communicate to the managed Unified CVP device.

Note: The Operations Console Server is the underlying software that manages component configurations. The Operations Console is the web-based interface you use to configure Unified CVP components. For more information, refer to **Operations Console User Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html).

Unified CVP installation uses a default JMX communications setting of *non-secured*, so communications are not encrypted. However, you can modify this setting to secure communications using Secure Sockets Layer (SSL).

Note: The procedure to modify this setting requires that you stop and restart services.

Before You Begin

You secure JMX communications by importing:

- Self-signed certificates that are created automatically from information that you specify during Unified CVP installation.
- Signed certificates available from a Certificate Authority (CA).

You manage certificates using:

- The *keystore*, a database for keys and trusted certificate information. For all keystore operations it is assumed that:

  For Windows 2003 Server machines:

  – Keystore resides in: **`%CVP_HOME%\conf\security\.keystore`**

  – Resource Manager keystore resides in: **`%CVP_HOME%\conf\security\.ormKeystore`**

  – Keystore password resides in: **`%CVP_HOME%\conf\security.properties`**

- The *keytool*, a command-line utility for managing keys and trusted certificates. The keytool is installed in:

  – For Windows 2003 Server machines:

  `%CVP_HOME%/jre`

Note:

- On Windows systems, the keystore and the keystore password are in a folder that is protected with Access Control List (ACL), so only a user who has Administrator privileges can import trusted certificates.
- For more information about the keytool and keystores, refer to your Java documentation.

How to Secure JMX Communications Between Unified CVP Components

Follow the steps below to secure JMX communication using SSL between the Unified CVP Operations Console service, a managed Unified CVP device, and other CVP-related JMX clients.

Step 1 Stop the *Unified CVP Operations Console* service. On a *Windows* system, do the following:

1. Select **Start > Settings > Control Panel > Administrative Tools > Services**.

   The Services window appears.

2. In the list of Service names, highlight the Cisco services that you need to stop to complete the secure communication setup.

   The services would include each device configured in the Operations Console, the Operations console itself, the Web Services Manager, and the Resource Manager as shown in the list that follows:

   – Cisco CVP CallServer

   – Cisco CVP OpsConsoleServer

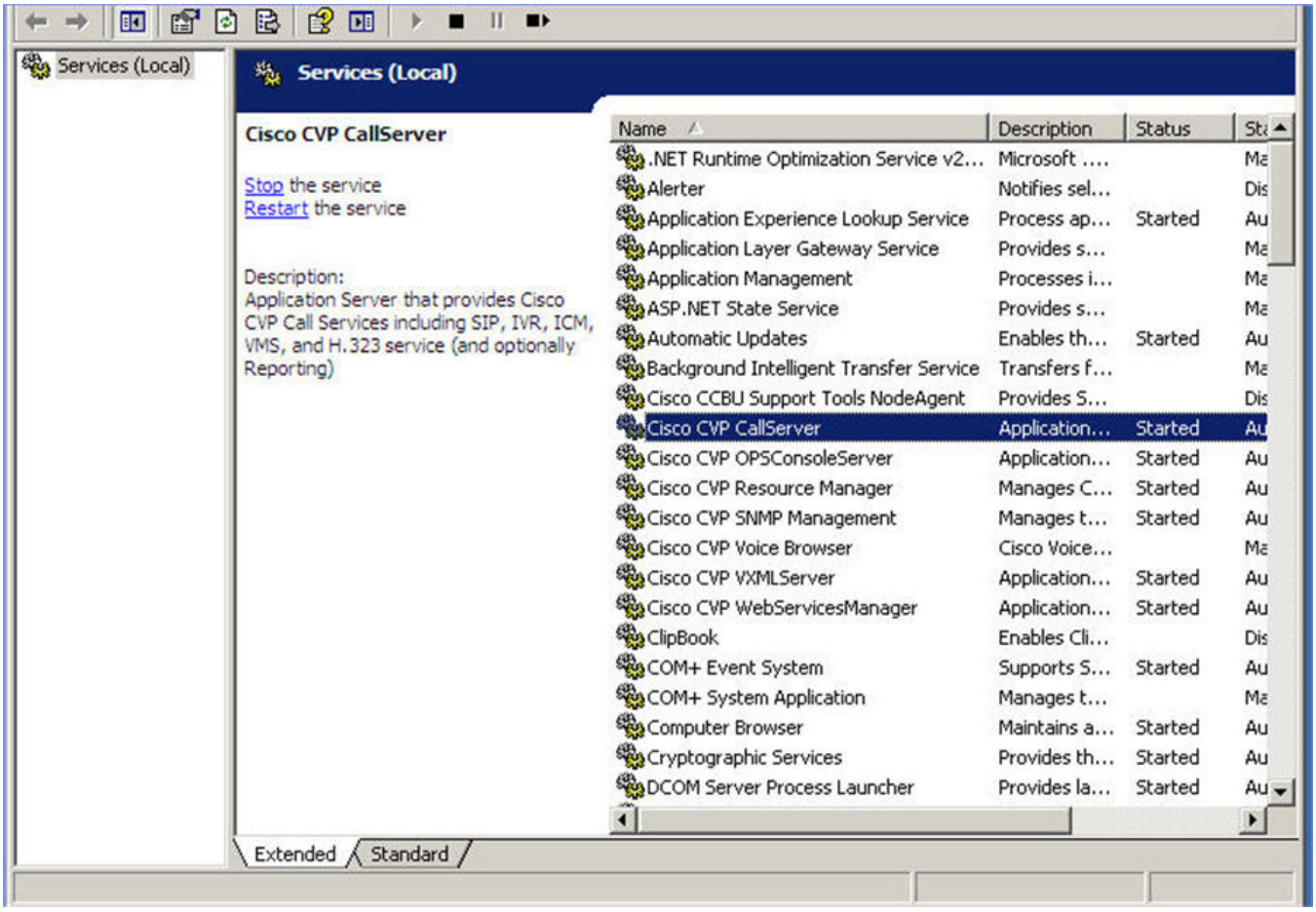
   – Cisco CVP Resource Manager

– Cisco CVP VXMLServer

– Cisco WebServicesManager

Note: Later, in Step #4, when you enable secure communication and save the setting, the Operations Console will provide a list of the services you must restart for secure communication to take effect.

Figure 32: Stop Services Before Configuring JMX



3. Click the **Stop** link.

   The service stops.

Step 2 Use the instructions in "How to Exchange Certificates Between Systems (page 240)" for details on using the keystore and keytool Java tools to exchange trusted certificates between the Operations Console and the device being managed.

Note: For information about prerequisites and assumptions regarding keystore and keytool, refer to "Before You Begin (page 236)." For detailed instructions about using these tools, refer to the Java documentation.

Step 3 Restart **just** the *Cisco CVP OpsConsoleServer* service.

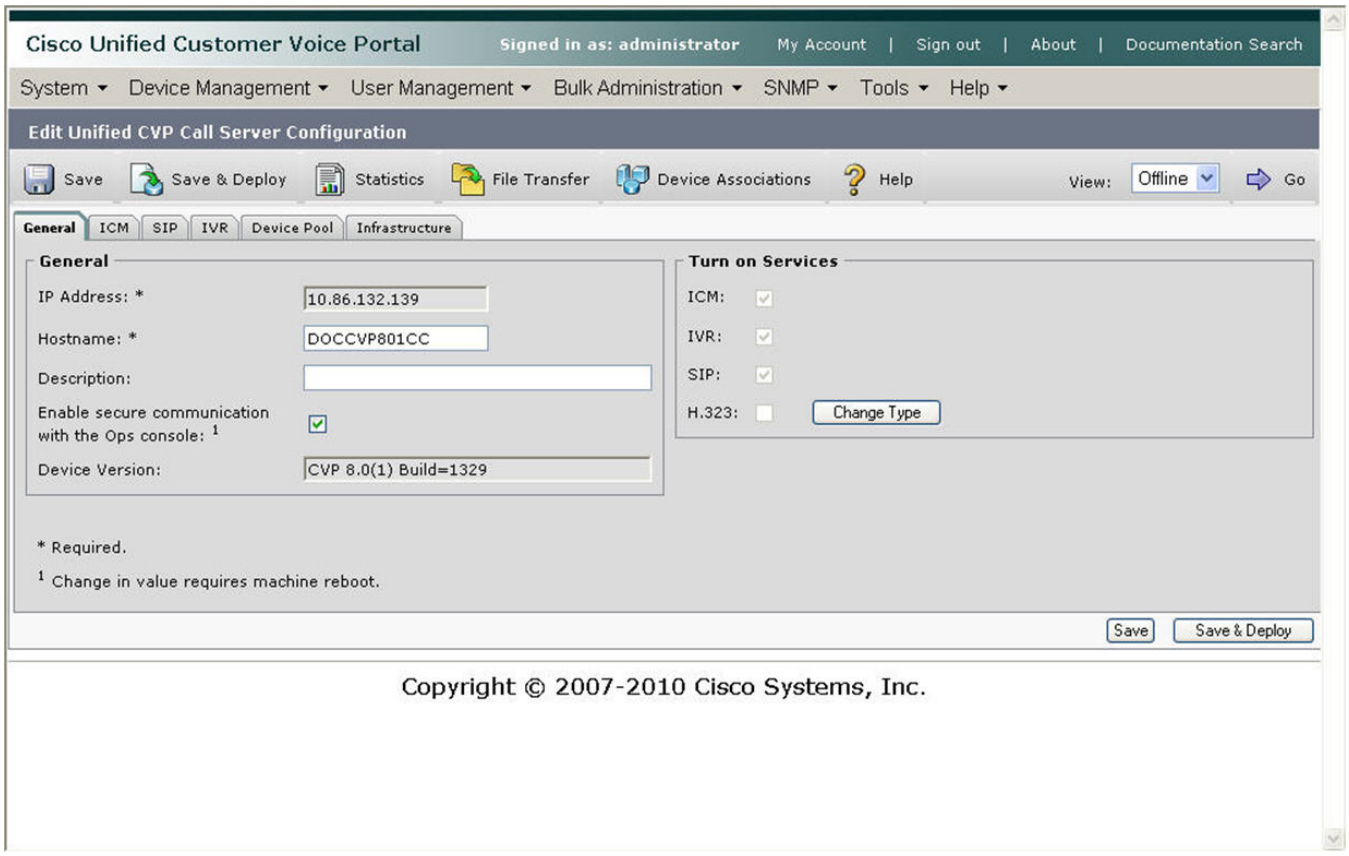
Follow the procedure described in Step 1, selecting the **Start** link instead of **Stop** on a Windows system.

Step 4 Use the **Enable secure communication with the Ops Console** checkbox on the Operations Console Device Management configuration page to enable security for devices that require secure communication. (Refer to "How to Enable Security on Unfied CVP Devices (page 241)" for more information.)

Note:

- Checking this box for any server on a specific box enables security for **all** the servers on that box. You will be asked to restart the servers that have security enabled.
- Once you have enabled secure communication between Unfied CVP components, any devices or clients that are not set up for secure communication will not work until modified for secure communication. Refer to How to Exchange Certificates Between Systems (page 240) to complete the setup.

Figure 33: Enabling Secure Communication with the Operations Console



Step 5 Restart the **Cisco CVP Resource Manager** service on the Unified CVP device machines on which communications needs to be secure by doing the following:

Restart the *Cisco CVP Resource Manager* service. Select **Start > Control Panel > Administrative Tools > Services**.

How to Exchange Certificates Between Systems

Follow the steps below to move certificates between keystores.

Note: The **`keytool`** commands shown below use the JRE relative path for the Windows platform.

Step 1 Import the Operations Console Server certificate as trusted on the managed Unified CVP device:

a. Retrieve the keystore password from the **`security.properties`** file (resides in the %CVP_HOME%\conf directory) on the *Operations Console* Server.

b. Export the certificate from the keystore on the Operations Console Server. Open a command prompt and navigate to the %CVP_HOME%\conf\security directory, then enter the following command:

```
.\..\jre\bin\keytool -export -v -keystore .keystore -storetype JCEKS
-alias oamp_certificate -file <oamp_cert_XXX>
```

Where the file argument in <> (angle brackets) is user-defined and unique.

Note: *Do not* modify the oamp_certificate alias name.

When prompted, enter the keystore password.

c. Copy the exported certificate file <oamp_cert_XXX> from the Operations Console service to the %CVP_HOME%\conf\security folder on the machine where the Unified CVP Resource Manager service is running.

d. Retrieve the keystore password from the **`security.properties`** file on the managed Unified CVP device.

e. For Windows, import the Operations Console certificate <oamp_cert_XXX> into the keystore on the managed Unified CVP device. Open a command prompt and navigate to the %CVP_HOME%\conf\security directory, then enter the following command:

```
..\..\jre\bin\keytool -import -keystore .keystore -storetype JCEKS
-trustcacerts -alias <orm_oamp_certificate> -file <oamp_cert_XXX>
```

Where the alias argument in <> (angle brackets) is user-defined and unique, and the file argument in <> (angle brackets) is the exported Operations Console certificate filename.

When prompted, enter the keystore password and then enter **`yes`** to confirm.

f. Repeat these steps for every machine where the Unified CVP Resource Manager service is running if the JMX communication from the Operations Server to that managed Unified CVP device needs to be secured.

Step 2 Import the managed Unified CVP device certificate as trusted in the keystore on the Operations Console Server:

a. Retrieve the keystore password from the **security.properties** file (resides in the %CVP_HOME%\conf directory) on the managed Unified CVP device.

b. For Windows, export the Unified CVP device certificate from the keystore. Open a command prompt and navigate to the %CVP_HOME%\conf\security directory, then enter the following command:

```
..\..\jre\bin\keytool -export -v -keystore .ormKeystore -storetype
JCEKS -alias orm_certificate -file <orm_cert_file_XXX>
```

Where the file argument in <> (angle brackets) is user-defined and unique.

Note:

- *Do not* modify the orm_certificate alias name.
- Append an IP address to the file name to make it unique. The IP address can be replaced with any value as long as it makes the filename unique when copied to the Operations Console Server.

c. Copy the exported certificate file <orm_cert_file> from the managed Unified CVP device to the %CVP_HOME%\conf\security folder on the Operations Console service.

d. Retrieve the keystore password from the **security.properties** file in the Operations Console Server.

e. Import the certificate <orm_cert_file> into the keystore on the Operations Console Server. Open a command prompt and navigate to the %CVP_HOME%\conf\security directory, then enter the following command:

```
..\..\jre\bin\keytool -import -keystore .keystore -storetype
JCEKS-trustcacerts -alias <oamp_orm_certificate_XXX> -file
<orm_cert_XXX>
```

Where the alias argument in <> (angle brackets) is user-defined and unique, and the file argument in <> angle brackets is the exported managed device certificate filename.

Note: Append an IP address to the certificate alias to make the alias unique in the keystore. The IP address can be replaced with any value as long as it makes the certificate name unique when imported to the keystore.

f. Repeat these steps for every machine where the Unified CVP Resource Manager service is running if the JMX communication from the Operations Console Server to that managed Unified CVP device needs to be secured.

How to Enable Security on Unified CVP Devices

Once you have completed the procedure described in How to Exchange Certificates Between Systems (page 240), you must *enable* security on the Unified CVP components that you want to accept only secure SSL communications.

Note: For information about enabling security on non-Unified CVP components that are part of the Unified CVP solution, refer to "Securing Communications Between Unified CVP and IOS Devices (page 244)."

By default, the communication channel between the Unified CVP Operations Console and the Resource Manager on Unified CVP devices is non-secure at the completion of Unified CVP installation. You use the Operations Console's Device Management configuration page to enable or disable secure SSL communications.

Note: Whenever this security setting is modified, you must restart the Unified CVP Resource Manager service on the machine where the device is running.

Caution: If you do not check the Enable secure communication with the Ops console checkbox, the communication link between the Operations Console and the managed Unified CVP device *will not* be secure.

Step 1 Open the Operations Console.

Step 2 Choose a device type from the **Device Management** menu.

Step 3 Click **Add New** or select an existing device name and click **Edit**.

The General tab displays.

Step 4 Select the **Enable secure communication with the Ops console** checkbox.

Step 5 Click **Save** to save the settings in the Operations Server database and click **Save and Deploy** to apply the changes to the device.

Step 6 Restart the Unified CVP Resource Manager service on the machine where the device is running.

Step 7 Repeat these steps for all Unified CVP components that are to accept only secure SSL communications.

Obtaining and Deploying Certificate Authority Signed Certificates

Follow the steps described in this section to:

- Generate a Certificate Signing Request.
- Obtain the signed certificate.
- Import the signed certificates on all machines managed by the Unified CVP Operations Console.

How to Add a Certificate Signed by a Certificate Authority to the Keystore

Follow the steps below to generate and import CA-signed certificates for secure communications between the Operations Console and the CVP ResourceManager on other devices in your Unified CVP solution.

Notes:

- This section *does not* discuss how to accommodate HTTPS connections to the Operations Console; for information on that topic, refer to "How to Add a Certificate Signed by a Certificate Authority for HTTPS Web Access (page 244)."
- The **`keytool`** commands shown below use the JRE relative path for the Windows platform.
- If you have already exchanged certificates to secure Unified CVP device communications, that process must be repeated after importing the signed certificates.

Step 1 Retrieve the keystore password from the **`security.properties`** file.

Step 2 Generate a Certificate Signing Request (CSR).

a. From the %CVP_HOME%\conf\security directory, enter the following:

```
..\..\jre\bin\keytool -keystore .keystore -storetype JCEKS -certreq
-keyalg RSA -sigalg MD5withRSA -alias orm_certificate -file
ormcertreq.csr
```

b. When prompted, enter the keystore password.

Step 3 Send the ormcertreq.csr certificate file to a Certificate Authority (CA) for sign-off. Once the certificate is signed, it will be returned with a root certificate of a CA and, depending on the signing CA, some optional intermediate certificates.

Step 4 Install the signed certificate into keystore.

a. Enter the following command to install the Intermediate CA Certificates (if any):

```
keytool -keystore .keystore -storetype JCEKS -import -alias root
-trustcacerts -file <filename_of_intermediate_CA_certs>
```

b. Enter the following command to install the root certificates (these are not in the Unified CVP keystore by default):

```
keytool -keystore .keystore -storetype JCEKS -import -alias root
-trustcacerts -file <filename_of_root_cert>
```

Note: Be careful to examine the contents of any root certificate file before installing it to your keystore as a trusted certificate.

The Java root certificates are installed in %CVP_HOME%\jre\lib\security\cacerts.

c. Enter the following command to install the CA Signed Certificate:

```
keytool -keystore .keystore -storetype JCEKS -import -alias
orm_certificate -trustcacerts -file
<filename_of_your_signed_cert_from_CA>
```

Step 5 Repeat these steps on every machine running Unified CVP Services.

How to Add a Certificate Signed by a Certificate Authority for HTTPS Web Access

Follow the steps below to present a CA-signed certificate to inbound Operations Console HTTPS clients.

Note: The OAMP and ORM certificates provided in the keystore **do not** provide TLS encryption for inbound HTTPS traffic; those certificates provide secure connections between the Operations Console and the CVP Resource Manager on other devices in your Unified CVP solution.

The certificate and private key used for Operations Console HTTPS are:

- Self-signed certificate: **`%CVP_HOME%\conf\security\oamp.crt`**
- Private key for self-signed certificate: **`%CVP_HOME%\conf\security\oamp.key`**

Step 1 Access the OpenSSL command line.

Note: You must first install OpenSSL (http://www.openssl.org), as it is not included with Unfied CVP. Refer to the OpenSSL documentation for details.

Step 2 Generate a Certificate Signing Request (CSR) by entering the following command:

```
openssl req -new -key xxxx.key -out xxxx.csr
```

Where *xxxx* represents the key and the certificate files.

Step 3 Send the xxxx.csr certificate file to a Certificate Authority (CA) for sign-off. Once the certificate is signed, it will be returned with a root certificate of a CA.

Step 4 Replace the original oamp.crt file with the signed certificate.

Securing Communications Between Unified CVP and IOS Devices

To secure file transfer between Cisco Gateways and/or Gatekeepers and the Unified CVP Operations Console, you need to import the Operations Console Server certificate on the IOS device during device configuration and enable SSH on the router; otherwise, any user-requested action through the Operations Console (for example, file transfer to an IOS device) will fail. For example, to copy a file to the IOS device, the Operations Console (when the security flag is set) expects SSH to be enabled on the device. If SSH is not enabled, a failure will occur.

How to Export the Operations Console Certificate to a File

Step 1 From the web browser, enter **`https://`***ServerIP***`:9443/oamp`**

The Security Alert dialog box appears.

Step 2 Click **View Certificate**.

The Certificate dialog box appears.

Step 3 Select the Details tab.

<All> will be highlighted in the Show drop-down list.

Step 4 Click **Copy to File**.

The Certificate Export Wizard dialog appears.

Step 5 Select the Base-64 encoded X.509 (.CER) radio button and click **Next**.

Step 6 Specify a file name in the File to Export dialog and click **Next**.

Step 7 Click **Finish**.

An Export was Successful message appears.

Step 8 Click **OK** and close the Security Alert dialog.

Step 9 Open the exported file in Notepad and copy the text that appears between the ----BEGIN CERTIFICATE--- and ----END CERTIFICATE--- tags.

You are now ready to copy the Operations Console certificate information to the IOS device.

How to Import the Operations Console Certificate to a Gatekeeper or Gateway

Step 1 Access the IOS device in privileged EXEC mode.

Note: For more information, refer to the Cisco IOS Command-Line Interface documentation.

Step 2 Access global configuration mode by entering:

```
configure terminal
```

Step 3 Create and enroll a trustpoint by entering the following commands:

```
crypto pki trustpoint xxxx
en terminal
exit
```

Where **xxxx** is a trustpoint name.

The IOS device exits conf t mode and returns to privileged EXEC mode.

Step 4 Do the following to copy the certificate exported to the Notepad to the IOS device:

1. Enter: **cryp pki auth xxxx**

   Where **xxxx** is the trustpoint name specified in the previous step.

2. Paste the certificate from the Notepad clipboard.

3. Enter: **quit**

The following happens:

- A message displays describing the certificate attributes.
- A confirmation prompt appears.

Step 5 Enter: **yes**

A message reports that the certificate was successfully imported.

How to Configure VXML Gateways to enable HTTPS

To enable HTTPS on VXML gateways perform the following steps:

1. Import the Operations Console Certificate to VXML Gateway.

2. Ensure that SSL is enabled on the VRU call leg.

   Note: The SSL is enabled by adding the configuration specified below for standalone or comprehensive deployments.

3. The ICM scripts and VXML server scripts that access media via direct calls with an URL using **http:** must be changed to **https:**.

- **Standalone Deployment**: Add the CVPSelfService-SSL parameter in VXML Gateway to enable HTTPS in VXML applications as:

```
Application
service vru-leg flash:CVPSelfService.tcl
  param CVPSelfService-port 7443
  param CVPSelfService-SSL 1
```

- **Comprehensive Deployment**: Add the cvpserverssl parameter in VXML Gateway to enable HTTPS in VXML applications as:

```
Application
service vru-leg flash:bootstrap.tcl
  param cvpserverssl 1
```

IVR Configuration for HTTPS with Microapps

Step 1 Login to CVP OAMP.

Step 2 Choose **Device Management** > **Call Server** > **IVR** tab, check the **Use Security for Media Fetch** checkbox.

How to Configure Secure Access to the Router Using Protocol SSH Version 2

The Unified CVP administrator can configure secure access to the router using the following example.

Step 1 Log into the IOS devices.

Step 2 On the gateway or gatekeeper, enter the command **`show ip ssh`** to determine if the SSH Server is enabled.

If the SSH Server is not enabled, perform the following steps.

Step 3 Access the configurable terminal and enter the following commands:

```
gw(config)# ip domain name <domain name>
gw(config)# crypto key generate rsa
gw(config) # ip ssh version 2
```

Where the domain name is your configured domain name.

Step 4 Enter **`768`** or higher for how many bits to use.

Note: Important! If you enter lower than 768, IOS automatically uses SSH Version 1, which is not compatible with the Operations Console, which is trying to connect with SSH Version 2.

Step 5 Choose the key modulus size in the range of **360 to 2048** for your General Purpose Keys.

Step 6 Enter the following commands:

```
gw(config)# ip ssh time-out 60
gw(config)# ip ssh authentication-retries 5
```

Step 7 Enter the **`show ip ssh`** command and confirm that "SSH version enabled -version 1.99" or "SSH version enabled - version 2.0" displays.

Step 8 Configure a user name and password in IOS by entering the following command:

```
gw(config)# username <username> password <password>
```

Where the username and password are your configured username and password.

Step 9 Configure the IOS settings to allow SSH logins by entering the following commands.

Note: These commands are located in the running configuration towards the end. Search for the line **vty configurations** in the configuration. The important settings are those indicated by an asterisk (*).

```
line vty 0 4*
exec-timeout 35 0
password <password>*
session-limit 10
login local*
transport input telnet*
transport output all
```

To configure these settings, access the configurable terminal and enter the following:

```
gw# configure terminal
gw(config)# line vty 0 4
gw(config-line)# password <password>
gw(config-line)# login local
gw(config-line)# transport input ssh
```

Where the password is your configured password.

Note: Important! Enabling these settings will *not* allow regular telnet anymore and will only allow SSH. If you want telnet and SSH available, then enter **`transport input all`** or **`transport input telnet`** for telnet-only access.

Securing Communications Between VoiceXML Gateways and VXML Server

To prevent the possibility of malicious requests being sent to VXML Server, configure the application server on which it runs to only accept connections from the IP addresses of the VoiceXML Gateways involved.

Note: The following instructions are specifically written for Tomcat users; for WebSphere, refer to the WebSphere documentation.

To configure Tomcat to only accept connections from a known set of IP addresses, do the following:

Step 1 Edit the CATALINA_HOME\conf\context.xml file, where *CATALINA_HOME* is the Tomcat installation directory.

Step 2 Between the existing <Context> tag and its closing tag, insert:

```
<Valve className="org.apache.catalina.valves.RemoteAddrValve" allow="X.X.X.X" deny=""/>
```

Where *X.X.X.X* is the IP address of your VoiceXML Gateway (or Content Services Switch IP, as appropriate).

If multiple IP addresses need access, indicate them as a comma-delimited list, such as:

```
allow="X.X.X.X,Y.Y.Y.Y"
```

Step 3 Restart Tomcat to enable the changes.

HTTPS Support for Unified CVP

Unified CVP can be configured to use HTTPS on the VXML Server and on the IVR leg of the Call Server. Only signed server certificates can be applied to the IOS gateway; self-signed certificates are not accepted. Unified CVP generates self-signed certificates for Tomcat applications, which must be signed by a Certificate Authority prior to use. Tomcat VXML Server users and Call Server users must follow these steps to have the certificate signed.

To sign a certificate, you must use Microsoft Certificate Services or send it to a third-party Certificate Authority company. For further instructions, contact a third-party Certificate Authority company.

Note: Refer to the 12.4(15)T IOS documentation for instructions on how to apply the certificates to the VXML gateway.

Using Tomcat to Present CA-signed Certificates to Inbound HTTPS Clients

Due to the large processing overhead of HTTPS, the Tomcat application server can only achieve up to 100 simultaneous connections dependent on the configuration (refer to **Cisco Unified Customer Voice Portal (CVP) Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) for further details). For better performance from the VXML Server with HTTPS, use WebSphere as the application server.

For higher performance in a Comprehensive call flow model, use VXML Server with WebSphere in conjunction with Call Server.

Tomcat users must follow the steps below to present a CA-signed certificate to inbound HTTPS clients.

Step 1 Open the **security.properties** file to retrieve the .keystore password. You will need to copy and paste the value of this property when managing the .keystore.

1. Open the **%CVP_HOME%\conf\security.properties** file where **%CVP_HOME%** is the target installation directory for Unified CVP. In the default case, this would be **C:\Cisco\CVP**.

   Note: The property file should contain one property: **Security.keystorePW**.

2. When managing the keystore, after entering a command, the keytool will ask for you to "Enter keystore password." To do this, copy the value of the **Security.keystorePW** property and paste it into the command-line window to enter your keystore password.

For example, consider the **%CVP_HOME%\conf\security.properties** file contains the property line:

- **Security.keystorePW = [3X}}E7@nhMXGy{ou.5AL!+4Ffm868**.
- The password to copy would be: **[3X}}E7@nhMXGy{ou.5AL!+4Ffm868**.

Step 2 Create a back up of the **%CVP_HOME%\conf\security** directory.

Step 3 Open a command-line prompt window and change to the security configuration directory: **cd\cisco\cvp\conf\security**.

Step 4 Create the certificate signing request using the private key entry for your certificate, remembering to enter in the keystore password when prompted. A new csr file will be created on the file system.

Examples:

- Call Server: **`%CVP_HOME%\jre\bin\keytool.exe -certreq -alias callserver_certificate -storetype JCEKS -keystore .keystore -file callserver_certificate.csr`**.
- VXML Server: **`%CVP_HOME%\jre\bin\keytool.exe -certreq -alias vxml_certificate -storetype JCEKS -keystore .keystore -file vxml_certificate.csr`**.

Step 5 Give the certificate signing request file to a trusted Certificate Authority. They will sign, returning one or more trusted certificates.

Step 6 Import the signed certificate file from your trusted Certificate Authority to the .keystore file, remembering to enter in the keystore password when prompted. If more than one certificate is delivered, certificates must be imported in order of the chained certificate hierarchy. For example: root, intermediate, signed certificate.

Examples:

- Call Server: **`%CVP_HOME%\jre\bin\keytool.exe -import -v -alias callserver_certificate -storetype JCEKS -trustcacerts -keystore . keystore -file signed_callserver_certificate.crt`**.
- VXML Server: **`%CVP_HOME%\jre\bin\keytool.exe -import -v -alias vxml_certificate -storetype JCEKS -trustcacerts -keystore .keystore -file signed_vxml_certificate.crt`**.

Using WebSphere to Present CA-signed Certificates to Inbound HTTPS Clients

WebSphere users must follow the steps below to present a CA-signed certificate to inbound HTTPS clients.

Step 1 Run IBM's ikeyman utility to create a keystore: C:\Program Files\IBM\HTTPServer\bin\ikeyman.bat.

Create a new Key database file:

- Select **Key Database File > New**.
- Database type: CMS
- Filename: key.kdb
- Location: C:\Program Files\IBM\HTTPServer\keys

Note: The "keys" directory will have to be explicitly created if it does not exist. Keystore will prompt for password and expiration information. Check off creates a Stash file.

Step 2 Create a new Certificate Request in ikeyman by selecting **Create > New Certificate Request**.

- Key Label: <hostname>
- Version: V509 v3
- Key Size: 1024
- Common Name: <hostname>
- Organization: <Company Name>
- Organization Unit: <Group Name>
- Locality: <Town>
- State: <State>
- Zip Code: <Zip Code>
- Country: <Country>
- Validity Period: 730 days

This saves a file certreq.arm.

Present this certificate request to a Certificate Authority for sign-off. Once the certificate is signed, it will be returned with the root certificate of a CA.

Step 3 Import the CA root certificate into the Key Database.

If the CA is not listed in the Key Database Content > Signer Certificates, then the root certificate of the CA must be imported into this list for the signed certificate to be installed into the database. Click **Add**, and provide the path of the DER certificate file of the Root CA certificate.

Step 4 Import the signed certificate into the key database.

Save the signed certificate from the CA as a DER file format and locate it on the machine in which you will install it into the key database. Then import the signed certificate into the database with the ikeyman utility. Key Database Content > Personal Certificates, click **Receive**. Provide the path of the signed certificate from the CA.

Set Up SSL

Follow the instructions below to set up SSL.

Step 1 Set IHS_HOME=<location of IBM HTTPServer>; for example, set IHS_HOME=C:\Program Files\IBM\HTTPServer.

Step 2 Modify the %IHS_HOME%\conf\httpd.conf file as follows:

Current configuration for HTTP/port 7000:

```
##
## To enable AFPA on supported Windows Operating Systems delete the "#" symbol in
## front of the "LoadModule ibm_afpa_module modules/mod_afpa_cache.so" directive
##
LoadModule ibm_afpa_module modules/mod_afpa_cache.so
<IfModule mod_afpa_cache.c>
AfpaEnable
AfpaCache on
AfpaPort 7000
AfpaLogFile "C:/Program Files/IBM/HTTPServer/logs/afpalog"
</IfModule></IfModule>
##
## Configuration with AFPA disabled
##
<IfModule !mod_afpa_cache.c>
Listen 0.0.0.0:7000
```

Comment out and change to HTTPS/port 7443:

```
##
#LoadModule ibm_afpa_module modules/mod_afpa_cache.so
#<IfModule mod_afpa_cache.c>
# AfpaEnable
# AfpaCache on
# AfpaPort 7000
# AfpaLogFile "C:/Program Files/IBM/HTTPServer/logs/afpalog"
#</IfModule>
##
## Configuration with AFPA disabled
##
#<IfModule !mod_afpa_cache.c>
# Listen 0.0.0.0:7000
# Use Win32DisableAcceptEx to downgrade to use winsock 1.1 network APIs.
```

```
# Note: You can use Win32DisableAcceptEx only if mod_afpa_cache.so is disabled.
# Win32DisableAcceptEx
#</IfModule>
LoadModule ibm_ssl_module modules/mod_ibm_ssl.so
Listen 0.0.0.0:7443
<VirtualHost 0.0.0.0:7443>
SSLEnable
SSLClientAuth none
KeyFile "c:\Program Files\IBM\HTTPServer\keys\key.kdb"
SSLStashfile "c:\Program Files\IBM\HTTPServer\keys\key.sth"
ErrorLog "c:\Program Files\IBM\HTTPServer\logs\sslerror.log"
TransferLog "c:\Program Files\IBM\HTTPServer\logs\sslaccess.log"
</VirtualHost>
SSLDisable
```

Step 3 Open the WebSphere Integrated Solutions Console (http://localhost:9080/ibm/console) and log in, if necessary.

Step 4 Using the WebSphere Integrated Solutions Console, navigate to **Servers > Web servers**. Delete "webserver1" by selecting and click **Delete**. (Be sure to click **Save** to save the configuration when prompted.)

Step 5 Using the WebSphere Integrated Solutions Console, navigate to **Environment > Virtual Hosts**. Add the port mapping: Hostname = *; Port = 7443. (Be sure to click **Save** to save the configuration when prompted.)

Step 6 Log out of the WebSphere Integrated Solutions Console.

Step 7 Copy: *%CVP_HOME%\bin\DefineWebContainer.bat*

To: *%CVP_HOME%\bin\DefineWebContainerHTTPS.bat*

Step 8 Edit *%CVP_HOME%\bin\DefineWebContainerHTTPS.bat* as follows:

Current HTTP port definition:

```
set PORT=7000
```

Change to new HTTPS port definition:

```
set PORT=7443
```

Step 9 Open up the WebSphere Integrated Solutions Console (http://localhost:9080/ibm/console) and log in, if necessary.

Step 10 Using the WebSphere Integrated Solutions Console, navigate to **Servers > Web servers**.

- Click **webserver1**.
- Click **Plugin properties** link.
- Click **Copy to Web server key store directory** and then click **OK**.
- Click **Save** to save the configuration when complete.

Step 11 Reboot the machine.

Step 12 After the machine restarts, try to load the standard HTTPServer HTTPS web page; for example, https://localhost:7443/.

Step 13 Try to load the CVP Licensing web page using HTTPS; for example, https://localhost:7443/CVP/Licensing.

Securing Communications Between Unified CVP and IOS Devices

To secure HTTPS between Cisco Gateways and/or Gatekeepers and the Call Server and VXML Server to the gateway for HTTPS, you need to import the Call Server or VXML Server certificate on the IOS device during device configuration.

Step 1 From the web browser, access the secure Call Server with [https://<ServerIP>:8443/] or the secure VXML Server with [https://<ServerIP>:7443/].

The Security Alert dialog box displays.

Step 2 Click **View Certificate**.

The Certificate dialog box displays.

Step 3 Select the Details tab.

<All> will be highlighted in the Show drop-down list.

Step 4 Click **Copy to File**.

The Certificate Export Wizard dialog appears.

Step 5 Click **Base-64 encoded X.509 (.CER)** and then click **Next**.

Step 6 Specify a file name in the File to Export dialog box and then click **Next**.

Step 7 Click **Finish**.

An Export was Successful message displays.

Step 8 Click **OK** and close the Security Alert dialog box.

Step 9 Open the exported file in Notepad and copy the text that appears between the ---BEGIN CERTIFICATE-- and --END CERTIFICATE-- tags.

You are now ready to copy the Operations Console certificate information to the IOS device.

Step 10 Access the IOS device in privileged EXEC mode.

Note: For more information, refer to the Cisco IOS Command-Line Interface documentation.

Step 11 Access global configuration mode by entering **`configuration terminal`**.

Step 12 Create and enroll a trustpoint by entering the following commands:

```
crypto pki trustpoint xxxx
en terminal
exit
```

Where *xxxx* is a trustpoint name.

The IOS device exits conf t mode and returns to privileged EXEC mode.

Step 13 Do the following to copy the certificate exported to the Notepad to the IOS device:

1. Enter **`crypto pki auth xxxx`**.

   Where *xxxx* is the trustpoint name specified in the previous step.

2. Paste the certificate from the Notepad clipboard.

3. Enter **`quit`**.

   The following happens:

   – A message displays describing the certificate attributes.

   – A confirmation prompt appears.

Step 14 Enter **`yes`**.

A message reports that the certificate was successfully imported.

Securing Sensitive Customer Information

Use the VXML Server Inclusive and Exclusive filters to control the sensitive customer information, such as PIN numbers, that are sent to the Reporting Server.

By default, all items except the Start and End element are filtered from information the VXML Server feeds to the Reporting Server unless they are added to an Inclusive Filter. If you create Inclusive filters that are broad enough to allow sensitive information to be passed, you then have the option to:

- Adjust the Inclusive filters so that the sensitive information is not included.
- Add Exclusive filters to prevent the sensitive information from being included.

See Also

Refer to the Operations Console online help topic *Configuring a VXML Server > Inclusive and Exclusive VoiceXML Filters for Reporting* for detailed information about configuring filters.

Part 2: Configuration Detail of Unified CVP Components

This part of the manual describes the Unified CVP components that you configure using the Operations Console.

Chapter 6

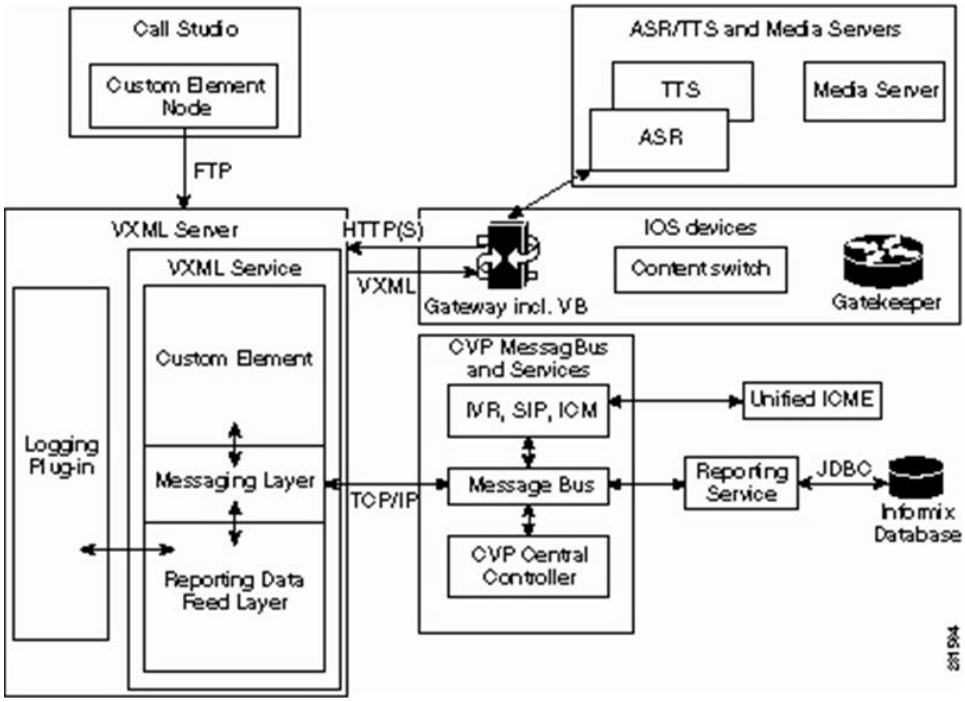
Configuring VXML Solution

Unified CVP provides a platform for developing powerful, speech-driven interactive applications accessible from any phone.

VXML solution consists of:

- *VXML Server*, a J2EE-compliant application server that dynamically drives the caller experience.
- *VoiceXML Service*, provides Unified ICME with call control capabilities and allows data to be sent to the Reporting Service.
- *Call Studio*, a drag-and-drop graphical user interface (GUI) for the rapid creation of advanced voice applications.

Figure 34: Unified CVP's VXML Solution



Note: This section focuses on how to *integrate* the Unified CVP solution with other Cisco products. For complete details on how to use the VXML Server software, refer to **User Guide for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html) .

This chapter contains the following topics:

About VXML Server

The VXML Server is a J2EE-compliant application server that provides a complete solution for rapidly creating and deploying dynamic VXML-based self service applications.

You can use the VXML Server:

- As an autonomous standalone component, without the Unified CVP Call Server and Unified ICME components. In this configuration, the VXML Server is designed to handle self-service VXML applications. Additionally, in a deployment that includes a Call Server the VXML Server is capable of consulting Unified ICME directly to obtain information or transfer destination labels.

- As an add-on component to a Unified CVP H.323 Service, SIP Service, or Call Server in an integrated Unified CVP deployment with Unified ICME. In this configuration, the H.323 or SIP Service:
  - Maintains control of the call flow, transferring calls to appropriate targets, under the direction of a script running on Unified ICME.
  - Can use the Cisco Voice Gateway's prompt-and-collect features by making requests to the VXML Server.

Note: For detailed information on deployment options for VXML Server, refer to "Configuration Overview (page 25)." For instructions on how to install a VXML Server, refer to *Installation and Upgrade Guide for Cisco Unified Customer Voice Portal*.

About VoiceXML Service

The VoiceXML Service provides Unified ICME call control capabilities and data to the Reporting Service.

The VoiceXML Service:

- Resides outside of the Call Server that gives call control capabilities to the Standalone Mode.
- Is the connection between the VXML Server and the ICM Service that feeds data to the Reporting Service.
- In a Standalone Mode with ICM Lookup deployment:
  - Interacts with the VXML Server and the ICM Service to do call control piece
  - Interacts with VXML Server and Reporting Service to populate the Reporting database.

Note: For more information, refer to Using Call Studio's ReqICMLabel Element to Pass Data (page 209).

About Call Studio

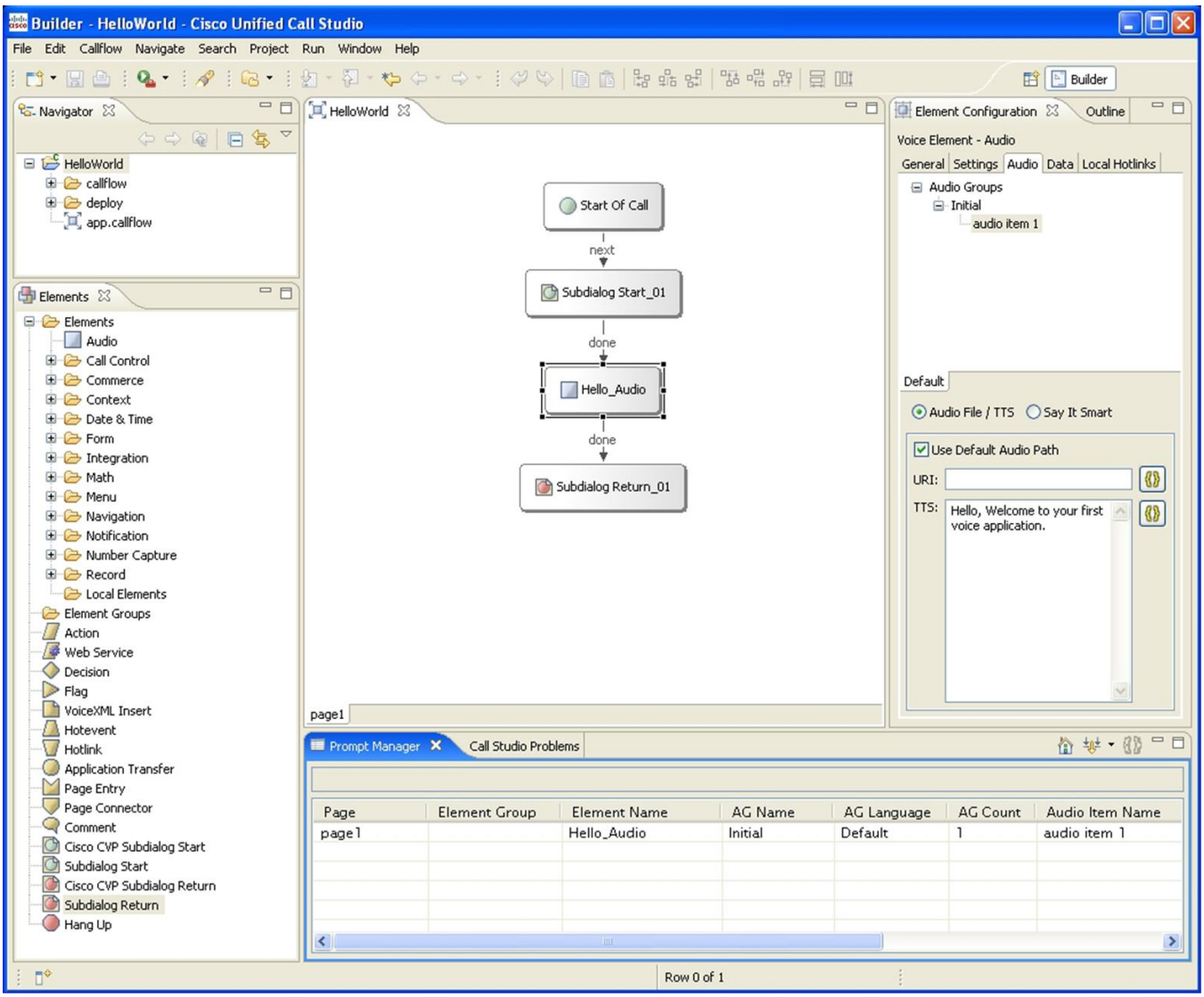
Call Studio is a development platform for the creation of voice applications.

Call Studio provides:

- A framework on which a whole host of Unified CVP and third-party tools will appear with a robust, consistent interface for voice application designers and developers to use.
- A true control panel for developing all aspects of a voice application, each function implemented as a plug-in to the greater Call Studio platform.

The following figure shows an example of a Call Studio application.

Figure 35: Call Studio Application



Note: For more information about Call Studio applications, refer to **User Guide for Cisco Unified CVP VXML Server and Cisco Unified Call Studio** (User Guide for Cisco Unified CVP VXML Server and Unified Call Studio).

VXML Server Reporting

VXML Server applications can be designed to function in a wide range of paradigms, from the VXML Server virtually controlling the entire user interaction to performing individual interactions on a scale similar to that of the Unified CVP micro-applications. Between these extremes, the VXML Server applications might be designed to implement specfic transactions. For example, in a banking application a transaction might consist of all the user interactions required to successfully complete a balance transfer or a telephone bill paymentThe high-level menus which the user might use to select a particular type of transaction would be controlled by the Unified ICME routing script, using standard Unified CVP micro-applications such as Menu and Play Media. Once a particular transaction type has been chosen, the Unified ICME routing script would issue an External VoiceXML micro-application to invoke the appropriate

VXML Server application which implements that transaction type. Once the VXML Server application completes, control returns to the Unified ICME routing script for further menus. Typically, audit information about the transaction would be returned, and can be stored in the Unified ICME database. It is also used to determine whether the transaction was successful or unsuccessful, needs to be transferred or queued to an agent, and so forth.

While Unified ICME VRU Progress reporting capabilities are always in effect, they compliment VXML Server applications most effectively when this transaction-oriented design is used. The customer would define a Unified ICME CallType for each type of transaction, and use the audit information returned from the VXML Server to determine how to set the Unified ICME's VRUProgress variable. The setting selected dictates how the transaction will be counted in the aggregate VRU reporting fields in the CallTypeHalfHour table.

VRU reporting enhancements are described in the Unified ICME 6.0(0) and later WebView Online Help.

WebView Reporting

This section lists the Unified CVP-related WebView reports that are available when using Unified CVP integrated with VXML Server.

Traditional VRU Services Reports

- persvc20:Peripheral Service for IVR Queue Half Hour Report
- persvc21:Peripheral Service IVR Queue Daily Report
- persvc22:Peripheral Service IVR Self-Service Half Hour Report
- persvc23:Peripheral Service IVR Self-Service Daily Report
- Caltyp35:VRU Calls Analysis Half Hour Report
- Caltyp36:VRU Calls Analysis Daily Report
- periph06:VRU Peripheral Capacity Report

Unified CVP Related Data

- trkgrp04:Trunks Real Time All Fields Report
- trkgrp12:Trunks Half Hour All Fields Report
- trkgrp20:All Port Busy Real Time
- trkgrp21:IVR Ports Idle & In Service Real Time Report
- trkgrp22:IVR Ports Status Real Time Ports
- trkgrp23:IVR Ports Performance Half Hour

- nettrk03:Network Trunk Group Real Time All Field Report
- nettrk13:Network Trunk Group Historical All Fields Report

Note: Currently, the VXML Server has the following reporting limitations:

- The VXML Server does not provide any pre-defined reports for Historical reporting
- The VXML Server provides real-time reporting capabilities that help monitor aggregate active sessions associated with a voice application. However, these active sessions cannot be mapped to the actual calls that triggered those sessions.

Configuring a VXML Server

The VXML Server is an optional J2EE-compliant application server that provides a complete solution for rapidly creating and deploying dynamic VoiceXML applications. If you installed a VXML Server, you must configure it before using it to deploy VoiceXML applications or licenses.

If you are using a VoiceXML gateway to route calls from the VXML Server but want to use the Unified CVP reporting feature, install the Call Server and Reporting Server on the same physical machine. Configure the Call Server with no call services enabled, then configure the Reporting Server and select the Call Server that is installed on the same machine (same IP address) as the primary Call Server for the Reporting Server.

If you need to make requests to an ICM server, without relinquishing control of the call or use Unified CVP reporting, you must configure the VXML Server to use a Call Server with at least the ICM Service enabled.

The Operations Console online help topic *Configuring a VXML Server* provides details for performing the following tasks:

- Adding a VXML Server
- Editing a VXML Server
- Deleting a VXML Server
- Uploading a Syslog XML File to a VXML Server
- Downloading a Syslog XML File from a VXML Server
- Editing the Log Messages XML File
- Transferring Script Files to a VXML Server
- Applying a License to a VXML Server

When you are creating a new VXML Server, you must apply a valid license file before using the server. You can browse for and upload the license file to the Operations Console, and then transfer the license to the VXML Server. Select either an existing license file in the Operations Console database or a new license file from your local PC. For more information on licensing, refer to *Installation and Upgrade Guide for Cisco Unified Customer Voice Portal*.

- Finding a VXML Server

  The Operations Console lets you locate a VXML Server from a list of configured servers or on the basis of specific search criteria.

- Viewing Device Status

The online help also contains two tables describing the VXML Server properties:

- VXML General Properties (refer to the help topic *VXML Server General Properties*

  The table includes settings that identify the VXML Server and choose a primary, and optionally, a backup Call Server to communicate with the Reporting Server. It also includes the setting to enable secure communications between the Operations Console and the VXML Server.

- VXML Server Configuration Properties (refer to the help topic *VXML Server Configuration Properties*

  From the VXML Server Configuration tab, you can enable reporting of VXML Server and call activities to the Reporting Server. When enabled, the VXML Server reports on call and application session summary data.

  Call summary data includes call identifier, start and end timestamp of calls, ANI, and DNIS. Application session data includes application names, sessionid, and session timestamps.

  If you choose detailed reporting, VXML Server application details are reported, including element access history, activities within the element, element variables and element exit state. Customized values added in the Add to Log element configuration area in Call Studio applications are also included in reporting data. Optionally, you can create report filters that define which data are included and excluded from being reported.

Use Inclusive and Exclusive VoiceXML filters to control the data that the VXML Server feeds to the Reporting Server. Data feed control is crucial for:

- Saving space in the reporting database

- Preserving messaging communication bandwidth

Refer to the Operations Console online help topic *Inclusive and Exclusive VoiceXML Filters for Reporting* for step-by-step configuration details.

You can also transfer script, and other files, to the VXML Server using the Operations Console. Refer to the Operations Console online help subtopic *Configuring a VXML Server > Transferring Script Files*.

Associating a VXML Server with a Call Server

The VXML Server uses the message service on a Call Server to communicate with the Reporting Server.

Steps to Associate a VXML Server with a Call Server

This section describes how to specify a primary and backup Call Server for the VXML Server.

Step 1 Open the Operations Console.

Step 2 Choose **Device Management > VXML Server**.

Step 3 Click **Add New** or select an existing VXML Server name and click **Edit**.

The General tab displays.

Step 4 Select the **Primary Call Server** from the drop-down list.

Step 5 Select the **Backup Call Server** from the drop-down list.

Step 6 Click **Save** to save the settings in the Operations Server database and click **Save & Deploy** to apply the changes to the VXML Server.

Step 7 Restart the VXML Server and the Call Servers..

Enabling Security for a VXML Server

By default, the communication channel between Operations Console and Unified CVP Resource Manager is non-secure at the completion of Unified CVP installation. To set up the VXML Server to accept *only* secure SSL connections from the Operations Console, you must manually enable a security setting.

Note:

- For more information, refer to Configuring and Modifying Unified CVP Security (page 235).
- Whenever this security setting is modified, you must restart the Cisco CVP Resource Manager service on the machine where the device is running.

Steps to Enable Security for a VXML Server

This section describes how to enable security for the VXML Server.

Step 1 Open the Operations Console.

Step 2 Choose **Device Management > VXML Server**.

Step 3 Click **Add New** or select an existing VXML Server name and click **Edit**.

The General tab displays.

Step 4 Select the **Security On** checkbox.

Step 5 Click **Save** to save the settings in the Operations Server database and click **Save & Deploy** to apply the changes to the VXML Server.

Step 6 Restart the VoiceXML Service.

Configuring VXML Server Applications: Filters

This section describes how to configure filters for VXML Server applications detail reporting.

Step 1 From the Operations Console.

Step 2 Choose **Device Management > VXML Server**.

Step 3 Click **Add New** or select an existing VXML Server name and click **Edit**.

Step 4 Select the **Configuration Tab** and use the **Inclusive Filters** and **Exclusive Filters** fields to specify the list of applications, element types, element names, element fields, and ECC variables to include and exclude from reporting data.

For information, refer to the Operations Console online help topic *Configuring a vXML Server > Inclusive and Exclusive VoiceXML Filters for Reporting*. This topic includes the subtopic *Rules for VoiceXML Inclusive and Exclusive Filters*.

Step 5 Click **Save**.

Step 6 Restart the VoiceXML Service.

Enabling Reporting for the VXML Server

This section describes how to configure reporting settings for the VXML Server.

Step 1 Open the Operations Console.

Step 2 Choose **Device Management > VXML Server**.

Step 3 Click **Add New** or select an existing VXML Server name and click **Edit**.

Step 4 Select the **Configuration Tab** and set the following:

- Click the **Yes** button to **Enable reporting for this VXML Server**. Indicates whether or not the VXML Server sends data to the Reporting Server. If disabled, no data is sent to the Reporting Server and reports will not contained any VoiceXML application data.
- Click the **Yes** button to **Enable reporting for VXML application details**. Indicates whether or not VoiceXML application details are reported.
- Specify a **Max. Number of Messages** to save in a file if failover occurs and the Reporting Server is unreachable. Limited by the amount of free disk space.

Step 5 Click **Save**.

Step 6 Restart the VoiceXML Service and the Call Server.

Configuring QoS for the VXML Server

Quality of Service (QoS) is the measure of transmission quality and service availability of a network (or internetworks).

Note: For more information about defining QoS criteria, refer to *Enterprise QoS Solution Reference Network Design Guide*.

Steps to Configure QoS for the VXML Server

This section describes how to configure QoS settings for the VXML Server.

Step 1 Open the Operations Console.

Step 2 Choose **Device Management > VXML Server**.

Step 3 Click **Add New** or select an existing VXML Server name and click **Edit**.

Step 4 Select the **Configuration Tab** and use the drop-down list to choose a **Select QoS Level**.

Note: For more information, refer to*Enterprise QoS Solution Reference Network Design Guide*.

Step 5 Click **Save**.

Step 6 Restart the VoiceXML Service.

Configuring a VXML Server (Standalone)

In the Unified CVP VXML Server (standalone) call flow model, the Call Server routes messages between the components. Calls arrive through a VoiceXML gateway and interact directly with

a VXML Server to execute VXML applications. The gateway performs both Ingress and VoiceXML functions. This call flow model provides a sophisticated VoiceXML-based VRU, for applications which in many cases do not need to interact with an ICM Server.

The VXML Server (standalone) does not provide statistics, and no database can be configured to capture data from standalone VoiceXML applications.

In this call flow model, you should configure the Call Server with no services enabled.

Refer to the Operations Console online help topic *Managing Devices > Configuring a VXML Server (Standalone)* for instructions on performing the following tasks:

- Adding a VXML Server (Standalone)

  Before adding a VXML Server (standalone) in the Operations Console, collect the hostname and IP address of the VXML Server.

- Editing a VXML Server (Standalone)

- Deleting a VXML Server (Standalone)

  Deleting a VXML Server (standalone) from the Operations Console deletes its configuration data in the Operations Console database and removes the VXML Server from the displayed list of VXML Servers.

- Finding a VXML Server (Standalone)

  You can locate a VXML Server on the basis of specific criteria entered into the search tool within the Operations Console, or you can identify the server from the list of servers displayed in the Operations Console.

- Applying a License to a VXML Server (Standalone)

Using VXML Server with Unified ICME

This section describes how to integrate VoiceXML and Unified ICME scripts.

Integrating VoiceXML Scripts with Unified ICME Scripts

This section describes how to integrate the VXML Server into the Unified CVP solution. This process involves:

- Creating a Unified ICME script with ECC variables configured for VXML Server.
- Creating a VRU Script to run in the Unified ICME script.

The following steps describe how to call the VXML Server from an Unified ICME script.

Step 1 Specify the URL (remove and port number) of the VXML Server that you want to reach, for example:

`http://12.34.567.890:7000/CVP/Server?application=HelloWorld`

In the example above, **`12.34.567.890`** is the IP address of the VXML Server, **`7000`** is the port number, and the application name is HelloWorld. The values are delimited by a colon (:).

Note: 7000 is the default port number for a VXML Server. The new port for Unified CVP 4.0 and later is 7000 for both Tomcat and WebSphere with VXML Server.

Step 2 In the Unified ICME script, first set the media_server ECC variable to:

`http://12.34.567.890:7000/CVP`

Step 3 Set the app_media_lib ECC Variable to "..", (literally two periods in quotes).

Step 4 Set the user.microapp.ToExtVXML[0] ECC variable to: **`application=HelloWorld`**

Note: This example indicates that theVXML Server will execute the “HelloWorld” application. To execute a different application, change the value of user.microapp.ToExtVXML[0] accordingly.

Step 5 Set the UseVXMLParams ECC Variable to "**`N`**."

Step 6 Create a Run External Script node within the Unified ICME script with a VRU Script Name value of **`GS,Server,V`**.

Note: Remember to link this node to the nodes configured in previous steps.

- The timeout value set in the Network VRU Script should be substantially greater than the length of the timeout in the VXML Server application. This timeout should only be used for recovery from a failed VXML Server.
- Always leave the **Interruptible** checkbox in the Network VRU Script Attributes tab checked; otherwise, calls queued to aVXML Server application might stay in the queue when an agent becomes available.

Step 7 After you configure the Unified ICME script, you need to configure a corresponding VXML Server script with Call Studio.

The VXML Server script must:

- Begin with a Unified CVP Subdialog_Start element (immediately after the Call Start element)
- Contain a Unified CVP Subdialog_Return element on all return points (script must end with a Subdialog_Return element)
- The Unified CVP Subdialog_Return element must include a value for the call input
- To enable reporting, you must add Data Feed/SNMP loggers

Correlating Unified CVP/Unified ICME Logs with VXML Server Logs

When using the VXML Server option in the Unified CVP solution, you can correlate Unified CVP/Unified ICME logs with VoiceXML logs. To do this, you need to pass the Call ID to the VXML Server by URL. Building upon the URL used in the previous example, the URL would be as follows:

```
http://12.34.567.890:7000/CVP/
Server?application=Chapter1_HelloWorld&callid=XXXXX-XXXXX-XXXXXX-XXXXXX
```

Note: Starting in Release 8.0(1), VXML Server (by default) receives callid (which contains the call GUID), _dnis, and _ani as session variables in comprehensive mode even if the variables are not explicitly configured as parameters in the ToExtVXML array. If the variables are configured in ToExtVXML then those values are used. These variables are available to VXML applications as session variables and they are displayed in the VXML Server log. This change is **backwards compatible** with the following script. That is, if you have added the following script, you do not need to change it.**However**, if you remove this script, you **save an estimated 40 bytes of ECC variable space**.

To configure logging, do the following:

Step 1 In the Unified ICME script, use the formula editor to set ToExtVXML[1]. Set the value of ToExtVXML[1] variable to:

```
concatenate("callid=",Call.user.media.id)
```

Note:

- Always include "callid" when sending the call to the VXML Server using the Comprehensive call flow model. The Call ID can also be used in VXML Server (standalone) solutions.
- When you concatenate multiple values, use a comma for the delimiter. For example:
  ```
  concatenate("ICMInfoKeys=
  ",Call.RouterCallKey,"-",Call.RouterCallDay,"-",Call.RouterCallKeySequenceNumber)
  ```
- Refer to "Passing Information to the External VoiceXML (page 200)" for more information.

Error Codes for VXML Server

The following are some of the error codes that you may see with the VXML Server application:

- Error Code 40 -- System Unavailable

  This is returned if the VXML Server is unavailable (shutdown, network connection disabled, and so forth)

- Error Code 41 -- App Error

This is returned if a VXML Server application error occurs (For example, a java exception).

- Error Code 42 -- App Hangup

  This is returned if the Hang Up element is used instead of the Unified CVP Subdialog_Return element.

  Note: If the application is configured correctly, this should not occur.

- Error Code 43 -- Suspended

  This is returned if the VXML Server application is suspended.

- Error Code 44 -- No Session Error

  This is returned when an emergency error occurs (for example, an application is called that has not been loaded in the VXML Server application).

- Error Code 45 -- Bad Fetch

  This is returned when the VXML Server encounters a bad fetch situation. This code is returned when either a .wav file or an external grammar file is not found.

VXML Server (Standalone) Solutions

The VXML Server is a J2EE-compliant application server that provides a complete solution for rapidly creating and deploying dynamic VoiceXML applications. You can install the VXML Server as a standalone component, without the Unified CVP H.323 Service or Call Server components. The VXML Server (Standalone) is designed to handle self-service VoiceXML applications.

Note: Refer to **Hardware and Software System Specification for Cisco Unified Customer Voice Portal Software** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html) for the most current information on supported versions of Cisco IOS software.

Configuring the VXML Server (Standalone) with ICM Lookup Call Flow Model

To configure the VXML Server (standalone) with ICM Lookup callflow model, do the following:

Step 1 Copy the following files from the VXML Server CD to the gateway flash memory using tftp:

CVPSelfService.tcl

critical_error.wav

For example:

```
copy tftp: flash:CVPSelfService.tcl

copy tftp: flash:CVPSelfServiceBootstrap.vxml

copy tftp: flash:critical_error.wav
```

Step 2 Define the VXML Server applications on the gateway. The following lines show an example configuration:

```
service CVPSelfService flash:CVPSelfServiceBootstrap.vxml
!
service [gateway application name] flash:CVPSelfService.tcl
param CVPBackupVXMLServer 12.34.567.890
param CVPSelfService-port 7000
param CVPSelfService-app [name of application on the VXML Server,
exactly how it appears]
param CVPPrimaryVXMLServer 12.34.567.891
```

Note: CVPSelfService is required. Backup server is optional. For both the Tomcat Application Server and the WebSphere Application Server, set the port to 7000.

After completing the gateway configuration, run the following to load and activate the applications:

```
call application voice load CVPSelfService
call application voice load [gateway application name]
```

Step 3 Define a dial-peer for the gateway application, for example:

```
dial-peer voice [dial-peer unique ID] voip /* for IP originated call
*/
service [gateway application name]
incoming called-number [dialed number]
dtmf-relay rtp-nte h245-signal h245-alphanumeric
codec g711ulaw
!
dial-peer voice [dial-peer unique ID] pots /* for TDM originated calls
 */
service [gateway application name]
incoming called-number [dialed number]
direct-inward-dial
```

Step 4 Optionally, create another dial peer to do transfers using the Unified ICME label that is returned.

Step 5 Create the application in Call Studio. In the Call Studio application, the ReqICMLabel has two exit states: error and done. The done path grabs a transfer element to transfer the caller to that label. The gateway needs another dial peer to transfer the label it gets from this process (refer to previous step). If you want to do real transfers, you must have the transfer element set up inside the Call Studio application.

Step 6 Drag the ReqICMLabel element onto the application created in Call Studio and configure it.

Note: This step is necessary to obtain a label from Unified ICME. For more information, refer to Using Call Studio's ReqICMLabel Element to Pass Data (page 209).

Step 7 Save and deploy the application from Call Studio using the VoiceXML Service on the Operations Console.

Step 8 Install the Call Server; select only the Core Software component.

Step 9 Configure the VXML Server to communicate with the Call Server through the Operations Console. Refer to "Configuring a VXML Server (page 264)."

Step 10 Transfer the application using File Transfer to the VXML Server. This will automatically deploy the application on the selected VXML Server.

See Also

Refer to **User Guide for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html) for complete instructions.

Configuring the VXML Server (Standalone) Call Flow Model (without ICM Lookup)

To configure the VXML Server (Standalone) call flow model, do the following:

Step 1 Copy the following files from the VXML Server CD to the gateway flash memory using tftp:

CVPSelfService.tcl

critical_error.wav

For example:

```
copy tftp: flash:CVPSelfService.tcl
copy tftp: flash:CVPSelfServiceBootstrap.vxml
copy tftp: flash:critical_error.wav
```

Step 2 Define the VXML Server applications on the gateway. The following lines show an example configuration:

```
service CVPSelfService flash:CVPSelfServiceBootstrap.vxml
!
service [gateway application name] flash:CVPSelfService.tcl
param CVPBackupVXMLServer 12.34.567.890
param CVPSelfService-port 7000
param CVPSelfService-app [name of application on the VXML Server,
exactly how it appears]
param CVPPrimaryVXMLServer 12.34.567.891
```

Note: CVPSelfService is required. Backup server is optional. For both the Tomcat Application Server and the WebSphere Application Server, set the port to 7000.

After completing the gateway configuration, run the following to load and activate the applications:

```
call application voice load CVPSelfService
call application voice load [gateway application name]
```

Step 3 Define a dial-peer for the gateway application, for example:

```
dial-peer voice [dial-peer unique ID] voip /* for IP originated call
* /
service [gateway application name]
incoming called-number [dialed number]
dtmf-relay rtp-nte h245-signal h245-alphanumeric
codec g711ulaw
!
dial-peer voice [dial-peer unique ID] pots /* for TDM originated calls
 */
service [gateway application name]
incoming called-number [dialed number]
direct-inward-dial
```

Step 4 Create the application in Call Studio. This application *must* have the same name as the CVPSelfService-app defined in the gateway configuration above.

Step 5

If: If there is an Operations Console

Then: Save and deploy the Call Studio application locally; create a VXML Server (Standalone) configuration; and upload and transfer the application script file to the desired VXML Server.

If: there is *not* an Operations Console

Then: Save and Deploy the Call Studio Application to the desired installed VXML Server. Then, on the VXML Server, run the deployallapps.bat file (c:/Cisco/CVP/VXMLServer/admin directory) .

See Also

Refer to **User Guide for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_user_guide_list.html) for complete instructions.

Sample Gateway Configuration

VXML Server:

```
application
  service CVPSelfService flash:CVPSelfServiceBootstrap.vxml

  service HelloWorld flash:CVPSelfService.tcl
    param CVPBackupVXMLServer 12.34.567.890

    param CVPSelfService-app HelloWorld
    param CVPSelfService-port 7000
    param CVPPrimaryVXMLServer 12.34.567.891
```

```
dial-peer voice 4109999 voip /* for IP originated call */
  service HelloWorld
  incoming called-number 88844410..
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  codec g711ulaw

dial-peer voice 4109999 voip /* for TDM originated call */
  service HelloWorld
  incoming called-number 88844420..
  direct-inward-dial
```

Activate the Gateway Configuration

Activate the gateway configuration by entering these commands:

Step 1 **`call application voice load CVPSelfService`**

Step 2 **`call application voice load HelloWorld`**

Using Takeback and Transfer in VoiceXML Scripts

Unified CVP provides the following takeback and transfer methods that you can invoke from a VoiceXML script:

- Two B-Channel Transfer (TBCT) - A call transfer standard for ISDN interfaces. This feature enables a Cisco voice gateway to request an NI-2 switch to directly connect two independent calls. The two calls can be served by the same PRI or by two different PRIs on the gateway.
- Hookflash Relay - A brief interruption in the loop current that the originating call entity (PBX or Public Switch Telephone Network switch) does not interpret as a call disconnect. Instead, once the PBX or Public Switch Telephone Network switch senses the hookflash, it puts the current call on hold and provides a secondary dial tone, which allows Unified CVP to transfer the caller to another destination.
- SIP Refer - VoiceXML applications can use a SIP REFER transfer instead of a blind or bridged transfer. This allows Unified CVP to remove itself from the call, thus freeing up licensed Unified CVP ports. Unified CVP cannot execute further call control or IVR operations after this kind of label has been executed.

Configuring Two B-Channel Transfer (TBCT)

To configure Two B-Channel Transfer (TBCT) for use with Unified CVP from a VoiceXML script, follow this procedure:

Step 1 Configure the originating gateway for TBCT call transfer, by following the instructions in "Configuring the Gateway for Two B-Channel Transfer (TBCT) (page 435)."

Step 2 Locate the following files on the VXML Server and copy them to flash memory on the gateway, using the tftp command:

en_error.wav

en_holdmusic.wav

en_pleasewait.wav

survivability.tcl

CVPSelfService.tcl

CVPSelfServiceBootstrap.vxml

Step 3 Add the following lines to the gateway:

```
service takeback flash:survivability.tcl
param icm-tbct 1
```

Step 4 Configure the CVPSelfService application, as follows:

```
service [gateway application name] flash:CVPSelfService.tcl
param CVPBackupVXMLServer 12.34.567.890
param CVPSelfService-port 7000
param CVPSelfService-app [name of application on the VXML Server,
exactly how it appears]
param CVPPrimaryVXMLServer 12.34.567.891
```

Note: CVPSelfService is required. Backup server is optional. For both the Tomcat Application Server and the WebSphere Application Server, set the port to 7000.

Step 5 From command line mode:

```
call application voice load takeback
call application voice load CVPSelfService
```

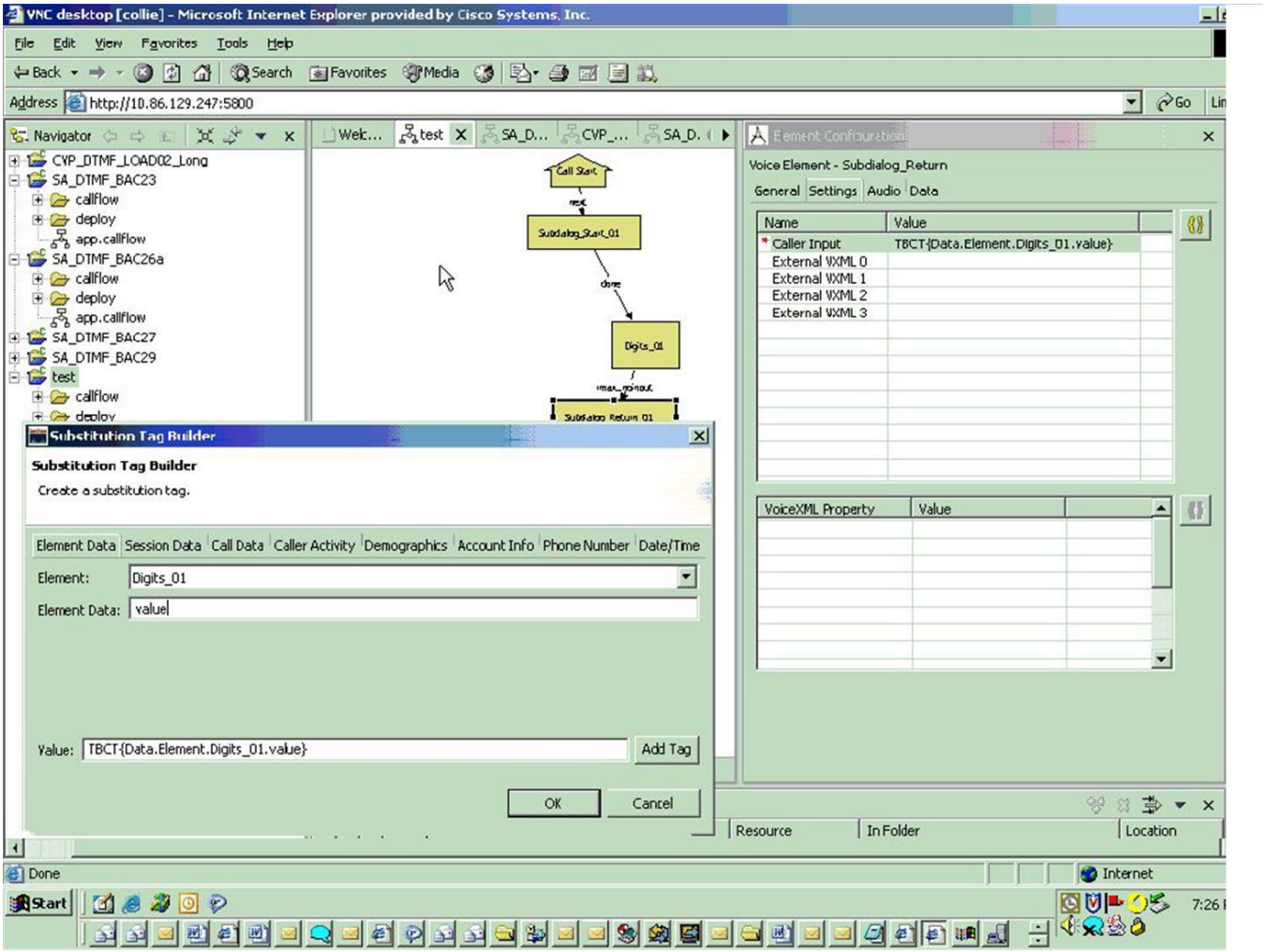
Step 6 Specify the target destination for the TBCT transfer either by entering the number manually, or dynamically by using caller input.

a. **Manually**. In the SubdialogReturn node in the VXML Server application, next to Caller Input in the Settings Tab, enter **`TBCT<target_destination_number>`**, where *target_destination_number* is the target destination of the TBCT transfer. For example:

TBCT8005551212

b. **Dynamically**. The target destination is created dynamically using input entered by the caller during the call. To do this, click the **Substitution** icon next to the Caller Input variable and select substitution values. For example:

Figure 36: Dynamic Target Destination



Step 7 In the event that the TBCT fails (for example, bad number, bad gateway config, etc.), you can specify alternate transfer targets under the survivability service according to survivability rules as defined in "Call Survivability (page 398)."

Configuring Hookflash Relay

To configure Hookflash Relay for use with Unified CVP from VoiceXML scripts, follow this procedure:

Step 1 Configure the originating gateway for Hookflash Relay call transfer, by following the instructions in "Configuring the Gateway for Hookflash Relay (page 434)."

Step 2 Locate the following files on the VXML Server and copy them to flash memory on the gateway.

en_error.wav

en_holdmusic.wav

en_pleasewait.wav

survivability.tcl

en_0.wav en_1.wav

en_2.wav en_3.wav

en_4.wav

en_5.wav

en_6.wav

en_7.wav

en_8.wav

en_9.wav

en_pound.wav

en_star.wav

Step 3 Add the following lines to the gateway:

```
service hookflash flash:survivability.tcl
```

Step 4 If you have not already done so, configure the CVPSelfService application as follows:

```
service [gateway application name] flash:CVPSelfService.tcl
param CVPBackupVXMLServer 12.34.567.890
param CVPSelfService-port 7000
param CVPSelfService-app [name of application on the VXML Server,
exactly how it appears]
param CVPPrimaryVXMLServer 12.34.567.891
```

Note: CVPSelfService is required. Backup server is optional. For both the Tomcat Application Server and the WebSphere Application Server, set the port to 7000.

Step 5 From command line mode:

```
call application voice load hookflash
call application voice load CVPSelfService
```

Step 6 In the SubdialogReturn node in your VXML Server application, next to Caller Input in the Settings Tab, enter **HF8005551212**, replacing 8005551212 with the target destination of the hookflash transfer.

The label can also be defined dynamically using digits entered by the caller in conjunction with the VXML Server substitution tags. If the switch requires a pause after the hookflash, commas can be inserted between the HF and the transfer number. Each comma represents 100ms.

Configuring SIP REFER

To configure SIP REFER for use with Unified CVP from a VoiceXML script, follow this procedure:

Step 1 Configure the gateway according to either "Configuring the VXML Server (Standalone) Call Flow Model (without ICM Lookup) (page274)" or "Configuring the VXML Server (Standalone) with ICM Lookup Call Flow Model (page 272)," whichever is appropriate for the implementation.

Note: The incoming dial-peer running the CVPSelfService application must be a voip dial-peer, not a pots dial-peer.

Step 2 Specify the target destination for the refer transfer in the Call Studio application either by entering the number manually, or dynamically using caller input.

a. **Manually**. In the SubdialogReturn node in the VXML Server application, next to CallerInput in the Settings tab, enter **`RF<target_destination_number>`**, where target_destination_number is the target destination of the REFER transfer; for example, RF8005551212.

b. **Dynamically**. The target destination is created dynamically using input entered by the caller during the call. To do this, click the **Substitution** icon next to the Caller Input variable and select the substitution values.

Step 3 The following configuration must be added to the gateway configuration for the handoff to survivability.tcl to occur and for the REFER to be sent:

```
service takeback flash:survivability.tcl
```

Configuring the Gateway for TDM to IP Calls

To transfer TDM call to IP, configure the gateway as follows:

Step 1 To configure the number dialed from theTDM phone to match, use the following configuration:

```
dial-peer voice [dialpeer name] pots
service [gateway application name]
incoming called-number [number dialed from phone]
direct-inward-dial
```

Step 2 To configure the number dialed from the IP phone to match, use the following configuration:

```
dial-peer voice [dialpeer name] voip
service [gateway application name]
incoming called-number [number dialed from phone]
dtmf-relay rtp-nte h245-signal h245-alphanumeric
codec g711ulaw
```

Step 3 To configure the transfer node to transfer to a specific extension (for example, 5080001):

Note: The gateway sends the call to Unified CM as a blind transfer.

```
dial-peer voice [dialpeer name] voip destination-pattern 508....
session target ipv4:[call manager subscriber IP]
dtmf-relay rtp-nte h245-signal h245-alphanumeric
codec g711ulaw
no vad
```

Step 4 To hear a ring tone on a blind transfer call, do the following:

- Select **Service > Service Parameters** in Cisco Unified Communications Manager.
- Select your server and the Unified CM Service.
- Set the Send H225 User Info Message setting to **H225 Info for Call Progress Tone**.

Passing Data to Unified ICME

In the VXML Server (standalone) with ICM Lookup call flow model, Unified ICME sends a label to Unified CVP. This process requires some configuration, which is explained below.

The Standalone with Request ICM Label variation of the Standalone call flow model performs a route request to Unified ICME, and then Unified ICME starts a script (new call). Unified ICME sees whatever the device puts in the new call message, then Unified ICME chooses a target, such as an agent, and sends a label back to the device. That route request to Unified ICME sends other information, such as ECC variables. Unified ICME can pass other ECC variables back to Unified CVP. Also, you need to configure a VXML Server in the Unified CVP Call Server for the call flow model.

Configuring the Connections

The instructions below describe how to set up aVXML Server that connects to the Unfied CVP Call Server through the ICM Service, and the connection from the ICM Service to the PG.

Note: The VRU PIM initiates the connection from the PG to the Call Server. The ICM Service listens for a connection from the VRU PIM.

Step 1 Start the VXML Server. The VXML Server starts the VoiceXML Service using the DataFeed mechanism or the ReqICMLabel element.

The ReqICMLabel element allows a Call Studio script to pass caller input, call variables, and External Call Context (ECC) variables to a Unified ICME script. The ReqICMLabel must be inserted into a Call Studio script as a decision element. In Call Studio, the returned Unified ICME label contains a result which can be used by other elements in the same application, such as the Transfer or Audio element. The Transfer element sends instructions to the IOS Voice Browser to transfer the caller to the desired location.

Once the VoiceXML Service starts, it begins communicating with the ICM Service.

Step 2 From the Operations Console, add a Call Server.

Step 3 On the General tab, enable the ICM Service by clicking the **ICM** checkbox.

Step 4 On the ICM General Configuration tab, specify the VRU Connection Port (by default this value is 5000). This is the port on which the ICM Service listens for a connection from the VRU PIM upon startup of the Call Server.

Note: Refer to the Unified ICME documentation for instructions on configuring the VRU PIM to connect to a VRU (for example, Unified CVP).

Step 5 Set up the following items in Unified ICME: Dialed Number (DN), customer records which must match the Network VRU, call type, and a script.

When setting up the script, be aware that only the following script nodes are allowed:

- Label
- Select node
- LAA and MED agents

Caution: The Run External Script node and queuing are not allowed.

See Also

Refer to the **Unified ICME Documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html) for more information.

Configuring the Gateway for IP to TDM Calls

The following components are required in order for the gateway to process IP to TDM calls:

- Phones and numbers must be configured on the TDM switch.
- Gateway must be defined on Unified CM.
- Route pattern on Unified CM that will send the call to the gateway.
- Need to configure a dial peer on the gateway that sends calls.

- Dial 888800605x on the IP phone (this is a specific physical phone extension).

Step 1 Configure the gateway to send the call to a particular VXML Server application, as follows:

```
dial-peer voice 8888 voip
service [gateway application name]
incoming called-number 888800....
dtmf-relay rtp-nte h245-signal h245-alphanumeric
codec g711ulaw
no vad
```

Step 2 To match the number in the VXML Server transfer node and send it out the T1 port to the G3 to its destination, use the following configuration:

```
dial-peer voice 8880 pots
destination-pattern 888800....
incoming called-number
direct-inward-dial
port 1/0:D
```

Configuring a Cisco Multiservice IP-to-IP Gateway for Unified CM Connections

Unified CM IP- to-IP calls that are routed through an H.323 endpoint or H.323 gateway, you must configure the following:

Step 1 Configure the IPIPGW to transport calls to and from Unified CM.

For detailed information on configuring the Cisco IOS gateway for Unified CM connections, refer to the Cisco Multiservice IP-to-IP Gateway Software documentation.

SNMP Monitoring for the VXML Server

SNMP monitoring for the VXML Server is provided by default when a Call Studio application is created. CVPSNMPLogger logs error events it receives from the VXML Server; for example, this process allows you to configure sending a page to a technical support representative when a particular error alert is triggered on the customer site.

Note: By default, CVPSNMPLogger is enabled when a new Call Studio application is created and deployed to the VXML Server.

Step 1 To view CVPSNMPLogger for the VXML Server, access the Call Studio interface.

Step 2 Right-click the application and select **Properties > Cisco Unified CVP > General Settings** from Call Studio for each Call Studio application. CVPSNMPLogger displays in the Loggers list box.

Caution: Do not remove CVPSNMPLogger. Doing so will disable viewing of SNMP events and alerts.

Chapter 7

Configuring Unified CVP Logging and Event Notifications

Unified CVP provides information about component device status and interaction through:

- Logs, which are presented in text format and can be viewed using Cisco Support Tools.
- Statistics, which can be viewed using the Unified CVP Operations Console.

This chapter also provides information about SNMP alerts for Cisco Unified Presence Server (page 309) and CVP SNMP-Raise/Clear Mappings (page 310).

This chapter contains the following topics:

Using Syslog

Unified CVP allows you to configure primary and backup syslog servers; however, it is important to note that failover from primary to backup server is not guaranteed. When the primary syslog server goes down (that is, the entire machine, not just the syslog receiver application), Unified CVP relies on the host operating system and the Java Runtime Environment for notification that the destination is not reachable. Because the semantics of this notification do not guarantee delivery, Unified CVP cannot guarantee failover.

Using Logs to Interpret Events

You can use the **CVPLogMessages.xml** file to help interpret events. This file contains all of the messages (or notifications) on SNMP events and/or through Syslog.

Note: The **CVPLogMessages.xml** file applies to all Unified CVP Services, except H.323.

Each event in the **CVPLogMessages.xml** field containing information that might be useful for correcting any problems indicated by the event.

Note: Be aware that the <resolution> field might not always contain as much information as the **Troubleshooting Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_troubleshooting_guides_list.html) or other Unified CVP documentation, and should be considered along with all other resources when troubleshooting a problem.

The sections that follow provide details about editing, uploading, and downloading the CVPLogMessages.xml file from the Operations Console.

Editing the Log Messages XML File

The log messages XML file, **CVPLogMessages.xml**, defines the severity, destination (SNMP management station or Syslog server) and possible resolution for Unified CVP log messages. This file also identifies an event type identifier and message text identifier for each event. The text for these identifiers is stored in the resource properties file, **CVPLogMessagesRes.properties**.

Each Unified CVP Call Server, VXML Server, and Reporting Server has a log messages XML file and log message file. You can edit the **CVPLogMessages.xml** file on a particular Unified CVP server to customize the severity, destination and possible resolution for each event that the server generates. You can also edit the **CVPLogMessagesRes.properties** file to change the text of the message that is generated when an event occurs on that server.

Use any plain-text editor (one that does not create any markup) or XML editor to edit the **CVPLogMessages.xml** file. Use a resource file editor, if available, to edit the **CVPLogMessagesRes.properties** file. If a resource file editor is not available, use a text editor.

Message Element	Possible Values	What it Means
Name	Resource="identifier"	Identifies the event type described in the **CVPLogMessagesRes.properties** file.
Body	Resource="identifier"	Identifies the message text described in the **CVPLogMessagesRes.properties** file.
Severity	0 to 6	Identifies the severity level (page ?) of the event.
SendToSNMP	True or false	Set to true, to send this message, when logged, to an SNMP manager, if one is configured.

Message Element	Possible Values	What it Means
SendToSyslog	True or false	Set to true to send this message, when logged, to a Syslog server, if one is configured.
SNMPRaise	True or false	Set to true to identify this message, when logged, as an SNMP raise event, which the SNMP management station can use to initiate a task or automatically take an action. Set to false to identify this message as an SNMP clear when sent to an SNMP management station. An SNMP clear event usually corresponds to an SNMP raise event, indicating that the problem causing the raise has been corrected. An administrator on an SNMP management station can correlate SNMP raise events with SNMP clear events.

VoiceXML Logs

Note: Refer to **Cisco Support Tools** (http://www.cisco.com/en/US/products/ps5905/tsd_products_support_series_home.html) for more information about viewing logs.

About VoiceXML Logs

VoiceXML logs record Unified CVP system-specific information, such as heartbeat status. By default, VoiceXML logs are stored in the \Cisco\CVP\logs\VXML folder.

Note: These logs are configured and viewed using Support Tools.

The table that follows describes the logs that VoiceXML creates.

Log Type	Log Name	Description
Infrastructure	CVP.<timestamp>.log	Unified CVP logs for the VoiceXML Service: This includes Notice, Info, and Debug logs. With Debug turned on, you can also see Call, Message, and Method trace types of logs.
Error messages	Error.<timestamp>.log	Unified CVP error log: This contains any error that Unified CVP Services and message layer has generated.

Correlating Unified CVP/Unified ICME Logs with VXML Server Logs

When using the VXML Server option in the Unified CVP solution, you can correlate Unified CVP/Unified ICME logs with VoiceXML logs. To do this, you need to pass the Call ID to the VXML Server by URL.

Note: Starting in Release 8.0(1), VXML Server (by default) receives callid (which contains the call GUID), _dnis, and _ani as session variables in comprehensive mode even if the variables

are not explicitly configured as parameters in the ToExtVXML array. If the variables are configured in ToExtVXML then those values are used. These variables are available to VXML applications as session variables and they are displayed in the VXML Server log. This change is **backwards compatible** with the following script. That is, if you have added the following script, you do not need to change it.**However**, if you remove this script, you **save an estimated 40 bytes of ECC variable space**.

Prior to Unified CVP release 8.0(1), to configure logging, do the following:

Step 1 In the Unified ICME script, use the formula editor to set ToExtVXML[1]. Set the value of ToExtVXML[1] variable to **`concatenate("callid=",Call.user.media.id)`**

Note:

- Always include "callid" when sending the call to theVXML Server using the Comprehensive call flow model. The Call ID can also be used in VXML Server (standalone) solutions.
- When you concatenate multiple values, use a comma for the delimiter; for example, **`concatenate("ICMInfoKeys= ",Call.RouterCallKey,"-",Call.RouterCallDay,"-",Call.RouterCallKeySequenceNumber).`**
- Refer to "Passing Information to the External VoiceXML (page 200)."

About VXML Server Logs

VXML Server logs record interactions between the VXML Server and the server that hosts the VoiceXML applications. By default, VXML Server logs are stored in the \Cisco\CVP\VXMLServer\logs folder.

The table that follows describes the logs that VXML Server creates.

Log Type	Log Name	Description
VXML Server Call Log	call_log<timestamp>.txt	Records a single line for every application visit handled by an install of VXML Server.
VXML Server Call Error Log	error_log<timestamp>.txt	Records errors that occur outside the realm of a particular application.
VXML Server Administration History Log	admin_history<timestamp>.txt	Records information from VXML Server administration scripts.

The VXML Server Call Error Log can contain the following error codes:

- Error Code 40 -- System Unavailable

  This is returned if the application server is unavailable (shutdown, network connection disabled, and so forth)

- Error Code 41 -- App Error

  This is returned if some VXML Server-specific error occurs (For example, java exception).

- Error Code 42 -- App Hangup

  This is returned to Unified CVP if the Hang Up element is used without being preceded by a Subdialog_Return element.

  Note: If the application is configured correctly, this should not occur.

- Error Code 43 -- Suspended

  This is returned if the VXML Server application is suspended.

- Error Code 44 -- No Session Error

  This is returned when an emergency error occurs (for example, an application is called that has not been loaded in the VXML Server application).

- Error Code 45 -- Bad Fetch

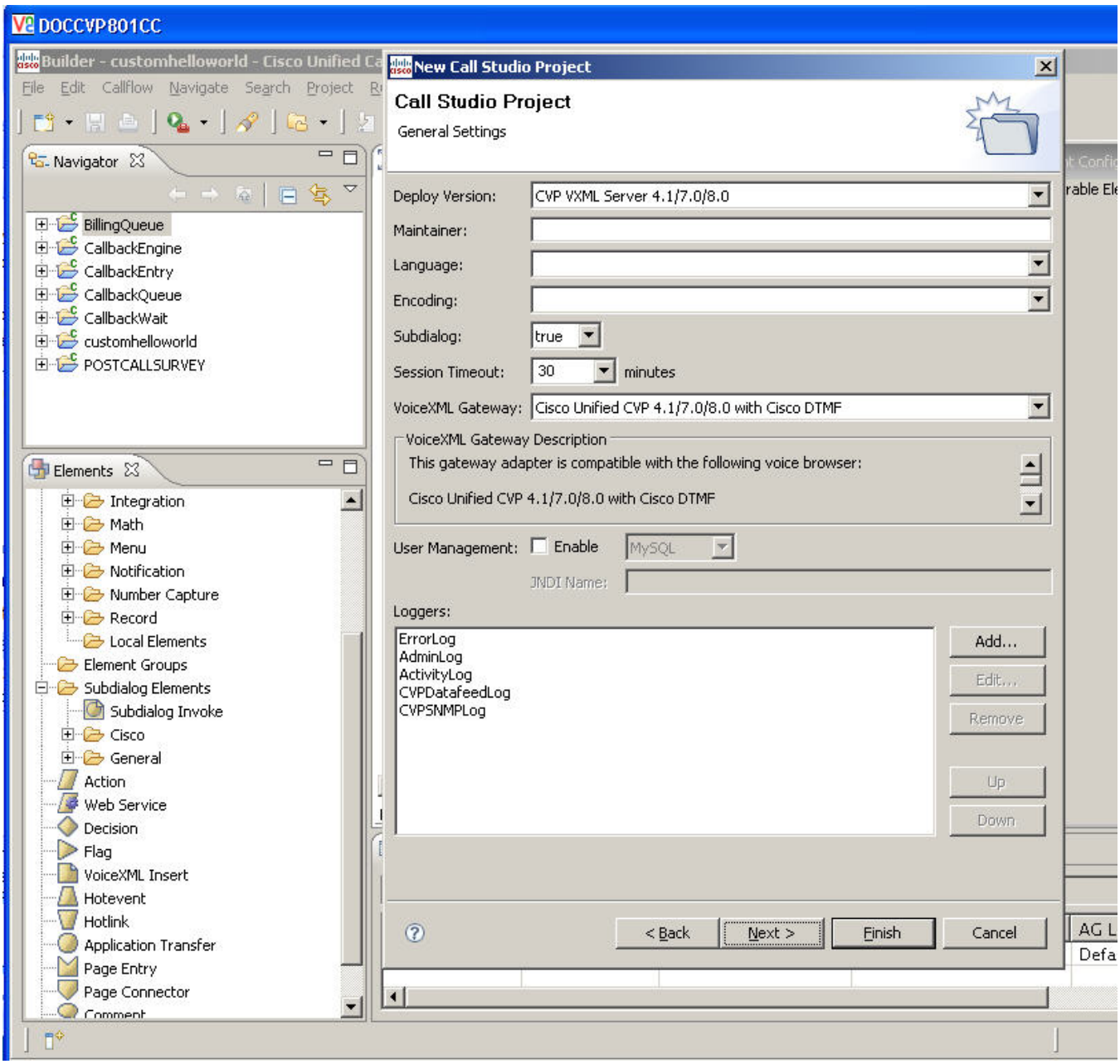
  This is returned when theVXML Server encounters a bad fetch situation.This code is returned when either a .wav file or an external grammar file is not found.

About VoiceXML Application Logging

The VXML Server creates several logs for each individual VoiceXML application. By default, these application logs–with the exception of CVPDatafeedLog and CVPSNMPLog–are stored in the \Cisco\CVP\VXMLServer\applications\<NAME of APPLICATION>\logs folder.

You configure these logs using Call Studio.

Figure 37: Call Studio Logger



Note: Refer to **Element Specifications for Cisco Unified CVP VXML Server and Unified Call Studio** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_programming_reference_guides_list.html) for details on configuring loggers.

The table that follows describes the logs that are created for each application.

Application Logger Type	Log Name	Description
ActivityLog	activity_log<timestamp>.txt **Note:** Log files are stored in the ActivityLog directory.	Records all application activity, which will show which elements were entered and exited during a call. Default setting: on

Application Logger Type	Log Name	Description
ErrorLog	error_log<timestamp>.txt **Note:** Log files are stored in the ErrorLog directory.	Records all error messages for the application. Default setting: on
AdminLog	admin_history<timestamp>.txt **Note:** Log files are stored in the AdminLog directory.	Records information from application-specific administration scripts. Default setting: on
CVPDatafeedLog	CVPDatafeed.log. **Note:** This log is stored in \Cisco\CVP\logs\VXML folder.	Listens for logging events and provides VXML Server and VoiceXML Service data to the Unified CVP Reporting Server. The Unified CVP Reporting Server stores this information in a reporting database so that it is available for later review. One CVPDatafeedLog is created per application. Default setting: on **Note:** The VoiceXML Service can be started by adding this logger in the VoiceXML application.
CVPSNMPLog	CVPSNMP.log. **Note:** This log is stored in \Cisco\CVP\logs\VXML folder.	Listens for a set of events and sends information about these events to the SNMP log, Syslog, or Unified CVP log. Default setting: on
DebugLog	debug_log<timestamp>.txt. **Note:** Log files are stored in the DebugLog directory.	Creates a single file per call that contains all HTTP requests and responses that occurred between a IOS Gateway and VXML Server during the call session. Default setting: off

About Event Statistics

You can monitor the following statistics through the Operations Console Control Center:

- Device statistics
- Infrastructure statistics
- ICM Service call statistics

- IVR Service call statistics
- SIP Service call statistics
- H.323 Service call statistics
- Gatekeeper statistics
- Gateway statistics
- VXML Server statistics
- Reporting Server statistics

Infrastructure Statistics

Unified CVP infrastructure statistics include realtime and interval data on the Java Virtual Machine (JVM), threading, and Licensing.

You can access these statistics by choosing Control Center from the System menu and then selecting a device. See the Operations Console topic *Viewing Infrastructure Statistics* for more information.

Access infrastructure statistics either by:

- Choosing **System > Control Center**, selecting a device, clicking the Statistics icon in the toolbar, and then selecting the **Infrastructure** tab.
- Selecting a device type from the **Device Management** menu, selecting a device, clicking the Statistics icon in the toolbar, and then selecting the **Infrastructure** tab.

The following table describes Licensing statistics.

Table 22: Licensing Statistics

Statistic	Description
Realtime Statistics	
Port Licenses Available	The number of port licenses available for the processing of new calls. Exactly one port license is used per call, independent of the call's traversal through the individual Call Server services.
Current Port Licenses in Use	The number of port licenses currently in use on the Call Server. Exactly one port license is used per call, independent of the call's traversal of the individual Call Server services.
Current Port Licenses State	The threshold level of port license usage. There are four levels: safe, warning, critical and failure. An administrator may set the required percentage of port licenses in use needed to reach a given threshold level, with the exception of the failure level which is reached when the number of ports checked out is equal to the total number of licenses ports.

Statistic	Description
Interval Statistics	
Start Time	The time the system started collecting statistics for the current interval.
Duration Elapsed	The amount of time that has elapsed since the start time in the current interval.
Interval Duration	The interval at which statistics are collected. The default value is 30 minutes.
Total New Port License Requests	The number of port license checkout requests made in the current interval. For each port license checkout request, whether it checks out new port license or not, this metric is increased by one.
Average License Requests/Minute	The average number of port license checkout requests made per minute in the current interval. This metric is calculated by dividing the port license requests metric by the number of minutes elapsed in the current interval.
Maximum Port Licenses Used	The maximum number of port licenses used during this time interval.
Aggregate Statistics	
Start Time	The time the service started collecting statistics.
Duration Elapsed	The amount of time that has elapsed since the service start time.
Total New Port License Requests	The number of port license checkout requests made since the system was started. For each port license checkout request, whether it checks out a new port license or not, this metric is increased by one.
Average License Requests /Minute	The average number of port license checkout requests made per minute since the system was started. This metric is calculated by dividing the aggregate port license requests metric by the number of minutes elapsed since the system was started.
Peak Port Licenses Used	The peak number of simultaneous port licenses used since the start of the system. When a port checkout occurs, this metric is set to the current port licenses in use metric if that value is greater than this metric's current peak value.
Total Denied Port License Requests	The number of port license checkout requests that were denied since the start of the system. The only reason a port license checkout request would be denied is if the number of port licenses checked out at the time of the request is equal to the total number of port license available. When a port license checkout is denied, the call does not receive regular treatment (the caller may hear a busy tone or an error message).

The following table describes thread pool system statistics.The thread pool is a cache of threads, used by Unified CVP components only, for processing of relatively short tasks. Using a thread pool eliminates the waste of resources encountered when rapidly creating and destroying threads for these types of tasks.

Table 23: Thread Pool Realtime Statistics

Statistic	Description
Realtime Statistics	
Idle Threads	The number of idle threads waiting for some work.

Statistic	Description
Running Threads	The number of running thread pool threads currently processing some work.
Core Threads	The number of thread pool threads that will never be destroyed no matter how long they remain idle.
Maximum Threads	The maximum number of thread pool threads that will ever exist simultaneously.
Peak Threads Used	The peak number of thread pool threads ever simultaneously tasked with some work to process.

The following table describes Java Virtual Machine statistics.

Table 24: Java Virtual Machine (JVM) Realtime Statistics

Statistic	Description
Realtime Statistics	
Peak Memory Usage	The greatest amount of memory used by the Java Virtual machine since startup. The number reported is in megabytes and indicates the peak amount of memory ever used simultaneously by this Java Virtual Machine.
Current Memory Usage	The current number of megabytes of memory used by the Java Virtual Machine.
Total Memory	The total amount of memory in megabytes available to the Java Virtual Machine. The number reported is in megabytes and indicates how much system memory is available for use by the Java Virtual Machine.
Available Memory	The amount of available memory in the Java Virtual Machine. The number reported is in megabytes and indicates how much of the current system memory claimed by the Java Virtual Machine is not currently being used.
Threads in Use	The number of threads currently in use in the Java Virtual Machine. This number includes all of the Unified CVP standalone and thread pool threads as well as those threads created by the Web Application Server running within the same JVM.
Peak Threads in Use	The greatest amount of threads ever used simultaneously in the Java Virtual Machine since startup. The peak number of threads ever used by the Java Virtual Machine includes all Unified CVP standalone and thread pool threads as well as threads created by the Web Application Server running within the same JVM.
Uptime	The length of time that the Java Virtual Machine has been running. This time is measured in hh:mm:ss and shows the amount of elapsed time since the Java Virtual Machine process began executing.

ICM Service Call Statistics

The ICM Service call statistics include data on calls currently being processed by the ICM service, new calls received during a specified interval, and total calls processed since start time.

Access ICM Service statistics either by:

- Choosing **System > Control Center**, selecting a CVP Call Server, clicking the **Statistics** icon in the toolbar, and then selecting the **ICM** tab.

- Choosing **Device Management > CVP Call Server**, selecting a Call Server, clicking the **Statistics** icon in the toolbar, and then selecting the **ICM** tab.

The following table describes ICM Service call statistics.

Table 25: ICM Service Call Statistics

Statistic	Description
Realtime Statistics	
Active Calls	The current number of calls being serviced by the ICM Server for a Unified CVP Call Server. This value represents a count of calls currently being serviced by the ICM for the Unified CVP Call Server for follow-on routing to a Contact Center agent.
Active SIP Call Legs	The ICM Server can accept Voice over IP (VoIP) calls that originate using either the Session Initiation Protocol (SIP) or H.323 protocolActive SIP Call Legs indicates the current number of calls received by the ICM Server from the Unified CVP Call Server using the SIP protocol.
Active H.323 Call Legs	The ICM Server can accept Voice over IP (VoIP) calls that originate using either the Session Initiation Protocol (SIP) or H.323 protocol. Active H.323 Call Legs indicates the current number of calls received by the ICM Server from the Unified CVP Call Server using the H.323 protocol.
Active VRU Call Legs	The current number of calls receiving Voice Response Unit (VRU) treatment from the ICM Server. The VRU treatment includes playing pre-recorded messages, asking for Caller Entered Digits (CED) or Speech Recognition Techniques to understand the customer request.
Active ICM Lookup Requests	Calls originating from an external VXML Server need call routing instructions from the ICM Server. Active Lookup Requests indicates the current number of external VXML Server call routing requests sent to the ICM Server.
Active Basic Service Video Calls Offered	The current number of simultaneous basic service video calls being processed by the ICM service where video capability was offered.
Active Basic Service Video Calls Accepted	The current number of simultaneous calls that were accepted as basic service video calls and are being processed by the ICM service.
Interval Statistics	
Start Time	The time at which the current interval has begun.
Duration Elapsed	The amount of time that has elapsed since the start time in the current interval.
Interval Duration	The interval at which statistics are collected. The default value is 30 minutes.
New Calls	The number of new calls received by the Intelligent Contact Management (ICM) application for follow-on Voice Response Unit (VRU) treatment and routing to a Contact Center agent during the current interval.
SIP Call Legs	The Intelligent Contact Management (ICM) application has the ability to accept Voice over IP (VoIP) calls that originate via the Session Initiation Protocol (SIP) or H.323 Protocol. Interval SIP Call Legs is an interval specific snapshot metric indicating the number of calls received by the ICM application via SIP during the current interval.

Statistic	Description
H.323 Call Legs	The Intelligent Contact Management (ICM) application has the ability to accept Voice over IP (VoIP) calls that originate via the Session Initiation Protocol (SIP) or H.323 Protocol. Interval H.323 Call Legs is an interval specific snapshot metric indicating the number of calls received by the ICM application via the H.323 protocol during the current interval.
VRU Call Legs	The number of calls receiving Voice Response Unit (VRU) treatment from the Intelligent Contact Management (ICM) application. The VRU treatment includes playing pre-recorded messages, asking for Caller Entered Digits (CED) or speech recognition techniques to understand the customer request during the current interval.
ICM Lookup Requests	Calls originating in an external VXML Server need call routing instructions from the Intelligent Contact Management (ICM) application. Interval Lookup Requests is an interval specific metric indicating the number of external VXML Server call routing requests sent to the ICM application during the current interval.
Basic Service Video Calls Offered	The number of offered basic service video calls processed by the ICM service during the current interval.
Basic Service Video Calls Accepted	The number of basic service video calls accepted and processed by the ICM service during the current interval.
Aggregate Statistics	
Start Time	The time the service started collecting statistics.
Duration Elapsed	The amount of time that has elapsed since the service start time.
Total Calls	The total number of new calls received by the Intelligent Contact Management (ICM) application for follow-on Voice Response Unit (VRU) treatment and routing to a Contact Center agent since system start time.
Total SIP Call Legs	The Intelligent Contact Management (ICM) application has the ability to accept Voice over IP (VoIP) calls that originate via the Session Initiation Protocol (SIP) or H.323 Protocol. Total SIP Switch Legs is a metric indicating the total number of calls received by the ICM application via SIP since system start time.
Total H.323 Call Legs	The Intelligent Contact Management (ICM) application has the ability to accept Voice over IP (VoIP) calls that originate via the Session Initiation Protocol (SIP) or H.323 Protocol. Total H.323 Switch Legs is a metric indicating the total number of calls received by the ICM application via the H.323 protocol since system start time.
Total VRU Call Legs	The total number of calls that have received Voice Response Unit (VRU) treatment from the Intelligent Contact Management (ICM) application since system start time. The VRU treatment includes playing pre-recorded messages, asking for Caller Entered Digits (CED) or Speech Recognition Techniques to understand the customer request.
Total ICM Lookup Requests	Calls originating in an external VXML Server need call routing instructions from the Intelligent Contact Management (ICM) application. Total Lookup Requests is a metric indicating the total number of external VXML Server call routing requests sent to the ICM application since system start time.

Statistic	Description
Total Basic Service Video Calls Offered	The total number of newly offered basic service video calls processed by the ICM service since system start time.
Total Basic Service Video Calls Accepted	The total number of new basic service video calls accepted and processed by the ICM service since system start time.

IVR Service Call Statistics

The IVR service call statistics include data on calls currently being processed by the IVR service, new calls received during a specified interval, and total calls processed since the IVR service started.

Access IVR Service statistics either by:

- Choosing **System > Control Center**, selecting a Call Server, clicking the **Statistics** icon in the toolbar, and then selecting the **IVR** tab.
- Choosing **Device Management > CVP Call Server**, selecting a Call Server, clicking the **Statistics** icon in the toolbar, and then selecting the **IVR** tab.

The following table describes the IVR Service call statistics.

Table 26: IVR Service Call Statistics

Statistic	Description
Realtime Call Statistics	
Active Calls	The number of active calls being serviced by the IVR service.
Active HTTP Requests[2]	The number of active HTTP requests being serviced by the IVR service.
Interval Statistics	
Start Time	The time the system started collecting statistics for the current interval.
Duration Elapsed	The amount of time that has elapsed since the start time in the current interval.
Interval Duration	The interval at which statistics are collected. The default value is 30 minutes.
Peak Active Calls	Maximum number of active calls handled by the IVR service at the same time during this interval.
New Calls	New Calls is a metric that counts the number of New Call requests received from the IOS Gateway or the CVP H323 Service. A New Call includes the Switch leg of the call and the IVR leg of the call. This metric counts the total number of New Call Requests received by the IVR Service during this interval.
Calls Finished	Calls Finished is a metric that counts the number of Unified CVP Calls that have finished during this interval. A Call, for the purpose of the

2) HTTP sessions are the number of HTTP requests currently being processed by the IVR SS. This # will be different than the # of calls being processed.

Statistic	Description
	Call Finished metric, includes both the Switch leg and the IVR leg of the Unified CVP call. When both legs of the call are finished, the *Calls Finished* metric increases.
Average Call Latency	The average amount of time in milliseconds it took the IVR Service to process a New Call or Call Result Request during this interval.
Maximum Call Latency	The maximum amount of time in milliseconds it has taken for the IVR Service to complete the processing of a New Call Request or a Request Instruction Request during this time interval.
Minimum Call Latency	The minimum amount of time in milliseconds it took for the IVR Service to complete the processing of a New Call Request or a Request Instruction Request during this time interval.
Peak Active HTTP Requests	Active HTTP Requests is a metric that indicates the current number of simultaneous HTTP requests being processed by the IVR Service. Peak Active Requests is a metric that represents the maximum simultaneous HTTP requests being processed by the IVR Service during this time interval.
Total HTTP Requests	The total number of HTTP Requests received from a client by the IVR Service during this time interval.
Average HTTP Requests/second	The average number of HTTP Requests the IVR Service receives per second during this time interval.
Peak Active HTTP Requests/second	HTTP Requests per Second is a metric that represents the number of HTTP Requests the IVR Service receives each second from all clients. Peak HTTP Requests per Second is the maximum number of HTTP Requests that were processed by the IVR Service in any given second. This is also known as high water marking.
Aggregate Statistics	
Start Time	The time the service started collecting statistics.
Duration Elapsed	The amount of time that has elapsed since the service start time.
Total New Calls	New Calls is a metric that counts the number of New Call requests received from the IOS Gateway or the H.323 Service. A New Call includes the Switch leg of the call and the IVR leg of the call. Total New Calls is a metric that represents the total number of new calls received by the IVR Service since system startup.
Peak Active Calls	The maximum number of simultaneous calls processed by the IVR Service since the service started.
Total HTTP Requests	Total HTTP Requests is a metric that represents the total number of HTTP Requests received from all clients. This metric is the total number of HTTP Requests received by the IVR Service since system startup.
Peak Active HTTP Requests	Active HTTP Requests is a metric that indicates the current number of simultaneous HTTP requests processed by the IVR Service. Maximum number of active HTTP requests processed at the same time since the IVR service started. This is also known as high water marking.

Statistic	Description
Total Agent Video Pushes	The number of videos pushed by agents since system start time.
Total Agent Initiated Recordings	The number of video recordings by agents since system start time.
Total Agent VCR Control Invocations	The number of video VCR controls invoked by agents since system start time.

SIP Service Call Statistics

The SIP service call statistics include data on calls currently being processed by the SIP service, new calls received during a specified interval, and total calls processed since the SIP service started.

Access SIP Service statistics either by:

- Choosing **System > Control Center**, selecting a Call Server, clicking the **Statistics** icon in the toolbar, and then selecting the **SIP** tab.
- Choosing **Device Management > CVP Call Server**, selecting a Call Server, clicking the **Statistics** icon in the toolbar, and then selecting the **SIP** tab.

The following table describes the SIP Service call statistics.

Table 27: SIP Service Call Statistics

Statistic	Description
Realtime Statistics	
Active Calls	A real time snapshot metric indicating the count of the number of current calls being handled by the SIP service. This value does not include H.323 calls.
Total Call Legs	The total number of SIP call legs being handled by the SIP service. A call leg is also known as a SIP dialog. The metric includes incoming, outgoing and ringtone type call legs. For each active call in the SIP service, there will be an incoming call leg, and an outgoing call leg to the destination of the transfer label.
Active Basic Service Video Calls Offered	The number of basic service video calls in progress where video capability was offered.
Active Basic Service Video Calls Answered	The number of basic service video calls in progress where video capability was answered.
Interval Statistics	
Start Time	The time the system started collecting statistics for the current interval.
Duration Elapsed	The amount of time that has elapsed since the start time in the current interval.
Interval Duration	The interval at which statistics are collected.The default value is 30 minutes.

Statistic	Description
New Calls	The number of SIP Invite messages received by Unified CVP in the current interval. It includes the failed calls as well as calls rejected due to the SIP service being out of service.
Connects Received	The number of CONNECT messages received by SIP service in order to perform a call Transfer, in the last statistics aggregation interval. Connects Received includes the regular Unified CVP transfers as well as Refer transfers. Any label coming from the ICM service is considered a CONNECT message, whether it is a label to send to the VRU or a label to transfer to an agent.
Avg Latency Connect to Answer	The period of time between the CONNECT from ICM and when the call is answered. The metric includes the average latency computation for all the calls that have been answered in the last statistics aggregation interval.
Failed SIP Transfers (Pre-Dialog)	The total number of failed SIP transfers since system start time. When Unified CVP attempts to make a transfer to the first destination of the call, it sends the initial INVITE request to set up the caller with the ICM routed destination label. The metric does not include rejections due to the SIP Service not running. The metric includes failed transfers that were made after a label was returned from the ICM Server in a CONNECT message.
Failed SIP Transfers (Post-Dialog)	The number of failed re-invite requests on either the inbound or outbound legs of the call during the interval. After a SIP dialog is established, re-INVITE messages are used to perform transfers. Re-invite requests can originate from the endpoints or else be initiated by a Unified CVP transfer from the Unified ICME script. This counter includes failures for both kinds of re-invite requests.
Basic Service Video Calls Offered	The number of basic service video calls offered in the current interval.
Basic Service Video Calls Answered	The number of basic service video calls answered in the current interval.
Aggregate Statistics	
Start Time	The time the service started collecting statistics.
Duration Elapsed	The amount of time that has elapsed since the service start time.
Total New Calls	The number of SIP Invite messages received by Unified CVP since system start time. It includes the failed calls as well as calls rejected due to the SIP service being out of service.
Connects Received	The number of CONNECT messages received by SIP service in order to perform a Unified CVP Transfer, since system start time. Connects Received includes the regular Unified CVP transfers as well as Refer transfers. Any label coming from the ICM service is considered a CONNECT message, whether it is a label to send to the VRU or a label to transfer to an agent.
Avg Latency Connect to Answer	The period of time between the CONNECT from ICM and when the call is answered. The metric includes the average latency computation for all the calls that have been answered since system start up time.
Failed SIP Transfers (Pre-Dialog)	The total number of failed transfers on the first CVP transfer since system start time. A SIP dialog is established after the first CVP transfer is completed. The metric does not include rejections due to SIP being out of service. The metric includes failed transfers that were made after a label was returned from the ICM in a CONNECT message.

Statistic	Description
Failed SIP Transfers (Post-Dialog)	The number of failed re-invite requests on either the inbound or outbound legs of the call since start time.After a SIP dialog is established, re-INVITE messages are used to perform transfers. Re-invite requests can originate from the endpoints or else be initiated by a Unified CVP transfer from the Unified ICME script. This counter includes failures for both kinds of re-invite requests.
Total Basic Service Video Calls Offered	The total number of basic service video calls ofered since system start time.
Total Basic Service Video Calls Answered	The total number of basic service video calls answered since system start time.

H.323 Service Call Statistics

The H.323 Service call statistics include data on calls currently being processed by the H.323 Service, new calls received during a specified interval, and total calls processed since the H.323 Service started. You can access these statistics by typing the **`ShowStatus`** command (or **`SS`**) in the VBAdmin tool.

Statistic	Description
New calls this interval	The number of new inbound calls handled by the H.323 Service during this interval.
Max call arrival per minute	The maximum number of inbound calls arriving per minute to the Unified CVP H.323 Service during this interval. This value represents the peak value over the interval and is expressed as "calls per minute."
Calls transferred this interval	The number of calls that have been transferred by the Unified CVP H.323 Service during this interval. The Unified CVP H.323 Service might handle multiple transfers for a single call, so the value might not equal the "New calls this interval" statistic.
Max IP Transfer per minute	The maximum number of calls transferred per minute by the Unified CVP H.323 Service during this interval. This value represents the peak value over the interval and is expressed as "calls transferred per minute."
Calls redirected this interval	The number of calls that have been redirected by the Unified CVP H.323 Service. The H.323 Service only redirects a call when it is in an "out of service" state. A redirected call means that the Unified CVP H.323 Service did not accept the call. This allows the originating gateway to redirect the call to an alternate Unified CVP H.323 Server, if so configured in its dial-peers.
Transfers not completed this interval	The number of calls that could not be transferred by the Unified CVP H.323 Service during this interval. The Unified CVP H.323 Service was unable to transfer the call to a particular destination due to one of several factors: ring-no-answer; busy; incorrect configuration; call admission control (CAC) denials. It is a recoverable error if proper configuration elements have been put in place, such as Unified ICME router requery, and/or gatekeeper alternate endpoints.
Prompts not found this interval	The number of calls to which prompts could not be played during this interval period. This statistic indicates a failure within the solution for each of these calls.

Statistic	Description
	It could be caused by an Unified ICME script error, a missing prompt, or failure of the HTTP media server.
Calls using critical media	The number of calls to which critical media needed to be played during the interval period. This usually, but not always, indicates an abnormal termination of the caller. In some benign cases, it can be caused by improper Unified ICME script termination practices.
Calls finished this interval	The number of calls to which the Unified CVP H.323 Service has completed servicing during this interval. These calls have ended and are no longer being handled by the Unified CVP H.323 Service.
Avg New Call Latency (ms)	The average elapsed time in milliseconds from when the Unified CVP H.323 Service sends a 'New Call' request to Unified ICME until the time when the Unified CVP H.323 Service receives a response from Unified ICME. This value is expressed in milliseconds and is averaged over the interval period.
Max New Call Latency (ms)	The maximum elapsed time in milliseconds from when the Unified CVP H.323 Service sends a 'New Call' request to Unified ICME until the time when the Unified CVP H.323 Service receives a response from Unified ICME. This value is expressed in milliseconds and indicates the high-water mark for the interval period.
Min New Call Latency (ms)	The minimum elapsed time in milliseconds from when the Unified CVP H.323 Service sends a 'New Call' request to Unified ICME until the time when the H.323 Service receives a response from Unified ICME. This value is expressed in milliseconds and indicates the low-water mark for the interval period.
Std Dev. New Call Latency	The standard deviation elapsed time in milliseconds from when the Unified CVP H.323 Service sends a 'New Call' request to Unified ICME until the time when the H.323 Service receives a response from Unified ICME.
Avg Transfer Time to Alert (ms)	The average amount of time elapsed for a transferred call to enter the "alerting" state once transferred. This value is expressed in milliseconds and is averaged over the interval.
Max Transfer Time to Alert (ms)	The maximum amount of time elapsed for a transferred call to enter the "alerting" state once transferred. This value is expressed in milliseconds and indicates the high-water mark for the interval.
Min Transfer Time to Alert (ms)	The minimum amount of time elapsed for a transferred call to enter the "alerting" state once transferred. This value is expressed in milliseconds and indicates the low-water mark for the interval period.
Std Dev. Transfer Time to Alert	The standard deviation elapsed time in milliseconds for a transferred call to enter the "alerting" state once transferred.
Avg Transfer Time to Answer (ms)	The average amount of elapsed time for a transferred call to be answered at the destination once transferred. This value is expressed in milliseconds and indicates the average time for the interval.
Max Transfer Time to Answer (ms)	The maximum amount of time elapsed for a transferred call to be answered at the destination once transferred. This value is expressed in milliseconds and indicates the high-water mark for the interval.
Min Transfer Time to Answer (ms)	The minimum amount of time elapsed for a transferred call to be answered at the destination once transferred. This value is expressed in milliseconds and indicates the low-water mark for the interval.

Statistic	Description
Std Dev. Transfer Time to Answer	The standard deviation or elapsed time in milliseconds for a transferred call to be answered at the destination once transferred.
Avg CPU Usage (percent)	The average server CPU usage for the Unified CVP H.323 Service computed during this interval. The CPU usage value is expressed as a percentage of potential usage available.
Max CPU Usage (percent)	The maximum server CPU usage for the Unified CVP H.323 Service during this interval. The CPU usage value is expressed as a percentage of potential usage available and represents a high-water mark of CPU usage during this interval.
Min CPU Usage (percent)	The minimum server CPU usage for the Unified CVP H.323 Service during this interval. The CPU usage value is expressed as a percentage of potential usage available and represents the low-water mark of CPU usage during this interval.
Std Dev. CPU Usage	The standard deviation server CPU usage for the Unified CVP H.323 Service during this interval. The CPU usage value is expressed as a percentage of potential usage available.
IVR Service: Avg Call Event Latency (ms)	Average latency statistics in processing a new call arrival.
IVR Service: Max Call Event Latency (ms)	Maximum latency statistics in processing a new call arrival.
IVR Service: Min Call Event Latency (ms)	Minimum latency statistics in processing a new call arrival.
IVR Service: Standard Deviation (theta)	Standard deviation latency statistics in processing a new call arrival.
IVR Service: Number of Call Events	Number of call latency statistics in processing a new call arrival.

Gatekeeper Statistics

Gatekeeper statistics include the number of active calls, available memory, CPU utilization and performance data.

Get Gatekeeper Statistics

To get gatekeeper statistics:

Step 1 Choose **System > Control Center**.

Step 2 Select the **Device Type** tab in the left pane, then select gatekeepers.

Gatekeepers are listed in the right pane.

Step 3 Select the gatekeeper by clicking on its link under the Hostname column.

the Edit Gatekeeper Configuration window opens.

Step 4 Select the Statistics icon in the toolbar.

See Also

Viewing Device Statistics in the Operations Console online help.

Gatekeeper Statistics Descriptions

The following table describes gatekeeper statistics.

Table 28: Gatekeeper Statistics

Statistic	Description
Active Calls	Number of currently active calls handled by the gatekeeper.
Free Memory	Free memory, for example: **`Processor memory free: 82%`** **`I/O memory free: 79%`**
CPU Utilization	CPU utilization, for example: **`CPU utilization for five seconds: 3%/3%; one minute: 3%; five minutes: 4%`**
Performance Statistics	Admission levels. Refer to Gateway Performance Statistics Example. (page 304)

Gateway Performance Statistics Example

```
Gatekeeper Level Admission Statistics:
ARQs received: 5635885
ARQs received from originating endpoints: 2663162
ACFs sent: 5635884
ACFs sent to the originating endpoint: 2663162
ARJs sent: 1
ARJs sent to the originating endpoint: 0
ARJs sent due to overload: 0
ARJs sent due to ARQ access-list denial: 0
Number of concurrent calls: 280
Number of concurrent originating calls: 140
Gatekeeper level Registration Statistics:
RRJ due to overload: 0
Total Registered Endpoints: 37
```

Gateway Statistics

Gateway statistics include the number of active calls, available memory, and CPU utilization.

Procedure

To get gateway statistics:

Step 1 Choose **System > Control Center**.

Step 2 Select the **Device Type** tab in the left pane, then select gateways.

Gateways are listed in the right pane.

Step 3 Select the gateway by clicking on its link under the Hostname column.

the Edit Gateway Configuration window opens.

Step 4 Select the Statistics icon in the toolbar.

See Also

Viewing Device Statistics in the Operations Console online help.

Gateway Statistics

The following table describes gateway statistics.

Statistic	Description
Active Calls	Number of currently active calls handled by the gateway. For example, Total call-legs: 0 no active calls
Free Memory	Free memory, for example: **`Processor memory free: 82%`** **`I/O memory free: 79%`**
CPU Utilization	CPU utilization, for example: **`CPU utilization for five seconds: 3%/3%; one minute: 3%; five minutes: 4%`**

Trunk Utilization Reporting

You can configure IOS gateways to report on truck utilization. The configuration involves two pieces:

- Configuring the Call Server using the Operations Console to request reporting from a given gateway.
- Configuring the gateway to respond to trunk utilization reporting requests.

To configure Unified CVP to provide trunk utilization reporting, complete these steps:

1. In the Operations Console, select: **Device Management > Call Server > ICM (tab) > Advanced Configuration** then under *Trunk Utilization*, select **Enable Gateway Trunk Reporting**.

2. In the same section, associate the gateway(s) that you want to send truck information to the Call Server.

3. Add the following configuration to the gateway configuration:

```
voice class resource-group 1
resource cpu 1-min-avg threshold high 80 low 60
resource ds0
resource dsp
resource mem total-mem
periodic-report interval 30

sip-ua
rai target ipv4:10.86.129.11 resource-group 1
rai target ipv4:10.86.129.24 resource-group 1
```

RAI information on SIP OPTIONS (CVP server group heartbeats)

If RAI is desired on SIP OPTIONS, the option override host setting can be used with server group heartbeating. When one or more CVPs are sending OPTIONS heartbeats to the gateway, RAI trunk utilization information is not normally sent in the 200 OK response, unless an RAI target is configured.

CLI like the following can be added in IOS in order to have RAI information sent to CVP in the response:

```
sip-ua
rai target dns:cvp.cisco.com resource-group 1
```

Note: Trunk Utilization data is only written to the CVP database when RAI OPTIONS are sent from the gateway to CVP targets. In the case when CVP is using server group heartbeats to the gateway, the RAI data in the response is only used to mark the element as UP or DOWN (overloaded resources) in the server group.

VXML Server Statistics

The Operations Console displays realtime, interval, and aggregate VXML Server statistics.

Access VXML Server statistics either by:

- Choosing **System > Control Center**, selecting a VXML Server, and then clicking the Statistics icon in the toolbar.

- Choosing **Device Management > VXML Server** (or VXML Server (Standalone)), selecting a VXML Server, and then clicking the Statistics icon in the toolbar.

Table 29: VXML Server Statistics

The following table describes the statistics reported by the VXML Server.

Statistic	Description
Real Time Statistics	
Active Sessions	The number of current sessions being handled by the VXML Server.
Active ICM Lookup Requests	The number of current ICM requests being handled by the VXML Server.
Interval Statistics	
Start Time	The time at which the current interval has begun.
Duration Elapsed	The amount of time that has elapsed since the start time in the current interval.
Interval Duration	The interval at which statistics are collected. The default value is 30 minutes.
Sessions	The total number of sessions in the VXML Server in the current interval.
Reporting Events	The number of events sent to the Reporting Server from the VXML Server in the current interval.
ICM Lookup Requests	The number of requests from the VXML Server to the ICM Service in the current interval.
ICM Lookup Responses	The number of responses to both failed and successful ICM Lookup Requests that the ICM Service has sent to the VXML Server in the current interval. In the case that multiple response messages are sent back to the VXML Server to a single request, this metric will increment per response message from the ICM Service.
ICM Lookup Successes	The number of successful requests from the VXML Server to the ICM Service in the current interval.
ICM Lookup Failures	The number of requests from the VXML Server to the ICM Service in the current interval. This metric will be incremented in the case an ICM failed message was received or in the case the VXML Server generates the failed message.
Aggregate Statistics	
Start Time	The time at which the current interval has begun.
Duration Elapsed	The amount of time that has elapsed since the start time in the current interval.
Total Sessions	The total number of sessions in the VXML Server since startup.
Total Reporting Events	The total number of reporting events sent from the VXML Server since startup.
Total ICM Lookup Requests	The total number of requests from the VXML Server to the ICM Service. For each ICM lookup request, whether the request succeeded or failed, this metric will be increased by one.
Total ICM Lookup Responses	The total number of responses the ICM Service has sent to the VXML Server since startup. For each ICM lookup response, whether the response is to a succeeded or failed request, this metric will be increased by one. In the case that multiple response messages are sent back to the VXML Server

Statistic	Description
	to a single request, this metric will increment per response message from the ICM Service.
Total ICM Lookup Success	The total number of requests from the VXML Server to the ICM Service since startup. For each ICM lookup request that succeeded, this metric will be increased by one.
Total ICM Lookup Failures	The total number of requests from the VXML Server to the ICM Service since startup. For each ICM lookup request that failed, this metric will be increased by one. This metric will be incremented in the case an ICM failed message was received or in the case the VXML Server generates a failed message.

Refer to the Operations Console online help topics:

- *Using the Control Center > Viewing Infrastructure Statistics*
- *Using the Control Center > Viewing Device Statistics*

Reporting Server Statistics

Reporting Server statistics include the total number of events received from the IVR, SIP, and VoiceXML services.

Access Reporting Server statistics either by:

- Choosing **System > Control Center**, selecting a Reporting Server, and then clicking the Statistics icon in the toolbar.
- Choosing **Device Management > CVP Reporting Server**, selecting a Reporting Server, and then clicking the Statistics icon in the toolbar.

The following table describes the Reporting Server statistics.

Table 30: Reporting Server Statistics

Statistic	Description
Interval Statistics	
Start Time	The time the system started collecting statistics for the current interval.
Duration Elapsed	The amount of time that has elapsed since the start time in the current interval.
Interval Duration	The interval at which statistics are collected. The default value is 30 minutes.
VXML Events Received	The total number of reporting events received from the VoiceXML Service during this interval. For each reporting event received from the VoiceXML Service, this metric will be increased by one.

Statistic	Description
SIP Events Received	The total number of reporting events received from the SIP Service during this interval. For each reporting event received from the SIP Service, this metric will be increased by one.
IVR Events Received	The total number of reporting events received from the IVR service in the interval. For each reporting event received from the IVR service, this metric will be increased by one.
Database Writes	The total number of writes to the database made by the Reporting server during the interval. For each write to the database by the Reporting server, this metric will be increased by one.
Aggregate Statistics	
Start Time	The time the service started collecting statistics.
Duration Elapsed	The amount of time that has elapsed since the service start time.
VXML Events Received	The total number of reporting events received from the VoiceXML Service since the service started. For each reporting event received from the VoiceXML Service, this metric will be increased by one.
SIP Events Received	The total number of reporting events received from the SIP Service since the service started. For each reporting event received from the SIP Service, this metric will be increased by one.
IVR Events Received	The total number of reporting events received from the IVR Service since the service started. For each reporting event received from the IVR Service, this metric will be increased by one.
Database Writes	The total number of writes to the database made by the Reporting server during since startup. For each write to the database by the Reporting server, this metric will be increased by one.

SNMP Alerts for Cisco Unified Presence Server (CUP Server)

Cisco Unified Presence Server (CUP Server) stores alarm definitions and actions in a structured query language (SQL) server database. The system administrator can search the database for definitions of all the alarms. The definitions include the alarm name, description, explanation, recommended action, severity, parameters, and monitors. This information aids the administrator in troubleshooting problems that Cisco Unified Presence Server encounters.

Note: You can track alerts for the CUP Server using the Real Time Monitoring Tool (RTMT), which is also used with Unified CM 5.0 alerts.

See Also

Refer to **Cisco Unified Presence Server Serviceability Administration Guide** (http://www.cisco.com/en/US/products/ps6837/prod_maintenance_guides_list.html) for a description of the alarm categories.

Unified CVP SNMP-Raise/Clear Mappings

The following log messages have been defined as SNMP-enabled by default. Administrators can define a unique alarm within their SNMP management station for all SNMP Raise events emitted by a system. These alarms are usually cleared automatically using one or more corresponding SNMP Clear events when the condition is resolved. The tables below list a mapping of Unified CVP SNMP Raise events with their corresponding SNMP Clears.

Note: Raises are listed first, with their corresponding clears below them.

Table 31: Messaging Layer

Raise ID	Clear ID	Event Name
7		ADAPTER_INITIALIZATION_FAILURE
	8	ADAPTER_INITIALIZATION_SUCCESS
9		PLUGIN_INITIALIZATION_FAILURE
	10	PLUGIN_INITIALIZATION_SUCCESS
15		SEND_QUEUE_THRESHOLD_REACHED
	20	SEND_QUEUE_SIZE_CLEAR

Table 32: Infrastructure

Raise ID	Clear ID	Event Name
9005		LICENSING
	1003	[AUDIT] "The system has started up."
9007		PORT_THRESHOLD
	9008	PORT_THRESHOLD
9014		SHUTDOWN
	1003	[AUDIT] "The system has started up."
	1004	[AUDIT] "The system has completely shutdown."
9016		SERVER_SETUP - "CCBUSNMPAgent Server setup failed because XXX" [3]
	9015	SERVER_SETUP - "CCBUSNMPAgent Server setup on port YYY"
1011		HEARTBEATS_STOPPED - "Heartbeats from XXX stopped..."[4]
	1014	RECEIVED_STATE_MSG - "StateManager: Subsystem [XXX] reported change to..."
1012		STATE_MANAGER_STARTUP_FAILURE

3) Where XXX must match in both messages

4) Where XXX must match in both messages

Raise ID	Clear ID	Event Name
	1003	[AUDIT] "The system has started up."
1020		STARTUP
	1003	[AUDIT] "The system has started up."
1024		SERVLET_STARTUP
	1003	[AUDIT] "The system has started up."
1025		START - "Could not start XXX due to: YYY" [5]
	1003	[AUDIT] "The system has started up."
1033		START - "No Subsystems have been started..."
	1026	START - "All Subsystems have been started."
1035		LICENSE_EXPIRATION
	1003	[AUDIT] "The system has started up."

Table 33: Unified ICME

Raise ID	Clear ID	Event Name
2001		LOGMSG_ICM_SS_MSGBUS_SHUTDOWN
	2003	LOGMSG_ICM_SS_MSGBUS_ACTIVE
2002		LOGMSG_ICM_SS_PIM_SHUTDOWN
	2004	LOGMSG_ICM_SS_PIM_ACTIVE
2005		LOGMSG_ICM_SS_HEARTBEAT_FAILURE
	2012	LOGMSG_ICM_SS_INSERVICE_STATE
2006		LOGMSG_ICM_SS_STATE
	2012	LOGMSG_ICM_SS_INSERVICE_STATE

Table 34: Reporting

Raise ID	Clear ID	Event Name
4005		REPORTING_SS_ERROR_RAISE
	1026	START - "All Subsystems have been started."
4006		REPORTING_DB_PURGE_FAILED
	4007	REPORTING_DB_PURGE_COMPLETED
4010		REPORTING_DB_BACKUP_FAILED
	4011	REPORTING_DB_BACKUP_COMPLETED
4014		REPORTING_DB_ALERT_MSG
	N/A	Not applicable
4017		REPORTING_DB_STARTING_PURGE
	4007	REPORTING_DB_PURGE_COMPLETED
	4009	REPORTING_DB_EMERGENCY_PURGE_COMPLETED

5) Where XXX must match in both messages

Raise ID	Clear ID	Event Name
4018		REPORTING_DB_REMAINDER_DATA
	4019	REPORTING_DB_NO_REMAINDER_DATA

Table 35: IVR

Raise ID	Clear ID	Event Name
3014		VBCLIENT_REMOVED
	3013	VBCLIENT_ADDED
3027		VBCLIENT_SHUT_DOWN
	3026	VBCLIENT_RESTARTED
3021		VBSERVLET_STATE_CHANGED
	3022	VBSERVLET_STATE_IN_SERVICE
3002		STATE_CHANGED
	3001	STATE_CHANGED_IN_SERVICE
3000		SHUTDOWN_NOTICE
	3001	STATE_CHANGED_IN_SERVICE

Table 36: SIP

Raise ID	Clear ID	Event Name
5001		SS_STATE; The SIP subsystem changed state to something other than the *in service* state.
	5002	SS_STATE; The SIP subsystem changed state to the *in service* state.

Table 37: VoiceXML

Raise ID	Clear ID	Event Name
6012		VXML_SERVER_APP_SHUTDOWN_ALERT
	6011	VXML_SERVER_APP_STARTUP_CLEAR
6013		VXML_SERVER_APPADMIN_ERROR
	1003	[AUDIT] "The system has started up."
	1004	[AUDIT] "The system has completely shutdown."
6014		VXML_SERVER_SYSTEM_ERROR
	1003	[AUDIT] "The system has started up."
	1004	[AUDIT] "The system has completely shutdown."

Chapter 8

Administering the Unified CVP H.323 Service

This chapter provides the following:

- An overview of the Unified CVP H.323 Service.
- Instructions for controlling the state of the Unified CVP H.323 service.
- Instructions for using the VBAdmin tool.
- Descriptions of each of the VBAdmin commands.
- Instructions for performing a H.323 blind transfer.

Note: The primary audience for this chapter is Unified CVP System Managers.

This chapter contains the following topics:

Unified CVP H.323 Service Overview

The Unified CVP H.323 Service, formerly known as the *CVP Voice Browser*, provides legacy H.323 call control between Unified CVP and other H.323 endpoints in the customer topology. These endpoints can be:

- Unified CM
- Cisco IOS gateways (refer to **Hardware and System Software Specification for Cisco Unified Customer Voice Portal (Unified CVP) Software** (http://www.cisco.com/en/US/

products/sw/custcosw/ps1006/prod_technical_reference_list.html) for the current list of gateways supported)

- PGW/HSI in call control mode (used as an Ingress Gateway only)

The H.323 Service interacts with Unified ICME through the Unified CVP Call Server to relay call arrival, disconnect, and transfer instructions between Unified ICME and the other H.323 endpoints.

Note: The H.323 Service is only used in H.323-based call flows.

The H.323 Service includes its own internal Voice Browser that serves several key roles in the Unified CVP architecture. It:

- Functions as the initial VoIP end-point for H.323 voice contacts.
- Coordinates the delivery of messages and prompts to a caller from the Media Server.

  Note: This is legacy functionality, only; it should not be used for new deployments. New deployments should *always* use the Unified CVP Comprehensive call flow model, where prompts are rendered from the IOS VXML gateway.

- Sends HTTP requests to the IVR Service.

In addition, the H.323 Service includes a configuration and administration tool—called **VBAdmin**.

On the *incoming* call side, the H.323 Service processes call and control signals from calls entering the system at one of the supported ingress H.323 endpoints. (The ingress H.323 endpoints converts them into H.323 messages before forwarding them to the H.323 Service.)

Once in control of the call, the H.323 Service converts the voice signals into events to be processed by a web server known as the *Call Server*, which houses the IVR Service. The H.323 Service connects to the IVR Service through a pre-defined URL address. (For more information, refer to "How Does the H.323 Service Connect to a Call Server? (page 315).")

The H.323 Service remains in the call control path until the call's logical completion. (The IVR Service never directly controls a telephone call.)

On its *outgoing* side, the H.323 Service acts upon VoiceXML commands received from the IVR Service. The VoiceXML commands contain instructions for:

- Transferring the telephone call.
- Disconnecting the telephone call.
- Delivering recorded messages and prompts and processing user responses to the prompts.

How Does the H.323 Service Connect to a Call Server?

The H.323 Service needs to find a Call Server to which to send requests to; Unified CVP installation sets the default Call Server address to **`localhost:8000/cvp/VBServlet`**.

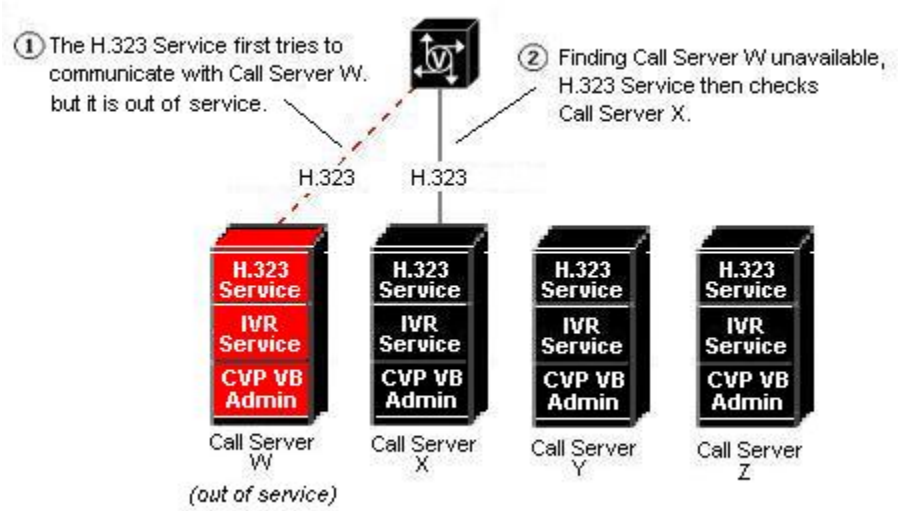
Note: In general, this value should not be changed. The H.323 Service configuration tool could be used to modify the list as needed. If the Call Server list changes, the first Call Server on the new list will be used.

There is one Call Server list for all calls.As shown in the figure below, the H.323 Service selects a Call Server as follows:

1. When the H.323 Service starts up, it attempts to connect with the first Call Server in the list, Call Server W.
2. If Call Server W does not respond, the H.323 Service will go to the next Call Server in the list, Call Server X.
3. If the H.323 Service can communicate with Call Server X, the next call is handled by that Call Server.

Note: If a call arrives and Call Servers X, Y, and Z do not respond, the H.323 Service goes back to the beginning of the list and tries the Call Server at the original starting point (that is, Call Server W). If the H.323 Service tries all Call Servers and there is no response, an alarm is generated. The H.323 Service takes itself out of service and refuses calls. It will continue to communicate with each of the Call Servers and, when one of them is available, goes back in service.

Figure 38: Call Server



If the Call Server list changes, the first Call Server on the new list will be used. The change will be used for new calls only.

Out of Band Communications

The H.323 Service has two types of connections to the Call Server:

- **Dynamic non-persistent HTTP connections**. Sends synchronous requests, such as call arrived, call disconnected, et cetera. The GUID (globally unique call id) is the mechanism used by the H.323 Service and the Call Server to identify a particular call. The GUID is always passed on these requests.
- **Semi-persistent connections**. Re-established every 60 seconds. Monitors the health of the Call Servers and processes asynchronous messages, such as transferring a queued call to an agent.

  Note: This is called *out of band communication*.

The H.323 Service Media File

The H.323 Service has one media file. This file contains a prompt to be played to a caller in case of a critical error during the call.

The name and location of this .wav file is one of the following:

- *<target directory>***`Critical_Error_Alaw.wav`**
- *<target directory>***`Critical_Error_Ulaw.wav`**

where *<target directory>* is the path specified during installation, typically **`C:\Cisco\CVP\VoiceBrowser`** for the target directory.

The customer should record the appropriate message for their installation and replace the placeholder one which was installed with the product—keeping the placeholder's name and codec. The file should be either the A-law or µ-law encoded file, depending on the encoding used in the system and configured in the H.323 Service.

For more information, refer to"Configuring the Media Servers (page 531)."

After the H.323 Service plays the message, it disconnects the call.

How to Control the State of the H.323 Service

You use the Service Control tool to change the state of the H.323 Service.

Step 1 To start the Service Control tool, select **Start > Programs > Cisco Unified Customer Voice Portal > H.323 Service**.

The Service Control window displays the following information:

- **Computer Name**. (Read-only.) Displays the name of the selected computer where Service Control is viewing and controlling services.
- **Select**. (Button.) Accesses the Select Computer dialog box. This option allows you to select a machine and remotely view and control the services on that machine. To select a remote computer, enter the computer name in the text field of the Select Computer dialogue box (or choose it from the list) and click **OK**.
- **Close**. (Button.) Click this button to exit and close the Service Control tool.

  Note: All services remain in their current state (running services continue to run) after you close the Service Control tool.
- **Services**. (Column.) Lists the name of the Windows 2003 service. This column also displays a traffic light icon which indicates the state of the service.
- **State**. (Column.) Describes the state of the service. Possible states include:
  - **Stopped**. The service is stopped.
  - **Stop Pending**. Service Control is in the process of stopping the service.
  - **Start Pending**. Service Control is in the process of starting the service.
  - **Running**. Service is running.
- **Startup**. (Column.) Indicates whether the Windows 2003 service is Automatic (service starts when you turn on the computer), Manual (user must start the service) or Disabled (user stopped the service).
- **All**. (Checkbox.) If not checked, the default, displays only Cisco CCBU Support Tools Node Agent, Cisco CVP SNMP Management, and Cisco CVP H.323 Service. If checked, displays all Windows 2003 services on the machine specified in the Computer Name.
- **Start/Start All**. (Button.) Use this button to start a service or services:
  - **Start All**. (The **All** checkbox is not checked and no services are selected.) Click **Start All** to start all Unified CVP services.

    Note: Start All only affects Cisco Support Tools Node Agent, Cisco Unified CVP SNMP Management, and Cisco Unified CVP H.323 Service (on a Call Server machine).
  - **Start**. (The **All** checkbox is checked or one service is selected.) Controls only one service at a time. Click **Start** to start a selected service.
- **Stop/Stop All**. (Button.) Use this button to stop a selected service or services.
  - **Stop All**. (The **All** checkbox is not checked and no services are selected.) Click **Stop All** to stop all Unified ICME services.

    Note: Stop All only affects Cisco Support Tools Node Agent, Cisco Unified CVP SNMP Management, and Cisco Unified CVP H.323 Service (on a Call Server machine).

  - **Stop**. (The **All** checkbox is checked or one service is selected.) Controls only one service at a time. Click **Stop** to stop a selected service.

- **Cycle**. (Button.) Click this button to stop and restart a selected service in a single action.
- **Manual/Automatic**. (Buttons.) These buttons allow you to switch the startup mode of a selected service between manual (user-initiated) and automatic (starts when the computer is turned on).
- **Help**. (Button) Click this button to access the Service Control online help.

Step 2 To remotely view and control services on another machine, click **Select**. The Select Computer dialog box appears.

Step 3 Type a computer name or select a name from the list and click **OK**. The Service Control window reappears, displaying the services on the computer you specified.

About the H.323 Service and the Gatekeeper

When the Unified CVP H.323 Service enters the "Out of Service" system state, it sends an RAI message to the gatekeeper stating that it is out of resources and an "O" flag displays in the gatekeeper flags column. The gatekeeper in turn will not send any further requests to that particular Unified CVP unless it is the only Unified CVP left available. That is, the gatekeeper will send all incoming calls to other Unified CVPs that are not marked as "O".

Note: Unified CVP does not unregister to the gatekeeper when it enters the "Out of Service" state.

VBAdmin Tool

The H.323 Service includes a configuration and administration tool—called **VBAdmin**—to help you keep track of the H.323 Service's interactions with related components. This tool provides a command line interface (CLI) you can use to:

- Gather statistics.
- Modify configuration settings.
- View system metrics and status.
- Control the H.323 Service.

Note: In order for the VBAdmin tool to function, the H.323 Service must be running.

The sections that follow describe the VBAdmin tool, its commands, and how to use them.

Running VBAdmin

Since there are often many H.323 Services in network installations, VBAdmin can be run locally or remotely (Remote VBAdmin), or redirected to monitor a different H.323 Service. However, Remote VBAdmin is supported *only* when being run on a server that has Unified ICME installed.

VBAdmin can also be used from any machine running Unified ICME to remotely administer the H.323 Service. The syntax to do so is: **`procmon Customer "Voice Browser" VB machine-name-or-IP-of-H323-Service`**.

Note: You can also use the Operations Console to send H.323 commands. Refer to the Operations Console online help.

VBAdmin Command Syntax

There are two types of VBAdmin commands: *Show* commands let you view the configuration settings; *Set* commands let you change the settings. The syntax for the VBAdmin commands is:

`show<`*Parameter***`>[ /? | /help]`**

`set<`*Parameter***`>[<NewValue> | /? | /help]`**

where:

- *show<Parameter>* or *set<Parameter>* is the command; for example, **ShowGateKeeper** or **SetCallServerNumTries**.
- *NewValue*—only used with Set commands—is the new setting for the parameter.
- */?* and */help* are two options for accessing VBAdmin Online Help.

Note:

- *NewValue* must be enclosed in quotes*only* if you are defining multiple settings for a parameter using one command. For example, to define one Call Server Name, you would enter: **`SetCallServerList CS_Lowell`**(no quotes). To define three Call Server Names at one time, you would enter: **`SetCallServerList “CS_Lowell CS_Salem CS_SanJose”`** (with quotes).
- VBAdmin Commands are not case-sensitive; for example, either **`ShowGateKeeper`** or **`showgatekeeper`** is an acceptable entry value.

After you enter a “Show” or “Set” command, VBAdmin responds with messages that fall into one of the following types:

- **Informational/Online Help**. Examples: “**`CallServerNumTries`** is currently **`3`**.” “Valid values: 256 character string.”

- **Confirmational**. Example: "`CallServerNumTries` has been changed from `3` to `5`."
- **Error**. Example: "Entered value needs to be a positive integer."

The first two message types are described in the Command tables in the "Using VBAdmin (page 320)" section. Error messages are described in "VBAdmin Error Handling (page 341)."

VBAdmin Online Help

When you enter `/?` or `/help` after a valid command, VBAdmin displays Help text consisting of:

- A description of the command, including any default setting and when changes to configuration take effect.
- The syntax of the command.

The following table shows some Online Help examples.

Table 38: VBAdmin Online Help Examples

Command	Help Message
`ShowCallServerNumTries /?`	Maximum number of times the H.323 Service will try to connect to the Call Server before failing and reporting an error. ShowCallServerNumTries [/? | /help] SetCallServerNumTries [<NewValue> | /? | /help]
`SetTraceMask /help`	For use by Technical Support, only. Default: 101003HEX Changes to this value take effect immediately. (System shutdown and startup unnecessary.) ShowTraceMask [/? | /help] SetTraceMask [<NewValue> | /? | /help]

Note: To obtain an alphabetical list of all commands supported by VBAdmin, enter `mhelp` at the VBAdmin CLI prompt.

Using VBAdmin

This section describes how to use the VBAdmin command language interface (CLI) tool.

How to Use the VBAdmin Tool Locally

Note: This section describes how to use the VBAdmin CLI tool locally. You can also use the Operations Console to send H.323 commands. Refer to the Operations Console online help.

Step 1 Select **Start > Programs > Cisco Unified Customer Voice Portal > H.323 Services > H.323 Service Admininstration**.

The H.323 Service Administration window appears.

Step 2 Enter a **`show<`***parameter***`>`** or **`set<`***parameter***`>`** command you want to execute.

Step 3 When finished using VBAdmin, enter **q**. The VBAdmin window closes.

Note: VBAdmin commands are not case-sensitive.

How to Use the VBAdmin Tool Remotely

You can use VBAdmin from any machine running Unified ICME to remotely administer the H.323 service.

Step 1 At the DOS prompt, enter the command: **`procmon Customer "Voice Browser" VB machine-name-or-IP-of-H323-Service`**.

Step 2 Enter a **`show<`***Parameter***`>`** or **`set<`***Parameter***`>`** command you want to execute.

Note: VBAdmin must be installed on any machine you want to access remotely.

The sections that follow describe the VBAdmin commands by category.

VBAdmin Configuration Commands

The following table describes the VBAdmin configuration commands:

Table 39: VBAdmin Configuration Commands

Show and Set Command Syntax	Description
`Alarm /?` **`ShowAlarms [/? | /help]`**	Use the command to raise or clear an alarm. Usage: **`Alarm [/msonline MediaServerMachineName]`** **`[/msoffline MediaServerMachineName]`** **`[/asonline CallServerMachineName]`**

Show and Set Command Syntax	Description
	[/asoffline CallServerMachineName] **[/running] [/shutdown]** **[/outofservice] [/inservice] [/gatekeeperonline]** **[/gatekeeperoffline] [/allcallserversdown]** **[/clearallcallserversdown] [/trace] [/help]** ShowAlarms [/?] Shows list of current alarms.
ShowCallServerConnectTimeout [/? | /help] **SetCallServerConnectTimeout [<New Value> | /? | /help]**	The number of seconds the H.323 Service waits for a TCP connect to the Call Server to complete before timing out and generating an error. Default: 2 seconds. Changes to this value take effect immediately.
ShowCallServerList [/? | /help] **SetCallServerList [<List of CallServer base URLs, delimited by spaces> | /? | /help]** **(Shorthand commands: sCSList, SetCSList)**	List of base URLs of Call Servers for the H.323 Service, delimited by spaces and the entire list enclosed in quotes. The syntax for the base URL is: <CallServer>:<Port>/cvp/VBServlet where: <*CallServer* is the hostname or IP address of the machine that is running the Call Server. (Default: localhost.) <*Port* is the port number the Call Server is listening on. (Default: 8000.) */cvp/VBServlet* is a fixed string that you must append to each name in the CallServerList. **Note:** A colon (:) must separate the <CallServer> and <Port> values. In addition, you must use the default Port number and treat "cvp/VBServlet" as a fixed string. Making changes to these value can render a Call Server unusable. IP Address and DNS name may be used in configuring the base URL portion of a Call Server. However, Unified CVP cannot currently support more than one IP address per DNS name, so you must have a one-to-one correspondence between a DNS name and IP address for each Call Server. When defining multiple *List of CallServer base URLs* settings, enclose them in quotes (for example: **SetCSList "CS_Boston:8000/cvp/VBServlet CS_SanJose:8000/cvp/VBServlet"**).

Show and Set Command Syntax	Description
	Changes to this value take effect immediately. **Note:** For more information, refer to "How Does the H.323 Service Connect to a Call Server? (page 315)."
ShowCallServerNumTries [/? | /help] **SetCallServerNumTries [<NewValue> | /? | /help]** **(Shorthand commands: sCSTries, SetCSTries)**	Maximum number of times the H.323 Service will try to connect to the Call Server before failing and reporting an error. Default: 3 System shutdown and startup necessary for changes to take effect.
ShowCallServerTimeout [/? | /help] **SetCallServerTimeout [<NewValue> | /? | /help]** **(Shorthand commands: sCSTimeout, SetCSTimeout)**	Number of seconds the H.323 Service should wait for a response from the Call Server before timing out and generating an error. **Note:** This setting *must* be greater than the IVR Service Timeout setting on the IVR tab in the Operations Console. (For information about setting the IVR ServiceTimeout setting, refer to the Operations Console online help.) Default: 7 seconds System shutdown and startup necessary for changes to take effect.
ShowCalledPartyTimeout [/? | /help] **SetCalledPartyTimeout [<NewValue> | /? | /help]** **(Shorthand commands: sCPT, SetCPT)**	Length of time in seconds to wait for additional instructions from Unified ICME after the Called Party (that is, agent) hangs up. If no instructions are received, the H.323 Service disconnects the caller. Default: 2 System shutdown and startup necessary for changes to take effect.
ShowCallTrace [/? | /help] **SetCallTrace [on | off | /? | /help]**	When this value is set to on, call events will be logged to the console and log files. This is the lowest level of debugging, recommended when you want to see basic call events (call arrived, call transferred, call disconnected). Valid Values: on and off Default: off.
ShowCLIDigitsToDelete [/? | /help] **SetCLIDigitsToDelete [<New Value>/? | /help]**	When a call arrives at the H.323 Service with the H323 presentationIndicator value set to “Restricted,” the H.323 Service needs to hide some portion or all of the caller ID before passing the CallerID to Unified ICME. This setting controls how many digits will be stripped from the rightmost portion of the callerID before passing it to Unified ICME. A value of 0 will leave the callerID as is, and a value that is greater than the length of the callerID will strip all the digits. Default: 30.

Show and Set Command Syntax	Description
ShowCodec [/? | /help] **SetCodec [g711Ulaw64k | g711Alaw64k | /? | /help]** (*Shorthand command: sCodec)*	Type of codec and rate used in the H.323 Service when communicating with another VoIP endpoint. Default: g711Ulaw64k **Note:** All H.245 channel signaling and RTP streams are based on the codec setting. The H.323 Service will only accept the prompt files with the same encoding as the codec setting. If instructed to play an audio file with a different encoding, the prompt will not be played and an error will be generated. System shutdown and startup necessary for changes to take effect.
ShowDefaultConnectTimeout [/? | /help] **SetDefaultConnectTimeout [<New Value> | /? | /help]**	The number of seconds the H.323 Service waits for a TCP connect to a server to complete before timing out and generating an error. Note: This does not apply to Call Servers and Media Servers. They have their own settings. Default: 3 seconds Changes to this value take effect immediately.
ShowExcludeIP [/? | /help] **SetExcludeIP [10.99.98.99, 10.99.99.100, ... | /? | /help]**	When SetTransferLabel is set, a call whose transfer label from Unified ICME matches one of the labels in that list would normally be sent back to the same gateway from which it arrived for IVR treatment, regardless of the route settings in the gatekeeper. Sometimes it is not desirable to send the call back to the same gateway from which it arrived, for instance, such as when the call originated from Unified CM. The SetExcludeIP command lets you specify a comma-delimited list of IP addresses that will ignore the SetTransferLabel directive. Valid value: A comma-delimited list of IP addresses. Default: "" Changes to this value take effect immediately.
ShowGatekeeper [/? | /help] **SetGateKeeper [<NewValue> | /? | /help]** *(Shorthand commands: SGK, SetGK)*	Specifies a list of gatekeepers to which the H.323 Service will register. **Note:** The H.323 Service will register to only one gatekeeper at a time. In the event that a gatekeeper failure is detected by the H.323 Service, it will register to the next gatekeeper in the list. **Note:** The H.323 Service will not automatically revert to the previous gatekeeper when that gatekeeper becomes available again. You will need to re-enter the setGK command with the original gatekeeper address to revert to the beginning of the list.

Show and Set Command Syntax	Description
	You can specify particular zones and ports on the gatekeeper. For example: • setGK "10.86.129.33" • setGK "10.86.129.33:zonename1" • setGK "10.86.129.33, 10.86.129.34, 10.86.129.35" • setGK "10.86.129.33:zonename1:portnum, 10.86.129.34" • setGK "10.86.129.33:zonename1:portnum, 10.86.129.34:zonename2:portnum" Valid values: IP address or “none” Default: none (or no value) Changes to this value take effect immediately.
ShowH323ID [/? | /help] **SetH323ID [<NewValue> | /? | /help]** *(Shorthand command: sH323ID)*	This value defines a unique identification for this H.323 endpoint in a H.323 network. By convention, this is the IP address of the machine. Valid values: 256 character string Default: Local IP address System shutdown and startup necessary for changes to take effect.
ShowH323Trace [/? | /help] **SetH323Trace [on | off | /? | /help]**	When this value is set to on, H323 details will be logged to the console and log files. Valid Values: on and off Default: off. Changes to this value take effect immediately.
ShowH323Zone [/? | /help] **SetH323Zone [<New Value> | /? | /help]**	This command is obsolete. Use setGatekeeper, instead.
ShowISNtoISN [/? | /help] **SetISNtoISN [on | off | /? | /help]**	When this value is set on, the H.323 Service is not allowed to send calls to another H.323 Service. Valid Values: on and off Default: off Changes to this value take effect immediately.
ShowInterfaceTrace [/? | /help]	When this value is set to on, Interface details will be logged to the console and log files.

Show and Set Command Syntax	Description
SetInterfaceTrace [on | off | /? | /help]	Valid Values: on and off Default: off. Changes to this value take effect immediately.
ShowLocationsBasedCAC [/? | /help] **SetLocationsBasedCAC [on | off | /? | /help]**	When this value is set on, the H.323 Service transfers calls to accommodate Unified CM (Unified CCM) Location-based CAC. Specifically, this means forcing the H.225 setup to port 1720 on Unified CCM and including the IP address of the originating gateway in the sourceCallSignalAddress of theARQ and H.225 setup message on the transfer. Valid Values: on and off Default: off Changes to this value take effect immediately.
ShowLogMeters [/? | /help] **SetLogMeters [on | off | /? | /help]**	Controls logging metrics. When "on," the H.323 Service will log metrics to the console and log files at the interval specified by the "Message Meter Interval." Metrics are always available on demand for the last interval(s). Valid Values: on and off Default: on Changes to this value take effect immediately.
ShowMaxIVRPorts [/? | /help] **SetMaxIVRPorts [<NewValue> | /? | /help]** *(Shorthand command: sMaxIVRPorts)*	Maximum number of calls in the H.323 Service which are allowed to receive IVR treatment at any given time. If the number of calls receiving IVR treatment at any given time exceed the MaxIVRPorts value, additional calls arriving at the H.323 Service is rejected. **Note:** Even when a Gateway is acting as the IVR—as can be the case in a Unified CVP Comprehensive call flow model—you still need a small number of IVR ports on the H.323 Service, based on the incoming calls per second and subsequent transfers. Use the Max IVR Ports listing in the H.323 Service Total Statistics log to help make decisions regarding IVR port sizing. Default: 50 System shutdown and startup necessary for changes to take effect.
ShowMaxTotalCalls [/? | /help] **SetMaxTotalCalls [<NewValue> | /? | /help]** (*Shorthand command: sMaxTotalCalls)*	Maximum total number of calls allowed in the H.323 Service. Default: 555 System shutdown and startup necessary for changes to take effect.

Show and Set Command Syntax	Description
ShowMediaServerConnectTimeout [/? | /help] **SetMediaServerConnectTimeout [<New Value> | /? | /help]**	The number of seconds the H.323 Service waits for a TCP connect to the Media Server to complete before timing out and generating an error. Default: 1 second Changes to this value take effect immediately.
ShowMediaServerTimeout [/? | /help] **SetMediaServerTimeout [<NewValue> | /? | /help]** (*Shorthand command: sMSTime)*	Number of seconds the H.323 Service should wait for a response from the Media Server before timing out and reporting an error. The H.323 Service will attempt to get the media file from all the IP addresses in sequence resolved by the DNS server for the Media Server URL. The timeout value is for every IP address resolved by DNS server for the media server URL. Default: 10 System shutdown and startup necessary for changes to take effect.
ShowMediaServerTries [/? | /help] **SetMediaServerTries [<NewValue> | /? | /help]** (*Shorthand commands: sMSTries, setMSTries)*	Maximum number of retries the H.323 Service attempts (in addition to the original attempt) in trying to connect to the media server before failing and reporting an error. The H.323 Service attempts to retry every IP address resolved by the DNS server for the Media Server URL. The MediaServerTries is the number of retries for each IP address resolved by the DNS server. The H.323 Service does all the tries on each media server before moving to the next media server. Default: 1 System shutdown and startup necessary for changes to take effect.
ShowMeterInterval [/? | /help] **SetMeterInterval [<New Value> | /? | /help]**	The time interval in which metrics are output to the console and log file. Setting this value to less than 300 seconds (5 minutes) might impact system performance. The meters setting must be set to "on" to activate logging. Changes to this value take effect immediately. Default: 1800 seconds
ShowNewCallOnly [/? | /help] **SetNewCallOnly [<on | off> | /? | /help]** (*Shorthand commands: sNco, setNco)*	Determines whether a call is restarted from the beginning if there is a NAM/Unified ICME or Call Server problem during the course of the call. Set this value to “on” to restart calls. (Because pre-routed calls cannot be restarted, turning this setting on means that you do not expect any pre-routed calls to come to this H.323 Service; that is, only new calls arrive at this H.323 Service.) **Note:** Do not set this value to “on” if you expect any pre-routed calls to come to this H.323 Service. Unified CVP cannot restart pre-routed calls.

Show and Set Command Syntax	Description
	Default: off Valid Values: on, off Changes to this value take effect immediately.
ShowNMRestartTimer [/? | /help] **SetNMRestartTimer [<New Value> | /? | /help]**	Number of seconds the Node Manager should wait to restart the H.323 Service after recognizing that it is not healthy. Delaying the restart gives transferred calls ample time to terminate conversations normally. Default: 1800 seconds. System shutdown and startup necessary for changes to take effect.
ShowNumOutOfBand [/? | /help] **SetNumOutOfBand [<NewValue> | /? | /help]** (*Shorthand commands: sOutofBand, SetOutofBand)*	Number of connections the H.323 Service should reserve per Call Server for performing out of band communications, such as transferring a queued call. Default: 3 System shutdown and startup necessary for changes to take effect.
ShowOutOfBandConsecFails [/? | /help] **SetOutOfBandConsecFails [<NewValue> |/? | /help]** (*Shorthand commands: sOBFail, SetOBFail)*	Maximum number of consecutive times the H.323 Service can try to connect to the out of band channels before failing and reporting an error. Default: 3 System shutdown and startup necessary for changes to take effect.
ShowOutOfBandTimeout [/? | /help] **SetOutOfBandTimeout [<NewValue> | /? | /help]** *(Shorthand commands: sOBTime, SetOBTime)*	Number of seconds the H.323 Service should wait to connect to out of band channels before failing and reporting an error. Default: 60 seconds **Note:** There is a relationship between this setting and the IVR Service's Heartbeat timeout setting in the Operations Console. The Heartbeat timeout should be two times the OutOfBandTimeout setting. (For information about setting the Heartbeat timeout, refer to the Operations Console online help.) System shutdown and startup necessary for changes to take effect.
ShowPassPresentationInd [/? | /help] **SetPassPresentationInd [<on or off> | /? | /help**	Controls whether the H.323 Service will pass the Calling Party Presentation Indicator to the outbound call leg. Valid values: on, off Default: on Changes to this value take effect immediately.
ShowRAIMaxThreshold [/? | /help]	The RAI indicator instructs the Gatekeeper to stop routing further incoming calls to the voice browser if the active incoming calls exceeds:

Show and Set Command Syntax	Description
SetRAIMaxThreshold [<NewValue> | /? | /help] *(Shorthand commands: sRaiMax, SetRaiMax)*	• (RAIMaxThreshold/100)*MaxTotalCalls • **OR:** • (RAIMaxThreshold/100)*maxIVRports - (currentTransferedCalls * (takebackPercentage/100)) Valid values: 0 - 100 Default: 90 **Note:** RaiMinThreshold must be less than RaiMaxThreshold. Changes to this value take effect immediately.
ShowRAIMinThreshold [/? | /help] **SetRAIMinThreshold [<NewValue> | /? | /help]** (*Shorthand commands: sRaiMin, SetRaiMin)*	The RAI indicator instructs the Gatekeeper to stop routing further incoming calls to the voice browser if the active incoming calls drops below: • (RAIMaxThreshold/100)*MaxTotalCalls • **OR:** • (RAIMaxThreshold/100)*maxIVRports - (currentTransferedCalls * (takebackPercentage/100)) Valid values: 0 - 100 Default: 80 **Note:** RaiMinThreshold must be less than RaiMaxThreshold. Changes to this value take effect immediately.
ShowRNATimeout [/? | /help] **SetRNATimeout [<integer >0 and <=1800> | /? | /help]** (*Shorthand commands: sRNATimeout)*	H.323 Service restart necessary for changes to take effect. The number of seconds the H.323 Service waits for the called party to answer before timing out and returning an RNA error back to Unified ICME. This value can be set on either a global basis (that is, all calls have the same RNA timeout) or on a per-transfer basis based on the value of the transfer DNIS. **Note:** This is a global setting and cannot be adjusted on a per-call basis. Global syntax: SetRNATimeout *:30 Sets the RNA timeout for all calls to be 30 seconds. Default is 15 seconds if no value is specified. Valid Values: 15 - 180 Default: 15 seconds

Show and Set Command Syntax	Description
	H.323 Service restart necessary for changes to take effect. Per-call syntax: setRNATimeout " 1*:30, 20*:60, 3...:20, *:300 " The value before the colon(:) represents the transfer DNIS.The value after the colon represents the RNA timeout for that DNIS. One "*" or multiple "." can be placed at the end of the DNIS value. So, in the above example: • Transfer DNIS's beginning with 1 will have a 30 sec RNA timeout. • Transfer DNIS's beginning with 20 will have a 60 sec RNA timeout. • Transfer DNIS's beginning with 3 and containing exactly 4 characters will have a 20 sec RNA timeout. • Transfer DNIS's that don't match any of the previous regular expressions will have a 300 sec RNA timeout. Precedence order for DNIS matching is as follows. Assume DNIS is 2100. In order of increasing matching precedence: * < 2* < 21* < 21.. < 2100 If a DNIS does not match any value in the list, it will default to the highest RNA timeout value specified in the list unless a specific "*:NNN" catch-all timeout is provided, in which case it will use that timeout value. Any RNA timeout value must be > 0 and <= 1800 seconds.
ShowSecurityMask [/? | /help] **SetSecurityMask [<New Value> | /? | /help]**	This setting controls various aspects of how and who can administer and configure the H.323 Service. Default: 1 Changes to this value take effect only after an H.323 Service restart. When set to value 0, remote administration of the H.323 Service using procmon is not allowed. All administration must be performed from the local machine.
ShowServiceMode [/? | /help] **SetServiceMode [in | out | /? | /help]**	When set to out, the H.323 Service does not accept new calls but processes all existing calls to completion. When set to in, the H.323 Service starts accepting new calls. Valid Values: in or out Changes to this value take effect immediately.
ShowSetupCallingNum [/? | /help]	Controls whether the Dialed Number (DNIS) or Calling Line Identification (CLI) is used as the Calling Party Number in VoIP

Show and Set Command Syntax	Description
SetSetupCallingNum [CLI|DNIS /? / help]	messages during IP Call Transfer. When this value is set to “CLI”, CLI (if present) will be passed in the Calling Party Number parameter of the Setup message during IP Transfer; when set to “DNIS”, DNIS are passed. Valid Values: CLI, DNIS Default: CLI Changes to this value take effect immediately.
ShowSigDigits [/? | /help] **SetSigDigits [integer >= 0 | /? | / help]**	The number of leading significant digits (not including the tech-prefix, if it exists), which Unified CVP will strip from the incoming DNIS. The value that is stripped off is saved. When the H.323 service receives a transfer command from Unified ICME, it first appends a # sign to this value and then prepends the resulting string to the transfer label. **Note:** This is a mechanism used to pair together Ingress Gateways to desired targeted transfer endpoints. Valid Values: an integer >= 0 Default: 0 Changes to this value take effect immediately.
ShowSurveyDnis [/? | /help] **SetSurveyDnis <original Dnis-survey Dnis, [original Dnis-survey Dnis, ...]> [/? | /help]**	One or more alternate DNIS numbers to which to restart the call. If the called party disconnects before the caller, and no further instructions are sent from Unified ICME within 2 seconds, the H.323 Service checks this list of DNIS numbers to see if the call should be restarted to an alternate DNIS. For more information on restarting a call to an alternate DNIS, refer to "Restart to Alternate DNIS (page 434)." You must specify at least one pair of values consisting of the original DNIS number and the alternate or survey DNIS number to which the call should be restarted. Wild cards are allowed. Each pair should be delimited by a comma and the entire string enclosed in quotes. Only [0-9,*] are acceptable characters, and the ‘*’, if present, must occur at the end of a DNIS value. For more information on wild cards, refer to the Operations Console online help. Select **System > Dialed Number Pattern > Add New** or Edit. Examples: SetSurveyDnis "9785551515-1234" If the original DNIS was 9785551515, the call will be restarted to DNIS 1234.

Show and Set Command Syntax	Description
	SetSurveyDnis "978*-1234" If the original DNIS was a DNIS beginning with 978, the call will be restarted to DNIS 1234. SetSurveyDnis "978*-800*" If the original DNIS was a DNIS beginning with 978, the call will be restarted to the same number but 978 would be replaced with 800, e.g. 9785551212 would be restarted to 8005551212. SetSurveyDnis "9785551515-1234,8003331111-9876" If the original DNIS was 9785551515, the call will be restarted to 1234, OR if the original DNIS was 8003331111, the call will be restarted to 9876. SetSurveyDnis "*-9000" If the original DNIS is anything, the call will be restarted to DNIS 8000. The survey DNIS feature can be used with Unified CVP that is defined as any VRU type, provided that the SurveyDnis invokes a new call in Unified ICME. That is, the survey DNIS must not fall in a Translation route DNIS range and must be less than or equal to the "Maximum Length of DNIS" field in the ICM Service tab in the Operations Console. Changes to this value take effect immediately.
ShowTakebackDelay [/? | /help] **SetTakebackDelay [<New Value in milliseconds> | /? | /help]**	This setting controls the number of delay milliseconds that each comma in a transfer label represents. For example, when doing an outpulse transfer, it is sometimes desirable to place a pause between the *8 and the digits, as in *8,,,9870987465. This setting controls how long the pause is for each comma. Hence, 3 commas = 300ms if this value is set to 100. Default:100 milliseconds Changes to this value take effect immediately.
ShowTakebackPercentage [/? | /help] **SetTakebackPercentage [<NewValue> | /? | /help]** (*Shorthand commands: sTakePct, SetTakePct)*	Percentage of currently transferred calls for which IVR ports should be reserved for queuing or other IVR treatment. Default: 0 (zero) Changes to this value take effect immediately. (System shutdown and startup unnecessary.) For example, if the MaxIVRPorts setting is 100, MaxTotalCalls is 300, and TakebackPercentage is 10, it would mean the following: • If no calls are currently transferred, the H.323 Service would accept up to 100 calls IVR.

Show and Set Command Syntax	Description
	• If 50 calls are currently transferred, the H.323 Service would accept up to 95 calls IVR (then there would be 145 total calls and the H.323 Service would stop accepting new ones). • If 200 calls currently transferred, the H.323 Service would accept up to 80 IVR (then there would be 280 total calls and the H.323 Service would stop accepting new ones). • At 300 calls, the H.323 Service would still have room for 70 IVR but would stop taking calls, anyway, because the total number of calls would have been reached.
ShowTechPrefix [/? | /help] **SetTechPrefix [<NewValue> | /? | /help]** (*Shorthand command: sTechPrefix)*	Specifies the tech-prefix the H.323 Service uses to register to the Gatekeeper. Default: 2# System shutdown and startup necessary for changes to take effect.
ShowTechPrefixRemoval [/? | /help] **SetTechPrefixRemoval [<on or off> | /? | /help]**	Controls whether the H.323 service will strip of the tech prefix before it sends the call to the Call Server. Valid values: on, off Default: on System shutdown and startup necessary for changes to take effect.
ShowTraceMask [/? | /help] **SetTraceMask [<New Value> | /? | /help]**	For Use by Technical Support only. Default: 0x101003 Changes to this value take effect immediately.
ShowTransferCLI [/? | /help] **SetTransferCLI [<on | off> | /? | /help]** (*Shorthand command: sTransferCLI)*	Controls whether the Dialed Number (DNIS) or Calling Line Identification (CLI) is used as the source address in VoIP messages during IP Call Transfer. When this value is set to “on,” the CLI (if present) will be passed in the SrcInfo parameter of the ARQ message during IP Transfer; when set to “off,” the DNIS will be passed. Valid Values: on, off Default: off Changes to this value take effect immediately.
ShowTransferLabel [/? | /help] **SetTransferLabel[<List of labels, delimited by spaces, all enclosed in quotes> | /? | /help]**	A list of label values (delimited by spaces and the entire list enclosed in quotes). If any of the values in this list match the VRU label configured in Unified ICME, it triggers the H.323 Service to send the transfer back to the originating Ingress Gateway instead of using the IP address returned by the gatekeeper lookup. If this value is not set or set to 'None', the H.323 Service uses the IP address returned by the gatekeeper lookup for the transfer.

Show and Set Command Syntax	Description
	Note: A gatekeeper is still needed even when this value is set. However, in that case, the IP address returned by the gatekeeper is ignored. Default: None Changes to this value take effect immediately.
ShowUUIParams[/? | /help] **SetUUIParams None | Full | XXX[N:N:...], [YYY[N:N:...]] | /? | /help]**	Lets you to specify which lines of GTD data to pass to Unified ICME. An incoming call (whether from the PSTN or from Unified CM) can contain GTD data. Generally, the data looks like this: **IAM,** **PRN,isdn*,,ATT5*,** **USI,rate,c,s,c,1** **CPN,00,,u,6005551212** **CGN,00,,u,y,1,12345** **UUS,3,303132333530** **GCI,2e456a598dcc11daa950000bfda207f2** **Note:** Refer to the NSS spec (Narrowband Signaling Syntax) Q.1980.1 ITU specification for specific details about the meaning and syntax of NSS (GTD) data. Use SetUUIParams to specify data to pass to Unified ICME and whether data should be converted from 2-byte hex representation to 1-byte ASCII. (For example, the UUS value would be "303132333530" in 2-byte hex representation; in 1-byte ASCII, the value would be "012350".) For example, the command **SetUUIParams CPN, UUS:2,CGN** would: • Pass the data in the CPN message to Unified ICME. • Convert the data in the UUS message to ASCII and pass it to Unified ICME. • Pass the data in the CGN message to Unified ICME. Default: None Changes to this value take effect immediately.

Show and Set Command Syntax	Description
ShowVbRestartTimer [/? /help] **SetVbRestartTimer [value /? /help]**	Number of seconds the H.323 Service should wait to restart itself if it is not operating properly. Delaying the restart gives transferred calls ample time to terminate conversations normally. Default: 1800 seconds Changes to this default take effect immediately.
ShowActiveCalls [/? | /help]	Shows calls the are currently active in the H.323 Service.
ShowAll [/? /help]	This command displays all configuration data and real-time data for the H.323 Service.

Delivering H.323 Incoming UUI to a Unified ICME Routing Script

The UUI extracted from the incoming call is passed to the Unified ICME scripting environment by the Unified CVP Call Server. This feature enables capturing data from an external system (for example, caller-entered digits from a third-party IVR) and passing that data to Unified ICME on a new call. This can be accomplished by populating the UUS parameter (often known as the UUI) in the IAM message of the GTD (Generic Transparency Descriptor) data that is sent to the gateway from the network in the Q.931 setup message. The gateway and Unified CVP can extract this data and send it to Unified ICME on a new call.

Additionally, other parameters in the GTD can also be extracted and sent to Unified ICME if the user chooses. Any parameter contained in the NSS IAM message can be extracted as long as the ingress IOS gateway also extracts it. Refer to **http://www.cisco.com/en/US/partner/products/sw/iosswrel/ps5207/products_configuration_guide_chapter09186a008020eccf.html** for a description of all the GTD fields IOS 12.4 will extract. Also, refer to the ITU-T Narrowband Signaling Syntax spec (Q.1980.1) for a detailed description of the IAM message. This feature is available in Unified CVP Comprehensive and VRU-Only call flow models and can be used with any Unified CVP VRU type.

How it Works

The external system inserts the desired values into the dat field of the UUS parameter of the NSS IAM message. These NSS messages are used as the basis for building the GTD data that ultimately arrives at the IOS gateway from the PSTN. Note that the UUS dat field is represented by pairs of hexadecimal digits. This means that if the external system wants to pass "12345" in the UUS dat field, it will arrive to the gateway with the following representation: "3132333435". By default, this 2-byte hexadecimal value is passed to Unified ICME. The Unified ICME script must manipulate this value using the Formula Editor. The user can specify an option on the gateway or the Unified CVP H.323 Service to convert to the 1-byte ASCII representation (that is, "12345") before passing to Unified ICME.

Any data that cannot be represented by a printable ASCII character will be replaced with a "." character. Therefore, if the incoming GTD data from the network contains binary data (such as counters), this option should not be used since it will result in a loss of valuable information. Additionally, other fields from the IAM message can also be utilized, if desired.

How that data moves from the gateway to the Unified ICME differs depending on the deployment model used. The format in which the data appears in Unified ICME will also be somewhat different based on the deployment model. In any case, Unified ICME 7.1(1) and beyond will accommodate a maximum length of 131 characters. Before that release, Unified ICME will only accommodate a maximum length of 36 characters.

VRU-Only Call Flow Model

The bootstrap.tcl file on the gateway extracts the GTD fields that the user has configured. By default, the UUS.dat is extracted, if user specifies nothing. That data is then passed to the Unfied CVP Call Server in the HTTP URL as the CALL_UUI.The Call Server places the CALL_UUI data in the UUI variable, which is then passed to Unified ICME. Unified ICME makes that data available in the Call.UserToUserInfo field in the Unified ICME script. It also stores it in the UserToUser column in the TCD (Termination Call Detail record) in the database.

Comprehensive Call Flow Model

GTD data is passed to the Unified CVP H.323 Service automatically in the H.323 setup in the NonStandardControl element if the following is configured on the ingress gateway:

```
voice service voip
    signaling forward unconditional
```

The Unified CVP H.323 Service extracts the GTD fields the user has configured using VBAdmin. The user must explicitly extract the UUI using the VBAdmin **setUUI**Unified CVP Call Server in the HTTP URL as the CALL_UUI.

The Call Server places the CALL_UUI data in the ICM Service UUI variable, which is then passed to Unified ICME. Unified ICME makes that data available in the Call.UserToUserInfo field in the Unified ICME script. It also stores it in the UserToUser column in the TCD (Termination Call Detail record) in the database.

VBAdmin Logging Commands

The following table describes VBAdmin logging commands. For more information about logging, refer to the Operations Console online help.

Table 40: VBAdmin Logging Commands

Show and Set Command Syntax	Description
`ShowCallTrace [/? | /help]` SetCallTrace [<on | off> | /? | /help] *(Shorthand command: sCallTrace)*	When “on,” the H.323 Service logs basic call detail information to the console and log files. Valid Values: on, off Default: off Changes to this value take effect immediately.

Show and Set Command Syntax	Description
ShowInterfaceTrace [/? | /help] **SetInterfaceTrace [<on | off> | /? | /help]** (*Shorthand command: sIntTrace, setIntTrace)*	When "on," the H.323 Service logs interface details to the console and log files. Valid Values: on, off Default: off Changes to this value take effect immediately.
ShowH323Trace [/? | /help] **SetH323Trace [<on | off> | /? | /help]**	When "on," H.323 details will be logged to the console and logfiles. Valid Values: on, off Default: off Changes to this value take effect immediately.
ShowTraceMask [/? | /help] **SetTraceMask [<NewValue> | /? | /help]** (*Shorthand command: sTraceMask)*	For use by Technical Support, only. Default: 101003HEX Changes to this value take effect immediately.

VBAdmin Metric and Control Commands

The H.323 Service keeps track of significant events and measurements; these metrics are logged at regular intervals and can be displayed on demand. Measurements of latency for particular actions (retrieving a voice file, for instance), are counted for the interval in which they complete, as opposed to the interval in which they start.

Most VBAdmin Metric commands are display-only (that is, in **show<***Parameter***>** syntax).

Note: For information on how H.323 Service metrics are logged, refer to "Using Cisco Support Tools with Unified CVP (page 221)."

Table 41: VBAdmin Metric and Control Commands

Command Syntax	Description
DisconnectCall [<LocalID> | /? | /help] *(Shorthand command: DisCall)*	Disconnects the call identified by Local ID from the active call list (after confirmation).
ShowActiveCalls [/? | /help]	Displays the following information about each call active in the H.323 Service: • Local ID • Unique Call ID • Creation Time

Command Syntax	Description
	• State • Duration • DNIS • ANI • Last Call Server the call accessed • Last Media Server the call accessed
`ShowCallHistory [<CallID> | /? | /help]` *(Shorthand command: sCallHist)*	For use by Technical Support, only. Can be used to print out a detailed call history of an active call. Either the long or short call ID may be entered as the argument. Default: none
`ShowIntervalStatistics [/? | /help]`	Displays the interval time, and a series of counts, averages and maximums during that interval. The statistics include: • Interval size. • Number of new calls. • Number of calls transferred. • Number of calls rejected (if H.323 Service is out of service). • Number of transfer errors. • Number of prompts not ready. This is the number of times a prompt was playing, and before the end of the prompt was reached, the H.323 Service had to stop playing the prompt because the next part of the file had not yet been received. • Number of prompts not found. • Number of calls using critical media, that is, that were prematurely terminated because of internal errors (in which case a "system error" message is played to the caller). **Note:** These do not always represent call drops. Sometimes, incorrect call release configuration in Unified ICME scripting and configuration can cause these errors; however, they should be considered suspect. • Number of calls terminated (call may have begun in a previous interval).

Command Syntax	Description
	• Percent of total system CPU used - average, minimum, maximum and standard deviation. • For each H.323 Service, average, minimum, maximum and standard deviation of the latency statistics in processing a new call arrival. This represents the round-trip time from when the H.323 service first receives the new call from the network to the time the H.323 service receives a response from Unified ICME. **Note:** This would include the time involved for the Unified ICME to perform back-end database lookups, if required. If there are no back-end database lookups, an average value of greater than 1000 ms indicates potential network problems. • For each H.323 Service, average, minimum, maximum and standard deviation of the latency statistics in processing a successful transfer, measuring the time from when the H.323 Service receives the Transfer command from the Call Server to the time of alerting. An average value of greater than 1000 ms indicates potential network problems. • For each H.323 Service, average, minimum, maximum and standard deviation of the latency statistics in processing a successful transfer, measuring the time from alerting to when the called party answers. • For each Call Server, average, minimum, maximum and standard deviation of the latency statistics in communicating with Call Server for call processing requests, and the number of message exchanges (requests to the Call Server from the H.323 Service). This represents the round-trip time from when the H.323 service sends a request (not including new call requests) to Unified ICME to the time the H.323 service receives a response from Unified ICME. **Note:** This would include the time involved for Unified ICME to do back-end database lookups, if required. If there are no back-end database lookups, an average value of greater than 1000 ms indicates potential network problems. **Note:** Averages, minimums, maximums and standard deviation statistics are calculated by using current data (occurring within this interval), not rolling averages (occurring over multiple intervals). Call latency is displayed in milliseconds.
`ShowLogMeters [/? | /help]` **`SetLogMeters [<on | off> | /? | /help]`** *(Shorthand command: sLogMeters)*	Controls logging metrics. When "on," the H.323 Service will log metrics to the console and log files at the interval specified by the Message Meter Interval setting. Metrics are always available on demand for the last interval(s). Valid Values: on, off

Command Syntax	Description
	Default: on Changes to this value take effect immediately.
ShowMeterInterval [/? | /help] **SetMeterInterval [<NewValue> | /? | /help]** *(Shorthand command: sMeterInterval)*	Time interval, in seconds, at which metrics will be calculated. Note that, if logging of metrics is "on," setting the interval to a short time will cause the log files to roll over more quickly. Also, the Meters parameter must be set to "on" to activate logging. Default: 1800 seconds (30 minutes) Changes to this value take effect immediately. (System shutdown and startup unnecessary.)
ShowServiceMode [/? | /help] **SetServiceMode [<in | out> | /? | /help]** *(Shorthand commands: sServMode, setServMode)*	Controls the processing of calls. When "out" (out of service), the H.323 Service will not accept new calls but will process all existing calls to completion. When "in" (in service), the H.323 Service will accept new calls. Valid Values: in, out Changes to this value take effect immediately. (System shutdown and startup unnecessary.)
ShowSnapshot [/? | /help]	Gives a count of all the calls in progress, and subtotals for the count in each state. The states include: • Waiting for VoiceXML (or for response from Call Server) • Playing Prompt • Waiting for DTMF • Transferred • Disconnecting • Disconnected (caller has been disconnected and software is just finishing up) • New (call has arrived, but processing has not started yet) • IVR Ports in use (a call is in one of two states at any point in time: receiving IVR treatment, or transferred. Time spent in all call setup—H.323 activity plus communications to Unified ICME—counts as being in the IVR state. Therefore, even though a call may be 'immediately' transferred upon arrival at Unified CVP, there is still a brief period—several seconds—that the call is in the IVR state until the endpoint answers the call. IVR Ports in use shows the number of calls in the IVR state at any point in time)

Command Syntax	Description
	• Wait (a transient internal state; calls should never stay in this state any length of time; this number increasing over time indicates a problem) • Other (calls should never be in this state; a catch-all for problem calls) • Internal (for debugging purposes only; shows the total number of calls in the internal call object map in the H.323 Service; includes all active call legs, plus calls that are marked for deletion). **Note:** The value of Internal Call Count should not be greater than the value of ShowMaxTotalCalls; if it is, it indicates that there is a memory leak in the H.323 Service. Also, this number can be greater than 0 when no calls are active in the system, as this value represents the number of call objects, not the number of active calls.
`ShowStatus [/? | /help]`	Status gives the overall status for the Call Server for the entire time the H.323 Service has been up, including: • Total calls • Disconnect Disposition (Rejected, Caller Hangup, Called Party Hangup, ICM Release, Critical Media) • Maximum Simultaneous Calls • Maximum IVR Ports • Total Prompts Not Found • Total Transfer Errors (Busy, Ring No Answer, Gatekeeper Problem, Destination Problem, Other) • System Startup Time • System Up Time • Current System State (In Service, Out of Service) • Packets Transmitted
`ShowVersion [/? | /help]`	Displays release number (e.g., 4.0) and build number of the software.

VBAdmin Error Handling

VBAdmin validates configuration commands and settings before accepting them. If a command is not valid, VBAdmin displays an appropriate error message.

Note: Although VBAdmin validates the syntax of the H.323 Service configuration commands, it performs no checks regarding the validity of IP addresses/DNS names.

Performing H.323 Blind Transfers

The Unified CVP H.323 Service can perform a H.323 blind transfer similar to the SIP Refer transfer, which allows Unified CVP to remove itself from the call, thus freeing up call control ports. Unified CVP cannot execute further call control operations after this kind of label has been executed; for example, it cannot perform subsequent transfers back to Unified CVP for self-service or queuing to another agent. However, Unified CVP survivability can still be executed in this scenario.

Using this feature, the call can be queued at the VoiceXML gateway and then sent to an agent with a Unified ICME label that begins with the letters "RF". Otherwise, standard Unfied ICME agent labels enable Unified CVP to remain in the signaling path for the duration of the call, and the Unified CVP resource will not be freed until the end of the call. RF transfers can be made to Unified CM or other H.323 endpoints, such as an ACD. The Unified CVP H.323 Service performs the digit outpulsing for H.323 RF and the Unified CVP survivability script detects that RF functionality has to be invoked based on the RF number, which should match in both the Unified CVP H.323 Service and the Unified CVP survivability service. The Unified CVP survivability script strips the RF number, terminates #, and connects the caller to the label using the configured dial-peer and releases the call control to the Unified CVP H.323 Service.

Note:

- This feature is only available for PSTN-originated calls via IOS GW running Unified CVP survivability service.
- This feature can be used in both the Comprehensive and the Call Director call flow models.

Configuring H.323 Blind Transfer in Unified ICME

Step 1 Set the user.h323.rftransfer ECC variable in a Unified ICME script. When using this ECC variable in a Unified ICME script, it must be set to the value of the single character "y" or "Y" and Unified CVP will use the RF method when transferring to the agents.

Step 2 Alternatively, use a fixed RF label in a Unified ICME script. The RF label format for H.323 service is "RF<Rf number>#<destination label>#" .An RF number (default is 88) identifies the type of transfer for Unified CVP survivability service. For example, if the destination label is 1002, then the corresponding RF label should be RF88#1002#. If the user.h323.rftransfer ECC variable is set to "y," then the Unified CVP H.323 Service will modify the received label automatically to conform to the format given above.

Configuring H.323 Blind Transfer in Ingress Gateway

Step 1 When using H.323 RF transfers, the Ingress gateway must include an outbound voip dial peer. This outbound dial peer is necessary because when the RF transfer message enters the gateway from the Call Server, it needs to match an outbound dial peer in order for the call to succeed. For example:

```
dial-peer voice 1050 voip
 destination-pattern 1...
 voice-class codec 1
 session target <your h323 destination>
 dtmf-relay rtp-nte h245-signal h245-alphanumeric
 no vad
!
```

Step 2 Enable the Unified CVP survivability service to execute the H,323 RF transfers by using the following parameter: **`param icm-rf 1`**. The default RF number in the Unified CVP survivability script can be changed using **`param rf-number <new RFnumber>`**.

Configuring H.323 Blind Transfer in H.323 Service

Step 1 Change the default RF number (88) using the Unified CVP VBAdmin **`setRFN <new RFNumber>`** parameter.

Note: Do not append any characters to the Temp connect using Label+corrId even if h323.rftransfer is set to "y" or "Y" in the Unified ICME script.

Part 3: Configuration Detail of Non-Unified CVP Components

This part of the manual describes the non-Unified CVP components that you configure outside the Operations Console.

Chapter 9

Using Cisco Unified ICME Warm Consult Transfer/Conference to Unified CVP

This chapter provides information about the minimal software component release requirements for the Unified ICME Warm Consult Transfer and Conference to Unified CVP feature for the H.323 Service and Type 2/7 VRUs. Resource sizing and configuration requirements are also included.

Note: For information about using the Warm Consult Transfer feature with SIP and Type 10 VRUs, refer to "Using the Warm Transfer Feature with SIP Calls (page 350)." Also, refer to the configuration instructions for the Call Director and Comprehensive call flow models using SIP in "High-level Configuration Instructions for Call Flow Models (page 25)."

This chapter contains the following topics:

About Unified ICME Warm Consult Transfer/Conference to Unified CVP Feature

When an agent attempts a warm consultative transfer/conference to another agent, but there is no agent available in the skill group to service the request, the first agent is placed in a queue to wait for the availability of an agent in the desired skill group. In order to place thefirst agent in queue, a call is initiated from Unified CM to Unified CVP (via a Translation Route to VRU) to provide queue music to the first agent. To Unified CVP, this appears as a new call from an IP phone.

Optionally, customer business call flows may require that IP phone users call Unified CVP directly. For example, you may have a corporate IP phone network that is serviced by a Unified CVP help desk call center. IP phone users with problems would call a Unified CVP number to open trouble tickets.

Configuring Unified ICME Warm Consult Transfer/Conference to Unified CVP Feature

To configure the Unified ICME Warm Consult Transfer/Conference to Unified CVP feature, do the following:

Step 1 Install a new Call Server (refer to **Installation and Upgrade Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html) for detailed information). It can be configured identically to all other Unified CVP machines, with the exception that you must add each Translation Route DNIS.

- Define it as a Type 2 VRU in the Network VRU Explorer tool in Unified ICME.
- Network Transfer Preferred must be disabled for this peripheral .
- Add a new DNIS in the **Add DNIS** box on the ICM tab in the Operations Console. Be sure to add each translation route DNIS.

Step 2 If the Unified CVP machine resides in a different location from the Unified CM cluster initiating the calls, WAN bandwidth will be a consideration since the prompts are played G.711 from the Unified CVP machine. In this case, size and configure the network appropriately. Where possible, Unified CVP should be co-located with Unified CM to eliminate these bandwidth requirements.

Step 3 Define a gateway device in Unified CM for the Unified CVP machine installed in Step 1. Under Device > Gateway, define an H.323 gateway using the Unified CVP IP address.

Step 4 (IP-originated calls only). Determine if customer business call flows require that IP phone users call Unified CVP directly (as described in the second paragraph of About Unified ICME Warm Consult Transfer/Conference to Unified CVP Feature (page 347)). In Unified CM administration under “Route Plan” using route groups/lists/patterns, route Unified CVP DNIS’s to the Unified CVP gateway installed in Step 1 above.

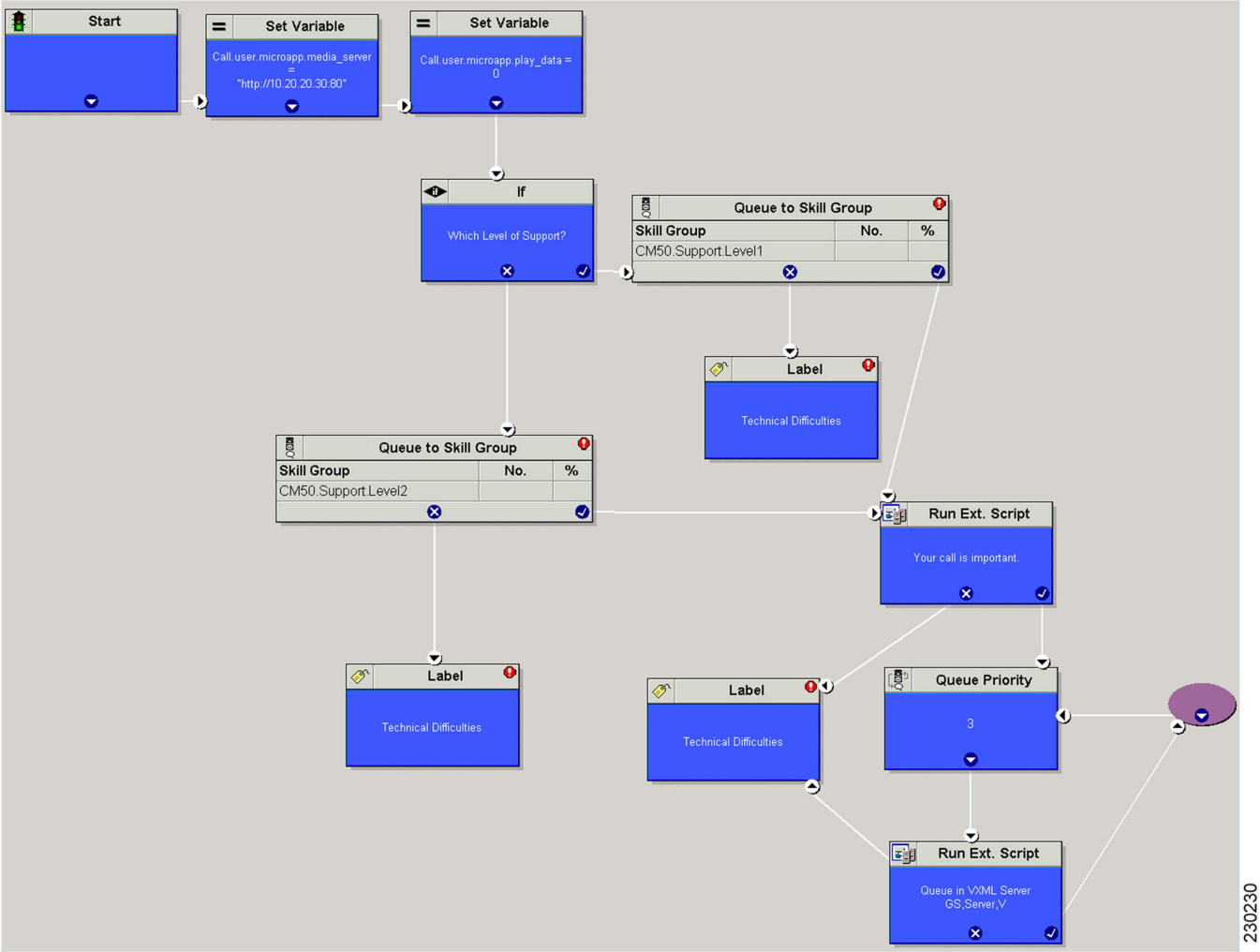
If you want to load-balance between two Unified CVP systems:

- Create a route group and put both of the Unified CVP gateways in the route group, both with order priority 1.
- Create a route list and put the route group in the route list.
- Create a route pattern and assign the route list to the route pattern.
- In Service Parameters for Unified CM, set “Reorder Route List” to **TRUE** and the 'H225 TCP timer' to **5**.

Note: The Reorder Route List setting applies only for Unified CM 3.3 and earlier.

Step 5 Create a Unified ICME script similar to the script below. (Refer to **Unified ICME documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html) for information.) This script should be tied to the Dialed number and call type that the agent invokes to do a warm consultative transfer/conference. This dialed number's Routing Client should be associated with a Unified CM peripheral from which the agent will be invoking the transfer or conference.

Figure 39: Unified ICME Script for Warm Consult Transfer



Minimal Component Version Required

Refer to **Hardware and System Software Specification for Cisco Unified Customer Voice Portal (Unified CVP) Software** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/prod_technical_reference_list.html) for the list of component versions required to use the Unified ICME Warm Consult Transfer and Conference to Unified CVP feature.

Using the Warm Transfer Feature with SIP Calls

In a scenario where an agent performs a warm transfer to another agent and then that agent is queued, and in addition a SendToVRU label returns to Unified CM using jtapi on the Unified CM PG connection, you must associate a Route Pattern for that label with a SIP TRUNK to send to Unified CVP or the Proxy Server to ensure the call returns to Unified CVP. Unified CVP then sends the "request instruction" message back to Unified ICME on the Unified CVP routing client and starts the queuing.

Note: These SIP calls do not require MTP enablement on the SIP trunks.

When using the Warm Transfer feature for SIP Calls with queuing, call flows does not require MTP enabled on the SIP trunk that is associated with the VRU label route pattern in the case where the agent completes consult transfer to caller while the call is still in the queue (VXML Gateway).

Note: The MTP is not required ifVXML GW version is IOS 12.4.(15)T8 or 12.4(20)T2 or later versions on these T releases. If the mobile agents supports KPML, then MTP is not required. In other cases, where there is SIP DTMF capability mismatch, MTP is required between CVP and CUCM.

Using the Unified ICME Warm Consult Transfer/Conference to Unified CVP Feature with a Type 10 VRU

Note: Unified CVP with a Type 10 VRU does not support multiple Network VRUs on the same Unified CVP peripheral device. Multiple customer instances can be used in order to address multiple Network VRUs, but they must then address different physical Unified CVP Call Servers as well. Calls that originate from an ACD or Unified CM, such as Warm Transfer/Conference, Helpdesk, or Outbound calls, are also limited to one Network VRU on any given Unified CVP Call Server. Note that the reverse is no problem: multiple Unified CVP Call Servers can share the same Network VRU.

In this scenario, an agent needs to transfer a call to another agent by dialing that agent's ID. If the agent is not available, the originating agent will be placed in a queue to wait for the second agent to pick up the call.

In order for the first agent to be queued while waiting for another agent, set up the following configuration:

Step 1 In the ICM Configuration Manager's PG Explorer tool Routing Client tabs, uncheck the NetworkTransferPreferred checkbox for Unified CM and Unified CVP routing clients.

Step 2 On the Advanced tab for the Unified CM routing client, select **None** for the Network VRU and your Type 10 VRU for the Unified CVP routing client.

Step 3 For your Type 10 VRU, in the ICM Configuration Manager's Network VRU Explorer tool define a label for the Unified CM routing client as well as the Unified CVP routing client and associate them with a customer instance.

Step 4 In the ICM Configuration Manager's Dialed Number List Tool, associate the dialed numbers for the incoming call as well as the transfer dialed number with the same customer instance.

When the second call is placed for the warm transfer and no agent is available, the label defined on the Unified CM RC plus the correlation ID will be sent back via EAPIM/JGW to Unified CM (for example, if the label is 7777777777, with a correlation ID it could be 777777777712345) since the call originated from the Unified CM RC and since the NetworkTransferPreferred checkbox is not checked.

Step 5 In Unified CM, select **Call Routing > Route/Hunt > Route Pattern > Add New**. Add a new route pattern to route the call to Unified CVP via either the SIP trunk if you are using SIP, or via a GK controlled trunk when using H.323 (for example, 777! where ! allows label plus arbitrary length correlation ID).

When Unified CVP sees this call, it perceives it as a pre-routed call with a correlation id and sends it back to Unified ICME to continue the script.

Unified ICME sends a temporary connection back to Unified CVP, which queues the agent call while the caller hears music on hold (MoH) from Unified CM.

Using Blind Transfer with Unified CVP as a Type 10 VRU

Blind Transfer, involving Unified ICME, does not work with Unified CVP as a Type 10 VRU. To use this feature, make the following configuration changes in Unified CM 5.0.

Step 1 Uncheck the **Wait for Far End H.245 Terminal Capability Set** parameter in CCMAdmin for the Unified CVP gateway device.

Note: When using Unified CVP H323 deployments with Unified CM 6.1(3) and above, you must check the **Wait for Far End H.245 Terminal Capability Set** parameter.

Step 2 Change the value of the Unified CM H323 Service Parameter called **Send H225 User InfoMessage** to the value **H225 Info for Call Progress**.

Chapter 10

Configuring Unified CM

This chapter describes how to configure Unified CM. Once Unified CM has been configured, you can add a pre-configured Unified CM server to the Operations Console network control panel. Once added, you can add the Unified CM Server to a device pool and access a Unified CM administration web page (refer to "Configuring a Cisco Unified Communications Manager Server (page 354)").

This chapter contains the following topics:

Configuring Unified CM for Use with Unified CVP

Prerequisite Configuration

Before configuring Unified CM for use with Unified CVP, you must do the following:

1. Configure a Unified CM server. (Refer to the instructions in the Unified CM documentation.)

2. To hear a ring tone on a blind transfer call, select**Service > Service Parameters** in Cisco Unified Communications Manager. Then select your server and the Cisco Unified Communications Manager Service.

3. Set the Send H225 User Info Message setting to **H225 Info for Call Progress Tone**.

4. Save your settings.

Configuring a Cisco Unified Communications Manager Server

From the Device Management menu, Communications Manager option, you can add a Unified CM Server to the Operations Console. Once added, you can add the Unified CM Server to a device pool and access a Unified CM administration web page, from which you can configure the Unified CM Server.

Unified CM manages and switches VoIP calls among IP phones. When combined with Unified ICME, Unified CM becomes the IPCC product. Unified CVP interacts with Unified CM primarily as a means for sending PSTN-originated calls to IPCC agents. However, several applications require that calls be originated by IPCC agents instead. Specifically, IPCC Outbound Option and calls that are being handled using the warm-consultative-transfer feature, from one agent to another are originated in this way. Help desk calls, in which an agent or other IP phone user calls Unified CVP (or calls a skill group and gets queued on Unified CVP), also fall into this category. A single Unified CM can originate and receive calls from either SIP or H.323 devices.

Refer to the Operations Console online help topics under *Managing Devices > Configuring a Unified Cm Server* for the following topics:

- Adding a Unified CM Server
- Editing a Unified CM Server
- Unified CM Configuration Settings
- Deleting a Unified CM Server
- Finding a Unified CM Server

Before You Begin

Configure the following SIP-specific actions on Unified CM. (Refer to the Unified CM documentation for more information.)

Complete the following tasks:

- Configure the SIP Proxy Server.
- Configure the SIP Trunk to the Proxy Server or the Unified CVP Call Server, if you are not using a SIP Proxy.
- Add call routing (route patterns) to send the call from Unified CM; for example, ring tone, playback dial patterns, and ICM route table calls.

Chapter 11

Configuring the SIP Devices

This chapter describes how to configure SIP calls on the gateway, Call Server, and Proxy Server.

This chapter contains the following topics:

Configuring a SIP Proxy Server

From the Operations Console *Device Management* menu, **SIP Proxy Server** option, you can add a pre-configured SIP Proxy Server. You must configure the SIP Proxy server before adding it to the Operations Console. Once added, you can add the SIP Proxy Server to a device pool. You can also configure a link to the administration web page for the SIP Proxy Server so that you can access that page from the Operations Console.

A SIP Proxy Server is a device that routes individual SIP transport messages among SIP endpoints. It plays a key role in high availability in a Unified CVP deployment for call switching. It is designed to support multiple SIP endpoints of various types, and implements load balancing and failover among those endpoints. SIP Proxy Servers can be deployed alone or as a pair. Also, smaller Unified CVP deployments can run without a SIP Proxy Server. In such cases, the Unified CVP SIP service assumes some of those functions because it provides the ability to configure a static table to look up destinations.

Before You Begin

Configure the following characteristics of the SIP Proxy Server for use with Unified CVP:

- A static route to the Unified CVP Call Server, Unified CM SIP trunks, VoiceXML gateway, and Ingress gateway for the transfer to the ringtone playback dialed number and error playback dialed number.

  Note: You must configure a Unified CM SIP trunk on Unified CM to point to the SIP Proxy Server. In a cluster with multi-subscribers and device pools, adding only one SIP trunk per proxy in the default device pool will be sufficient.

- Incoming Access Control Lists (ACLs) for Unified CVP calls.

Unified CVP works with RFC-3261-compliant SIP Proxy Servers and has been qualified with the following:

- Unified CVP works with the CUP server, not the presence engine (PE).

Refer to the Operations Console online help topics under *Managing Devices > Configuring a SIP Proxy Server* for details about performing the following tasks:

- Adding a SIP Proxy Server

  This topic also describes the SIP Proxy Server configuration settings.

- Deleting a SIP Proxy Server

- Editing a SIP Proxy Server

- Finding a SIP Proxy Server

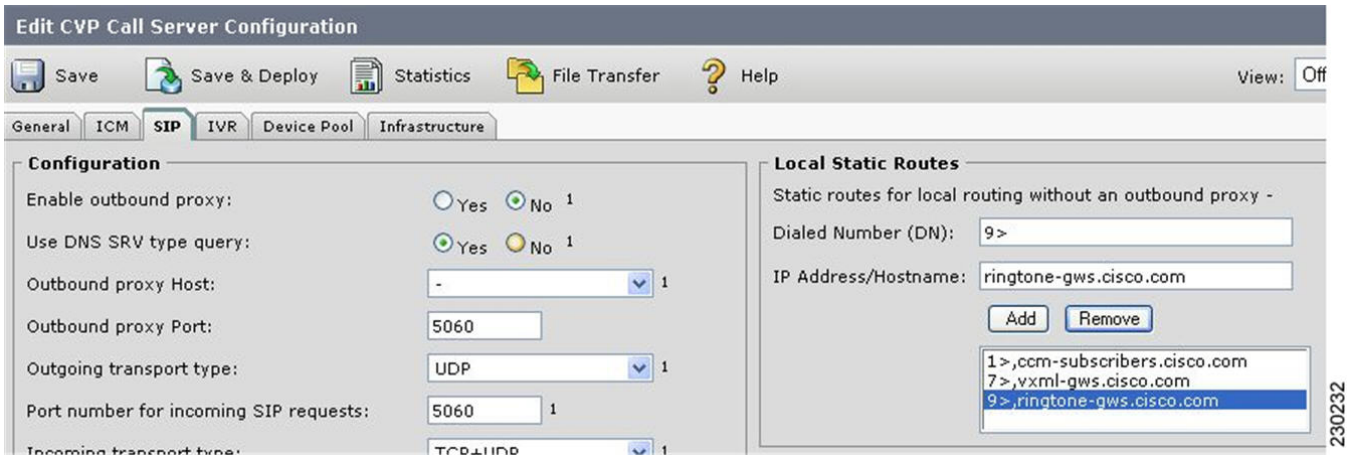
Load-balancing SIP Calls

SIP calls can be load balanced across destinations in several different ways as outlined below:

- Using the CUP server, define several static routes with the same route pattern and priorities and weights.
- Using DNS, configure SRV records with priorities and weights. Both the DNS client and the server settings must be configured and operating successfully for DNS "A" and "SRV" type queries to work. Configure SRV queries to be used wherever outbound SIP calls are made, such as on the IOS Ingress gateway, on the Call Server itself, and on Unified CM.

Note: Refer to DNS Zone File Configuration (page 59) for information about load balancing and failover without a Proxy Server. Only the DNS SRV method is supported for load balancing and failover without a Proxy Server. The following screen shot example displays this configuration in the Operations Console:

Figure 40: DNS SRV Method for Load Balancing and Failover without a Proxy Server



How to Set Up the Ingress Gateway to Use Redundant Proxy Servers

Step 1 Configure the gateway with the following to send calls to redundant proxy servers as resolved using DNS SRV lookup:

```
ip domain name <your domain name>
ip name-server <your DNS server>
sip-ua
sip-server dns:<your SRV cluster domain name>
dial-peer voice 1000 voip
session target sip-server
```

How to Set Up the Unified CVP Call Server with Redundant Proxy Servers

You may use redundant proxy servers for CVP outbound calls by using a DNS-based SRV cluster name or a non-DNS SRV cluster name (also known as Server Group Name).

Refer to the Operations Console online help topic **Using SIP Server Groups > Configuring Server Groups** for information on how to configure local based SRV records.

The SRV cluster name should be configured to be accepted by the proxy server or else the call will be rejected. In the CUP Server, the SRV domain name should be configured in the service parameters section.

Note: The CUSP proxy server (blade proxy) does not require this configuration.

Cisco Unified SIP Proxy (CUSP) Configuration

The following configuration shows a CUSP proxy in Unified CVP. The highlighted lines are specific to a Unified CVP solution. For additional configuration details refer to the **Configuring Cisco Unified SIP Proxy Server** (http://www.cisco.com/en/US/docs/voice_ip_comm/sip/proxies/1.0/administration/guide/config.html) guide.

Configuration Example:

```
server-group sip global-load-balance call-id
server-group sip retry-after 0
server-group sip element-retries udp 1
server-group sip element-retries tls 1
server-group sip element-retries tcp 1
sip dns-srv
no enable
no naptr
end dns
!
no sip header-compaction
no sip logging
!
sip max-forwards 70
sip network netA noicmp
non-invite-provisional 200
allow-connections
retransmit-count invite-server-transaction 9
retransmit-count non-invite-client-transaction 9
retransmit-count invite-client-transaction 2
retransmit-timer T4 5000
retransmit-timer T2 4000
retransmit-timer T1 500
retransmit-timer TU2 32000
retransmit-timer TU1 5000
retransmit-timer clientTn 64000
retransmit-timer serverTn 64000
```

```
end network
!
no sip peg-counting
!
sip privacy service
sip queue message
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue radius
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue request
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue response
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue st-callback
drop-policy head
low-threshold 80
size 2000
thread-count 10
end queue
!
sip queue timer
drop-policy none
low-threshold 80
size 2500
thread-count 8
end queue
!
sip queue xcl
drop-policy head
low-threshold 80
size 2000
thread-count 2
end queue
!
route recursion
!
```

```
sip tcp connection-timeout 240
sip tcp max-connections 256
!
no sip tls
!
trigger condition in-netA
sequence 1
in-network netA
end sequence
end trigger condition
!
trigger condition mid-dialog
sequence 1
mid-dialog
end sequence
end trigger condition
!
trigger condition out-netA
sequence 1
out-network netA
end sequence
end trigger condition
!
accounting
no enable
no client-side
no server-side
end accounting
!
server-group sip group cucm-cluster.cisco.com netA
element ip-address 10.86.129.219 5060 udp q-value 1.0 weight 10
element ip-address 10.86.129.62 5060 udp q-value 1.0 weight 10
element ip-address 10.86.129.63 5060 udp q-value 1.0 weight 10
failover-resp-codes 503
lbtype global
ping
end server-group
!
server-group sip group cvp-call-servers.cisco.com netA
element ip-address 10.86.129.220 5060 udp q-value 1.0 weight 10
element ip-address 10.86.129.224 5060 udp q-value 0.9 weight 10
failover-resp-codes 503
lbtype global
ping
end server-group
!
server-group sip group vxml-gws.cisco.com netA
element ip-address 10.86.129.229 5060 udp q-value 1.0 weight 10
element ip-address 10.86.129.228 5060 udp q-value 1.0 weight 10
failover-resp-codes 503
lbtype global
ping
end server-group
!
route table cvp-route-table
key 9 target-destination vxml-gws.cisco.com netA
```

```
key 8 target-destination cvp-call-servers.cisco.com netA
key 7 target-destination vxml-gws.cisco.com netA
key 700699 target-destination cvp-call-servers.cisco.com netA
key 2 target-destination cucm-cluster.cisco.com netA
key 1 target-destination cucm-cluster.cisco.com netA
key 7000 target-destination 172.19.151.41 netA
key 777333 target-destination cvp-call-servers.cisco.com netA
key 1004 target-destination 10.86.139.84 netA
key 7105 target-destination dialer-gws netA
end route table
!
policy lookup cvp-policy
sequence 1 cvp-route-table request-uri uri-component user
rule prefix
end sequence
end policy
!
trigger routing sequence 1 by-pass condition mid-dialog
trigger routing sequence 10 policy cvp-policy condition in-netA
!
server-group sip ping-options netA 10.86.129.200 5038
method OPTIONS
ping-type adaptive 5000 10000
timeout 500
end ping
!
server-group sip global-ping
sip listen netA udp 10.86.129.200 5060
!
end
```

REFER Transfers Using "RFXXXX" Type Unified ICME Labels

Unified CVP SIP Service can perform a SIP REFER transfer instead of using SIP re-invites, which allows Unified CVP to remove itself from the call, thus freeing up licensed Unfied CVP ports. (Unified CVP cannot execute further call control operations after this kind of label has been executed; for example, it cannot perform subsequent transfers back to Unified CVP for self service or queuing to another agent.)

However, if the transfer fails, configure survivability to transfer the call elsewhere. This process is not the same as an ICM router requery; for example, as it will appear as a new call to Unified ICME, but it is a way to take an alternate action, if the transfer fails.

Note:

- This feature can be used in Comprehensive (SIP only), Call Director, and Standalone call flow models.
- Router requery can be performed with a REFER transfer *only* if the NOTIFY messages are sent back to Unified CVP with the result of the REFER operation. Since Unified CVP does not hang up the call after sending REFER, it is possible to requery Unified ICM and get another label, and send another REFER.

- The use of the survivability tcl service on the ingress gateway cannot currently support sending the NOTIFY messages with a failed transfer result, so router requery cannot be used with REFER when it is handled by the survivability service. The Survivability service can handle REFER, except that it will always report a successful transfer to Unified CVP, even when the transfer failed. This is a known limitation of the TCL IVR API for REFER handling in IOS, including ingress and CUBE gateways.

- The use of the survivability tcl service on the ingress gateway cannot currently support sending the NOTIFY messages with a failed transfer result, so router requery cannot be used with REFER when it is handled by the survivability service. Survivability service can handle REFER, except that it will always report a successful transfer to Unified CVP, even when the transfer failed. This is a known limitation of the TCL IVR API for REFER handling in IOS, including ingress and CUBE gateways.

Using this feature, the call can be queued at the VoiceXML gateway and then sent to an agent with a Unified ICME label that begins with the letters "rf." Otherwise, standard Unified ICME agent labels enable Unified CVP to remain in the signaling path for the duration of the call, and the licensed Unified CVP resource will not be freed until the end of the call. REFER transfers can be made to Unified CM or other SIP endpoints in the SIP cloud, such as anACD. The ECC variable "user.sip.refertransfer" can also be set in Unified ICME scripts. (When using this ECC variable in an Unified ICME script, it must be set to the value of the single character "y" and Unified CVP will use REFERs when transferring to the agents.)

When using REFER transfers, including the REFER used to play back critical_error.wav for abnormal disconnects, the Ingress gateway must include an outbound voip dial peer. This outbound dial peer is necessary because when the REFER message enters the gateway from theUnified CVP Call Server, it needs to match an outbound dial peer in order for the call to succeed; otherwise, a 503 rejection occurs if no dial peers match the REFER-TO header URI. Dial peer destination targets must match the labels in the REFER-TO SIP URI; that is, <errorDN>@<sip-server> and other labels that might be used in the Unified ICME routing label. For example:

```
dial-peer voice 1050 voip destination-pattern
1... voice-class codec 1 session protocol sipv2 session target <your
sip-server destination> dtmf-relay rtp-nte h245-signal h245-alphanumeric
 no
vad
```

When configuring Route Patterns on Unified CM for REFERs to destinations outside of the cluster, such as to the CUP Server or the gateways directly, you must add **SIP Route Pattern** for the SIP Trunk associated with that endpoint. For example, if you use REFER to Error DN to the IP Originated caller on Cisco Unified Communications Manager, and the host of the REFER To header SIP URL is the CUP Server, you must create a SIP Route Pattern with that IP address or domain name and associate it with your SIP Trunk for the CUP Server.

Note:

- When a TDM gateway handles REFER, and not Cisco Unified Border Element (CUBE), a REFER triggered INVITE is sent out. The REFER triggered INVITE requires a dial peer with a session target and typical codec information. The Refer-To header URI host that was formulated by CVP routing algorithm configuration is ignored.

- When CUBE receives a CVP initiated REFER, it does not send it transparently through to the originator. A dial peer is required to match the DN (user portion of the Refer-To header URI) and the host portion of the URI is rewritten to match the session target of the dial peer. The REFER is passed to the originator using cli "supplementary-service sip refer", otherwise CUBE will handle the REFER and send the triggered invite to the refer DN on its own as a back to back user agent.

Configuring Custom Streaming Ringtones

You can configure custom ringtone patterns that will enable you to play an audio stream to a caller in place of usual ringtone. Customized streaming ringtones are based on the dialed number destination and, when configured, play an in-progress broadcast stream to the caller while the call is transferred an agent.

Procedure

To customize ringtone patterns for streaming audio:

Step 1 Configure Helix for streaming audio.

The default installation and configuration of Helix server is all that is required for use with Unified CVP. Refer to *Helix Server Administration Guide* for information about installing and configuring Helix Server.

Step 2 In the Operations Console, perform the following steps to configure custom streaming ringtones:

a. Select **System** > **Dialed Number Pattern**.

b. Click **Add New**.

c. Complete the following fields to assoicate a dialed number pattern with a custom ringtone.

Table 42: Dialed Number Pattern Configuration Settings

Property	Description	Default	Value	
General Configuration				
Dialed Number Pattern	The actual Dialed Number Pattern.	None	Must be unique Maximum length of 24 characters Can contain alphanumeric characters,	

Property	Description	Default	Value
			wildcard characters such as exclamation point (!) or asterisk (*),single digit matches such as the letter X or period (.) Can end with an optional greater than (>) wildcard character
Description	Information about the Dialed Number Pattern.	None	Maximum length of 1024 characters
Enable Custom Ringtone	Enables customized ring tone. • **Ringtone media filename -** Enter the name of the file that is to be played for the respective dialed number pattern. Provide the URL for the stream name in the following format: **`rtsp://<streaming server IP address>:<port>/<directory>/<filename>.rm`**	Disabled none	Maximum length of 256 characters Cannot contain whitespace characters

d. Click **Save** to save the Dialed Number Pattern.

You are returned to the **Dialed Number Pattern** page. To deploy the Dialed Number Pattern configuration, click **Deploy** to deploy the configuration to all Unified CVP Call Server devices.

e. Access the IOS device in global configuration mode and add the following commands on your VXML Gateway:

```
rtsp client timeout 10

rtsp message timeout 10
```

The range is 1 to 20; the recommended value is 10 seconds.

Step 3 Add a Send to VRU node in your ICM script before any Queue node.

The explicit Send to VRU node is used to establish theVRU leg before the transfer to the agent; this is required to play streaming audio ringtones to a caller.

See Also

How to Configure the Play Media Micro-Application to Use Streaming Audio (page 164)

Using the Helix Server (page 553)

Using the sendtooriginator Setting in the SIP Service

Note:

- The information in this section is applicable to *all* Unified CVP call flow models.
- The setting on the IOS gateway for "signaling forward unconditional" is only required if ISDN call variables need to be available in the Unified ICME scripting environment. If these call variables are not required, then omitting this setting is acceptable.The setting makes the SIP INVITE message larger in terms of bytes due to the extra payload in the message body for GTD variables. If the packet size is significantly greater than 1300 bytes, then TCP transport might be recommended over UDP transport due to the possibility of a network "fragmentation" of messages. Refer to the Operations Console online help for more information.
- If the pattern matches the label returned from ICM, the call is routed to the originating host derived from the incoming calls remote party ID header or contact header.
- The call will be sent to the origination gateway if the following is true: the remote party ID header is present on the incoming SIP invite; the user agent header of the INVITE indicates an IOS gateway; and the pattern matcher on the label is configured for send-to-origin.

For the Unified CVP Branch call flow model, incoming calls into the Unified CVP Call Server from the gateway might be automatically routed back to the originating gateway at the branch using the"sendtooriginator" setting in the SIP Service of the Call Server. This setting overrides sending the call to the outbound proxy or to any locally configured static routes on Unified CVP. If the label returned from Unified ICME for the Unified CVP transfer matches one of the configured patterns under the Unified CVP "sendtoorginator" settings, then the call will be routed to the SIP URL of "sip:<label>@<host portion from header of incoming invite>." This functionality is similar to the Unified CVP H.323 "SetTransferLabel" setting for the H.323 calls.

100rel and SIP Outbound Dialer

100rel is not supported in Unified CVP, but it is required for the SIP Outbound Dialer product. If the gateway is being used for calls on both Unified CVP and the SIP Outbound Dialer, then the rel1xx setting under the global **`voice service voip`** must not be set to **`rel1xx`**

`required`, which is not supported in Unified CVP, or **`rel1xx disabled`**, which is not supported with SIP Outbound Dialer. The rel1xx settings must be set on the level of the dial peers, not at the global level. All the examples in this document show the rel1xx setting at the dial-peer level.

Expiration Timeout Setting Using Dialed Number Patterns in SIP

The Unified CVP SIP Service can be configured with Dialed Number patterns to set the expiration timeout of an outbound call for a Unified ICME label. Longer or shorter times can be configured before the call is cancelled with no pickup (ring no answer timeout). Use the **DN Pattern Outbound Invite Timeout** option in the Operation Console's SIP Service configuration tab to add the expiration timeout for a particular dialed number pattern. For example, you can configure Unified ICME labels to Cisco Unified Communications Manager agents with shorter timeouts to perform a router requery, if desired.

Note: It is possible IOS will timeout before INVITE number from Unified CVP (to Unified CM, gateway, etc.) As a practical setup, set the gateway sip-ua expires timer to higher than the highest setting on the Unified CVP configuration. The *minimum* ring timeout is 60 seconds on the IOS side. Unified CVP can set a minimum ring timeout on its outbound calls as low as necessary.

Critical Error Message Playback on Abnormal Disconnects with SIP

Unified CVP is designed to play a critical error message to the caller when a call rejection occurs. The critical error message's playback works differently for SIP calls than it does for H.323 calls because the Call Server will not perform media termination by itself, as the H.323 Service does. Therefore, the SIP Service in the Call Server uses the SIP REFER method to refer the calls to the IOS gateway "cvp-error" TCL scripted service for message playback.The purpose of this is to define the ERROR DN (default 92929292) on the SIP Service of the Call Server in the Operations Console.

Note:

- Critical Error messages will not be played if REFER blind transfers fail since the Unified CVP SIP Service is not longer holding onto the call; it cannot take the call back and refer to the error DN. (This also includes 302 redirect responses.)
- Anything other than a code 200 (SIP 200 OK) or a 16 (Q.850;cause=16) is considered an abnormal disconnect. Error message playback only occurs if Unified CVP disconnects the caller abnormally. If the disconnect arrives on Unified CVP on the caller leg from the ingress gateway, it appears that the caller is disconnecting so Unified CVP does not count this as an abnormal disconnect even though the Unified CVP survivability script on the ingress gateway might detect this event.

How Critical Error Message Playback Works

Critical Error Message playback works as follows:

- *For typical TDM calls using the survivability service on the IOS Gateway*, the error message is played using the survivability script when it detects the non-normal disconnect code after Unified CVP has disconnected the call (refer to SIP BYE message). The display name of "--CVP" is appended in a Remote Party ID header of the SIP call. This is how the SIP Service can identify whether or not the incoming call uses survivability. For example, IP-originated Cisco Unified Communications Manager callers will not have this setting. All TDM callers must have survivability service.

- *For IP-originated calls that the SIP Service detects is not using the survivability service*, when the call is disconnected and the reason code is abnormal, Unified CVP sends a REFER message with the SIP URL of the error DN. The error DN must be configured in the dial plan to point to an IOS gateway that has the cvp-error TCL service configured. When the caller gets the REFER TO destination, it attempts to blind transfer the call to the Unified CVP Error service on the gateway. Once Unified CVP sends a REFER for error message playback, it cannot retrieve the call, so it is important to confirm this mechanism is configured correctly during deployment. For any call that Unified CVP sees is not using the survivability service, it will attempt the REFER method to the error DN for critical error message playback for abnormal disconnects.

- For third party SIP gateways, where the survivability service is not running, and you desire to avoid the use of the SIP REFER to error service, it may be turned off in the SIP section of the Call Server configuration. Turning off this property allows Unified CVP to pass through the rejection response as-is to the caller, or else the BYE with reason header will be passed, depending on the call state. It may be that the SIP gateway does not implement SIP REFER and you desire to turn this setting off.

Delivering SIP Incoming UUI to Unified ICME Routing Script

For SIP Calls, ISDN call parameters such as "user to user information" (also known as "UUI" or "UUS") can be passed to the Unified ICME script. (ISDN call data is only forwarded in SIP calls from the TDM calls using a gateway.) On the gateway, the "signaling forward unconditional" setting is required on the "voice service voip" section of the IOS configuration. This allows forwarding of the Generic Type Descriptor (GTD) data of the call in the SIP INVITE to Unified CVP in a multipart mime content type media format.

The Unified CVP SIP Service extracts the GTD section of the call and parses out the parameters that are configured to be passed to Unified ICME in the payload field of the ICM Service's "NEW CALL" message "usertouserinfo." ISDN call parameters, which are forwarded in the GTD payload, as set in the Unified ICME script, cannot be sent in outbound SIP calls. In addition to sending ISDN data inbound to Unified ICME, the GTD section of the inbound call leg will be passed along untouched in the outbound call leg by the Unified CVP SIP Service.

Note:

- The SIP Service only reads the data and sends it to Unified ICME (it does not modify the data).

- **Session Timer RFC 4028 support in Unified CVP.** Unified CVP, acting as a Back-to-Back User Agent (b2bua) will transparently pass thru the headers for Supported, Session Expires,

and Min-SE on the call legs. However, since the IOS gateway does not set a Session Expires header by itself, it will not request session refreshing. By default, Unified CVP adds a *Session-Expires: 1800;refresher=uac* value to the 200OK response on the initial invite sent into Unified CVP by the caller, if the Session Expires header is not already present in the 200OK of the outbound leg. This value will cause the gateway, as UAC, to send the refesher invites in mid call every 1800 seconds. This configuration allows the gateway to clean up a zombie call, due to a failed refresher reinvite, if the Unified Call Server is taken down in mid call. The setting on Unified CVP to disable adding a Session Expires header is configurable.

How UUI is Delivered

Assume the following ISDN data is sent in the call to Unified CVP:

```
PRN,isdn*,,ATT5*,
USI,rate,c,s,c,1
USI,lay1,ulaw
TMR,00
CPN,00,,u,5900
CPC,09
FCI,,,,,,,y,
UUS,3,3132333435
GCI,87c0c79d91dd11daa9c4000bfda207f2
```

Note: By default, UUS field data is converted from 2-byte hex representation to 1-byte ASCII; for example, the UUS value would be"303132333530" in 2-byte hex representation; in 1-byte ASCII, the value would be "012350."

The **Generic Type Descriptor (GTD) Parameter Forwarding** configuration setting in the Operations Console is used for passing GTD (UUI) data to Unified ICME in a new call. (The default is UUS.) Additionally, other parameters in the GTD can also be extracted and sent to Unified ICME. (UUS,PRN,GCI) use comma-separated values. Any parameter contained in the NSS IAM message can be extracted. (Refer to the ITU-T Narrowband Signaling Syntax spec(Q.1980.1).)

Configure the SIP Service by selecting **Device Management > CVP Call Server > SIP tab** in the Operations Console and make an entry for the **Generic Type Descriptor (GTD) Parameter Forwarding** with the string to forward using the parameter names delimited by commas. For example, configure the SIP Service with the string **`UUS,PRN,GCI`** to forward these three parameters in a concatenated string to Unified ICME. The Unified ICME script needs to retrieve the call variable on the route request message called "UsertoUserInfo," and parse out the needed information.

G.722 Codec Support

Starting in Release 8.0(1), Unified CVP supports the use of the G.722 codec. G.722 is an ITU standard codec that offers a significant improvement in speech quality over older narrowband-codecs, such as G.711, without an excessive increase in implementation complexity, but with an increase in the bandwidth required. G.722 renders a more natural sounding human voice than G.711, in part because it uses a higher sampling rate than G.711. It is useful in

fixed-network VOIP applications where ever its greater required bandwidth does not make it prohibitive.

G.722 enables bitrates of 64, 56 and 48 kbits/s; however, only the 64 kbits/s bitrate is supported.

Note:

- The Unified CVP IVR service does not support G.722. Unified CVP endpoints that connect with the VXML gateway still require support of G.711.
- There is no transcoding support between G.722 and G.711, but media server .WAV files do not need to be converted to G.722 because renegotiation takes place automatically when necessary.

G.722 Example Configuration

This task details an example configuration for using G.722 codec in the Unified CVP environment.

No configuration changes are required on Unified CVP for G.722 support. However, you must configure Cisco Unified Communications Manager and your Gateway(s) to support G.722.

Step 1 Configuration for Cisco Unified Communications Manager.

On Unified Communications Manager:

a. Select: **System > Service Parameters**

b. Select the Server, then select: **Cisco CallManager Service**

c. Scroll down to **Clusterwide Parameters (System - Location and Region)** and verify **G722 Codec Enabled** is set to **Enabled for All Devices**.

d. Select: **System > Enterprise Parameters**

e. Verify **Advertise G.722 Codec** is set to **Enabled**, then click **Save**.

f. Select: **System > Region**

g. Click **Find** to list the regions defined. For each region that should support G.722, select the region and verify that **Audio Codec** is set to **Use System Default**.

h. Click: **Save**

Step 2 Configuration on your TDM Ingress gateway(s).

a. For voice class code x (where x is some number), add g722-64 as the preferred codec and add g711ulaw as a secondary codec.

Example:

```
 voice class codec 1
  codec preference 1 g722-64
  codec preference 2 g711ulaw
```

b. On the voip dial peer(s), add the voice class you configured for G.722

Example:

```
 dial-peer voice XXXXX voip
 description CVP SIP Comprehensive dial-peer
  destination-pattern XXXXXXXX
  session target XXXXXXX
  session protocol sipv2
  dtmf-relay rtp-nte h245-signal h245-alphanumeric
  voice-class codec 1
  no vad
!
dial-peer voice XXXXXXX pots
 description CVP TDM dial-peer
 service survivability
```

KPML Support

Starting in release 8.0(1), Unified CVP supports Key Press Markup Language (KPML) for SIP. KPML is a SIP feature that enables monitoring of DTMF signals "out-of-band" (OOB), as opposed to the (RFC2833) in-band DTMF monitoring method that Unified CVP typically uses.

Not all endpoints (such as CTI Ports) support in-band DTMF monitoring. For these endpoints, the "DTMF Signaling Method" on the SIP Trunk can be set with RFC2833. In this case, the Unified Communications Manager must allocate an MTP resource to translate DTMF from the in-band packets to the OOB signaling for the end points that do not support in-band DTMF.

Using KPML enables you to avoid having to set RFC2833 on the SIP trunk (and thus having Unified CM dynamically allocate an MTP resource due to the DTMF mismatch on endpoints). Instead, you set the DTMF Signaling Method to no preference on the SIP trunk and set the bootstrap dial-peer on the gateway to use SIP and KPML.

There are a number of design considerations for using KPML. Refer to **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) guide for details.

KPML Example Configuration

This task details an example configuration for KPML.

Step 1 On Unified Communications Manager:

a. On Unified CM Administration, select **Device > Trunk**, then click **Find** to list the trunks.

b. Select the SIP trunk that you want to configure for KPML.

c. Scroll down to the **SIP Information** section and set **DTMF Signaling Method** to **No Preference**, then click **Save**.

d. Associate the routing pattern for the IP calls with the SIP trunk you modified above.

Step 2 On the VXML Gateway, for the dial-peer:

a. Add: **session protocol sipv2**

b. Add: **dtmf-relay sip-kpml**

VXML Gateway Configuration Example:

```
dial-peer voice 777 voip
...
 service bootstrap
 session protocol sipv2

 dtmf-relay sip-kpml
```

c. Remove the DTMF information on the ringtone and cvperror DNs.

Step 3 On the Ingress Gateway, for the dial-peer:

a. Add: **session protocol sipv2**

b. Add: **dtmf-relay sip-kpml**

Ingress Gateway Configuration Example

```
dial-peer voice 5001 voip
 destination-pattern 5001
 voice-class codec 1
 session protocol sipv2
 session target sip-server
 dtmf-relay sip-kpml
 no vad
 !
```

Step 4 In the Unified ICM routing script, add a set variable node before the Get Digit (GD) microapp with the value: **`user.microapp.input_type = "D"`**.

SIP Hookflash

Hookflash is a Signaling mechanism typically associated with aTDM PBX or ACD. Hookflash sends a quick off-hook/on-hook/off-hook signal to the PBX or ACD. This signaling causes the voice gateway to send a string of routing digits to the PBX or ACD. Upon collection of the routing digits, the PBX orACD transfers the caller to the new termination, such as another agent or an ACD queue or service on that same PBX or ACD.

Starting in release 8.0(1), Unified CVP supports using hookflash with SIP. Using this feature you can transfer SIP calls using a hookflash followed by the DTMF destination. This is useful for deployments in which a PBX "front-ends" the Unified CVP ingress gateway, and in which the PBX provides non-VOIP connectivity to agents.

In a typical scenario, a caller calls into a system and is transferred to an agent who is associated with a non-Cisco ACD. Unified ICM returns a label to Unified CVP to perform a hookflash transfer to the PSTN so that the caller can be routed to the correct agent (the label returned must have "HF" prepended to the hookflash routing digits). The caller is then transferred to the agent and Unified CVP is no longer in control of the call.

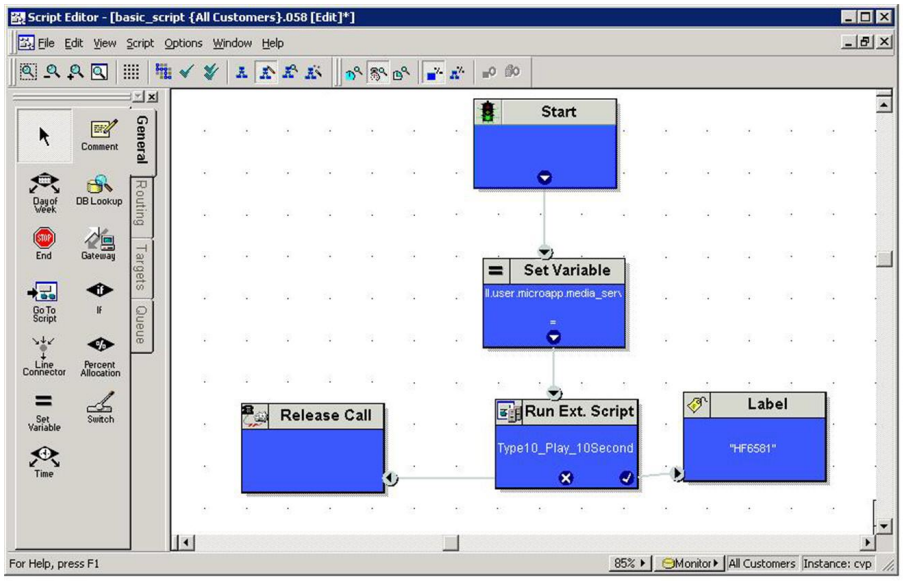
SIP Hookflash Configuration

To configure your system to use SIP hookflash:

The gateway T1 controller must be configured for hook flash relay. Refer to "Configuring Hookflash Relay (page 435)."

Step 1 On Unified ICM, in your script, use a label node such as "HF,5551212", or, in the following example, HF6581.

Figure 41: SIP Hookflash Configuration Script



Unified CVP automatically disconnects the call two seconds after sending the digits. If the switch needs more time to complete the hook flash sequence, increase the delay by appending more commas after the transfer number to extend the two-second timeout.

If the switch requires a pause after the hookflash, insert commas between the HF and the transfer number. Each comma represents 100 milliseconds. For example, to use the hook flash feature to transfer the call to the number "5551212" with a 500- millisecond pause after the hookflash, configure the Unified ICM label as "HF,,,,,5551212".

Step 2 On your ingress and VXML gateways configure the survivability service on your POTS dial-peers. Refer to Call Survivability (page 398).

For example:

```
 service cvp-survivability flash:survivability.tcl
  param icm-hf 1
 !

dial-peer voice 800555.... pots
 destination-pattern 800555....
 service cvp-survivability
! etc...
```

```
ip rtcp report interval 2000
gateway
media-inactivity-criteria all
timer receive-rtcp 4
timer receive-rtp 1200
sip-ua
retry bye 2
!
```

End of the media code example.

Step 3 Configure your gateway T1 controller for hookflash relay. Refer to Configuring Hookflash Relay (page 435).

For example, for an analog configuration (assume incoming number is 6708):

```
voice-port 1/0/1
   signal groundStart
   connection plar 6708


dial-peer voice 6708 pots
   service cvp-survivability
   incoming called-number 6708
   direct-inward-dial
   port 1/0/1
   forward-digits 0
```

For a digital configuration (assume incoming number is 6708):

```
controller T1 2/0
   framing esf
   linecode b8zs
   ds0-group 1 timeslots 1-24 type fxo-loop-start
   voice-port 2/0:1
   output attenuation 0
   connection plar 6708


dial-peer voice 6708 pots
   service cvp-survivability
   incoming called-number 6708
   direct-inward-dial
   port 2/0:1
   forward-digits 0
```

Passing Information with User-to-User Information (UUI)

You can pass generic data to and from Unified ICM through Unified CVP using User-to-User Information (UUI). For example, it is sometimes desirable to capture data from an external

system (such as caller-entered digits from a third-party IVR) and pass that data to Unified ICM on a new call.

This is accomplished by populating the UUS parameter (often known as the UUI) in the Initial Address Message (IAM) of the Generic Transparency Descriptor (GTD) data that is sent to the gateway from the network in the Q.931 setup message. The gateway sends this data to Unified CVP through SIP messages. Unified CVP can then send the data to Unified ICM on a new call.

UUI can also be set by ICM scripts and extracted by Unified CVP to be resent in SIP messages.

UUI processing scenarios:

- When GTD data is present in the inbound call leg of the SIP INVITE message, Unified CVP saves the GTD data as *inbound GTD* and the UUI portion (if present) is passed to Unified ICM.

  If Unified ICM modifies the data, it sends the modified UUI back to Unified CVP. Unified CVP converts the UUI data it receives from Unified ICM into Hex and modifies the UUS (if it is present) and overwrites the *inbound GTD* value. Only the UUS portion will be modified, using the format:

  `UUS,3,<converted Hex value of data from ICM>`

  The rest of the GTD parameter values are preserved, keeping the values as they arrived from caller GTD.

- When GTD is not present in the inbound call leg, Unified CVP prints an informational message on the trace stating *No GTD Body present in Caller Body* and the call continues as a regular call.

  The modified UUI from Unified ICM is passed using the ICM variable **`Call.UsertoUserInfo`**.

- REFER and 302 Redirects and UUI.

  If UUI is set in the Unified ICME script, and a refer call flow is being used, then the UUI will be placed in a mime body and hex encoded according to aATT IP Toll Free NSS format. This applies to 302 redirect responses as well.

```
VER,1.00
PRN,t1113,*,att**,1993
FAC,
UUS,0,(hex encoded UUI string here)
```

When sending GTD, if a regular label is received from Unified ICM, then the modified GTD from the scenarios above is passed on the outbound INVITE from Unified CVP.

If a DTMF label for outpulse transfer is received on a connected call, then the BYE is sent with the following GTD only if UUI is passed by Unified ICM. The BYE is sent immediately after the SIP INFO with DTMF.

```
Content-Type: application/gtd
Content-Disposition: signal;handling=optional
REL,
PRN,isdn*,,NI***,
UUS,3,<converted Hex value of UUI from ICM>
```

Note:

- Unified CVP does not send BYE on the DTMF label if UUI is not received from Unified ICM. If a BYE message is received, then the GTD from the received BYE is used to send it on the other leg.
- The Ingress gateway must be configured with "**`signaling forward unconditional`**", for example:

```
!
voice service voip
 signaling forward unconditional
```

Passing Information with SIP Headers

Unified CVP enables the passing of one or more SIP headers to Unified ICM for use within ICM scripts. You configure which headers are passed to Unified CVP through the Operations Console, on the Call Server SIP tab. You can also modify SIP headers in the Unified ICM script and pass these back to Unified CVP.

When sending SIP headers to Unified ICM, in the Operations Console, you can specify a specific header, or a header and specific parameters within that header. For example:

- **Header:** *Supported*
- **Header:** *Remote-Party-ID* **Parameter:** *privacy*
- **Header:** *Call-Info* **Parameter:** *purpose*

These SIP headers are passed to Unified ICM in the **SIPHeaderInfo** field of New Call and Request Instruction messages. To access the SIP Header variable in the ICM script, read the **Call.SIPHeader** call variable.

The amount of space available to send header data to Unified ICM is limited to 255 bytes. The SIP protocol RFC provides a mechanism to represent common header field names in an abbreviated form. The compact header format, as defined in **RFC 3261** (http://www.ietf.org/rfc/rfc3261.txt) and other RFCs for newly defined headers, is used for the header titles before passing the header to Unified ICM.

Note: Not all headers have a compact format. For example, P-Headers (private headers, such as **P-Asserted-Identity**) may not have a compact form. In this case, Unified CVP passes the full header name to Unified ICM.

You can also write to headers using the ICM script and pass the modified headers back to Unified CVP. This feature provides a scriptable option to modify SIP headers on the outgoing Unified

CVP transfer. You can specify SIP header values in outgoing SIP INVITES (only the initial INVITE, not reinvites). You can add, modify or remove header values. Changes to the INVITE are applied just before it is sent out.

Writing to **Call.SIPHeader** using the Unified ICM script causes Unified CVP to use that data in outbound SIP calls to IVR or Agents, or in REFER or 302 redirect messages.

Unified CVP uses the following format to send headers to ICM:

```
"f:Name <sip:from@127.0.0.1:6666>;param1;param2|e:tar|v:SIP/2.0/UDP
viaHost"
```

Note: In the example above, each header field is separated by the vertical var character (|).

Unified CVP provides the flexibility to add/modify/remove outgoing SIP header in the INVITE message only. This enables you to deploy Unified CVP in many scenarios to facilitate inter-op with third party devices.

For example, if there are inter-op problems in your call flows, you may not be able to enable inter-working with the default SIP signaling. Passing information using SIP headers enables Unified CVP to provide a work-around for inter-op issues.

Warning: SIP header modification feature is a powerful tool which can tweak SIP headers as needed. You should exercise caution when applying SIP Profiles and ensure that the profiles do not "create" interoperability issues, rather than solving them.

Warning: The outgoing header modification feature enables you to remove, modify, or add any SIP header. These include SIP headers such as To, From, Via, CSeq, Call-Id, and Max-Forwards. However, there is no validation for your changes in the ICM script editor interface. If your changes are not configured correctly, they may cause unexpected errors and a call failure. You must test your modifications before using them in your production environment. A header modification that does not comply with the RFC specification for that header ABNF format will encounter a Java SIP stack parser exception, and will not be performed.

If there is a problem updating or adding a header with the string given from ICM script, then in the Unified CVP Call Server logs, a WARN type message displays if there is a **DsSipParserException**, or Unified CVP sends the INVITE as is with possible unexpected results on the receiver end.

Configuration for SIP Headers

There is no syntax checking while adding or modifying headersYou must be careful that headers retain their correct syntax.

Note: When modifying the headers, do not use the semicolon and the comma characters.These two characters are used internally to store the configuration in the sip.properties file and are reserved.

Refer to the examples that follow the header modification steps.

Step 1 To send SIP Headers from Unified CVP to Unified ICM:

a. In the Unified CVP Operations Console, select **Device Management > CVP Call Server**.

b. On the Edit CVP Call Server Configuration window, click the Call Server that contains the SIP headers you want to configure.

c. Click the **SIP** tab and scroll down to the section **SIP Header Passing (to ICM)**.

d. Provide the header name, and, optionally, a parameter, then click **Add**.

In your Unified ICM scripts, the call variable **Call.SIPHeader** now contains the SIP Headers you specified in the Operations Console.

Step 2 If you are going to send SIP headers back to Unified CVP, then you can modify the **Call.SIPHeader** variable using a Set node in the Unified ICM script.

Note:

- The header length (including header name) after modification should not exceed 200.
- In the Unified ICM script, you delimit the header, operation, and value with a tilde character, and use the bar character to concatenate operations.

Examples of modifying headers.

Adds a Call-Info header with the proper call info syntax as per RFC3261:

```
"Call-Info~add~<sip:x@y>;parm1=value1"
```

Short Form notation, plus concatenated operations. Adds a VIA header and modifies the From header:

```
"Via~add~SIP/2.0/UDP viaHost"  Adds a Via header to the message.
"v~add~SIP/2.0/UDP viaHost|f~mod~<sip:123@host>;parm1=value1"
```

The following operation will fail due to incorrect syntax of Call-Info header per RFC 3261. You will see a WARN message in the CVP log. This is enforced in the stack.

```
"Call-Info~add~parm1=value1"
```

From header add and modify will do the same thing, since only one From header is allowed in a message per RFC 3261. This is enforced in the stack.

```
"From~add~<sip:x@y>;parm1=value1"

"Call-ID~add~12345@xyz"  Same as From header, only one allowed.

"Call-ID~mod~12345@abc" Same as From header, only one allowed.
```

Configuring the Cisco Integrated 3G-H324M Gateway for Unified CVP

The Cisco Integrated 3G-H324M Gateway - or Video Gateway - allows multimedia communications (H.324M) between 3G (third generation) mobile handsets and Cisco AS5xxx Universal Gateways.

Before you Start

You must configure the Cisco Integrated 3G-H324M Gateway, by following standard IOS procedures. Refer to the information on Cisco.com for more indepth information about the Cisco Integrated 3G-324M Gateway:**Cisco Integrated 3G-H324M Gateway** (http://www.cisco.com/en/US/docs/video/milticomm/3g324m.html).

How to Configure the Cisco Integrated 3G-H324M Gateway for Unified CVP

To configure the Cisco Integrated 3G-H324M Gateway, follow this procedure:

Step 1 Log in to the Operations Console and select Device Management > Unified CVP Call Server.

The *Find, Add, Delete, Edit Call Servers* page displays.

Step 2 Click the Call Server for which you want to configure the Cisco Integrated 3G-H324M Gateway.

The *Edit CVP Call Server Configuration* page displays.

Step 3 Select the SIP tab and scroll down to the section *Advanced Configuration*.

Step 4 Set the default UDP Transmission Count timer to a higher value than the default for thisconfiguration because video call setup can require more time.

Caution: IOS Gateway can only support G.711 codec for the audio when connectingtoH324M video devices on PSTN.

Step 5 Click **Save** and **Deploy** to deploy the Unified CVP Call Server device.

Example Dial-Peer Configuration for Connecting the Cisco Integrated 3G-H324M Gateway

This example shows the configuration needed to connect a dial-peer to an Integrated 3G-H324MGateway. You will need to add the lines of code identified in the commented code sample below to configure video capabilities on the 3G-H324M gateway.

```
voice class codec 100
codec preference 1 g711ulaw
video codec h263 profile 10
voice class sip-profiles 1
```

```
request INVITE sip-header Contact modify "$" ";video;audio"
license feature gsmamrnb-codec-pack
!
codec profile 10 h263
fmtp "fmtp:34 QCIF=1;MAXBR=521"
!
dial-peer voice 101 pots
description From LiveLine to CVP

! The next command assigns the video information type.
information-type video
incoming called-number 3556.
direct-inward-dial
port 6/0:D
forward-digits all
!
dial-peer voice 35561 voip
description To CVP for Basic Video Calls
destination-pattern 35561
session protocol sipv2
session target ipv4:10.30.30.111

! The next command assigns a previously configured codec selection preference list to
a dial-peer.
voice-class codec 100

! The next command assigns a previously configured SIP profile to a dial-peer.
voice-class sip profiles 1
dtmf-relay rtp-nte h245-signal h245-alphanumeric
no vad
```

Chapter 12

Transferring and Queuing Calls with Unified CVP

IVRs from Unified ICME's Perspective

Essentially, Unified ICME categorizes IVRs into one of two types:

- **Intelligent Peripheral IVRs**, where—under Unified ICME's control—the carrier network routes calls to the IVR and then removes calls from the IVR for delivery to agents. With Intelligent Peripheral IVRs, once the IVR's prompting or queuing treatment has been completed, the IVR typically has no further role to play for that call.
- **Service Node IVRs**, where—following prompting/queuing treatment—the IVR initiates call delivery to agents, who are under Unified ICME's control. When functioning as a Service Node IVR, Unified CVP can stay involved with a call even after it has been transferred to another VoIP endpoint.

Unified CVP can act as either IVR type.

Note: For complete information about the call flow models available for Unified CVP, refer to "High-level Configuration Instructions for Call Flow Models (page 25)."

Transferring Calls Using Unified CVP in Comprehensive Mode

This section provides examples of Unified CVP call transfer scripts.

Note: The Script Editor Busy and Ring nodes are not supported.

Transferring Calls Using SIP Service

The SIP Service can be configured to operate in two modes to perform Unified CVP transfers. Typically, Unified CVP remains in the signaling path for the duration of the call, and in this normal mode it uses SIP re-INVITE messages to perform the transfers. This will also cause Unified CVP to hold the port license for the call duration.

To operate in standard re-INVITE mode, you do not need to modify the Unified ICME script; however, to send a REFER transfer, send a dynamic label with the letters "rf" prepended to it. Or, when using a Queue node in the Unified ICME script, define an ECC variable called "user.sip.refertransfer" and set it to the value of the lowercase "y." Unified CVP then uses the REFER method to blind transfer to agent labels.

Alternatively, Unified CVP might perform a SIP REFER type transfer where it moves out of the signaling path after sending a referral to the caller to the label provided by Unified ICME. This allows Unified CVP to release the port license after the REFER is sent. Unified CVP will receive notification of the outcome of the call using SIP NOTIFY messages, and this will be included in the reporting database as well.

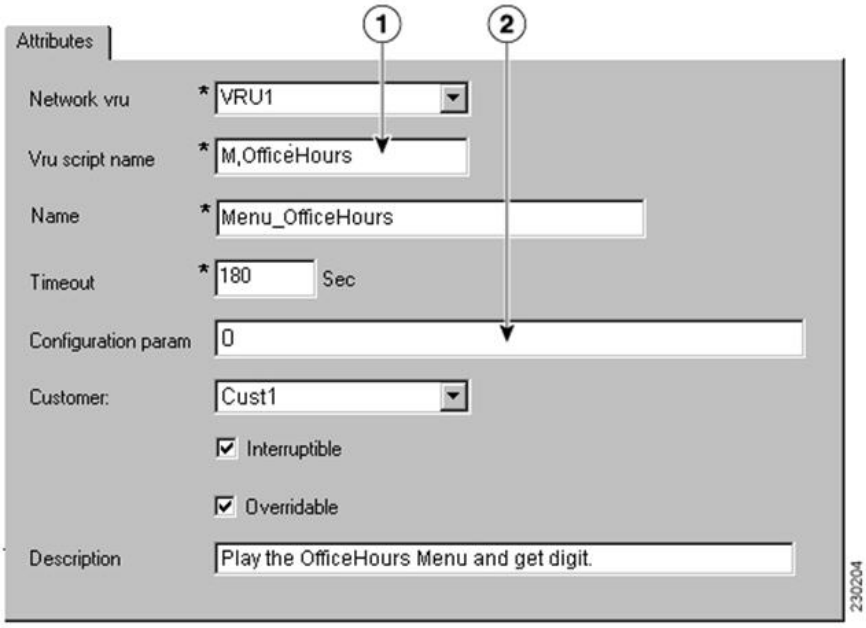
Caution: When using REFER, do not apply the survivability script for TDM callers on the Ingress gateway. Also, SIP transfers to VoiceXML gateways for micro-applications will not use the REFER method. It will only be used for non-"SEND TO VRU" type transfers. When using REFERs, note that the survivability script does not currently support REFER messaging events, so when using REFER with TDM calls on the IOS gateway, the survivability service *must* be removed from the pots dial peer for those calls. REFER is used as a "blind refer" operation and can typically be used when sending calls to third-party ACD agents, but it can also be used to send calls to the Cisco Unified Communications Manager extensions as well, if desired.

Example: Transfer to a Label

This example shows sample ICM Configuration Manager and Script Editor screen captures for a Menu application that plays a prompt presenting a menu ("Our office hours are between 8 AM and 6 PM. If you would like to talk to a customer service representative, press 0 at any time.") and then does the following:

- If the caller presses 0, collects the digit, and then routes and queues the call.
- If the caller does not press 0, releases the call.

Figure 42: Office Hours Menu Configuration



The Network VRU Script List tool's Attribute tab in the figure above shows:

1. The VRU Script Name field contains two Unified CVP parameters:

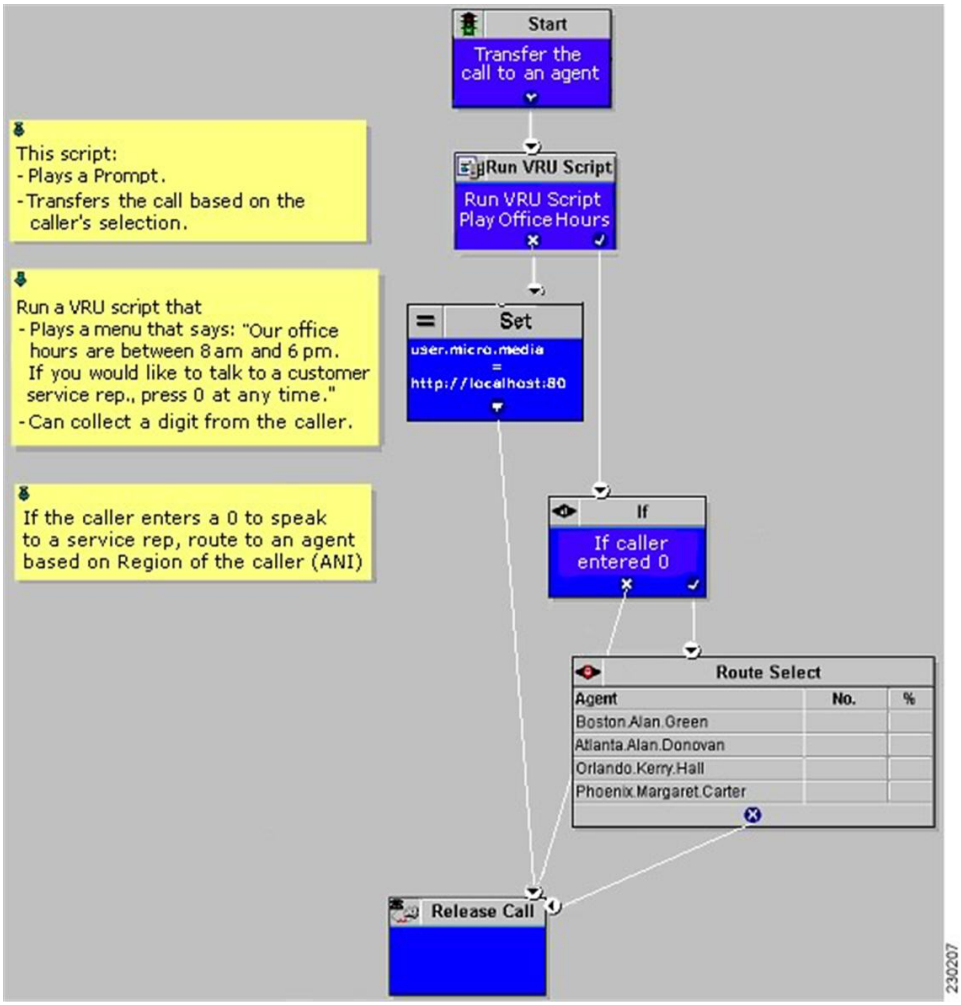
   M. Menu

   OfficeHours. Media File name

2. The Config Params field contains the following Unified CVP parameter:

   0. The number 0 is the only valid option.

Figure 43: Transfer to Skill Group in Comprehensive Mode

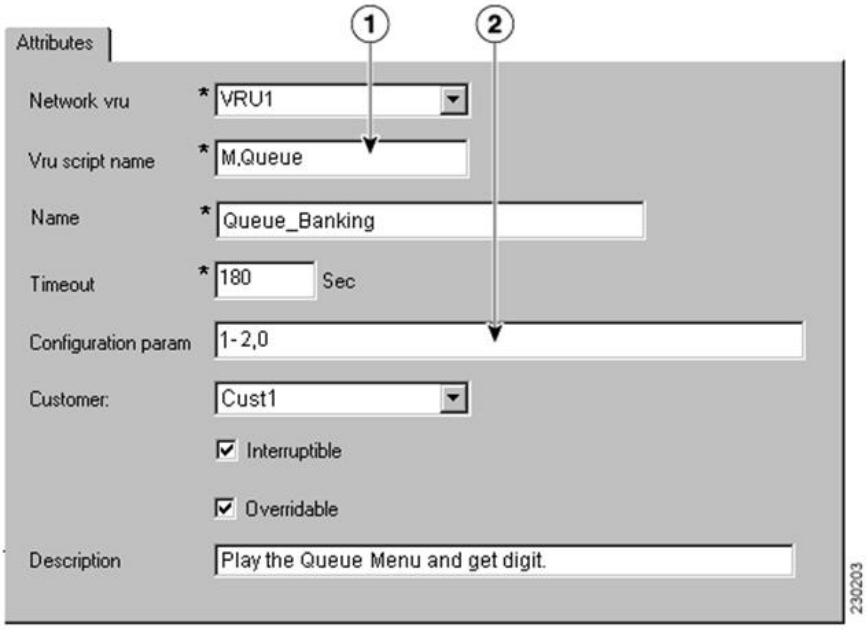


Example: Queue and Transfer to a Skill Group

Unified ICME can queue a call to an agent group and instruct Unified CVP to entertain the caller with IVR scripting using the RunVRU Script and other nodes. When the resource becomes available, Unified ICME tells Unified CVP to cancel the original request, Unified CVP then confirms the cancel request, Unified ICME sends the label for the destination, and Unified CVP or the network will transfer the call to a freed-up agent.

This example shows sample ICM Configuration Manager and Script Editor screen captures for a Menu application that plays a prompt presenting a menu ("For Checking, press 1. For Savings, press 2. To speak to a customer service representative, press 0."), retrieves any caller-entered digits, then routes and queues the call.

Figure 44: Queue Menu Configuration



The Network VRU Script List tool's Attribute tab in the figure above shows:

1. The VRU Script Name field containing two Unified CVP parameters:

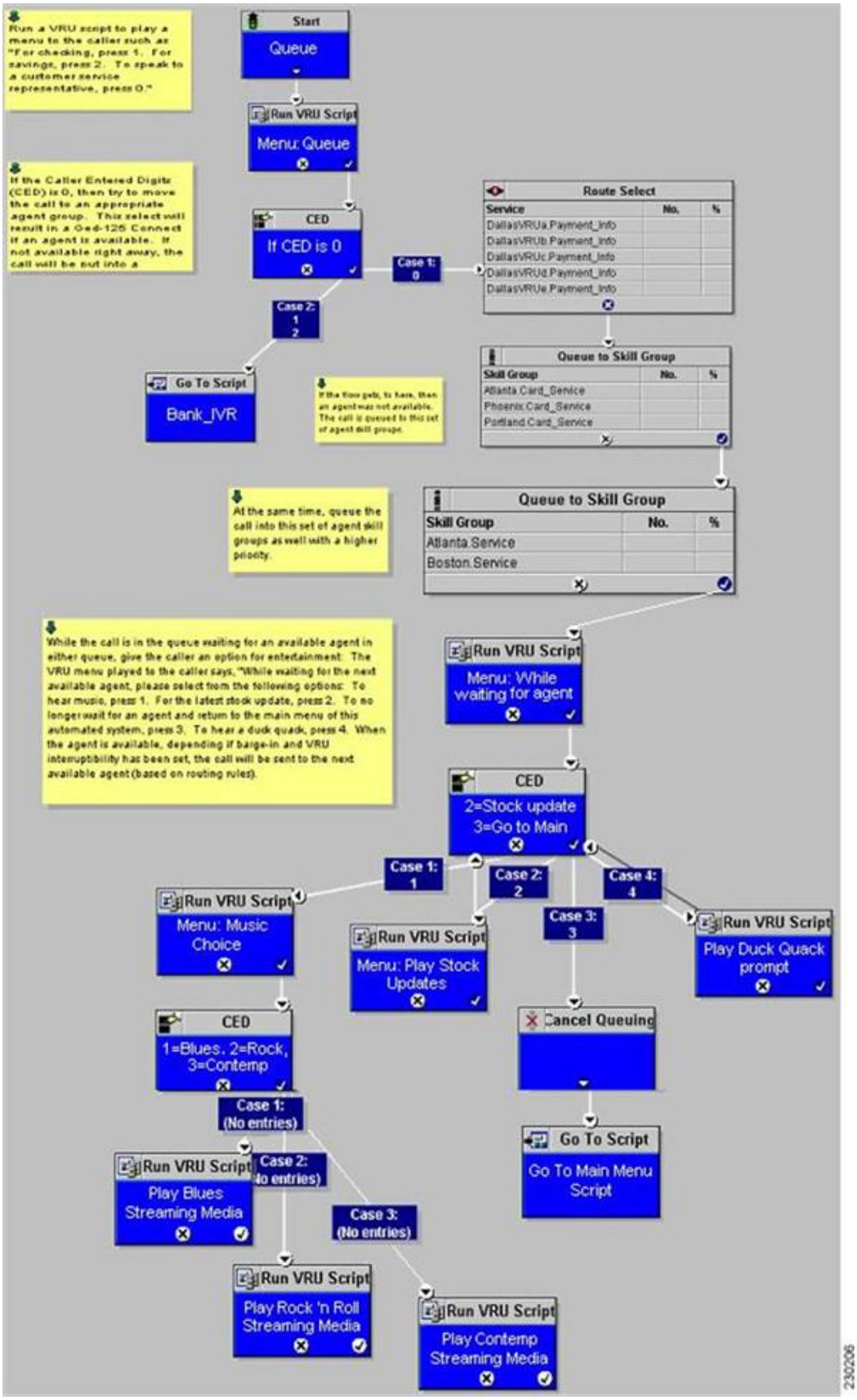
   M. Menu

   Queue. Media File name

2. The Configuration Param field containing the following Unified CVP parameters:

   1-2,0. The numbers. 1, 2, and 0 are valid options

Figure 45: Queue and Transfer to a Skill Group



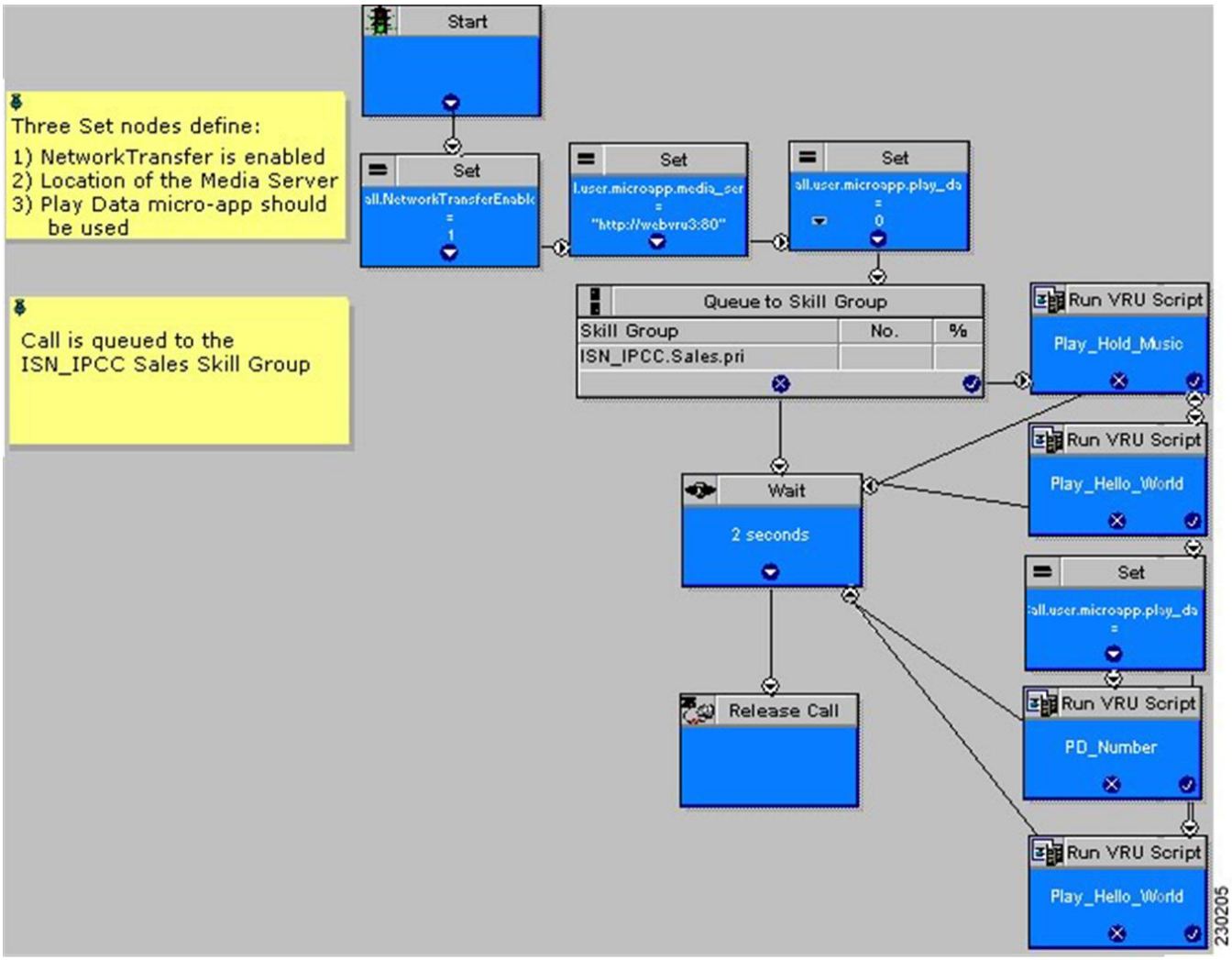
Example: Network Transfer Script

Unified CVP provides capabilities to transfer calls to another destination after they have been answered by an agent. These capabilities are referred to as Network Transfer. The Network Transfer feature does not require any special installation on the part of Unified CVP. The feature is disabled by default for all PG types except Enterprise Agent (EA).

To change the Network Transfer setting, do the following:

- Use the Script Editor's Set node to specify the **Call.NetworkTransferEnabled** variable. If you set this variable to 1, Network Transfer is enabled; if you set it to 0, Network Transfer is not enabled.
- In EA PG setups where the EA is behind a PBX, use the **Network Transfer Preferred** checkbox on the PG Explorer's Routing Client tab. If this box is "checked," Network Transfer is enabled; if "unchecked," Network Transfer is not enabled.

Figure 46: Network Transfer Script



Transferring a Call from Agent to Agent

When a call is transferred from Unified CVP to an agent, and that agent wants to transfer the call to another agent, the agent can make that transfer using either the agent IP phone or agent desktop. Transfers from the IP phone are made using CTI route points that point to a Unified ICME script. Transfers from the agent desktop are made using the Dialed Number Plan.

For network transfer from either the IP phone or CTI OS Agent Desktop, you must Queue the call to skill group in the first Unified ICME script, for example "NetXfer1", to create the call context. In this script, the "networkTransferEnabled" flag must be set to "1".

Note: The NetworkTransferEnabled setting must explicitly be set to 1 in all post-route scripts.

Configuring Network Transfer from IP Phone

Step 1 Define a CTI Route Point, for example "9999", in Unified CM. Associate it with the JTAPI User that is connected to Unified CCE PIM in Unified ICME.

Step 2 In the ICM Admin Workstation, define a Dialed Number for Unified CCE PIM and a call Type for that dialed number. This call type can then be associated with a Unified ICME Script, for example "NetXfer2".

Note: Do not define the labels of agents for the Unified CCE PIM. Instead, define the labels for VRU PIM so that the route result will be returned to VRU instead of Unified CCE PIM. If you do define the agent labels for the Unified CCE PIM, the Unified ICME router returns the route result to the VRU PIM if "Network Transfer Preferred" is enabled on the Unified CCE PIM and VRU PIM and returns the route result to the Unified CCE PIM if "Network Transfer Preferred" is disabled on the Unified CCE PIM and VRU PIM.

Step 3 When the call is delivered to Agent 1 using the Unified ICME Script "NetXfer1", the agent can dial the number 9999 to send the call to another script "NetXfer2."

Configuring Network Transfer from CTI OS Agent Desktop

Step 1 Define a Dialed Number Plan in Unified ICME. The routing client is the Unified CCE PIM and dialed number will be the one defined before for the Unified CCE PIM; that is, IPCC_PIM.9999.

Step 2 Set Post Route to **Yes** and Plan to be **international**.

Step 3 In the Agent Desk Settings, check all the **Outbound access** check boxes.

IP Transfer Example (Unified CCE Routing)

An IP transfer to an Unified CCE agent is very similar to an IP transfer to an ACD (TDM) agent with the following exceptions:

- The egress Gateway for this case is Unified CM. In the Gatekeeper, the agent extensions are configured to point to the Unified CM's IP address.
- When Unified CM receives the new call, it uses the "Skinny protocol" to connect to the agent at an IP phone. The voice channels are then connected from the Ingress Gateway to the IP phone.

Overriding the CLI Field on Outgoing Transfers

Calling Line Identification (CLI) is a set of digits and related indicators (type of number, numbering, plan identification, screening indicator, presentation indicator) that provide numbering information related to the calling party. This feature allows customers to override the CLI field on outgoing transfers, using either a Label node or an ECC variable in the Unified ICME routing script. This feature is required for transfers into Unity, which uses both ANI and DNIS to determine the appropriate mailbox to access. CLI is passed through most networks and into most call-handling devices, so this feature provides a back-door method to transmit arbitrary data during transfers when translation routing is not feasible.

The following section describes how to enable the **call.user.microapp.override_cli** ECC variable, which you must configure to enable this feature.

Configuring CLI Override

CLI override is controlled from the Unified ICME routing script.

Note:

- For IP originated calls, the "Asserted-Identity" checkbox will need to be unchecked on the CUCM SIP Trunk configuration.
- For SIP calls, the CLI Override feature is only supported using the ECC variable as shown in Method #2 below. Using a dynamic label as in Method #1 with "CLI" prepended is not supported.

You can configure CLI override one of two ways:

Step 1 **(First Method)** Append **CLI=NNNNNNNN** to the label in a LABEL node. Setting NNNNNNNN to the word **null** will blank out the CLI on the transfer.

Example: Setting a label node to **1111;CLI=9876543** results in a transfer to 1111 using a CLI of 9876543.

Example: Setting a label node to **1111;CLI=null** results in a transfer to 1111 using an empty CLI.

Step 2 **(Second Method)** Set the **call.user.microapp.override_cli** ECC variable *before* invoking a transfer using Queue to Skill Group, Label node, et ceteraThe **call.user.microapp.override_cli** Expanded Call Variable List. Set the maximum length to the maximum length of the data that will be used for CLI override. The Unified CVP Call Server must be restarted after adding this variable to Unified ICME. Setting the variable to the word **null** will blank out the CLI on the transfer.

Example: Setting **call.user.microapp.override_cli** ECC variable to **9876543** prior to a Queue to SkillGroup where agent **1111** becomes available results in a transfer to 1111 using a CLI of 9876543.

Example: Setting **call.user.microapp.override_cli=null** ECC variable *prior* to a Queue to Skill Group where agent **1111** becomes available results in a transfer to 1111 using an empty CLI.

If both of the above methods are used in one routing script, the LABEL node CLI value takes precedence over the ECC variable.

CLI override takes precedence over the SetSetupCallingNum command in VBAdmin. That is, the new CLI will always be propagated to the transfer call leg regardless of the value of ShowSetupCallingNum.

CLI override also forces the presentationIndicator to presentationAllowed on the transfer call leg.

Note: For *SIP* calls, the CLI Override feature is only supported using the ECC variable. Using a dynamic label with "CLI" prepended is not supported.

Unified CVP H.323 Service Debugging

In the VBAdmin tool, execute the following command:

```
setIntTrace on
```

The Unified CVP H.323 Service log displays the following message when it is executing CLI Override:

```
03:32:52 VoiceBrowser-VB Trace: 00000003: INTF: Overriding old CLI
with new CLI 876543 : DNIS = 5900 : CID =
20efcd51-92fb-11da-8b50-ccd458cce346
```

Most Significant Digit

Configure the H.323 Service to transfer calls to a specific Unified CM cluster so that locations-based CAC can function properly.

To make this configuration, enter the following command at the VBAdmin command line interface:

```
SetSigDigits 1
```

Note: This command takes effect immediately. You do not need to restart the H.323 Service.

With this configuration, Unified CVP will transfer calls to a specfic Unified CM cluster. Unified CVP will strip one leading significant digit (excluding the tech-prefix, if it exists) from the dialed number (provided by DNIS). The stripped value is saved and prepended with a pound sign (#) before the call is transferred to Unified CM.

For example, assume that Unified CVP receives a call with the DNIS-provided dialed number of 2#38005551234. This call is processed as follows:

1. Unified CVP strips the tech-prefix, 2#, leaving 38005551234.
2. Unified CVP strips one digit (3) from the beginning of the dialed number, leaving 8005551234.
3. Unified CVP passes 8005551234 to Unified ICME for routing.
4. When Unified ICME transfers the call to an agent, assume that Unified ICME returns the agent device label 3201.
5. Unified CVP prepends 3# to this device label and then passes 3#3201 to the gatekeeper for address resolution.
6. The gatekeeper resolves this label to a specific Unified CM cluster.

   This cluster is identified by the Unified CM gatekeeper.
7. Unified CM strips 3# from the label, leaving 3201 as the destination phone address.

Configuring Unified CCE Re-route On No Answer for Unified CVP

This section describes how to use the Re-route On No Answer function when using Unified CVP as a queue point for Unified CCE.

Summary

When using Unified CCE with Unified CVP as a queuing point and routing client, Re-route On No Answer needs to be configured differently than when using it with Unified IP IVR . The difference is caused by the fact that when using Unified IP IVR the call control is with Unified CM, whereas with Unified CVP, the call control is with Unified CVP.

Re-route On No Answer Operation for Unified CCE with Unified IP IVR

The Re-route On No Answer function ensures that when an agent does not answer a call – for example, because he walked away from his desk without making himself Not Ready – the call is taken away after ringing for a configurable number of seconds, is presented to another agent or put back in queue, and the agent is put in Not Ready state.

This function is implemented by setting a Re-route On No Answer timeout in the agent desk settings. When the call has been ringing for the configured number of seconds, the Unified CM PG will make the agent unavailable and send a post-route request to Unified ICME using a dialed number that is also configured in the Agent Desk Settings. A routing script is executed that determines a new destination for the call. This can be another agent, or the script can put the call back in a queue. When using Re-route On No Answer with Unified IP IVR , Unified ICME software responds back to Unified CM with the new destination for the call. Unified CM is responsible for sending the call to the right destination (IP IVR for queuing or new agent).

Re-route On No Answer Operation with Unified CVP

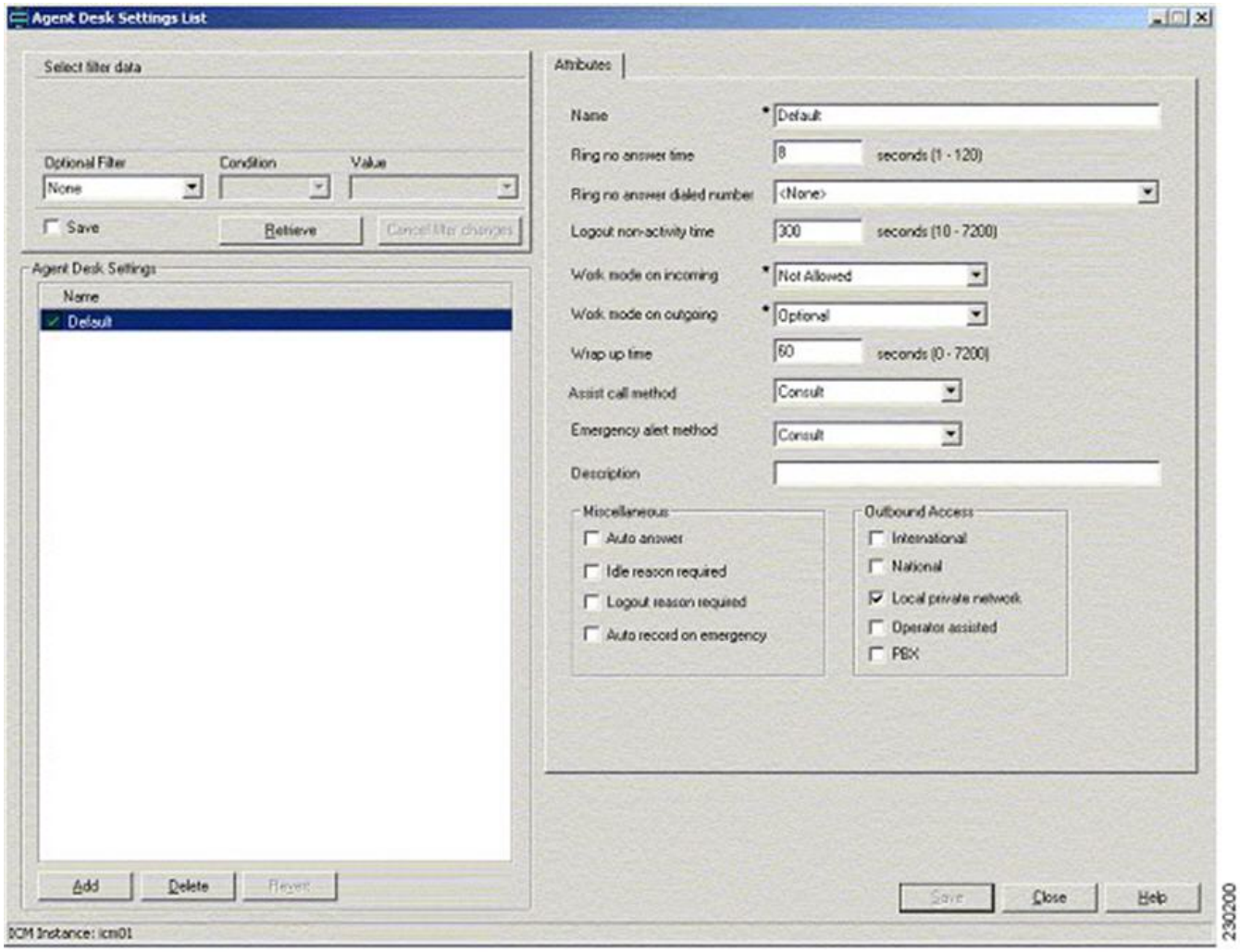
When using Unified CCE with Unified CVP, Unified CM does not control the queuing platform (Unified CVP) and can therefore not send the call back to Unified CVP for re-queuing. Instead, Unified CVP controls the call and needs to take action.

The solution is to use the Re-route On No Answer function only to make the agent unavailable when the agent does not answer the call, and to use the ICM Router Requery function to take the call away from the non-answering agent.

Re-route On No Answer Agent Desk Settings Configuration

Set a "Ring no answer time" in the Agent Desk Settings configuration, but do *not* set a "Ring no answer dialed number." Set the timeout to the maximum time you want to allow the agent to answer a call; for example, 2 rings = 8 seconds. Set this timer shorter than the no answer timeout for router requery (refer to the following example).

Figure 47: Agent Desk Settings Configuration



This will cause the agent to be made unavailable after the Re-route On No Answer timer expires, but will not invoke the Re-route On No Answer mechanism to re-route the call.

Router Requery Configuration

Router Requery is triggered by the routing client (Unified CVP) when a NoAnswer timer expires (a different timer than the Re-route On No Answer timer).

- The No Answer timer for Router Requery is not controlled by Unified ICME, but by the switching fabric, which is Unified CVP in this case. CVP 1.0 has a fixed No Answer timer of 15 seconds. The Unified CVP SIP and H.323 Services have a configurable No Answer timer (RNATimeout) with a default value of 15 seconds. This value can be set in the Operations Console for SIP and in the VBAdmin tool for H.323.

  When using Unified CVP, set RNATimeout to the desired number of seconds that the agent phone should ring before being taken away. This would probably be less than 15 seconds (4 rings), perhaps 10 seconds. In any case, this timeout **must be longer than the Re-route On No Answer timeout** set in the Agent Desk Settings.

After the Unified CVP VB RNATimeout expires, the VB/AS/PG sends an **EventReport=NoAnswer** to the router. The router picks another target according to the routing script and sends the Connect message to Unified CVP. The target might be another agent or it might be a VRU label to requeue the call. When the call disappears from the first agent, he is put in Not Ready state, due to the expiration of the No Answer Timeout in the desk setting.

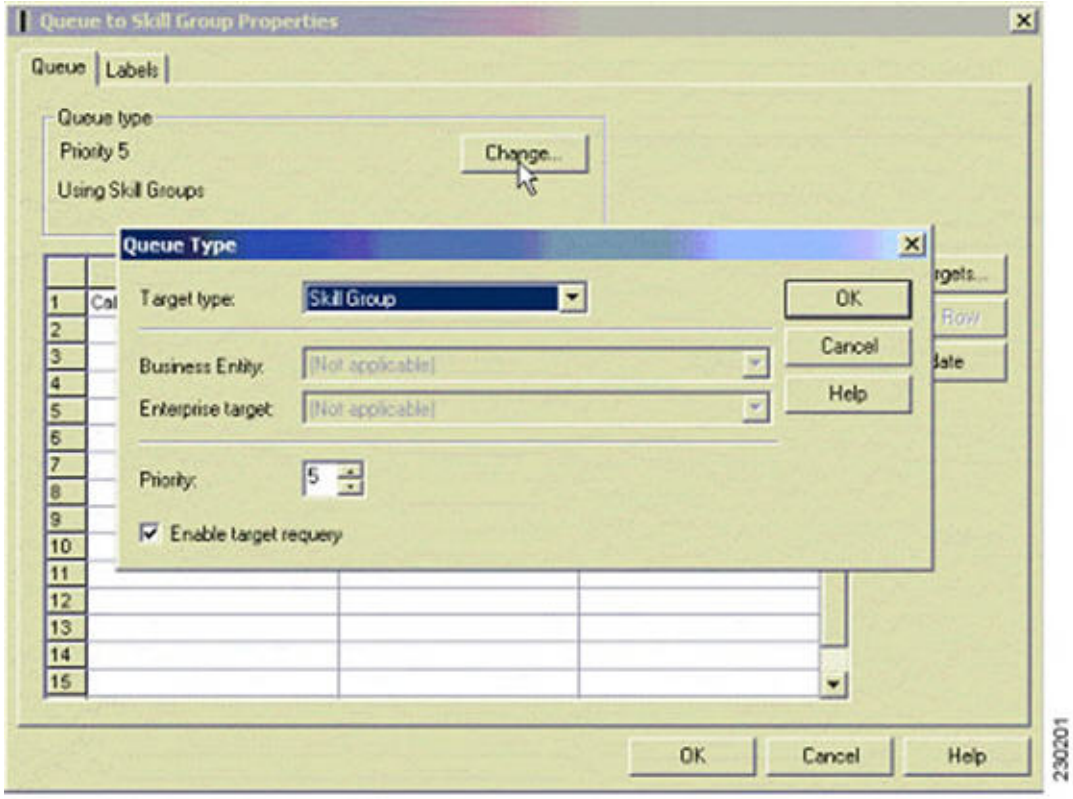
Note: Do not set the No Answer DN in the desk setting, because this is a global Unified ICME setting for all scripts, and may not be suitable for all scripts depending on the complexity of the deployment. Instead, each script should have the X path of the queue node set appropriately for each script.

- Enable Requery on the node in the script that selects the first agent. Depending on the type of node used, the Requery mechanism will select a new target from the available agents or will require additional scripting. The **Scripting and Media Routing Guide for Cisco Unified ICM/Contact Center Enterprise & Hosted** (http://www.cisco.com/en/US/products/sw/custcosw/ps1844/products_user_guide_list.html) describes how Requery works for the different nodes.

In most cases Unified CCE will use the Queue node. The Queue node requires additional scripting to handle the requeuing of the call in front of the queue. The script example below provides a standard way of handling this.

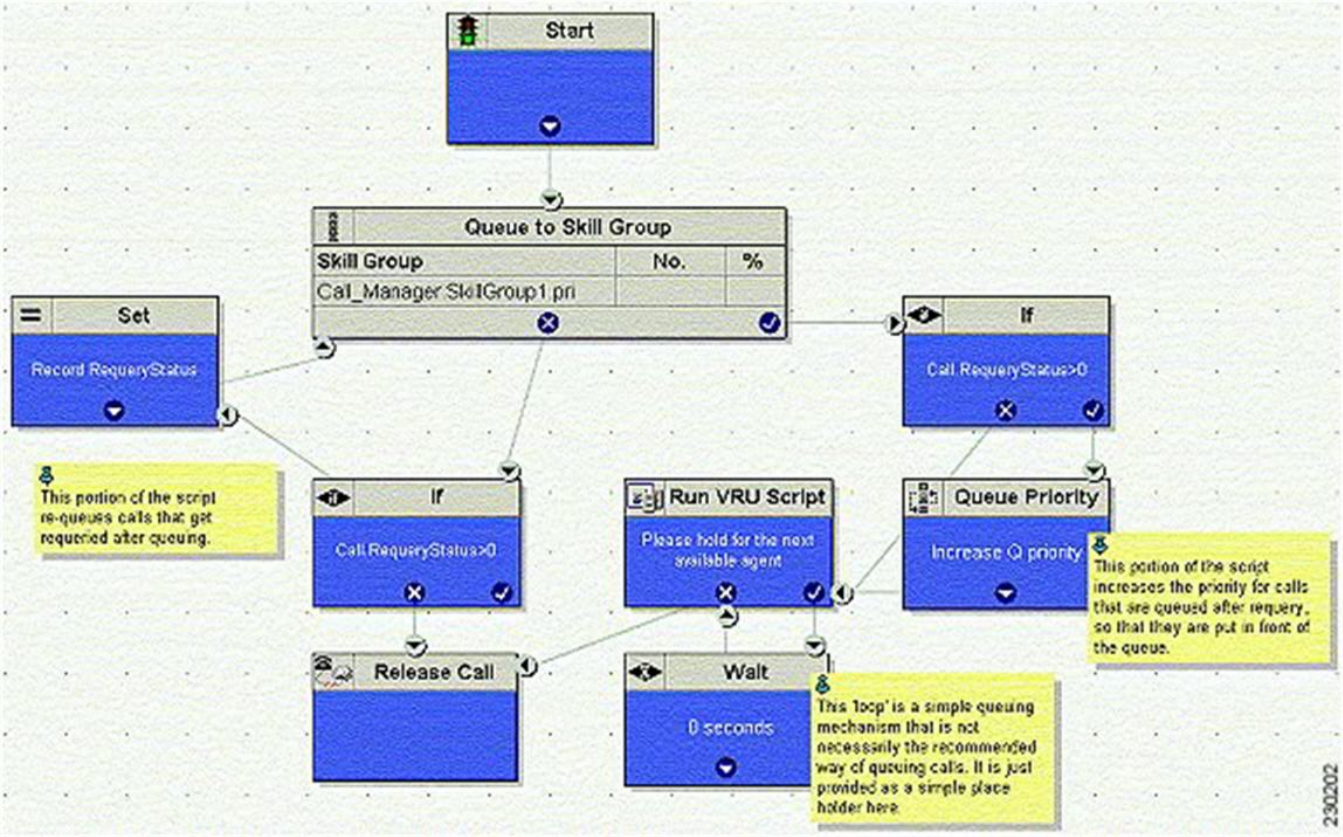
The Queue node selects the longest available agent from the skill groups configured, if there is an available agent. If there is no available agent, it queues the call with a priority set in the node (refer to the following screen shot) and continues down the success exit of the node. When an agent becomes available, Unified ICME always selects the longest queued call from the ones with the highest priority. When the Queue node connects the call to an agent and the agent does not answer the call, Unified CVP Ring-No-Answer timeout will expire, causing the Requery mechanism to kick in.

Figure 48: Queue to Skill Group Properties



When this happens, the script immediately continues through the failure exit of the Queue node with the Requery Status variable set to 'No Answer' (= 3). The typical treatment is to put the call back into the same queue but with a higher priority than all other calls, since the call needs to go in the front of the queue, not the back.

Figure 49: Sample Script for Re-Route on No Answer



In this script, when the Queue node selects an agent that does not answer the call, the script exits through the failure exit (X) of the Queue node. The If node tests the RequeryStatus variable. If it has value of greater than zero, this is a requery call, and the script re-queues the call. In the preceding example it also sets a flag using a call variable for reporting purposes. Assuming that there are no agents available, the Queue node immediately exits through the success exit (Checkmark). The If node checks to see if this is a requeried call. If so, it increases the Queue Priority of the call so that it will be handled before any other calls in queue. It then enters the normal wait loop with RunScripts.

The call flow is as follows:

- Script connects call to agent by sending connect message to Unified CVP (with requery enabled).
- Agent phone rings.
- After the Re-route On NoAnswer timeout expires, Unified ICME makes the agent unavailable. The agent state does not actually change until the call gets taken away from the agent. The agent phone continues to ring and the agent can still pick up the phone (if he does pick up the phone, he will be left in Ready state after the call, even if it was after the Re-route On No Answer timer expired).
- After the Unified CVP VB RNATimeout expires, the VB/AS/PG sends an EventReport=No Answer to the router. The router picks another target according to the routing script and sends

the Connect message to Unified CVP. The target might be another agent or it might be a VRU label to requeue the call.

- When the call disappears from the first agent he is put in Not Ready state.

Limitations

The configuration described in this section has the following limitation:

- Each call that is redirected by this mechanism is counted twice in the Skill Group, once as redirected and once as handled (if the call is finally handled). But the Call Type will only count this call once. Although it will be counted Handled and Requeried, Requeried is not used to balance CallsOffered in the Call Type. If you want to see this call counted twice in the Call Types, you can address this by changing the call type in the error path before the second queue to skill group node.

Configuring Re-route On No Answer for Unified CM with Unified CVP

When calls are originated from Unified CM to a CTI Route Point, in the case of an agent transfer, routing client responsibilities should be passed back to Unified CVP as soon as possible upon entering the Unified ICM script. To ensure that Unified ICM Router directs calls to Unified CVP, you need to include a SendtoVRU node in the Unified ICM script before any Runscript or SkillGroup node executes. When the routing script executes the SendToVRU node, the ICM Router instructs Unified CVP to become the routing client to handle for any subsequent transfers or VRU call processing.

RONA Operation to a script CTI Route Point Transfer

The Go to Script node is used as a RONA destination when "enable target requery" is configured on the Queue to Skill Group node and the agent does not answer. When the ICM script executes the Go To Script node, script execution proceeds to the specified script. For example, when an agent does not answer, the X-path out of the Queue to Skill Group Node will target a Go To Script node with script "CTI_Route_Point_Transfer" specified. Script processing then continues from the beginning of the CTI_Route_Point_Transfer"script and proceeds normally.

Valid destinations out of the X-path of Queue to Skill Group node are:

- Another skill group
- A prompt
- GoTo node (do not use "Dynamic Label")

See Also

The Ring on NoAnswer (RONA) timeout setting used for Unified ICM are configured differently for Unified CVP. To ensure these setting are configured properly. Refer to VBAdmin Configuration Commands (page 321) and Router Requery Configuration (page 393) for additional

configuration information. You can also refer to the Operations Console online help for detailed information about configuring CVP H.323 Service and SIP Service settings.

Limitations

The configuration described in this section has the following limitations:

- The disposition of the requeried call is not correctly reported. The Redirect No Answer field in the agent and skill group reports do not show calls that are redirected by this mechanism. Each call that is redirected by this mechanism is counted twice, once as abandoned and once as handled (if the call is finally handled). There are two Unified CCE TerminationCallDetail records for this call, one for the rerouted call (with CallDisposition 'Abandoned while Ringing', code 3) and one for the handled call with a CallDisposition depending on how the call was finally handled. The scripting example above shows how a Peripheral Call Variable can be used to mark and count calls Requeried because of no answer. A custom reporting template can be written to report on this data.

Call Survivability

This section describes how to install and configure Unified CVP with a script that allows the gateway to transfer a call in the event of a critical Unified CVP application error or WAN failure. This application should be placed on the incoming pots dial-peer that is destined for Unified CVP. Call survivability is supported in all Unified CVP call flow models except the VRU-only call flow model. In the Unified CVP Standalone call flow model, survivability will be invoked if the gateway encounters an error from the CVP Voice Server, the "param survive" parameter has been included and a survivability service has been defined.

In the event of critical Unified CVP application errors or a WAN failure that would normally disconnect the caller, this script allows the gateway to attempt a transfer to some alternate location after the failure occurs instead of disconnecting the caller. In the event that the call cannot be transferred to an alternate agent, the script will play a "call-back-later" message and disconnect.

This script provides the following capabilities:

- Ability to do multiple types of transfer in call failure conditions:
  - *8 transfer connect (outpulse)
  - Hairpin
  - SRST
  - Hookflash Relay
  - Two B-Channel Transfer (TBCT)
- Ability to differentiate call recovery behavior by incoming DNIS.

- Ability to differentiate call recovery behavior by incoming DNIS and how long the call had been in Unified CVP prior to failure.
- Ability to differentiate call recovery behavior based on time of day and date.
- Ability to hand off to an auto-attendant type application in the event of some downstream failure (for example, WAN failure, Unified ICME failure, Unified CVP failure). This auto-attendant functionality could be CME's BACD, a Unified CVP Standalone call flow model, a VXML Server application, or a custom-written VXML application.

Caution: This script is a component of the Unified CVP software. Do not make any modifications to this script. Modifications to this script not made as part of an official Unified CVP release nullify Cisco support responsibility for this script.

Installing the Call Survivability Script

To install the Call Survivability script, perform the following steps:

Step 1 Using the Operations Console, copy all script/prompt files to the gateway.

Step 2 On the gateway, do the following:

For a **Unified CVP Comprehensive** call flow model, first define two services:

```
application
service survive flash:survivability.tcl
paramspace callfeature med-inact-det enable
service handoff flash:handoff.tcl
```

Then add the following parameters:

```
 ip rtcp report interval 2000
gateway
timer receive-rtcp 4
```

Note:

- This will cause survivability to be invoked between 8 and 16 seconds ((2000 ms *4) * 2) for an active call after a WAN failure. If IOS detects the absence of both RTP and RTCP packets after 8 to 16 seconds, it will raise an error event and survivability will be invoked. (The **factor of 2** is a built-in IOS factor that cannot be configured. *Do not* adjust these values lower as this could cause the survivability event to be prematurely invoked.)
- The timer receive-rtcp command configures a media activity timer that is common to both H.323 and SIP. If set, it affects both H.323 and SIP calls.

For a **Unified CVP Standalone** call flow model, first define one service:

```
application
 service my-survivability-service flash:survivability.tcl
```

Note: You can replace "my-survivability-service" with any name you choose.

Then associate the my-survivability-service that you just created as a parameter on the CVPSelfService.tcl service associated with the incoming pots dial-peer Note that the text "param survive" must be entered exactly as shown, but the my-survivability-service can be renamed to the service name of your choice. For example:

```
dial-peer voice XXXX pots
service my-CVP-service
incoming called-number NNNNN
service my-CVP-service flash:CVPSelfService.tcl
param CVPPrimaryVXMLServer my-VXML-server-IP
param CVPBackupVXMLServer my-backup-VXML-server-IP
param CVPSelfService-app my-VXML-server-app
param keepalive my-CVP-service
param survive my-survivability-service
service my-survivability-service flash:survivability.tcl
```

Optionally, start a background keepalive service to the VXML Server. For example, for a service name of "my-standalone-service":

```
service my-standalone-service
param keepalive my-standalone-service
```

Note: This service prevents the caller from hearing a period of silence at the start of each call if the VXML Server is down, as the gateway will know the current status of the VXML Server.

Step 3 On the gateway, do a "call appl voice load my-survivability-service" and "call appl voice load handoff."

Step 4 Do the following:

On a **Unified CVP Comprehensive** call flow model:

- Create a Unified CVP pots dial-peer on the gateway, placing the Unified CVP called number on an incoming-called-number parameter.
- Assign the my-survivability-service service to this dial-peer.

On a **Unified CVP Standalone** call flow model, no special survivability dial-peer needs to be created. However, the parameter "param survive my-survivability-service" must be included on the CVPSelfService.tcl service. This parameter indicates which service to run in the event of a system failure. In this way, different survivability services can be invoked depending on the incoming pots dial-peer invoked.

How to Configure the Gateway for Call Survivability

You can configure the following parameters on the gateway for call survivability:

- **open-hours-agent** - The destination recovery target DNIS to be used when the current time matches any open-hours-time parameter. The script will cycle through all agents sequentially until one answers. If no one answers, (or in the case of a takeback transfer, the PSTN does not take back the call), the script cycles through all after-hours-agent's (maximum of 50 agents).
  - **Syntax:** open-hours-agentX DNIS
  - **Arguments:** X = a number from 0 to 49, DNIS = target destination for the recovery transfer.
  - **Example 1:** `DTMF*8,9875551212` (When PSTN *8 takeback is desired) where:

    DTMF - Indicates takeback and transfer via DTMF tones

    ***8** - The sequence the switch recognizes to perform the takeback.

    Zero or more commas - Each comma represents a pause of 100 ms. Some switches require a pause between the takeback sequence and the DNIS.

    9875551212 - The actual DNIS to which the PSTN should transfer the call.
  - **Example 2:** `HF,,,,,9875551212` (when hookflash transfer is desired) where:

    HF - Indicates takeback and transfer via hookflash relay

    Zero or more commas - Each comma represents a pause of 100 ms. Some switches require a pause between the hookflash and the DNIS.

    9875551212 - The actual DNIS to which the switch should transfer the call.

    Note: When using either DTMF or hookflash takeback, you need to configure the following additional parameters on the gateway voice ports:

    ```
    voice-port 7/1:0

    no echo-cancel enable

    no non-linear

    no vad

    playout-delay maximum 250

    playout-delay nominal 200

    playout-delay minimum high

    playout-delay mode fixed
    ```
  - **Example 3:** `9875551212` (when hairpin or SRST transfer is desired)
  - **Example 4:** `TBCT9875551212` (when TBCT is desired)

  - **Example 5: <retry>** (when a retry to the original CVP DNIS is desired)

    Assuming the original Unified CVP DNIS was *4444*:

    <retry> will send the call to CVP using DNIS 4444

    56<retry>78 will send the call to CVP using DNIS 56444478

- **after-hours-agent** - The destination recovery target DNIS to be used when the current time matches any after-hours-time parameter or as a default destination if transfers to the open-hours-agent's fail. The script will cycle through all agents sequentially until one answers (maximum of 50 agents). If no one answers, a call-back-later message will be played to the caller and then disconnected.

  - Syntax: identical to open-hours-agent

- **open-hours-time** - A string representing the date or days of week and time of day that open-hours-agent's will be used for the recovery transfer (maximum of 20 values). Month/day has higher selection priority than days of the week.

  - **Syntax:** open-hours-timeX {month/day | days-of-week}[:HHMM-HHMM]

  - **Arguments:**

    X = a number from 0 to 19.

    month/day = month of year and day of month (no year)

    days-of-week = a string of up to seven digits representing the days of the week (Sunday = 0, Saturday = 6)

    HHMM-HHMM = the starting and ending time of the period, expressed in 24-hour clock notation.

- **after-hours-time** - A string representing the date or days of week and time of day that after-hours-agent's will be used for the transfer. These do not explicitly need to be listed. If the current date/time does not fall in an open-hours-time slot, it will default to an after-hours agent. A typical use would be to specify holidays that would normally fall on working weekdays. A maximum of 20 values are allowed.

  - **Syntax:** identical to open-hours-time

- **open-hours-cvptime** - You may want to choose a particular recovery agent based on how long the call had been in Unified CVP before the failure occurred. If no open-hours-cvptime is specified, the associated open-hours-agent will be used regardless.

  - **Syntax:**number-of-seconds

  - **Arguments:**

    X = a number from 0 to 19, corresponding to the associated open-hours-agent

number-of-seconds55 would use open-hours-agent0 only when the call had been in Unified CVP less than 55 secs.

- **after-hours-cvptime** - Same as open-hours-cvptime, but applies instead to after-hours-agents.
- **alert-timeout** - A numeric value indicating the maximum number of seconds the destination phone should ring before aborting the call attempt.
  - **Syntax:** alert-timeout 20
- **setup-timeout** - A numeric value indicating the maximum number of seconds that the tcl script will wait in establishing a tcp connection to Unified CVP before aborting the call attempt. This value should be greater than the "h225 timeout tcp establish" parameter under the voice class h323 configuration on the gateway.
  - **Syntax:** setup-timeout 7
- **aa-name** - If non-blank, indicates that when a failure occurs, the Unified CVP survivability script hands off the caller to the BACD auto-attendant application. Enter the following:

```
service <survivability-servicename>
param aa-name <BACD-servicename>
service <BACD-servicename>
param isn-name <survivability-servicename>
```

  Where **`servicename`** is the service name of the BACD auto-attendant script to which control should be passed.

- **standalone** - If non-blank, indicates that when a failure occurs, this Unified CVP survivability script passes control to the service name specified. Typically, that service would reference the CVPSelfService.tcl script to invoke a Call Studio application to provide IVR treatment to the caller; for example:

```
service survivability flash:survivability.tcl
param standalone vxmlapp
service vxmlapp flash:CVPSelfService.tcl
```

- **standalone-isntime** - Select the standalone option depending on how long the call had been in Unified CVP before the failure occurred. If no standalone-isntime is specified, the standalone option is invoked if it is *non-blank*.
  - **Syntax**: **`standalone-isntime {> OR <}number-of-seconds`**
  - **Arguments**: number-of-seconds = number of seconds the call was in Unified CVP before the call failed, prefixed with > or <. For example, standalone-isntime <2 would use standalone only when the call had been in Unified CVP less than 2 seconds.
- **icm-tbct** - A numeric boolean value (0 or 1) indicating whether or not Unified ICME scripts will issue TBCT transfers. Default is 0 (by default, Unified ICME does not handle TBCT transfers). Set this value to 1 to enable TBCT transfers issued from a TBCT label in an Unified ICME script.
  - **Syntax Example:** icm-tbct 1

Call Survivability Examples

In the first Call Survivability example, the following configurations are used:

```
service survivability flash:survivability.tcl

param open-hours-agent0 9777123400
param open-hours-agent1 4444888
param open-hours-time0 12345:0900-1730
param open-hours-time1 12/18:0600-2300

param after-hours-agent0 7777008
param after-hours-agent1 8766008
param after-hours-time0 7/21:0700-0800
param after-hours-time1 11/25

param setup-timeout 7
param alert-timeout
dial-peer voice 800232 pots
application survivability
incoming called-number 8002321765
direct-inward-dial
```

Using the above survivability configurations, review the following cases:

- Case 1: Assume today is a holiday, Thursday, 11/25 at 1300 hours. Since 11/25 is defined as a specific after-hours-time, it will be selected before the 12345:0900-1730 open-hours-time which also falls on a Thursday. If the WAN fails, this script will first try a transfer to 7777008 and then to 8766008.

- Case 2: Assume today is Saturday, 12/18 at 0900 hours, peak of the holiday shopping season. Since 12/18 is defined as a specific open-hours-time, it will be selected for an open-hours-agent even though it falls on a Saturday which would normally be an after hours time. If the WAN fails, this script will first try a transfer to 9777123400, then try 4444888, 7777008, 8766008.

- Case 3: If you don't care about time-of-day routing, and simply want a last-resort transfer mechanism, put one of more DNIS's in the after-hours-agent slots and don't define any times. Any failed call will always be directed to the list of after-hours-agents.

The next example illustrates how to organize call survivability functionality by incoming DNIS, create a separate application for each DNIS and apply desired call recovery properties to each application. For example:

- Assume billing callers dial 45XX and sales callers dial 55XX to access Unified CVP.

- Assume that a billing call fails somewhere in the course of the call:

  – If the call fails and the call had been in Unified CVP less than 30 seconds (this would also include the case where the call had *never* made it to Unified CVP, i.e., 0 seconds), send the caller back through the PSTN via a *8 takeback to 8005556666.

  - If the call fails and the call had been in Unified CVP greater than or equal to 30 seconds, send the caller back through the PSTN via a *8 takeback to 8007778888.

- Assume that a sales call fails somewhere in the course of the call:
  - If the call fails (in this case, the amount of time the call had been in Unified CVP is irrelevant), send the caller back through the PSTN via a hairpin transfer to 8009990000.

- Assume the PSTN switch is sending ANI and DNIS in such a way that the ANI and DNIS are concatenated together in the DNIS field. Assume that ANI length is 10 and DNIS length is 4. Also assume that ANI can be blank, e.g. blocked callerID.

The IOS configuration elements necessary to accomplish these cases is shown below.

Note: Dial-peers 2 and 4 are necessary in the event of no ANI (blocked caller ID). The lower preferences of dial-peers 2 and 4 is to protect against the case where a caller's ANI begins with, say, 45. For example, assume caller with ANI 4521111111 dials the sales DNIS. Without lower preferences, the caller would have matched dial-peer 2 and gone to the billing application instead of sales (you wanted it to match dial-peer 3).

The following are the configuration elements for the second example:

```
dial-peer voice 1 pots
preference 1
application billing
incoming called-number 45..
#-----------------------------------------
dial-peer voice 2 pots
preference 2
application billing
incoming called-number 45..
#-----------------------------------------
dial-peer voice 3 pots
preference 1
application sales
incoming called-number 55..
#-----------------------------------------
dial-peer voice 4 pots
preference 2
application sales
incoming called-number 55..
#-----------------------------------------
dial-peer voice 5 pots
destination-pattern 8009990000
port 7/0:D (or whatever port is desired)
#-----------------------------------------
dial-peer voice 6 voip
incoming called-number 8009990000
codec g711ulaw (To force the call to g711ulaw on the outgoing hairpin)

#-----------------------------------------
service billing flash:survivability.tcl
param after-hours-agent0 DTMF*8,,,8005556666
param after-hours-cvptime0 <30
```

```
param after-hours-agent1 DTMF*8,,,8007778888
param after-hours-cvptime1 >29
param ani-dnis-split 10:4
#-----------------------------------------
service sales flash:survivability.tcl
param after-hours-agent0 8009990000
param ani-dnis-split 10:4
```

Location-based Call Admission Control (LBCAC -- Queue-at-the-Edge)

LBCAC is used to maximize local branch resources, keeping a call within the branch whenever possible and limiting the number of calls that go over the WAN. Unified CVP revision 8.0(1) supports *queue-at-the-edge*, a simpler and more effective configuration of LBCAC than the legacy configuration (page 381). Using the queue-at-the-edge functionality, the call originating from a specific branch office is deterministically routed to a local VXML Gateway based on priority. That is, always choose a local branch agent if possible.

Note: For design discussion and design considerations when using LBCAC, refer to **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) guide.

LBCAC Concept Definitions

The following definitions are used in the configuration of LBCAC:

- **Phantom Location.** A default location with unlimited bandwidth used when calculating calls that are hairpinned over an H.323 or SIP trunk or when the H.323 or SIP call is queued at the local branch, to enable correct bandwidth calculations. The Phantom location should be assigned to the gateway or trunk for CVP.
- **SiteID.** The SiteID is a string of numbers that is appended to the label from Unified ICM so that the dial plan can be configured to route the call to a specific destination, such as the branch VXML gateway or egress gateway, or Unified CM node. The SiteID can be appended at the front of the label, at the end, or not at all. This configuration is separate from the Unified CM location configuration, and is specific to Unified CVP. The SiteID is used to indicate the real location of the call and allow the bandwidth to be deducted from the correct location.

Configuration of LBCAC Queue-at-the-Edge

The following steps provide an example configuration for LBCAC with queue-at-the-edge functionality. There are several sets of steps. Each set is identified as to whether its steps apply to SIP, H.323, or both protocols. Complete the *common* steps, then the steps for the protocol you are using.

COMMON STEPS FOR SIP and H.323

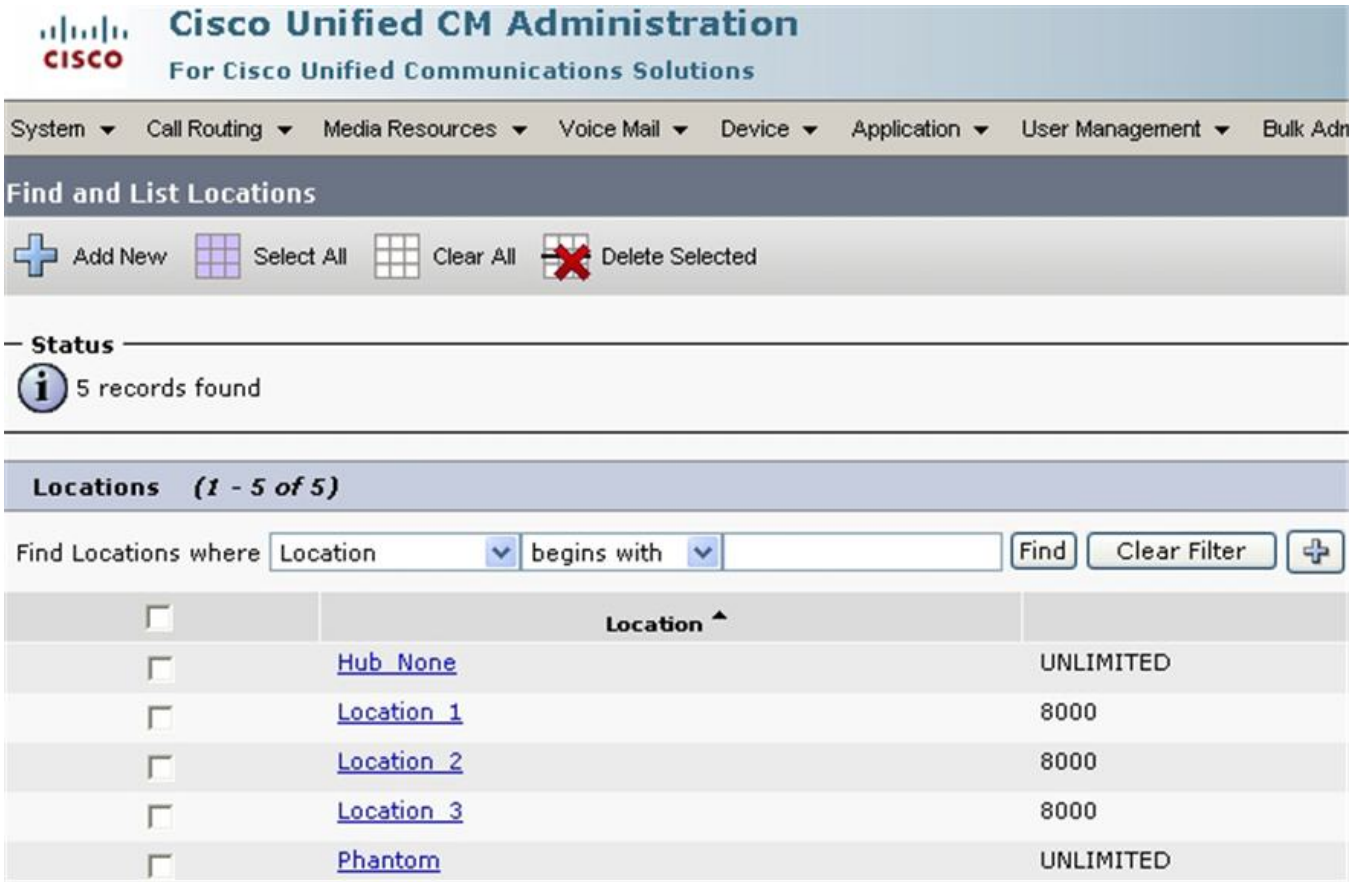
Using the Unified Communications Manager, configure all branches so that Location and Bandwidth is defined:

1. From Unified CM Administration select **System > Location**. Click **Find** to list the locations and add new ones as appropriate.

2. If you are using H.323, verify that a location of *Phantom* exists. If not, add it also.

   Note: Unlimited must be *unchecked* for each branch (the box to the left of the location name); otherwise bandwidth will not be deducted for that branch. (The Phantom location will still have unlimited bandwidth even when unchecked.)

Figure 50: Specify Bandwidth for Each Branch Location

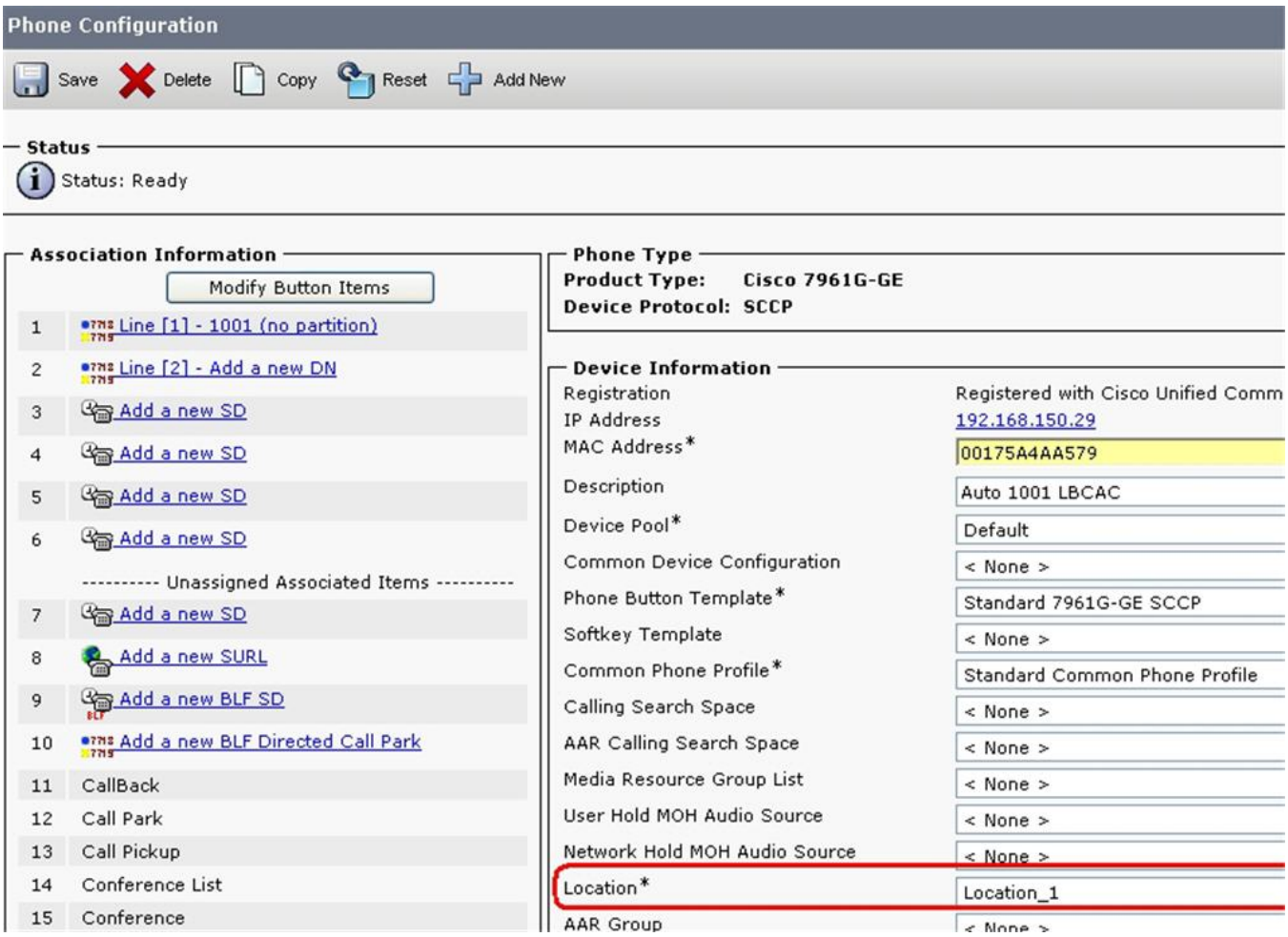


3. For the branch phones, configure each phone so that it is assigned the branch location for that phone.

   a. Select **Device > Phone**. Click **Find** to list the phones.

   b. Select a phone and set the *Location* field.

Figure 51: Associate Phone with Branches



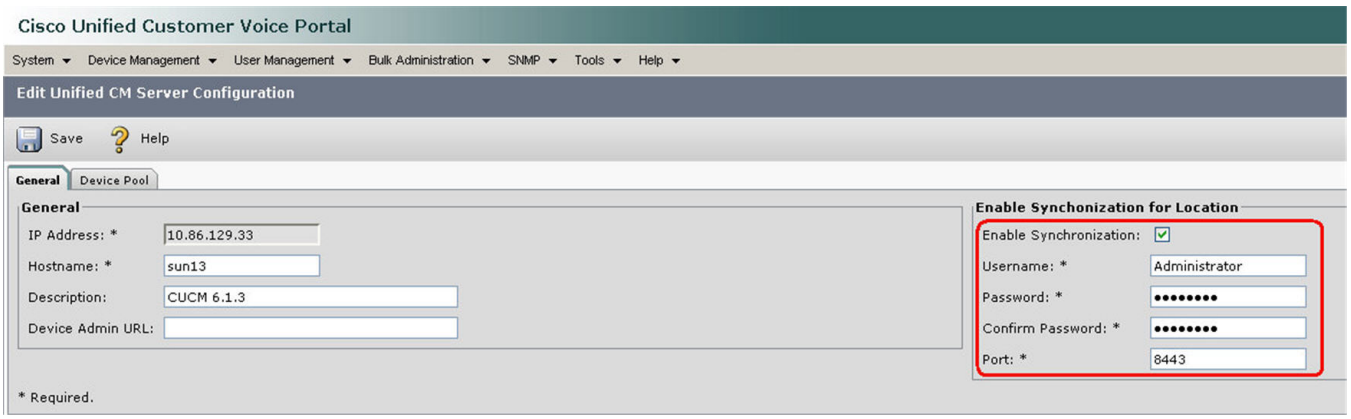
4. Verify that the Cisco AXL Web Service is started and that an Application User is defined and has a role of *Standard AXL API Access*.

   a. From Cisco Unified Servicability, select **Tools > Control Center > Feature Services**

   b. Start the Cisco AXL Web Service, if it is not started.

   c. From Cisco Unified CM Administration, select **User Management > Application User**. Verify you have a user with the role of *Standard AXL API Access*, or create a new one and add that user to a group that has the role of *Standard AXL API Access*.

Figure 52: Verify Standard AXL API Access User



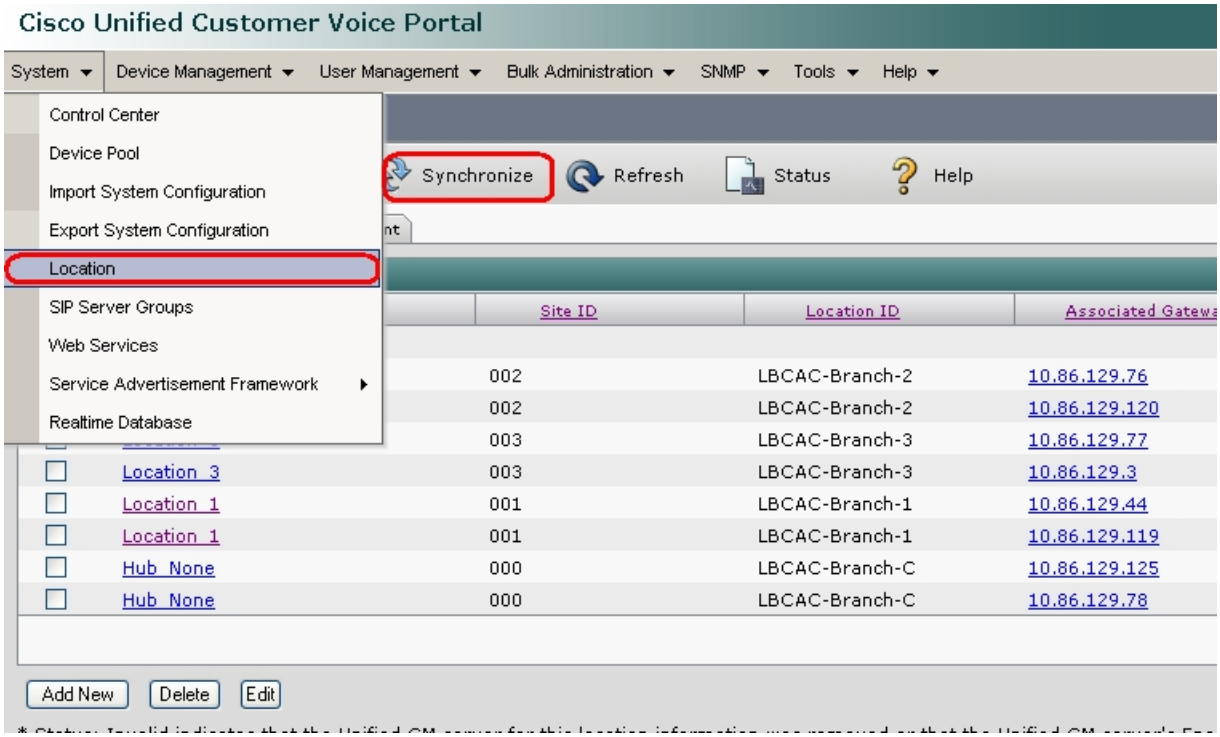
MORE COMMON STEPS: SIP and H.323 for Unified CVP

On Unified CVP, using the Operations Console:

1. In **Device Management > Unified CM**, in the section *Enable Synchronization for Location*, enable synchronization and provide the credentials required to log in.

2. In **System > Location**, click **Synchronize** to retrieve the locations defined on Unified CM.

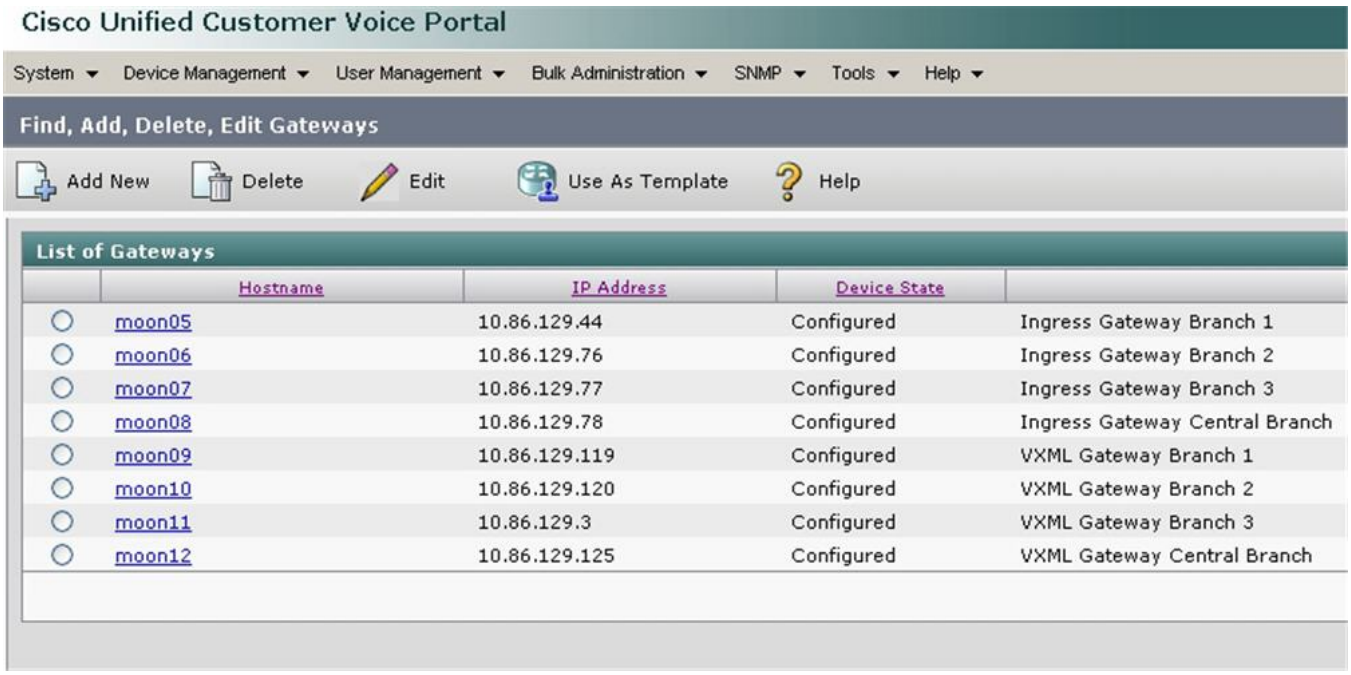
Figure 53: Enable Syncronization and Provide Credentials



Select **System > Location** and verify that the locations have been synchronized from Unified CCM.

3. In **Device Management > Gateway**, define the Ingress and VXML gateways.

Figure 54: Define Ingress and VXML Gateways

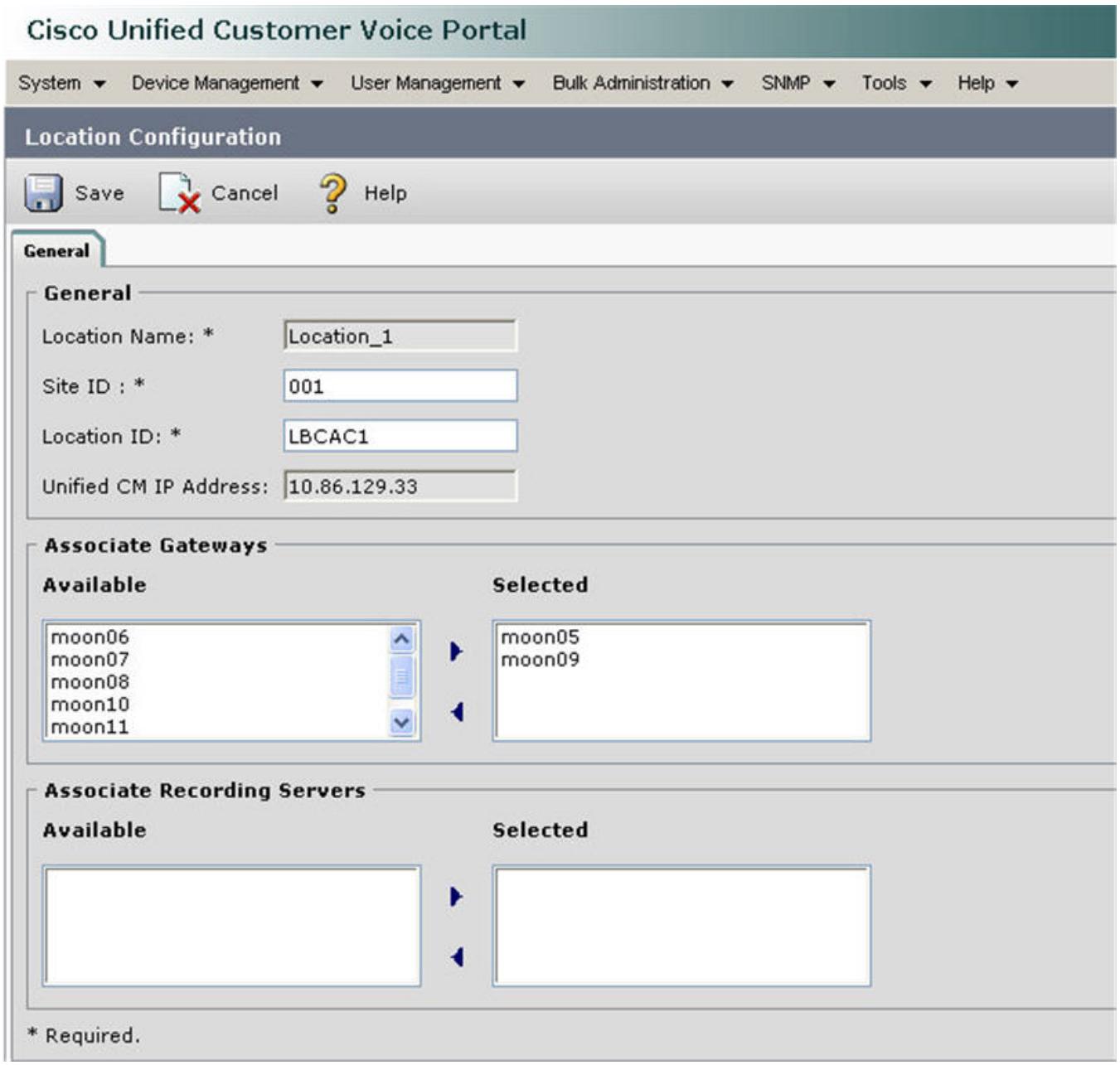


4. **Assign IDs.** In **System > Location**, select a location.

   a. Assign a Site ID and Location ID to the location, then add the associated gateways to the location.

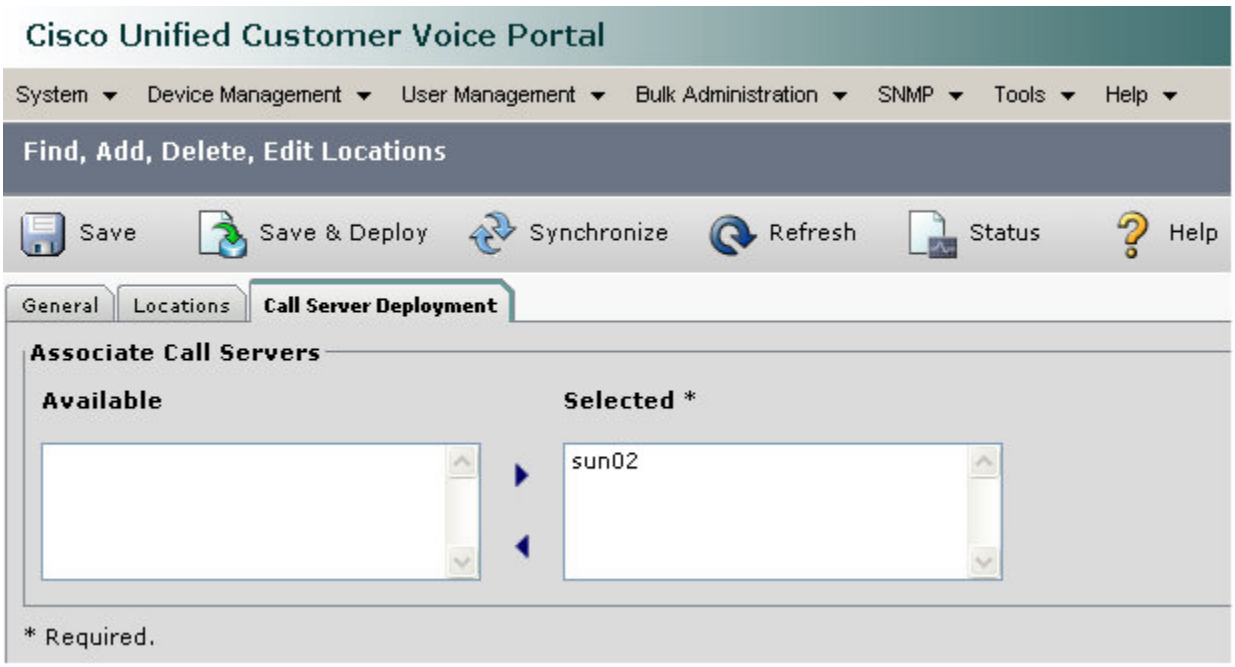
   b. Repeat for each of the locations.

Figure 55: Assign Site ID and Location



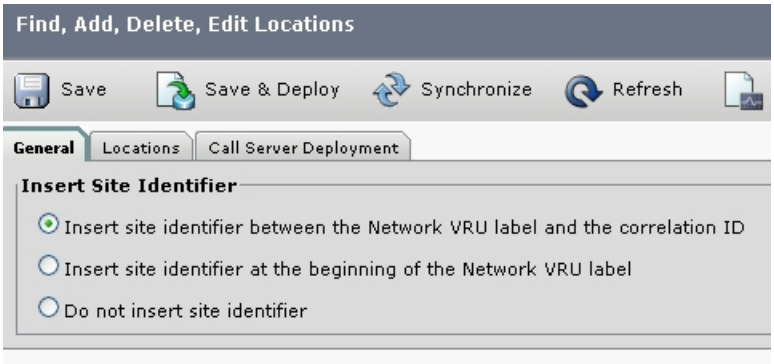
5. In **System > Location**, navigate to *Call Server Deployment* and select the Call Servers where the configuration is to be deployed. Click **Save and Deploy**.

Figure 56: Select Call Servers for LBCAC



6. For the insertion point of the SiteID, use the default location *between the Network VRU label and the correlation ID* as shown in the following screenshot.

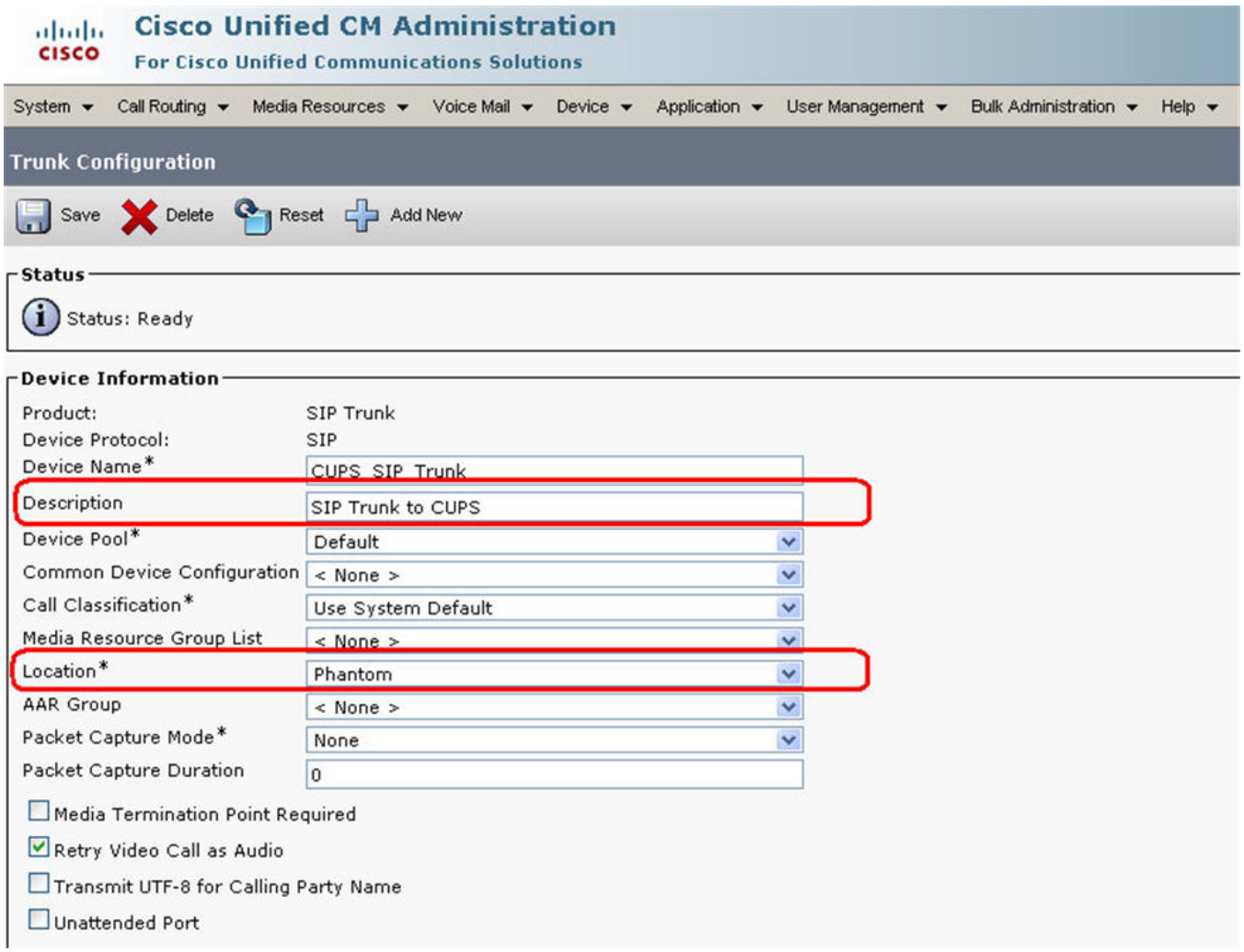
Figure 57: Define Location to Insert SiteID



SIP Deployments: Unified Communications Manager Steps

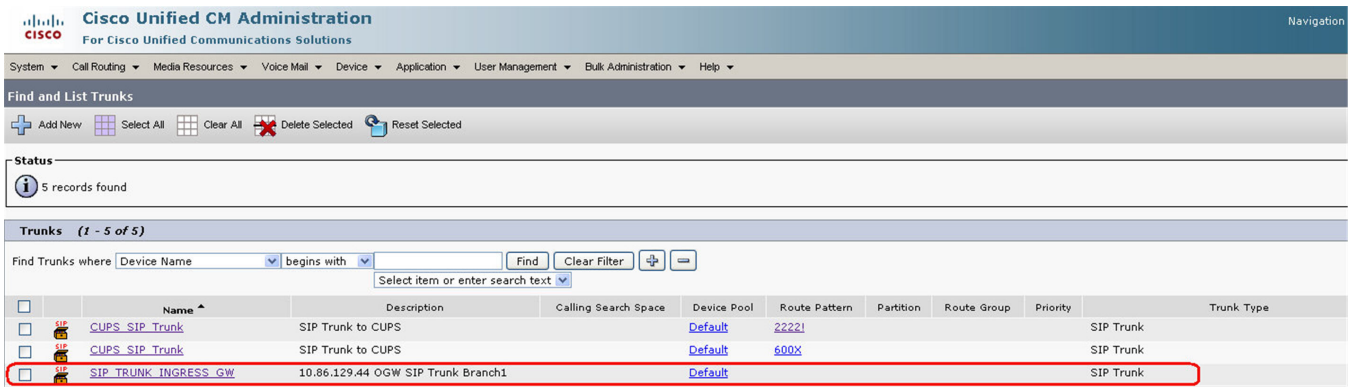
1. Using the Unified Communications Manager, create a SIP trunk toward the SIP proxy server and select the *Phantom* location.

Figure 58: Create SIP Trunks to Proxy Server for LBCAC



2. Create a SIP trunk for each ingress gateway and make the location of these ingressTDM-IP gateways the actual branch location.

Figure 59: Create SIP Trunks for Ingress Gateways



3. Create a route pattern pointing the Network VRU Label of the CCM routing client to the SIP trunk toward the SIP proxy you created in Step 1.

   The SIP proxy should route the Network RRU label of CCM routing client to the farm of CVP Call Servers.

4. For any IP-originated calls, the CCM route pattern should be associated with the SIP trunk created in Step 1.

5. Using the Unified Communications Manager Administration, select **Device -> Device Settings -> SIP Profile -> Trunk Specific Configuration -> Reroute Incoming Request to new Trunk based on -> Call-Info header with the purpose equal to x-cisco-origIP**.

6. Associate the new SIP profile from Step 3 with the trunk defined in Step 1 and each Ingress gateway defined in Step 2.

SIP Deployments: CUP SIP Proxy Configuration Steps

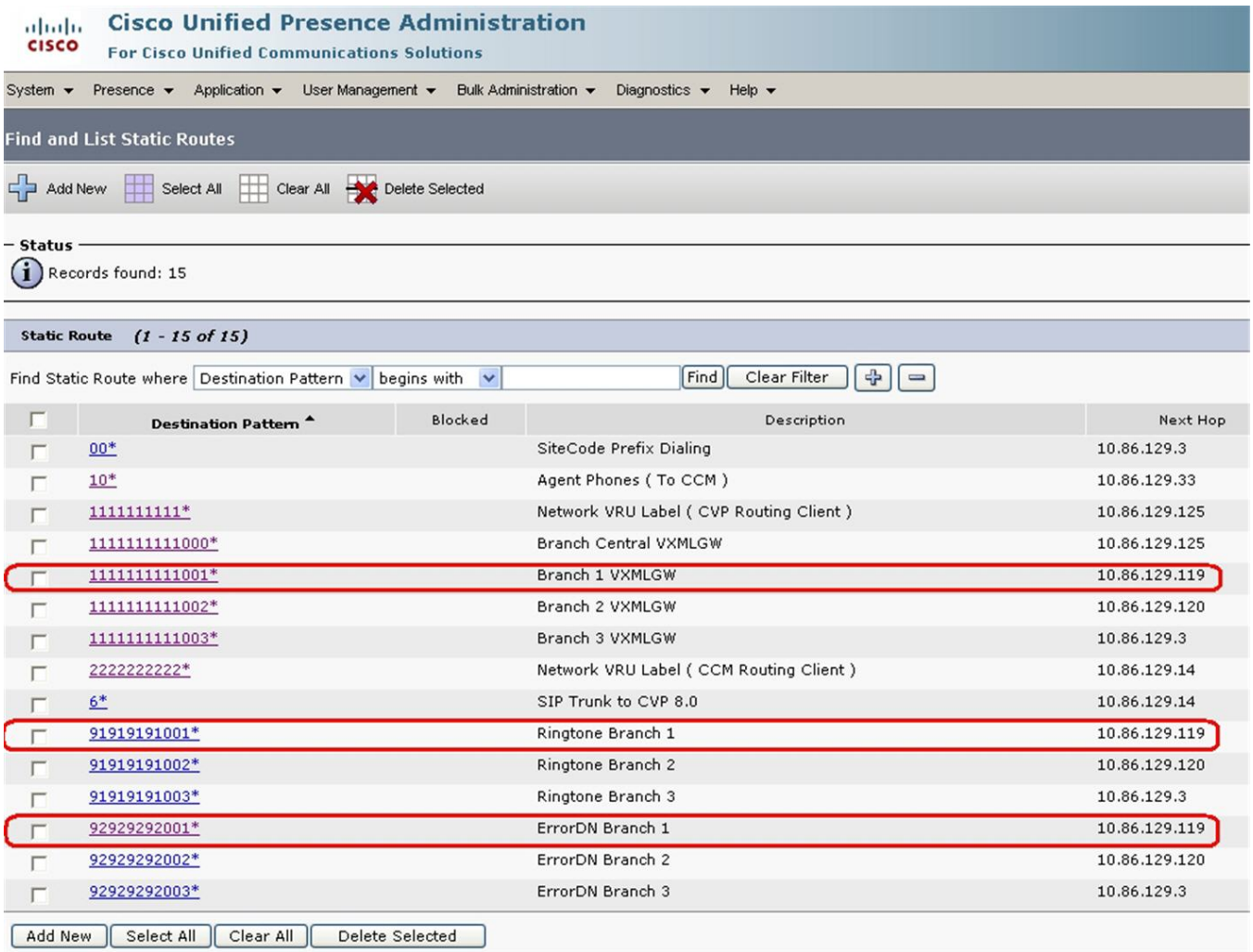
1. Create static routes to send calls to the branch VXML GW by appending the siteId to the Network VRU label of CVP Routing client.

   For example, if the Network VRU label in ICM for CVP Routing client is "1111111111", then the route shown below in the proxy will send the call originated from a branch1 phone back to a branch1 VXML GW (001 as the sitecode for branch1) for queuing.

   Refer to "Configuring the SIP Devices (page 355)," and to SIP DN Pattern Matching Algorithm (page 19) for detailed information.

2. Also define the routes for ringtone and error to send them to the local branch VXML gateway.

Figure 60: Define Ringtone Routes for LBCAC



3. In **System > Service Parameters > SIP Proxy Settings**, raise the value of **Maximum MTU Size** to 1800. This change prevents "482-Loop Detected" error on a Requery when there is an LBCAC rejection.

H.323 Deployments: Unified Communications Manager Steps

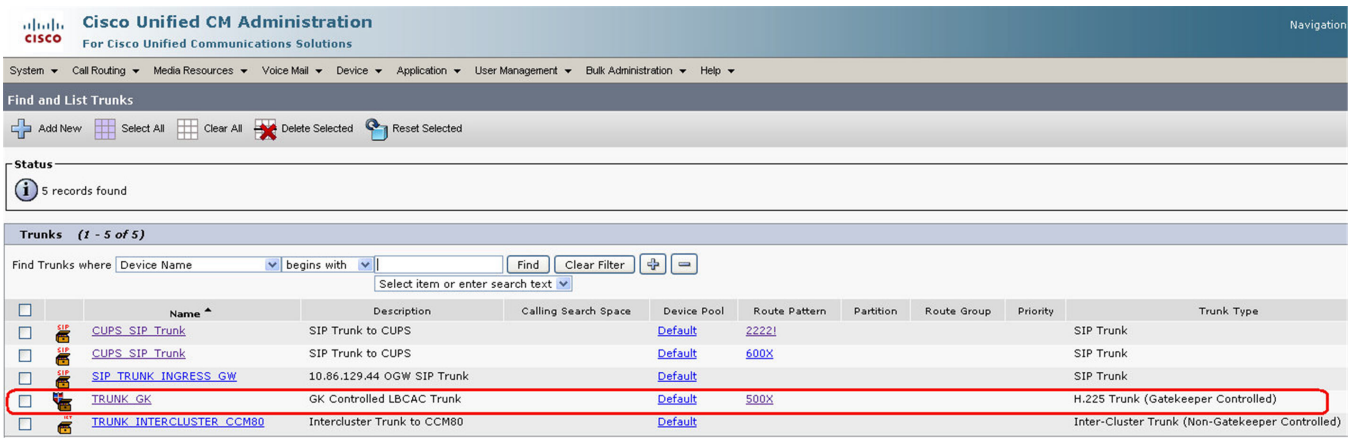
On the Unified Communications Manager:

1. Create a Gatekeeper controlled H.225 trunk and set the location of the trunk to *Phantom*.

   a. Select **Device > Trunk**. Click **Add New**.

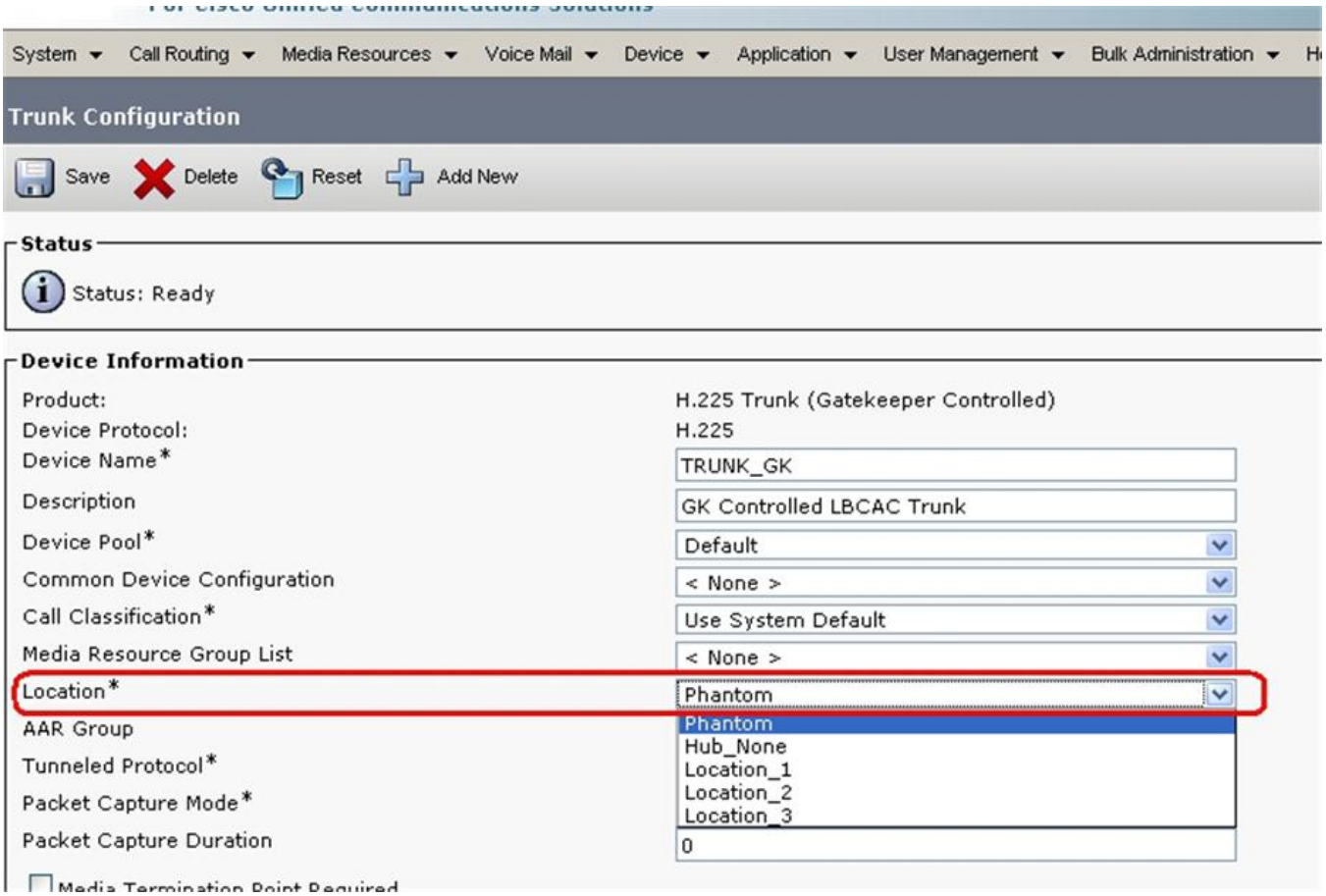
   b. Select **H.225 Trunk (Gatekeeper Controlled)** and click **Next**.

Figure 61: Create Gatekeeper for LBCAC



c. Set the Location.

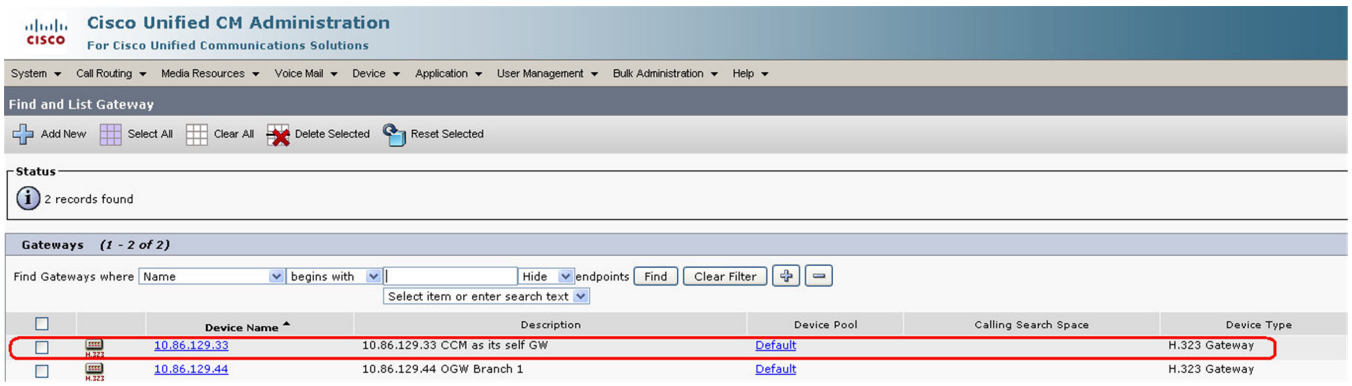
Figure 62: Select Gatekeeper Branch Location for LBCAC



2. Create an H.323 gateway for each individual Unified CM call processing node.

   Set the location of these trunks to *Phantom*.

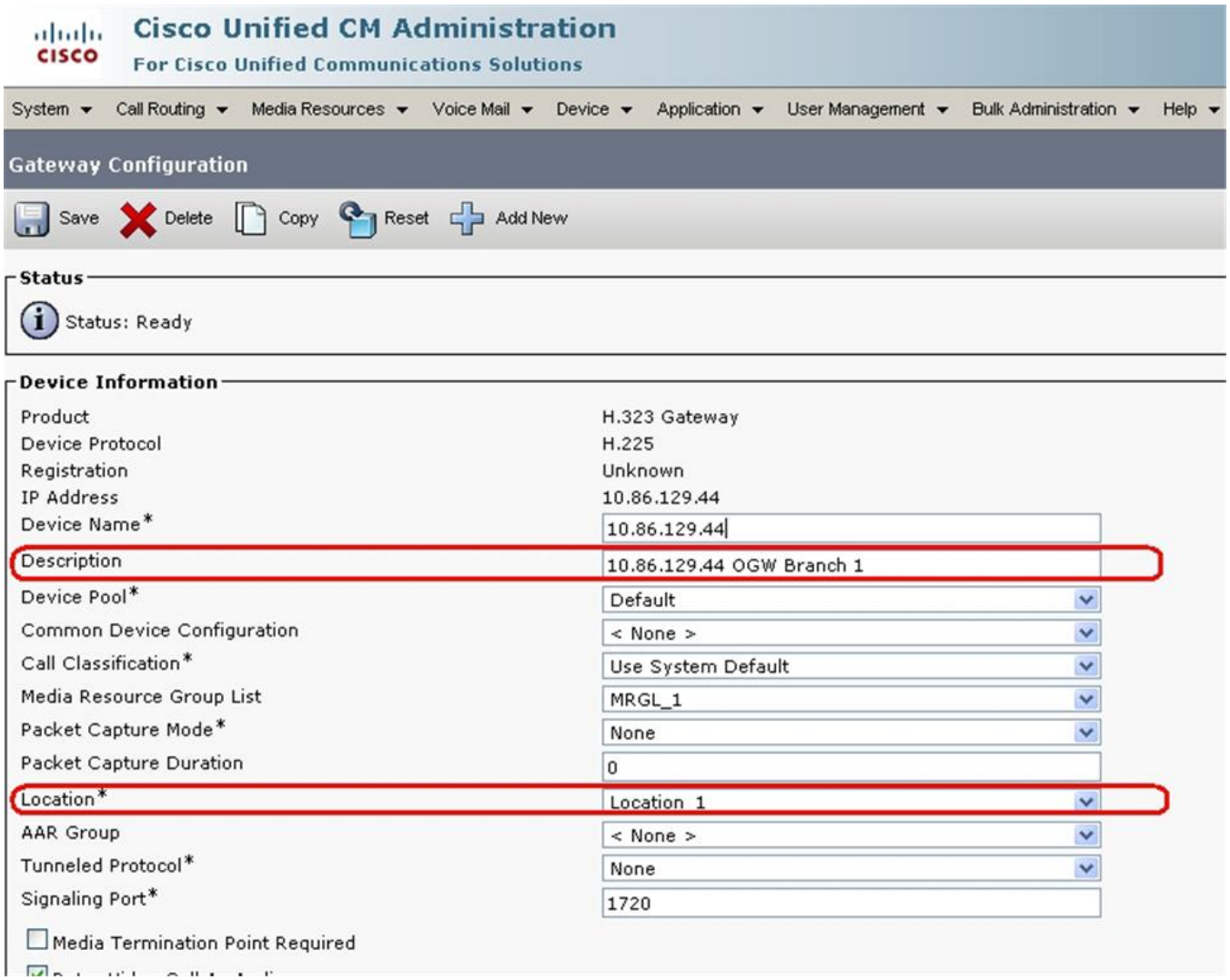
Figure 63: Set Gatekeeper Trunk Location to Phantom for LBCAC Configuration



3. Create an H.323 gateway for each of the ingress gateways.

   Set the locations of the ingress TDM-IP gateways to be the actual branch locations.

Figure 64: Set Ingress Gateways to Branch Locations



4. Create a route pattern pointing the Network VRU Label of Unified CM routing client to the H.225 Gatekeeper Controlled trunk created in Step 1.

The Gatekeeper should route the Network VRU Label of Unified CM routing client to the Unified CVP Call Servers.

For any IP originated calls, the Unified CM route pattern should be associated with the Gatekeeper controlled trunk created in Step 1.

5. Select **System > Service Parameters** and select the appropriate server.

   Select the Cisco CallManager service; set the following values, then click **Save**.

   – Accept Unknown TCP Connection: **True**

   – Send H225 User Info Message: **H225 Info for Call Progress Tone**

   – Device Name of GK-Controlled Trunk that will use Port 1720: **None**

   – Clusterwide Parameters (Service) > Media Exchange Timer: **20**

   – Clusterwide Parameters (Service) > Media Resource Allocation Timer: **20**

H.323 Deployments: Gatekeeper Configuration Steps

You can define routes to deterministically send the calls to the branch VXML gateway by appending the siteID to the Network VRU label of Unified CVP Routing client.

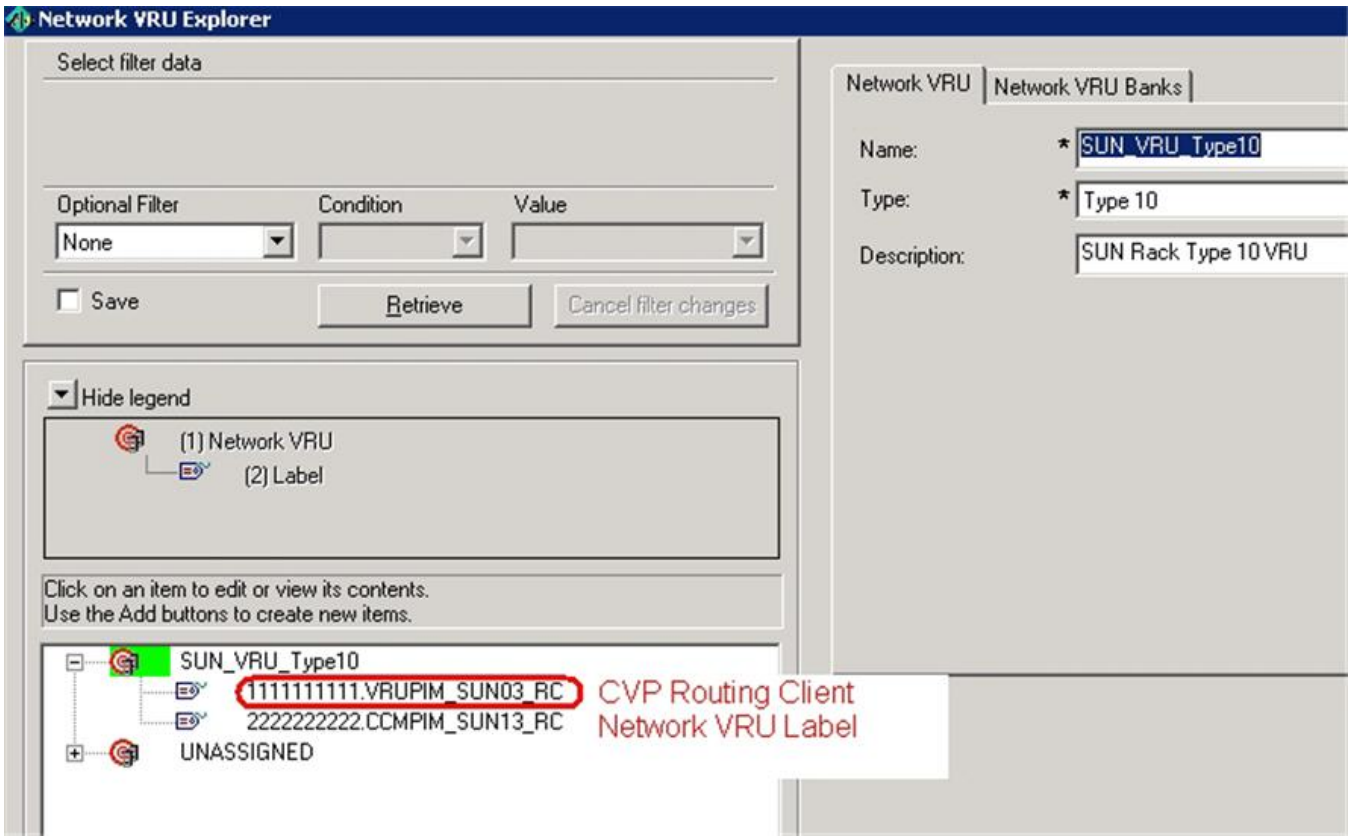
For example, if the Network VRU label in Unified ICM for Unified CVP Routing client is "1111111111", then the route given below for the gatekeeper sends the call originated from a Branch1 phone back to a branch1VXML gateway (001 as the sitecode for branch1) for queuing.

Gatekeeper Configuration.

1. On the gatekeeper, create a route. For example:

```
zone prefix <gk-zone> 1111111111001* gw-priority 10 <branch-1 VXML
GW>
```

Figure 65: Configure Gatekeeper for LBCAC



2. On Unified ICM, create the labels for the Network VRU to correspond to the DN created on the gatekeeper.

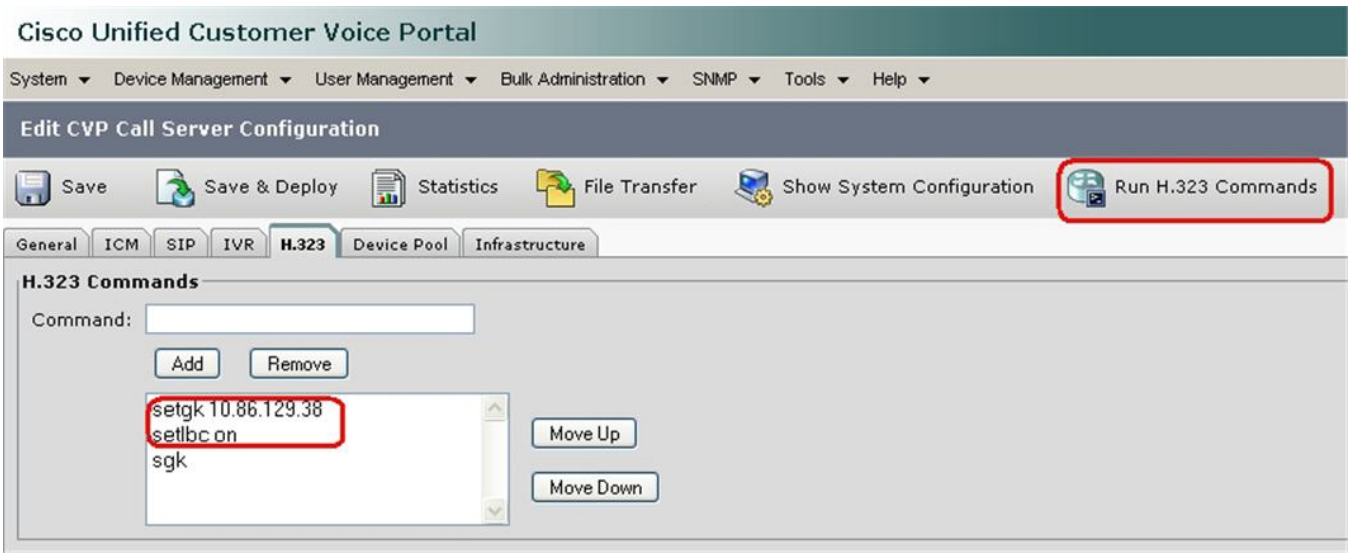
H.323 Deployments: Unified CVP Configuration Steps

From the Operations Console:

1. Add the following commands to the Call Server:
   a. Select **Device > CallServer > H.323 tab**.
   b. Add: setgk <gk-1 IP>, <gk-2 IP>

      Where: gk-1 and gk-2 are the two gatekeepers used for redundancy and failover.
   c. setlbc on
   d. Click **Save** or **Save and Deploy**.
2. To process the configured sequence of commands, click the **Run H.323 Commands** icon in the Edit Call Server Configuration page's toolbar.

Figure 66: Run H323 Commands for LBCAC Configuration



Cisco Unified Presence Server Setting

The following MTU Size setting change is required to perform Requery for LBCAC when there is a call rejection.

Make the following change to the CUP server:

1. Log into the Cisco Unified Presence Server Administration tool.
2. Select **System >Service Parameters** and select the service from the Server drop-down box.
3. From the Service drop-down box, select **Cisco UP SIP Proxy**
4. Scroll to the bottom of the **Sip Parameters** list to find **Maximum MTU Size - upconvert (bytes)**.
5. Change the default 1300 value to **1800** and click **Save**.

Configuring Locations-Based Call Admission Control (LEGACY IMPLEMENTATION)

Locations-based call admission control (CAC) is used in the Cisco Unified Contact Center Enterprise branch-office call flow model (also known as the Centralized Model). This means that all severs (Unified CVP, Unified ICME, Unified CM, SIP Proxy server, gatekeepers, and media servers) are centralized in one or two data centers and each branch office (of which there can hundreds or thousands) contain only a gateway and IP phones.

This section provides an overview of how to configure Unified CVP to:

- Accommodate Unified CM locations-based CAC

- Minimize bandwidth usage on the WAN

This section also describes other call flow and bandwidth utilization issues to consider.

The following sections do not include detailed installation and configuration instructions. Rather, they are intended to provide you with guidance as you set up the Unified CVP solutions in your network. For additional information about how to install, set up, run, and administer Unified CVP, refer to **Installation and Upgrade Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html).

This section includes the following topics:

Configuring Unified CM Service Configuration Settings

Set the following configuration parameters to make Unified CM use the Ingress gateway instead of Unified CVP as the originating location of the call.

- Set "Accept Unknown TCP connection" in Unified CM Service parameters.
- Set the Unified CM Service parameter "GK controlled trunk that will listen to 1720" to “None”.
- *Do not* define Unified CVP as a gateway device in Cisco Unified Communications Manager.
- Define the Ingress gateway(s) as gateway devices in Unified CM. Assign the correct location to the device(s).

These settings will ensure that CAC can be properly adjusted based on the locations of the calling endpoint and the phone.

Configuring Ingress Gateway H.323 Settings

To help the Unified CVP H.323 Service identify the Unified CM cluster the Ingress gateway is registered to, a translation rule on the Ingress gateway is used. The rule will prefix a digit to the called number prior to initiating the call set-up to Unified CVP. The prefixed digits represent the tech-prefix number used by the Unified CM Cluster gatekeeper controlled trunk when registering with the gatekeeper.

Note: This technique is a special application of the tech-prefix call routing described in Configuring the H.323 Devices and VoIP (page 465)."

In the example below:

- The Unified CM cluster registers with tech-prefix 7#.
- The Ingress gateway prepends the digit **7** to the DNIS.
- The VBAdmin command **setSigDigits 1** strips off the first digit before passing the DNIS to Unified ICME.
- When the Unified ICME transfers the call to the agent, Unified CVP prepends **7#** to the transfer DNIS before initiating the transfer.
- The gatekeeper sees that the Unified CM cluster is registered with **7#**Unified CM cluster; CAC is adjusted correctly.

```
translation-rule 99
    Rule 1 ^6951000 76951000


dial-peer voice 99 voip
    destination-pattern 695100.
    translate-outgoing called 99
```

Configuring Unified CVP H.323 Service Configuration Settings

The following sections describe some of the Unified CVP H.323 Service configuration settings that you must make to accommodate Unified CM locations-based CAC and to minimize the bandwidth used in your telephony solution:

- Hairpinning
- Call signaling Address

You make these settings using the H.323 Service configuration and administration tool, called VBAdmin. For detailed information about VBAdmin and its command line interface, refer to "Administering the H.323 Service (page 313)."

Hairpinning

Configure the H.323 Service to hairpin a voice response unit (VRU) call leg to its originating gateway. In this way, the playing of prompts does not consume WAN bandwidth and the bearer path stays entirely within the originating gateway.

To make this configuration, enter the following command at the VBAdmin command line interface:

`SetTransferLabel` *NetworkRoutingNumber*

where you replace *NetworkRoutingNumber* with the label value that you configured in Unified ICME with the Type 3, 7, or 10 VRU.

Note: This command takes effect immediately. You do not need to restart the H.323 Service.

With this configuration, a call is handled as follows:

1. When the H.323 Service receives a call, it notifies Unified ICME that a call has arrived.
2. Unified ICME might return a transfer request to Unified CVP that instructs Unified CVP to send a second call (the *VRU call leg*) to an IOS Voice Browser for prompt and collect treatment.
3. When Unified CVP receives the transfer request, it sends an admission request (ARQ) to the gatekeeper, but it ignores the reply and instead sends the VRU call leg back to the originating gateway.

Call Signaling Address

Configure the H.323 Service to transfer calls to accommodate the Unified CM locations-based CAC.

To make this configuration, enter the following command at the VBAdmin command line interface:

`SetLocationsBasedCAC on`

Note: This command takes effect immediately. You do not need to restart the H.323 Service.

With this configuration:

- The H.323 Service forces the H.225 SETUP message to the Unified CM port 1720, which is the default H.225 listening port.
- The H.323 Service includes the IP address of the originating gateway in the ARQ and in the H.225 SETUP message that it sends to Unified CM.

In this way, Unified CM will identify calls as originating from the remote gateway instead of from Unified CVP and will adjust its CAC counters accordingly.

Unified CVP Bandwidth Utilization

The following factors contribute to WAN bandwidth usage by Unified CVP in a Centralized Call Control with Distributed Queuing call flow model:

- VoiceXML Documents (page 424)

- Prompt Retrieval (page 424)
- H.323 Signaling (page 425)

The following sections describe the bandwidth requirements of these factors in an example Centralized Call Control with Distributed Queuing call flow model. The examples in these sections are based on data that Cisco obtained from testing.

In these examples, assume that:

- Each call begins with some IVR treatment followed by a transfer to an agent.
- Each branch has 20 agents and each agent handles 30 calls per hour. Thus, the total number of calls is:

  20 * 30 = 600 calls per hour = 0.166 calls per second (CPS).

VoiceXML Documents

A VoiceXML document corresponds approximately to a Run External node in a Unified ICME script.

A round trip of a VoiceXML document between Unified CVP and the gateway consumes an average of 7 KB (7,000 bytes). If each call includes approximately 20 VoiceXML documents, the WAN bandwidth consumed by VoiceXML documents can be calculated as follows:

1. 7,000 bytes * 20 VoiceXML documents * 8 bits = 1,120,000 bits per call
2. 0.166 CPS * 1,120,000 bits per call = 185.9 Kbps per remote site

Prompt Retrieval

Voice prompts can be stored in the following locations:

- In flash memory on each local site gateway—In this way, gateways do not need to retrieve .wav files for prompts and WAN bandwidth is not affected. However, if a prompt needs to change, you must change it on every gateway.
- On an HTTP media server—In this way, each local site gateway (if properly configured) can cache many or all prompts, depending on the number and size of the prompts.

When prompts are stored on an HTTP media server, the refresh period for the prompts is defined on that server. The bandwidth consumed by prompts consists of the initial loading of the prompts at each gateway and of the periodic updates at the expiration of the refresh interval.

As an example of determining the bandwidth consumed by prompts, assume that a call flow has 50 prompts with an average size of 50 KB (50,000 bytes) each. Also, assume that the refresh period for the prompts is defined as 15 minutes (900 seconds) on the HTTP media server.

The WAN bandwidth required for prompts in this call flow can be calculated as follows:

1. 50 prompts * 50,000 bytes * 8 bits = 20,000,000 bits
2. 20,000,000 bits / 900 seconds = 22.2 Kbps per branch

H.323 Signaling Bandwidth

Each call that a local site gateway processes requires approximately 6,000 bytes, plus approximately 1,000 bytes each time the call is transferred to an agent. Thus, each call requires:

7,000 bytes * 8 bits = 56,000 bits per call

The bandwidth required for the WAN link to a remote branch can be calculated as follows:

0.166 CPS * 56,000 bits per call = 9.3 Kbps for the WAN link to a remote branch

Gateway Prompt Caching Considerations

When audio prompts are stored on an HTTP media server, proper gateway prompt caching methods are necessary to optimize both the performance of the gateway and network bandwidth consumption. Gateway performance decreases by approximately 35-40% if caching is disabled entirely.

How to Configure Caching on the Gateway

To configure caching on the gateway, do the following:

1. Set the following on the gateway:
   - ivr prompt memory 15000
   - http client cache memory file 500
   - http client cache memory pool 15000

   Note: The 'http client cache memory file' represents the largest size prompt file (in Kbytes) that can be cached. In general, break up customer prompts larger than 500K (about a minute in length) into smaller, more manageable pieces to facilitate loading and caching. For example, queue music could be a repetitive loop of a 30 second prompt. Note also that because the prompts are streamed, the prompt will not be cached unless the whole prompt is played. Therefore, you must make prompts a manageable size.

2. Synchronize the datetime between the gateway and the HTTP media server.

Note: Synchronization does not have to be exact, but at least within a minute or two. Times that are not synchronized can cause prompts to never refresh or they will refresh with every call, both of which are undesirable behaviors.

3. On the media server, set the content expiration (for example 15 minutes).

Note: In IIS, this is done under the "HTTP Header" tab. The gateway prompt will be refreshed after this time period. The period chosen reflects how often the record prompts and how long you are willing to wait to have the new prompt load after modification.

How to Determine if the Gateway is Caching Properly

To determine if you have properly configured gateway caching, do one of the following:

- The IIS log on the media server records every time a client requests a prompt. If caching is set up correctly, these requests appear approximately every X minutes (X is whatever was defined as the refresh interval in Step 3 above (page 426) for any particular prompt. The log is located at: **`C:\WINNT\system32\LogFiles\W3SVC1\ex*`**
- Do ‘show http client cache’ on the gateway. The 'Fresh Time' column equals the refresh time period set on the HTTP media server. For example, if the refresh period was set to 15 minutes, it says 900 seconds. The 'Age' column shows how many seconds have passed since the prompt was last refreshed. In general, this number will be less than the 'Fresh Time'. However, if no call has ever accessed the prompt recently, this number could be greater than the fresh time. Prompts are only refreshed when triggered by a call *and* the prompt 'Fresh Time' has expired. If the Fresh Time is a very high value, the only way to remove the prompt from cache is to reload the gateway.

Incoming UUI to be Used as the Correlation ID

Unified CVP uses the User-to-User Information (UUI) from the incoming call as a Correlation ID in the VRU-Only call flow model. This feature allows customers to transfer Correlation IDs through their network, for example, when using a Call Routing Service Protocol (CRSP) NIC for call control.

Note: This feature applies only to the Unified CVP VRU-Only call flow model.

The network typically has no place to store a Correlation ID so it must be "hidden" in the ISDN setup that arrives at the IOS gateway and then extracted by the gateway. The UUS parameter (often known as the User-to-User Information (UUI) of the Generic Transparency Descriptor (GTD) data) can be used to "hide" the Correlation ID, provided the call control client has the capability of inserting a Correlation ID value into the GTD.

When the call arrives at the gateway from the network, the call control client extracts the value and appends it to the DNIS before sending an HTTP request to the Type3 Unified CVP Call Server.

How It Works

The call control client (such as the CRSP NIC) inserts the desired Correlation ID value into the dat field of the UUS parameter of the NSS IAM message. These NSS messages are used as the basis of building the GTD data that ultimately arrives at the IOS gateway from the PSTN. Refer to the ITU-T Narrowband Signaling Syntax spec (Q.1980.1) for a detailed description of the IAM message and UUS parameter, included below for convenience. Note that the dat field contains pairs of hexadecimal digits. This means that if the Correlation ID is "12345", the dat field must be populated as "3132333435". The gateway bootstrap.tcl script converts back to "12345" before appending to the DNIS and passing to the Unified CVP Call Server in the HTTP URL.

Configuring the Gateway

Follow the instructions below to configure the gateway to enable incoming UUI to be used as the Correlation ID.

```
conf t
    application
        service <your-cvp-service-name>
            param use-uui-as-corrid Y (Refer to Note 1)
            param correlation-gtd-attribute XXX (Refer to Note 2)
            param correlation-gtd-instance N (Refer to Note 2)
            param correlation-gtd-field YYY (Refer to Note 2)
            dial-peer voice 123 pots
    service <your-cvp-service-name>
```

Note:

- This is a mandatory parameter to enable this feature.
- These parameters are optional. They need only be specified if the call control client placed the Correlation ID in a GTD parameter other than uus.dat.

Debugging Tips

The following two sections provide debugging tips.

Debug Trace Settings for the Gateway

On the gateway, enter the following:

```
debug voip application script
debug gtd
```

GTD Values in the Gateway Log

In the gateway log, look for the GTD values:

```
6616806: *Jan 31 17:12:41.220: cdapi_find_tsm:
Found a gtdmsg of length144:6616807: *Jan 31 17:12:41.220:
gtd msg = "IAM,PRN,isdn*,,
ATT5*,USI,rate,c,s,c,1USI,lay1,ulawTMR,00CPN,00
,,u,5900CPC,09FCI,,,,,,,y,UUS,3,3132333435

---> This is the UUI that will become the Correlation ID12345GCI,
87c0c79d91dd11daa9c4000bfda207f2"
```

Performing External Transfers in Unified ICME

Defining a Unified ICME Script Label for Outpulse Transfer

Labels in Unified ICME scripts for Unified CVP calls that require outpulse transfer mode must be prepended with the characters **DTMF** followed by ***8** and some number of commas, where each comma represents a pause of 100 milliseconds. By configuring the target label with the form **DTMFnnnnn** (where **nnnnn** are the digits to outpulse) Unified CVP sends the digits out-of-band using H.245 signaling to the Ingress gateway for outpulsing.

For example, to use the AT&T Transfer Connect feature to transfer the call to the number “4441234”, configure the label as **DTMF*8,,,4441234**.

Note:

- Usually the PSTN switch expects a delay between the *8 and the phone number. Each comma represents 100ms by default. It can be changed with the SetTakebackDelay command in VBAdmin.
- In outpulse transfer mode, Unified CVP will send whatever digits are in the label to the Gateway for outpulsing. It is the customer’s responsibility to confirm interoperability with the target switch.
- In your Unified ICME script when using outpulse transfers with SIP calls, digits can only be outpulsed on a call that has already been established. This means that it is necessary to transfer the call to the VXML gateway with a run external script node *before* you can send the DTMF*8 label. The Unified ICME script cannot send the DTMF*8 label back to Unified CVP for the first connect message in the call because the call has not been answered at this point. The Unified CVP Call Server uses SIP INFO messages to send the digits to the gateway for outpulsing.
- When using outpulse transfers with SIP, you can also use the comma duration as the default interdigit pause duration. For example, with the default 100 msec comma duration, a label such as "DTMF*8,,,8009785001" will have 300 msecs between the first 8 and the second 8.

The interdigit pause will also be 100 msecs.The tone duration is also configurable and defaults to 100 msecs.

- Outpulse transfer with SIP utilizes SIP INFO messages being sent to the TDM gateway, where the outpulsing of digits occurs. If the agent using the CTI desktop performs a blind transfer (single step transfer), and the scheduled script for the transfer DN returns a DTMF type label, the CUCM SIP Trunk can loop the CVP DTMF label through the bridged call using an UPDATE message, and then Unified CVP can get the label back and convert the digits to SIP INFO messages to forward to the ingress gateway. This only works on blind transfers, and is not supported on consult transfers.

Defining a Unified ICME Script Label for Two B-Channel Transfer (TBCT)

For Unified CVP calls that require Two B-Channel Transfer (TBCT) mode, add a label node to your Unified ICME script with the following syntax:

`TBCT99#8005551212#`

Replace 8005551212 with your transfer destination target; **`TBCT99`** and the **`#`** sign are mandatory.

By configuring the target label in this form, Unified CVP sends the digits to the Ingress endpoint for Two B-Channel transfer.

Defining a Unified ICME Script Label for Hookflash Transfer

Labels in Unified ICME scripts for Unified CVP calls that require hookflash transfer mode must be prepended with the characters **`HF`**. By configuring the target label with the form **`HFnnnnn`** (where **`nnnnn`** are the digits to call) Unified CVP sends the digits to the Ingress endpoint for hookflash transfer.

If the switch requires a pause after the hookflash, insert commas between the **`HF`** and the transfer number. (Each comma represents 100 milliseconds.)

For example, to use the hookflash feature to transfer the call to the number “4441234” with a 500- millisecond pause after the hookflash, configure the Unified ICME label as **`HF,,,,,4441234`**.

Using Takeback and Transfer in VoiceXML Scripts

Unified CVP provides the following takeback and transfer methods that you can invoke from a VoiceXML script:

- Two B-Channel Transfer (TBCT) - A call transfer standard for ISDN interfaces. This feature enables a Cisco voice gateway to request an NI-2 switch to directly connect two independent calls. The two calls can be served by the same PRI or by two different PRIs on the gateway.

- Hookflash Relay - A brief interruption in the loop current that the originating call entity (PBX or Public Switch Telephone Network switch) does not interpret as a call disconnect. Instead,

once the PBX or Public Switch Telephone Network switch senses the hookflash, it puts the current call on hold and provides a secondary dial tone, which allows Unified CVP to transfer the caller to another destination.

- SIP Refer - VoiceXML applications can use a SIP REFER transfer instead of a blind or bridged transfer. This allows Unified CVP to remove itself from the call, thus freeing up licensed Unified CVP ports. Unified CVP cannot execute further call control or IVR operations after this kind of label has been executed.

Configuring Two B-Channel Transfer (TBCT)

To configure Two B-Channel Transfer (TBCT) for use with Unified CVP from a VoiceXML script, follow this procedure:

Step 1 Configure the originating gateway for TBCT call transfer, by following the instructions in "Configuring the Gateway for Two B-Channel Transfer (TBCT) (page 435)."

Step 2 Locate the following files on the VXML Server and copy them to flash memory on the gateway, using the tftp command:

en_error.wav

en_holdmusic.wav

en_pleasewait.wav

survivability.tcl

CVPSelfService.tcl

CVPSelfServiceBootstrap.vxml

Step 3 Add the following lines to the gateway:

```
service takeback flash:survivability.tcl
  param icm-tbct 1
```

Step 4 Configure the CVPSelfService application, as follows:

```
service [gateway application name] flash:CVPSelfService.tcl
  param CVPBackupVXMLServer 12.34.567.890
  param CVPSelfService-port 7000
  param CVPSelfService-app [name of application on the VXML Server,
exactly how it appears]
  param CVPPrimaryVXMLServer 12.34.567.891
```

Note: CVPSelfService is required. Backup server is optional. For both the Tomcat Application Server and the WebSphere Application Server, set the port to 7000.

Step 5 From command line mode:

```
call application voice load takeback
call application voice load CVPSelfService
```

`call application voice load CVPSelfService`

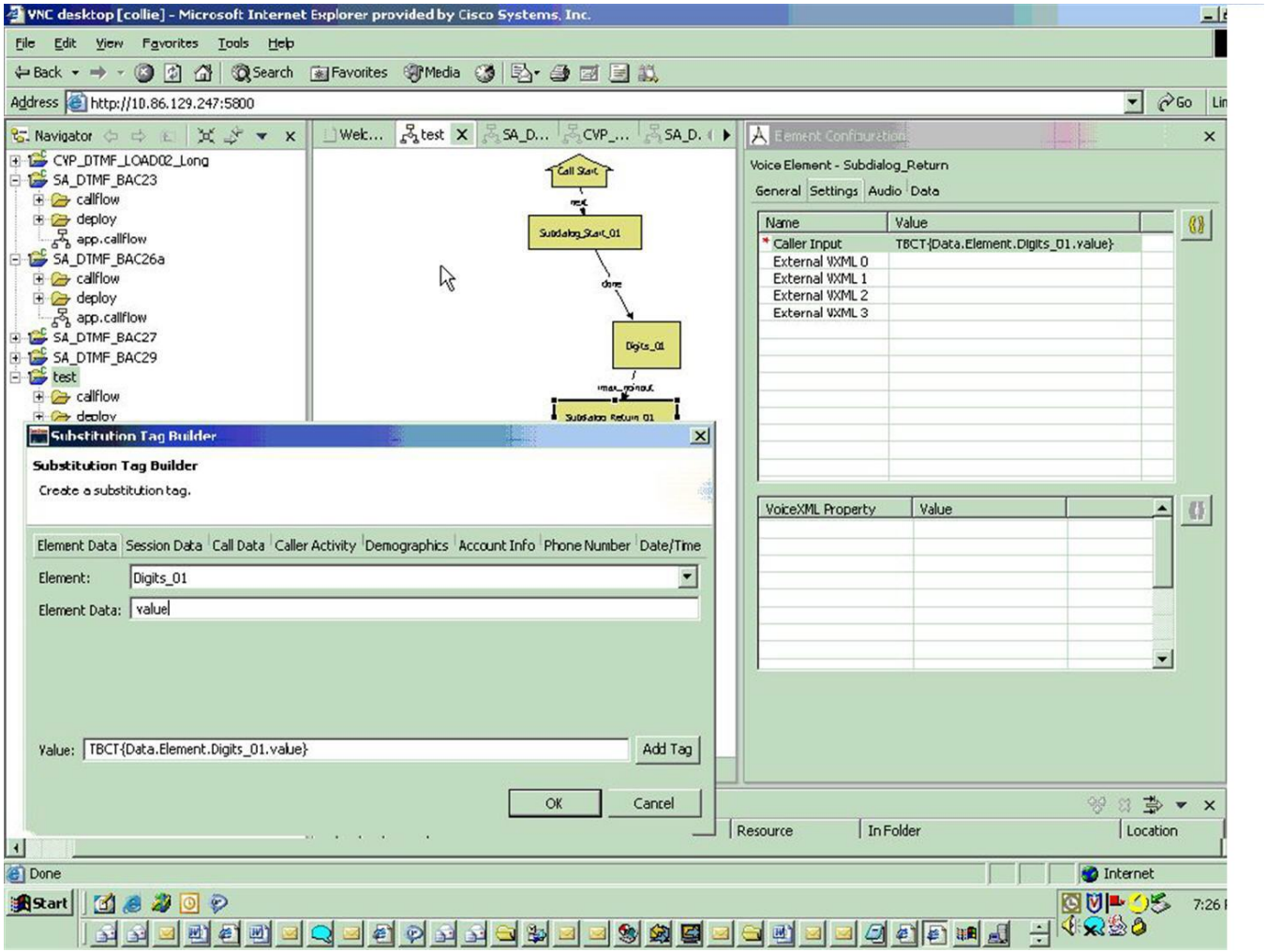
Step 6 Specify the target destination for the TBCT transfer either by entering the number manually, or dynamically by using caller input.

a. **Manually**. In the SubdialogReturn node in the VXML Server application, next to Caller Input in the Settings Tab, enter **`TBCT<target_destination_number>`**, where *target_destination_number* is the target destination of the TBCT transfer. For example:

TBCT8005551212

b. **Dynamically**. The target destination is created dynamically using input entered by the caller during the call. To do this, click the **Substitution** icon next to the Caller Input variable and select substitution values. For example:

Figure 67: Dynamic Target Destination



Step 7 In the event that the TBCT fails (for example, bad number, bad gateway config, etc.), you can specify alternate transfer targets under the survivability service according to survivability rules

as defined in the "Call Survivability (page 398)" section in Chapter 19, "Configuring the H.323 Devices and VoIP," in Part 3.

Configuring Hookflash Relay

To configure Hookflash Relay for use with Unified CVP from VoiceXML scripts, follow this procedure:

Step 1 Configure the originating gateway for Hookflash Relay call transfer, by following the instructions in the "Configuring the Gateway for Hookflash Relay (page 434)" section in Chapter 18, "Transferring and Queuing Calls with Unified CVP," in Part 3.

Step 2 Locate the following files on the VXML Server and copy them to flash memory on the gateway.

en_error.wav

en_holdmusic.wav

en_pleasewait.wav

survivability.tcl

en_0.wav en_1.wav

en_2.wav en_3.wav

en_4.wav

en_5.wav

en_6.wav

en_7.wav

en_8.wav

en_9.wav

en_pound.wav

en_star.wav

Step 3 Add the following lines to the gateway:

```
service hookflash flash:survivability.tcl
```

Step 4 If you have not already done so, configure the CVPSelfService application as follows:

```
service [gateway application name] flash:CVPSelfService.tcl
  param CVPBackupVXMLServer 12.34.567.890
```

```
  param CVPSelfService-port 7000
  param CVPSelfService-app [name of application on the VXML Server,
exactly how it appears]
  param CVPPrimaryVXMLServer 12.34.567.891
```

Note: CVPSelfService is required. Backup server is optional. For both the Tomcat Application Server and the WebSphere Application Server, set the port to 7000.

Step 5 From command line mode:

```
call application voice load hookflash
call application voice load CVPSelfService
```

Step 6 In the SubdialogReturn node in your VXML Server application, next to Caller Input in the Settings Tab, enter **`HF8005551212`**, replacing 8005551212 with the target destination of the hookflash transfer.

The label can also be defined dynamically using digits entered by the caller in conjunction with the VXML Server substitution tags. If the switch requires a pause after the hookflash, commas can be inserted between the HF and the transfer number. Each comma represents 100ms.

Configuring SIP REFER

To configure SIP REFER for use with Unified CVP from a VoiceXML script, follow this procedure:

Step 1 Configure the gateway according to either "Configuring the VXML Server (Standalone) Call Flow Model (without ICM Lookup) (page274)" or "Configuring the VXML Server (Standalone) with ICM Lookup Call Flow Model (page 272)," whichever is appropriate for the implementation.

Note: The incoming dial-peer running the CVPSelfService application must be a voip dial-peer, not a pots dial-peer. Additionally, the IOS image that is used must contain the IPIPGW feature. SIP REFER will not work with the IPIPGW feature.

Step 2 Specify the target destination for the refer transfer in the Call Studio application either by entering the number manually, or dynamically using caller input.

a. **Manually**. In the SubdialogReturn node in the VXML Server application, next to CallerInput in the Settings tab, enter **`RF<target_destination_number>`**, where target_destination_number is the target destination of the REFER transfer; for example, RF8005551212.

b. **Dynamically**. The target destination is created dynamically using input entered by the caller during the call. To do this, click the **Substitution** icon next to the Caller Input variable and select the substitution values.

Restart to Alternate DNIS for H.323

In the event that the called party disconnects before the caller, and no further instructions are sent from Unified ICME within 2 seconds, the H.323 Service checks a list of Dialed Number Identification Services to see if the call should be restarted to an alternate DNIS number. The called party can be any Unified CVP transfer agent, including, a TDM IVR, an Unified CCE agent, or even the transfer to the IVR leg in the Comprehensive call flow model. The H.323 Service does not differentiate the type of transfer target that disconnected.

The original call and the restarted call to Unified ICME each contain the same call ID (media_id ECC variable) so that the Unified ICME scripts can correlate the two calls for database query purposes.

The timeout value that the caller waits before being connected to the post-call Unified ICME script can be changed using the SetCalledPartyTimeout command in VBAdmin. The default value is 2 seconds.

The H.323 Service can distinguish between a called party who hangs up normally or a called party who is disconnected abnormally, such as that caused by a crash of the target IOS gateway or Unified CM. In the event of an abnormal disconnect of the called party, the H.323 Service will not invoke the SurveyDnis restart unless SetNewCallOnly command in VBAdmin is set to 'on' (default is 'off'). If Unified CVP surviviability.tcl functionality is being used, it may be desirable to simply let the call default to the recovery actions defined in survivability rather than restarting the call to the surveyDnis.

To limit the possibility of putting the caller into an infinite restart loop, the call can be restarted only once.

Note: For the SIP version of this feature, refer to Post Call Survey for SIP (page 381).

Configuring the Gateway for Hookflash Relay

Note: Hookflash transfers with Unified CVP are supported only on 2X and 3X gateways. They are not supported on 4X gateways due to the fact that the 5X gateway cannot generate a line-side hookflash.

Most switches and Private Branch Exchange (PBX) networks support only line-side hookflash, as opposed to trunk-side hookflash). (A line is an electrical connection between a telephone service provider's switch and a telephone terminal or Key system. A trunk is an electrical connection between a telephone service provider's switch and another switch. Line-side hookflash limits the signaling options available on the gateway to only digital or analog FXO signaling.

The following example shows a sample gateway configuration for T1/CAS FXO signaling. Only the pertinent gateway configuration elements are shown.

Example: Gateway Configuration for Hookflash Relay

```
controller T1 2/0
framing esf
linecode b8zs
```

```
ds0-group 1 timeslots 1-24 type fxo-loop-start

voice-port 0/1/0:1
output attenuation 0
timing hookflash-out 600 */ Can be adjusted for the specific needs of
 PBX /*

dial-peer voice 11 pots
port 0/1/0:1
forward-digits all */ Necessary to forward DNIS to CVP for T1/CAS FXO
 /*
```

Configuring Hookflash Relay When Using a Unified ICME Label Node

To configure hookflash relay:

Step 1 Follow the gateway configuration instructions as described in Configuring the Gateway for Hookflash Relay (page 434).

Step 2 Add a label node to your Unified ICME script with the following syntax:

HF8005551212

Replace *8005551212* with the number to which the switch will transfer the call. If the switch requires a pause after the hookflash, insert commas between the HF and the transfer number. Each comma represents 100 milliseconds.

Note: Unified CVP automatically disconnects the call two seconds after sending the digits. If the switch needs more time to complete the hookflash sequence, you can increase the delay by appending more commas after the transfer number to extend the two-second timeout.

Configuring Hookflash Relay for Survivability

To configure hookflash relay for survivability:

Step 1 Configure the originating gateway for hookflash relay call transfer, by following the instructions in Configuring the Gateway for Hookflash Relay (page 434)" .

Step 2 Follow the instructions for configuring survivability in "How to Configure the Gateway for Call Survivability (page 400)." Refer to the "HF" target for open-hours-agent and after-hours-agent.

Configuring the Gateway for Two B-Channel Transfer (TBCT)

The originating gateway and the PSTN switch must be configured for TBCT:

Step 1 Save a copy of the current running configuration to flash memory in case you need to roll back to the earlier version.

Step 2 If needed, upgrade the IOS version on the originating gateway. Consult the *Hardware and Software System Specification for Cisco Unified Customer Voice Portal Software Release 7.0(1)* for the required version of Cisco IOS on the Gateway.

Step 3 Configure trunks, switches, and dial-peers to process TBCT call transfers.

Consult your service provider for switch configuration details as they will vary between switches (DMS100, 5ESS). TBCT supports the National ISDN-2 (NI-2) standard. One sample originating gateway configuration is shown in the Gateway Configuration for Two B-Channel Transfer example, below. Only the pertinent gateway configuration elements are shown. For more information on the TBCT feature, refer to *Cisco IOS Tcl IVR and VoiceXML Application Guide*.

The following is an example of a gateway configuration for Two B-Channel Transfer:

```
isdn switch-type primary-ni

trunk group 1
isdn supp-service tbct notify-on-clear

interface Serial6/7:23
no ip address
trunk-group 1
isdn switch-type primary-ni
isdn incoming-voice modem
no cdp enable

dial-peer voice 111111 pots
trunkgroup 1
service someName
destination-pattern 978.......
incoming called-number 35500
```

Note:

- The trunk group is what ties the incoming and outgoing call legs together.
- The notify-on-clear parameter is optional. When set on both the gateway and the switch, the gateway will monitor the status of the call if required for billing purposesThe notify-on-clear parameter is only available on 5XXX gateways.
- You can use the "tbct max" gateway configuration parameters to limit the number of TBCT calls. For more information on the TBCT feature, refer to *Cisco IOS Tcl IVR and VoiceXML Application Guide*. The tbct max parameter is only available on 5XXX gateways.

Debugging TBCT Gateway Configuration Problems

Note: If the gateway is instructed to perform a TBCT transfer (either from a Unified ICME script label, a VXML Server label, or from survivability), if the gateway trunk group and dial-peer

are not correctly configured for TBCT according to the instructions in this guide, the caller will be disconnected when the called party answers the phone

The following are useful settings for debugging TBCT Gateway configuration problems:

- **`debug isdn q931`**
- **`debug isdn err`**
- **`debug cdapi event`**
- **`debug voip application script`**

Verifying that TBCT Is Working Correctly

The TBCT feature is working correctly if:

- The caller was connected to the transfer destination, and
- The gateway log (using the above debug settings) shows the following (abbreviated) messages:

```
014349: *Jul 4 11:20:23.483: ISDN Se6/7:23 Q931: RX <- ALERTING
014387: *Jul 4 11:20:23.487: ISDN Se6/7:23 Q931: TX -> FACILITY
Operation = EnhancedExplicitECTExecute (TBCT)
014388: *Jul 4 11:20:23.587: ISDN Se6/7:23 Q931: RX <- FACILITY
Operation = SetCallTag (TBCT)
014434: *Jul 4 11:20:23.599: //-1//TCL2:HN0B1453B0:/tcl_PutsObjCmd:
******* CallID = F2B1AAC0.EBD511D9.803D000B.FDA20840
```

Configuring TBCT When Using a Unified CVP Label Node

1. Configure the originating gateway for TBCT call transfer by following the instructions in the "Configuring the Gateway for Two B-Channel Transfer (TBCT) (page 435)" section.
2. Configure your Unified CVP incoming pots dial-peer to use the Unified CVP survivability Tcl script. Refer to "How to Configure the Gateway for Call Survivability (page 400)." If survivability is already configured on your gateways, go to the next step

```
dial-peer voice 111111 pots
trunkgroup 1
service someName
dial-peer voice 222222 voip
session target [your Unified CVP H.323 Service]

service someName flash:survivability.tcl
param icm-tbct 1
```

3. In command line mode:

   `'call application voice load someName'`

4. Add a label node to your Unified ICME script with the following syntax:

```
TBCT99#8005551212#
```

replacing 8005551212 with your transfer destination target. Note that the TBCT99 and the # signs are mandatory.

5. In the event that the TBCT fails (for example, bad number, bad gateway config, et cetera), you can specify alternate transfer targets under the survivability service according to survivability rules as defined in "How to Configure the Gateway for Call Survivability (page 400)."

Configuring TBCT for Survivability

1. Configure the originating gateway for TBCT call transfer, by following the instructions in "Configuring the Gateway for Two B-Channel Transfer (TBCT) (page 435)."
2. Follow the instructions for configuring survivability in "How to Configure the Gateway for Call Survivability (page 400)." Refer to the "TBCT" target for open-hours-agent and after-hours-agent.

Transferring Calls with the Select Node

A Unified ICME script can use a combination of a Select node and ECC variables to perform a transfer within Unified CVP. The label includes information to indicate the type of transfer to be performed as well as the number (or representation of the number) to use for the destination. The Call Server then forwards this information to the H.323 Service.

Since the transfer is happening as a result of a Select node, there is no way for Unified ICME or Unified CVP's software to check if the customer label and associated variable is valid or not. Therefore, if the transfer cannot occur, no matter what the reason, the call flow diverts to the Select node error branch.

Label Definition

Labels are part of the Unified ICME configuration. Unified ICME determines which label to send and includes it in the request to transfer the caller. Then, for IP transfers, Unified CVP does a Gatekeeper look-up on the label from Unified ICME to determine the IP address of the called party. For more information on this process, refer to "Configuring the H.323 Devices and VoIP (page 465)."

There are some caveats regarding label definition for use with Unified CVP:

- For requery and network transfer to work, you must define "dummy" VRU labels, even if Unified CVP is the main routing client. In addition, in a NAM/CICM configuration, these labels must be defined identically on both the NAM and CICM.
- You cannot use Unified ICME's feature of substitution variables (such as %1).

  Labels for Unified CVP calls which will require outpulse transfer mode must be pre-pended with the characters DTMF. By configuring the target label with the form DTMF*nnnnn* (where

nnnnn are the digits to outpulse), Unified CVP will send the digits to the Ingress gateway for outpulsing. (Conversely, a label not intended for outpulse transfer mode use cannot begin with DTMF.)

For example, to use the AT&T Transfer Connect feature to transfer the call to the number "4441234" the label would be configured as DTMF*84441234.

Note: In outpulse transfer mode, Unified CVP will send whatever digits are in the label to the Gateway for outpulsing. It is the customer's responsibility to confirm interpretability with the target switch.

Courtesy Callback

The Courtesy Callback feature is available in Unified CVP starting in Release 8.0(1). Courtesy Callback reduces the time callers have to physically wait on hold or in a queue. The feature enables your system to offer callers who meet your criteria the option to receive a courtesy callback by the system instead of waiting on the phone for an agent. The caller who has been queued by Unified CVP can hang up and subsequently be called back when an agent is close to becoming available (preemptive callback).

Preemptive callback does not change the time a customer must wait to be connected to an agent, but rather enables the caller hang up and not be required to remain in queue listening to music. Callers who have remained in queue or have undergone the callback treatment appears the same to agents answering the call.

If the caller decides to be called back by the system, they leave their name and phone number. Their request remains in the system and when the system determines that an agent will be available soon (or is available), then the system places a call back to the caller. The caller answers the call and confirms that they are the original caller and the system connects the caller to the agent after a brief wait.

In the event that the caller cannot be reached after a configurable max number and frequency of retries, the callback is aborted and the database status is updated appropriately. You can run reports to determine if any manual callbacks are necessary based on your business rules.

Note:

- You cannot schedule a callback for a specific time.
- There are a number of prerequisites and design considerations for using this feature. Be sure to review them in the Cisco Unified Customer Voice Portal Solution Reference Network Design guide.
- The **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) contains a *typical use scenario* section that walks through a caller experience with the callback process, including alternate results for different caller decisions.

Callback Criteria

In your callback script, you can establish criteria for offering a caller a courtesy callback. Examples of callback criteria include:

- Number of minutes a customer is expected to be waiting *in queue* exceeds some maximum number of minutes (based on your average call handling time per customer)

  Note: The included example scripts use this method for determining callback eligibility.

- Assigned status of a customer (*gold* customers may be offered the opportunity to be called back instead of remaining on the line)

- The service a customer has requested (sales calls, or system upgrades, etc. may be established as callback criteria)

Modifiable Example Scripts and Sample Audio Files

The courtesy callback feature is implemented using Unified ICM scripts. Modifiable example scripts are provided on the Unified CVP 8.0(1) install media in the **`\CVP\Downloads and Samples\`** folder. These scripts determine whether or not to ofer the caller a callback, depending on the callback criteria (previously described).

The files provided are:

- **CourtesyCallback.ICMS**, the ICM script

- **CourtesyCallbackStudioScripts.zip**, a collection of Call Studio scripts

  The following example scripts are provided:

  – **BillingQueue:** Plays queue music to callers. Can be customized.

  – **Callback Engine:** Keeps the VoIP leg of the call alive when the caller elects to receive the callback (and hangs up) and when the caller actually receives the callback. Do **not** modify this script.

  – **CallbackEntry:** Initial IVR when caller enters the system and is presented with opportunity for a callback. Can be customized.

  – **CallbackQueue:** Handles the keepalive mechanism for the call when callers are in queue and listening to the music played by BillingQueue. Do **not** modify this script.

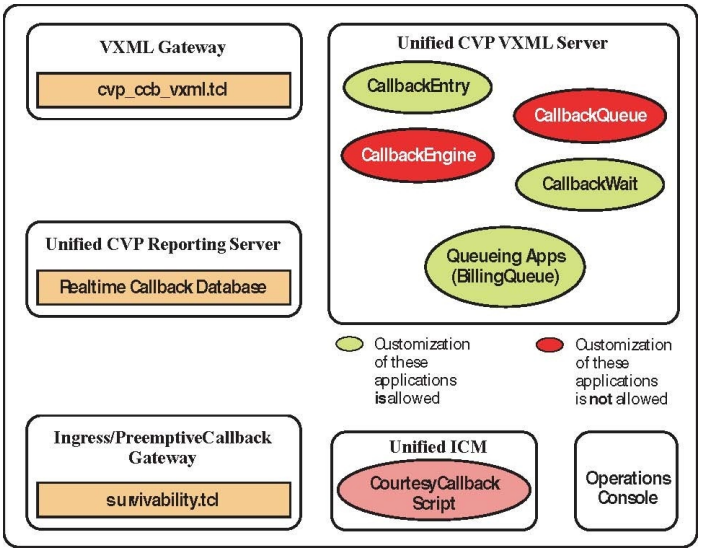
  – **CallbackWait:** Handles IVR portion of call when caller is called back. Can be customized.

Sample audio files that accompany the sample studio scripts are installed to the **%CVP_Home%\OPSConsoleServer\CCBDownloads** folder as **CCBAudioFiles.zip** and also as part of the Media Files installation option.

Courtesy Callback Configuration

The following diagram shows the components that must be configured for Courtesy Callback:

Figure 68: Courtesy Callback Components



The Courtesy Callback feature must be configured on the following servers/gateways:

- Ingress Gateway (IOS configuration)
- VXML Gateway (IOS configuration)
- Reporting Server (through the Unified CVP Operations Console)
- Media Server (upload of Courtesy Callback media files)
- VXML Server (upload of Call Studio Scripts)
- Unified ICM (through the ICM script)

Configuring the Ingress Gateway for Courtesy Callback

The ingress gateway at which the call arrives is the gateway that processes the preemptive callback for the call, if the caller elects to receive a callback.

Note: Only TDM-originated calls can be offered a courtesy callback.

To configure an ingress gateway for Courtesy Callback:

Step 1 Copy **`surviability.tcl`** from the Operations Console to the flash memory of the gateway. Using the Operations Console:

a. Select: **Bulk Administration > File Transfer > Scripts and Media**

b. In Device Association, for *Select Device Type* select: **Gateway**

c. From the default gateway files, highlight: **survivability.tcl**.

d. Click the **right arrow icon** to move the file to the Selected pane.

e. Click **Transfer**.

Step 2 Log into the ingress gateway.

Step 3 If survivability is not already configured, configure it as described in Call Survivability (page 398).

Step 4 Add the following to the survivability service:

```
param ccb id:<host name or ip of this gateway>:loc:<location
name>;trunks:<number of callback trunks>
```

Where the definitions of the preceding fields are:

- **id:** A unique identifier for this gateway and is logged to the database to show which gateway processed the original callback request.
- **loc:** An arbitrary location name specifying the location of this gateway.
- **trunks:** The number of DS0's reserved for callbacks on this gateway. Limit the number of T1/E1 trunks to enable the system to limit the resources allowed for callbacks.

The following example shows a basic configuration:

```
service cvp-survivability flash:survivability.tcl
 param ccb id:10.86.132.177;loc:doclab;trunks:1
!
```

If you are updating the survivability service, or if this is thefirst time you created the survivability service, remember to load the application using the command:

```
call application voice load cvp-survivability
```

Step 5 Create the incoming POTS dial peer, or verify that the survivability service is being used on your incoming POTS dial peer. For example:

```
dial-peer voice 978555 pots
 service cvp-survivability
 incoming called-number 9785551234
 direct-inward-dial
!
```

Step 6 Create outgoing POTS dial peers for the callbacks.These are the dial peers that place the actual call back out to the PSTN. For example:

```
dial-peer voice 978555 pots
 destination-pattern 978555....
 no digit-strip
 port 0/0/1:23
!
```

Step 7 Use the following configuration to ensure that SIP is set up to forward SIP INFO messaging:

```
voice service voip
 signaling forward unconditional
```

Configuring the VXML Gateway for Courtesy Callback

To configure the VXML gateway for Courtesy Callback:

Step 1 Copy **cvp_ccb_vxml.tcl** from the Operations Console to the flash memory of the gateway. Using the Operations Console:

a. Select: **Bulk Administration > File Transfer > Scripts and Media**

b. In Device Association, for *Select Device Type* select: **Gateway**

c. From the default gateway files, highlight: **cvp_ccb_vxml.tcl**.

d. Click the **right arrow icon** to move the file to the Selected pane.

e. Click **Transfer**.

Step 2 Add the cvp_cc service to the configuration:

`service cvp_cc flash:cvp_ccb_vxml.tcl`

The service does not require any parameters.

Load the application with the command:

`call application voice load cvp_cc`.

Step 3 On the VoIP dial-peer that defines the VRU leg from Unified ICM, verify that the codec used, can be used for recording; the following example verifies that **`g711ulaw`** can be used for recording in Courtesy Callback:

```
dial-peer voice 123 voip
 service bootstrap
 incoming called-number 123T
 dtmf-relay rtp-nte h245-signal h245-alphanumeric
 codec g711ulaw
 no vad
!
```

Step 4 Use the following configuration to ensure that SIP is setup to forward SIP INFO messaging:

```
voice service voip
 signaling forward unconditional
```

Step 5 VXML 2.0 is required to play the beep to prompt the caller to record their name in the *BillingQueue* example script. Add the following text to the configuration so the VXML Server uses VXML 2.0:

```
vxml version 2.0
```

Note: Whenever *vxml version 2.0* is enabled on the gateway,*vxml audioerror* is **off** by default. When an audio file cannot be played, error.badfetch will **not** generate an audio error event. To have the gateway generate an error.badfetch event when a file cannot be played, enable *vxml audioerror* in your gateway configuration. The following example uses "config terminal mode" to add both commands:

```
config t
vxml version 2.0
vxml audioerror
exit
```

Configuring the Reporting Server for Courtesy Callback

A Reporting Server is required for the Courtesy Callback feature. The Reporting Server must be installed and configured prior to completing the following task.

- **Installation:** If you have not **installed** a Reporting Server, refer to **Installation and Upgrade Guide for Cisco Unified Customer Voice Portal** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/tsd_products_support_series_home.html).
- **Configuration:** If you have not configured a Reporting Server in the Operations Console, refer to the Operations Console Online Help,**Managing Devices > Configuring a Reporting Server**.

Once you have added the Reporting Server, configure the Reporting Server for courtesy callback as follows:

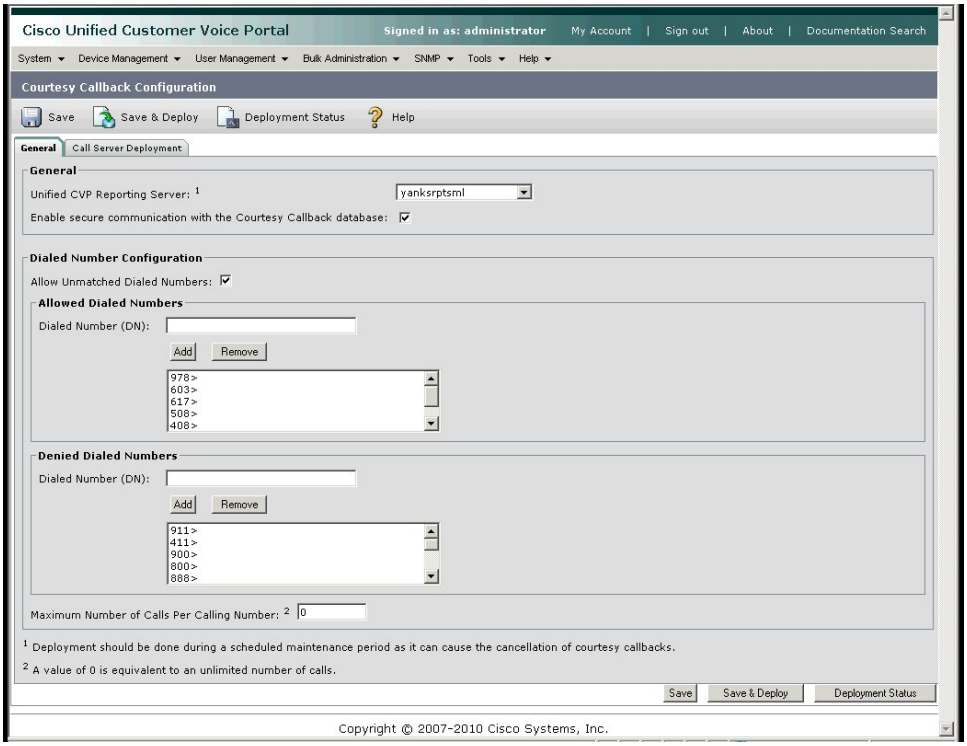
Step 1 In the Operations Console, select **System > Courtesy Callback**. The *Courtesy Callback Configuration* page displays.

From this window, on the General tab you can:

- Select the Reporting Server for Courtesy Callback
- Enable secure communication with the Courtesy Callback database
- Configure allowed and disallowed dialed numbers

These operations are described in the following steps.

Figure 69: Courtesy Callback Operations Console Main Window



Step 2 On the Courtesy Callback Configuration page, click the **Unified CVP Reporting Server drop-down**, and select the Reporting Server to use for storing Courtesy Callback data.

Note: If you do not have a Reporting Server configured, refer to the notes at the beginning of this procedure.

Step 3 If desired, enable secure communication with the callback reporting database. Check **Enable secure communication with the Courtesy Callback database**.

Step 4 Configure allowed and disabled dialed numbers. These are the numbers that the system **should and should not** call when it is making a courtesy callback to a caller.

Initially, there are **no** allowed dialed numbers for the Courtesy Callback feature. That is:

- Allow Unmatched Dialed Numbers is **deselected**.
- And, the Allowed Dialed Numbers window is empty.

This initial configuration is intentional; you must specifically enable the dialed numbers allowed for your deployment.

If you wish to allow **all** dialed numbers **except** those that are specifically listed in the *Denied Dialed Numbers* box, check **Allow Unmatched Dialed Numbers**.

Otherwise, add specific allowed number to the Allowed Dialed Numbers box. Refer to the Operations Console online help for details, and for valid dialed number shortcut patterns that are allowed.

Note: The Denied Dialed Numbers window is prepopulated if your local language is "en-us"(United States, English). Be sure to add any additional numbers you want to deny.

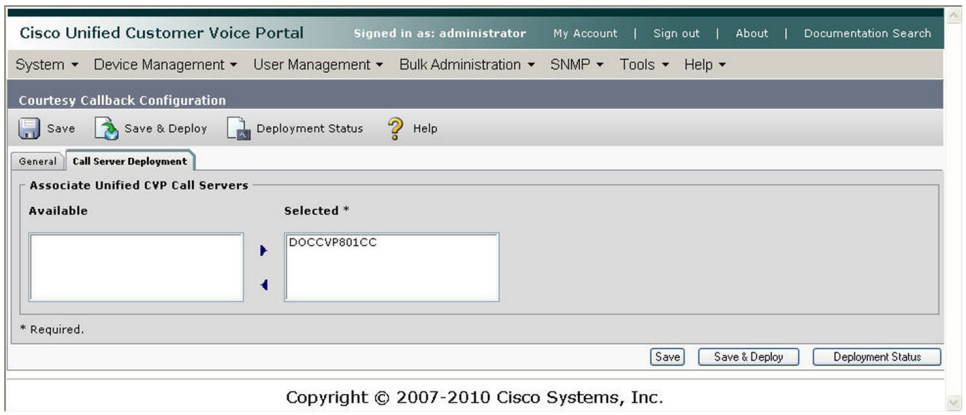
- Wildcarded DN patterns can contain "." and "X" in any position to match a single wildcard character.
- Any of the wildcard characters in the set ">*!T" will match multiple characters but can only be used trailing values because they will always match all remaining characters in the string.
- The highest precedence of pattern matching is an exact match, followed by the most specific wildcard match.
- When the number of characters are matched equally by wildcarded patterns in both the Allowed Dialed Numbers and Denied Dialed Numbers lists, precedence is given to the one in the Denied Dialed Numbers list.

Step 5 Adjust the "Maximum Number of Calls per Calling Number" to the desired number. By default, this is set to 0 and no limit is imposed.

This setting allows you to limit the number of calls, from the same calling number, that are eligible to receive a callback. If this field is set to a positive number (X), then the courtesy callback "Validate" element only allows X callbacks per calling number to go through the "preemptive" exit state at any time. If there are already X callbacks offered for a calling number, new calls go through the "none" exit state of the "Validate" element. In addition, if no calling number is available for a call, the call always goes through the "none" exit state of the "Validate" element."

Step 6 Click the **Call Server Deployment** tab and move the Call Server you want to use for courtesy callbacks from the *Available* box to the *Selected* box, as shown in the following screen shot

Figure 70: Courtesy Callback Select Reporting Server



Step 7 Click **Save** to save the configuration. The configuration becomes active (is deployed) the next time the Reporting Server is restarted. You can also deploy the new Reporting Server configuration immediately by clicking **Save & Deploy**.

Note: If you are updating the courtesy callback configuration (for example, changing to a different Reporting Server), perform deployment during a scheduled maintenance period. Otherwise, restarting the Reporting Server could cause the cancellation of currently scheduled courtesy callbacks.

Configuring the Media Server for Courtesy Callback

Several Courtesy-Callback-specific media files are included with the sample scripts for Courtesy Callback. During the Unified CVP installation , these files are copied as:

%CVP_HOME%\OPSConsoleServer\CCBDownloads\CCBAudioFiles.zip

CCBAudioFiles.zip has callback-specific application media files under *en-us\app* and media files for Say It Smart under *en-us\sys*.

The special audio files should be unzipped and copied to your media server.

Note:

- If you selected the Media File installation option, during the Unified CVP install, the audio files were unzipped and copied to **C:\inetpub\wwwroot\en-us\app** on the installation server.
- CCBAudioFiles.zip also contains media files for Say It Smart. During installation, these files are copied to **C:\inetpub\wwwroot\en-us\sys**. Copy these files to your media server, if you do not have them there already.
- If you are using the VXML Server as a media server (a lab environment configuration), then copy all of the files to **`VXMLServer\Tomcat\webapps\CVP\audio`**.
- The sample scripts are set up to use the default location of "http://<server>:<port>/en-us/app" for the audio files. Later in this configuration process you will change the <server> and <port> parameters in the default location of the audio files in the example scripts to be your media server IP address and port number.

Configuring Call Studio Scripts for Courtesy Callback

The Courtesy Callback feature is controlled by a combination of Call Studio scripts and ICM scripts. Refer to **Cisco Unified Customer Voice Portal Release Solution Reference Network Design (SRND)** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_implementation_design_guides_list.html) for a discussion of the script logic.

To configure the Call Studio scripts:

Note: This example follows the *BillingQueue* example application.

Step 1 Extract the example Call Studio Courtesy Callback scripts contained in **CourtesyCallbackStudioScripts.zip** to a folder of your choice on the **computer running Call Studio**.

You can access the .zip file from the following two locations:

- From the Unified CVP install media in **\CVP\Downloads & Samples\Studio Samples\CourtesyCallbackStudioScripts.zip**

- From the Operations Console server in **%CVP_HOME%\OPSConsoleServer\StudioDownloads**.

Step 2 The extraction process should have copied the following folders. Each folder contains a Call Studio project having the same name as the folder. The five individual project comprise the Courtesy Callback feature.

Do **not** modify the following scripts:

- **CallbackEngine:** Keeps the VoIP leg of the call alive when the caller elects to receive the callback (and hangs up) and when the caller actually receives the callback. Do **not** modify this script.
- **CallbackQueue:** Handles the keepalive mechanism for the call when callers are in queue and listening to the music played by BillingQueue. Do **not** modify this script.

Modify the following scripts to suit your business needs:

- **BillingQueue:** Determines the queue music played to callers.
- **CallbackEntry:** Modify the initial IVR treatment a caller receives when entering the system and is presented with opportunity for a callback.
- **CallbackWait:** Modify the IVR treatment a caller receives when they respond to the callback.

Step 3 Start Call Studio by selecting **Start > Programs > Cisco > Cisco Unified Call Studio**.

Step 4 In Call Studio select **File > Import**.

Step 5 In the Import dialog box, expand the **Call Studio** folder and select **Existing Call Studio Project Into Workspace**

Step 6 Click **Next**.

Step 7 In the Import Call Studio Project From File System dialog, browse to the location where you extracted the call studio projects. For each of the folders that were unzipped, select the folder (for example *BillingQueue*) and select **Finish**.

The project is imported into Call Studio. **Repeat this action** for each of the five folders.

When you are finished importing the five folders, you should see five projects in the *Navigator* window in the upper left.

Step 8 Update the *Default Audio Path URI* field in Call Studio to contain the IP address and port value for your media server.

For **each** of the Call Studio projects previously unzipped, complete the following steps:

a. Select the project in the Navigator window of Call Studio.

b. Click **Project > Properties > Call Studio > Audio Settings**.

c. On the **Audio Settings** window, modify the **Default Audio Path URI** field by supplying your server IP address and port number for the **<Server>** and **<Port>** placeholders.

d. Click **Apply** then **OK**.

Step 9 **BillingQueue Project:** If desired, you can change the music played to the caller while on hold.

Note:

- You can also create multiple instances of this project if you want to have different hold music for different clients, for example: BillingQueue with music for people waiting for billing, and SalesQueue with music for people waiting for sales. You also need to point to the proper version (BillingQueue or SalesQueue) in the ICM script. In the ICM script, the parameter queueapp=BillingQueue would also have a counterpart, queueapp=SalesQueue.
- The CallbackEntry Project (in the following step) contains a node called SetQueueDefaults. This node contains the value Keepalive Interval which must be **greater** than the length of the queue music you use. Refer to Keepalive Interval in the next step for details.

Step 10 **CallbackEntry Project:** If desired, in the **CallbackEntry** project, modify the caller interaction settings in the **SetQueueDefaults** node.

Note: This step defines values for the default queue. You can insert multiple SetQueueDefaults elements here for each queue name, if it is necessary to customize configuration values for a particular queue. If you do not have a SetQueueDefaults element for a given queue, the configuration values in the default queue are used.

a. In the Call Studio Navigator panel, open the **CallBackEntry** project and double click **app.callflow** to display the application elements in the script window.

b. Open the **Start of Call** page of the script using the tab at the bottom of the script display window.

c. Select the **SetQueueDefaults** node.

d. In the Element Configuration panel, select the **Setting** tab and modify the following default settings as desired:

For the SetQueueDefaults element, the caller interaction values in the Start of Call and the Wants Callback elements, may be edited. For more information on the caller interaction values, please refer to the Settings table in Chapter 10, Callback_Set_Queue_Defaults, in the **Element Specifications for Cisco Unified CVP VXML Server and Cisco Unified Call Studio Release 8.5(1),** (http://www.cisco.com/en/US/products/sw/custcosw/ps1006/products_programming_reference_guides_list.html)

Step 11 Set the path for the storage of recorded caller names.

In the **CallbackEntry** project, on the **Wants Callback** page, highlight the **Record Name** node and click the **Settings** tab in the **Element Configuration** window of Call Studio.

In the **Path** setting, change the path to the location where you want to store the recorded names of the callers. By default, Call Studio saves the path string in your VXML Server audio folder.

In a lab environment, if you are using the default path, you can create a new folder called *recordings* in the **`%CVP_HOME%\VXMLServer\Tomcat\webapps\CVP\audio\`** folder on the VXML Server.

Step 12 Set the name of the **Record name file**.

On the **CallbackEntry** project, the **Wants Callback** page, highlight the **Add Callback to DB** node and click the **Settings** tab in the **Element Configuration** window of Call Studio.

Change the **Recorded name file** setting to match the location of the recording folder you created.

Note:

- This setting references the URL of the recordings folder, whereas the *Path* setting references the file system path.
- The AddCallback element setting in the CallbackEntry project is configured to do automatic recorded file deletions. If automatic recorded file deletion is not desired, then remove the value of the *Recorded name path* setting in the AddCallback element. This removal action assumes that you will be doing the deletion or management of the recorded file yourself.

Step 13 In the CallbackEntry project, the *Callback_Set_Queue_Defaults* node, be sure the **keepalive** type (in seconds) is greater than the length of the queue music being played. The default is 120 seconds.

Step 14 Save the CallbackEntry project.

Step 15 **CallbackWait Project:** Modifying values in the **CallbackWait** application.

In this application, you can change the IVR interaction that the caller receives at the time of the actual callback. The caller interaction elements in **CallbackWait > AskIfCallerReady (page)** may be modified. Save the project after you modify it.

Step 16 Validate each of the five projects associated with the Courtesy Callback feature and deploy them to your VXML Server.

a. Right-click each Courtesy Callback project in the **Navigator** window and select **Validate**.

b. Right click each of the projects and click **Deploy**, then click **Finish**.

Step 17 Using windows explorer, navigate to **`%CVP_HOME%\VXMLServer\applications`**.

Step 18 For each of the five Courtesy Callback applications, open the project's *admin* folder, in %CVP_Home%\VXMLServer\applications, and double-click **deployApp.bat** to deploy the application to the VXML Server.

Step 19 Verify that all the applications are running by going into **`%CVP_HOME%\VXMLServer\admin`** and double-clicking **status.bat**. All five applications should be listed under *Application Name* and the status for each one should be *Running*.

Note: As an alternative to following steps 16-19 above, to deploy a VXML application to the VXML server, you can also use the Bulk Administration **VXML Applications** feature. This way, you can deploy all the applications into a single archive, and then deploy them from OAMP in one click. This process is simpler and can save time. Bulk Administration deploys the application to the VXML server, then executes update-all-apps batch file, then executes deploy-all-new-apps batch file.

To deploy a VXML application to the VXML server using the Bulk Administration **VXML Applications** feature:

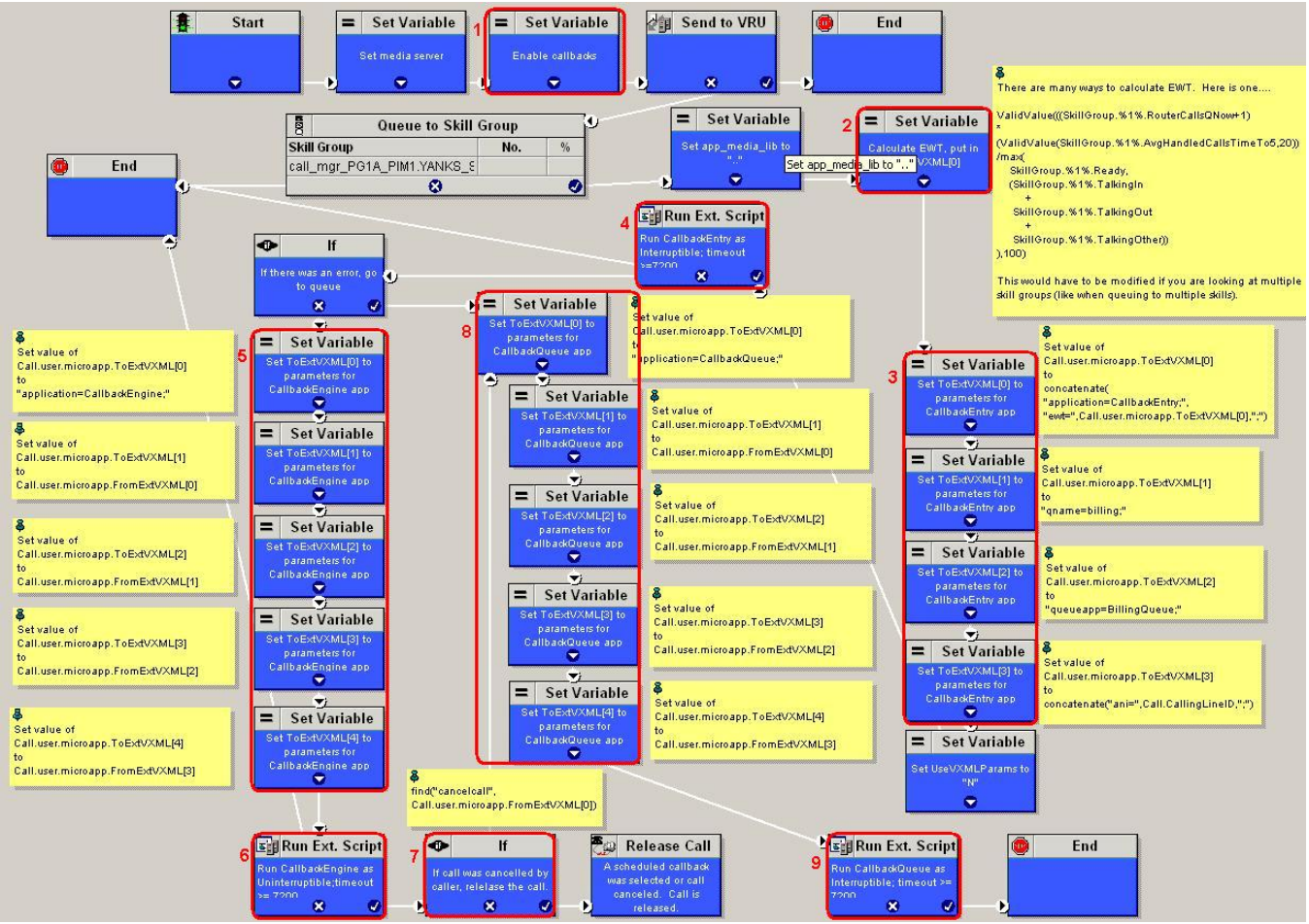
1. After validating and saving your applications, in the Navigator panel of Call Studio (top left), **right-click** and select all the applications you want to deploy.
2. Click **Deploy**.
3. In the Deploy Destination area, select **Archive File** and click **Browse**.
4. Navigate to the archive folder that you have set up, for example: **C:\Users\Administrator\Desktop\Sample**.
5. Enter the name of the file. For example: **Samplefile.zip**.
6. Click **Save**.
7. In the Deploy Destination area click **Finish**.
8. Log in to OAMP and navigate to: **Bulk Administration\File Transfer\VXML Applications**.
9. Select the VXML Server to which you want to deploy the applications.
10. Select the zip file that contains the applications. For example: **Samplefile.zip**.
11. Click **Transfer**.

Understanding the ICM Script for Courtesy Callback

The following discussion provides an overview of the script used for the courtesy callback feature. There are nine numbered blocks, or sets of blocks identified below. The numbered descriptions, following this screen shot, correspond to the numbered items in the graphic below.

Note: In the example below, the yellow comment blocks describe first the value being set and then the place that value is being sent to.

Figure 71: ICM Script Example for Courtesy Callback



The following bullets provide descriptions for the numbered blocks in the preceding graphic:

- Block 1: Enable callback or shut it off.
- Block 2: Compute average wait time. Once the caller is *in queue*, calculate the Estimated Wait Time (EWT) for that queue and place the value in ToExtVXML[0].

  If there is poor statistical sampling because of sparse queues and the wait time cannot be calculated in the VXML Server, use the ICM-calculated estimated wait time.

  One method of calculating EWT (the method used in this example) is:

```
ValidValue(((SkillGroup.%1%.RouterCallsQNow+1)
*
(ValidValue(SkillGroup.%1%.AvgHandledCallsTimeTo5,20))
/max(
   SkillGroup.%1%.Ready,
   (SkillGroup.%1%.TalkingIn
   +
   SkillGroup.%1%.TalkingOut
   +
   SkillGroup.%1%.TalkingOther))
```

```
),100)
```

This method would have to be modified if you are looking at multiple skill groups (when queuing to multiple skills).

- Block 3: Set up parameters to be passed.
- Block 4: run this block; prompt the caller; if caller does not accept the offer for a callback, keep the caller in the queue and provide queue music.
- Block 5: Set up variables. Call flow returns to this block if the caller elects to receive a callback. Otherwise, the call remains queuing in the queuing application (BillingQueue in this example) on the VXML Server.
- Block 6: Run external to Callback engine; keeps the call alive; if agent becomes available and there is no caller, then agent can't interrupt (do not want an agent to pick up and have no one there).
- Block 7: Has the caller rejected the callback call? If no, then go to block 8.
- Block 8: Set up variables.
- Block 9: Put caller briefly into queue (after caller accepts the actual callback call).

Overview of ICM Script Configuration for Courtesy Callback

The ICM script elements needed to enable Courtesy Callback can be found on the CVP 8.0(1) Installation CD in *CVP\Downloads and Samples\ICM Downloads*. The script sample found there (CourtesyCallback) contains the necessary sample elements for the courtesy callback feature. However you must merge this script into your existing ICM scripts.

As a starting point and to run a simple test, import the script into ICM script editor, validate it with the ICM script editor validation tool to locate nodes that need extra configuration (such as for Network VRU scripts and ECC variables), and then modify the script according to your existing ICM environment.

The general process is as follows:

1. Locate each queue point in every ICM script. This can be Queue To Skill Group, Queue to Enterprise Skill Group, Queue to Scheduled Target or Queue to Agent.
2. Categorize each queue point according to the pool of resources that it is queuing for. Each unique pool of resources will ultimately require a queue in VXML Server if Courtesy Callback is going to be offered for that resource pool. For example, using the following example, QueueToSkill X and QueueToSkill Z are queuing for the exact same resource pool (despite the different queuing order). Queue to Skill Y, however, is queuing to a different pool because it includes Skill Group D.
   - QueueToSkillGroup X is queuing for Skill Group A, B, C in that order.

– QueueToSkillGroup Y is queuing for Skill Group A, C and D in that order.

– QueueToSkillGroup Z is queuing for Skill Group C, B, A in that order.

3. Assign a unique name to each unique resource pool. In the above example, we can use names ABC and ACD as example names.

4. For each resource pool, decide whether callbacks will be allowed in that resource pool. If yes, then every occurrence of that resource pool in all ICM scripts must be set up to use VXML Server for queuing. This is to ensure that the Courtesy Callback mechanism in the VXML Server gets a full, accurate picture of each resource pool's queue.

5. For any queue point where Courtesy Callback will be offered, modify all ICM scripts that contain this queue point according to the guidelines in the following ICM script examples.

Configuring the ICM Script for Courtesy Callback

Many of the configuration items below relate to the numbered blocks in the diagram and provide understanding for ICM Script for Courtesy Callback (page 451). Steps that refer to specific blocks are noted at the beginning of the each step.

To configure ICM to use the sample Courtesy Callback ICM script:

Step 1 Copy the ICM example script, **CourtesyCallback.ICMS** to the ICM Admin Workstation.

The example ICM script is available in the following locations:

- On the CVP install media in **\CVP\ICM Downloads**
- From the Operations Console in **%CVP_HOME%\OPSConsoleServer\ICMDownloads**

Step 2 Map the route and skill group to the route and skill group available for courtesy callback.

a. In Script Editor, select **File > Import Script...**.

b. In the script location dialog, select the **CourtesyCallback.ICMS** script and click **Open**.

c. In the Import Script - Manual Object Mapping window, map the route and skill group to the route and skill group available for courtesy callback (identified previously).

Step 3 Once the script is open in Script Editor, open the **Set media server** node and specify the URL for your VXML Server.

For example: **http://10.86.132.139:7000/CVP**

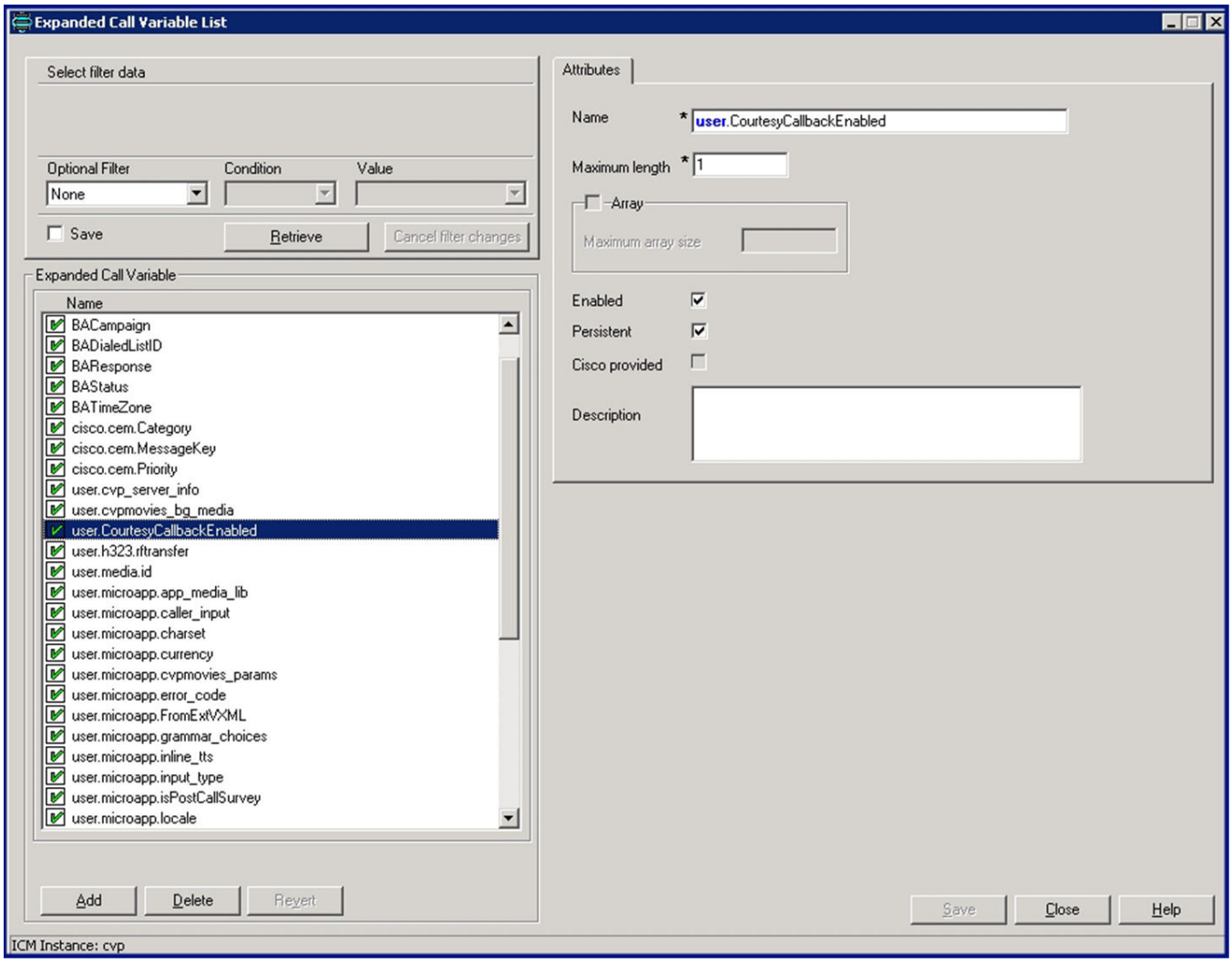
Step 4 **Refer to Block #1:** A new ECC variable is used when determining if a caller is in queue and can be offered a callback. Define a new ECC variable for courtesy callback.

On the ICM Admin Workstation, in the ICM Configuration Manager, Expanded Call Variable List tool, create the following ECC Variable specific to Courtesy Callback:

- **user.CourtesyCallbackEnabled** - Maximum Length 1, Enabled, Persistent

Note: This step assumes you have already created the standard ECC variables required for any Unified CVP installation. Refer to "Common Unified ICMH Configuration: Define Unified CVP ECC Variables (page 131)."

Figure 72: Set Courtesy Callback ECC Variables



Step 5 **Block #2:** If you wish to use a different estimated wait time (EWT), modify the calculation in block #2; you will need to do this if you use a different method for calculating EWT or if you are queuing to multiple skill groups.

Step 6 **Block #3:** Set up the parameters that will be passed to CallbackEntry (VXML application).

Note: This step assumes you have already configured the ICM and ECC variables not related to Courtesy Callback.

Variable values specific to Courtesy callback include:

ToExtVXML[0] =
concatenate("application=CallbackEntry",";ewt=",Call.user.microapp.ToExtVXML[0])

ToExtVXML[1] = "qname=billing";

ToExtVXML[2] = "queueapp=BillingQueue;"

ToExtVXML[3] = concatenate("ani=",Call.CallingLineID,";");

Definitions related to these variables are:

- CallbackEntry is the name of the VXML Server application that will be executed.
- ewt is calculated in **Block #2**.
- qname is the name of theVXML Server queue into which the call will be placed.There must be a unique qname for each unique resource pool queue.
- queueapp is the name of theVXML Server queuing application that will be executed for this queue.
- ani is the caller's calling Line Identifier.

Step 7 Create Network VRU Scripts.

Using the ICM Configuration Manager, Network VRU Script List tool, create the following Network VRU Scripts:

Block #4: Interruptible Script (agent can interrupt the caller on hold):

- Name: **VXML_Server_Interruptible**
- Network VRU: Select your Type 10 CVP VRU
- VRU Script Name: **GS,Server,V,interrupt**
- Timeout: **9000 seconds**
- Interruptible: **Checked**

Block #6: Noninteruptible Script (agent cannot interrupt because no caller is available):

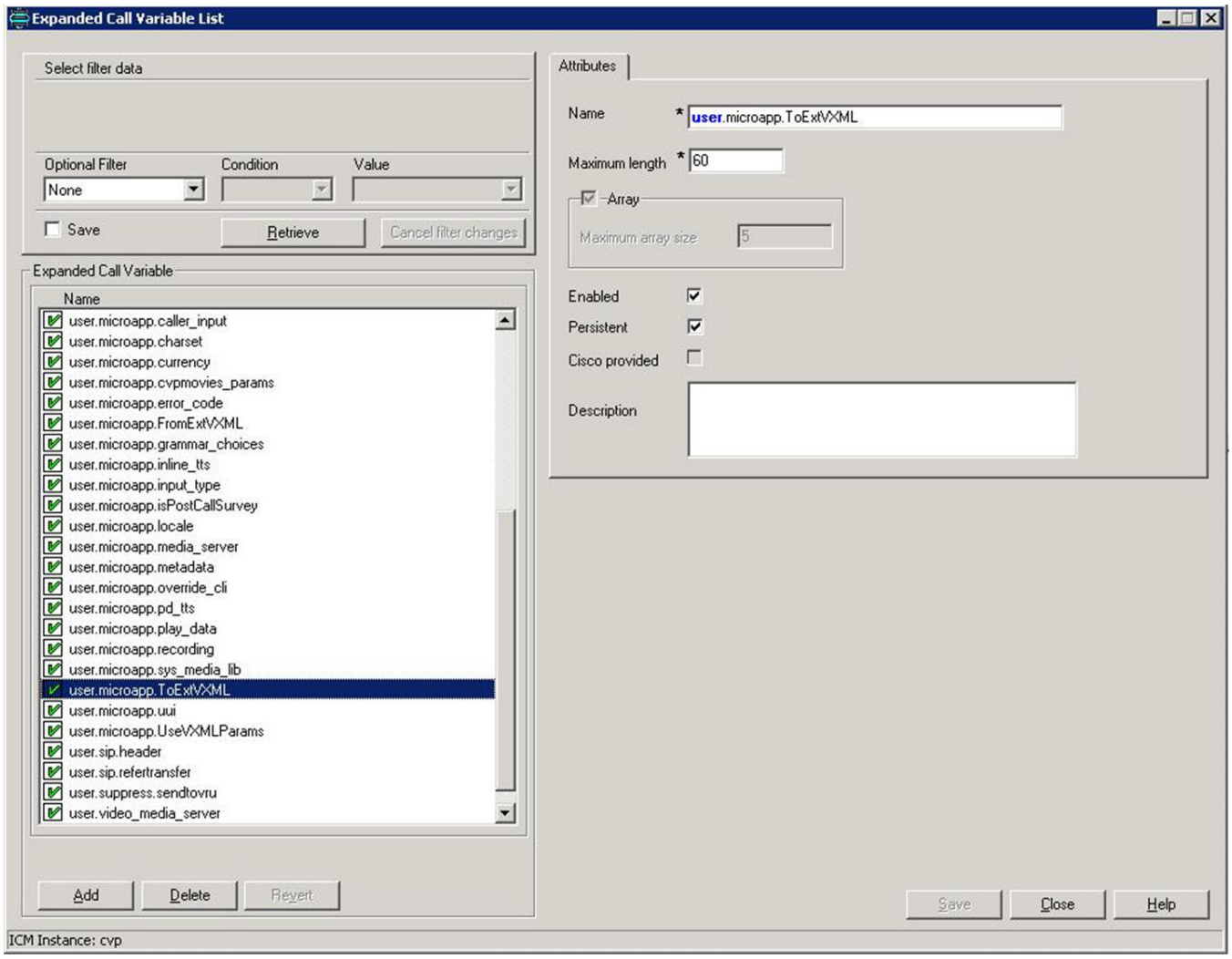
- Name: **VXML_Server_Noninterruptible**
- Network VRU: Select your Type 10 CVP VRU
- VRU Script Name: **GS,Server,V,nointerrupt**
- Timeout: **9000 seconds** (must be greater than the maximum possible call life in Unified CVP)
- Interruptible: **Not Checked**

Step 8 As part of setting up the Courtesy Callback feature, verify that the **user.microapp.ToExtVXML ECC** variable is set up for an array of 5 items with a minimum size of 60 chars and the

user.microapp.FromExtVXML variable is set up for an array of 4 with a minimum size of 60 chars, as shown in the following to screen shots.

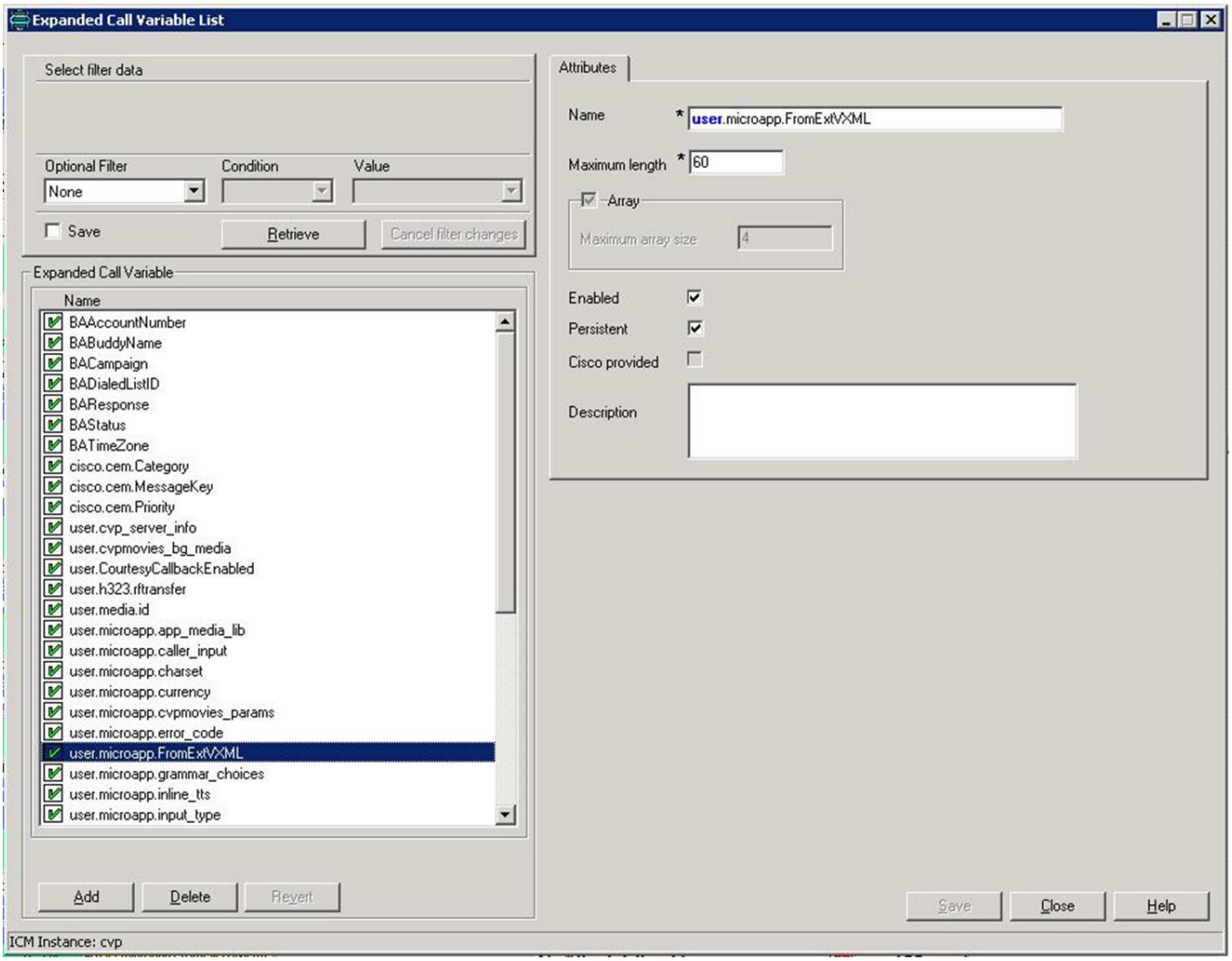
user.microapp.ToExtVXML configuration:

Figure 73: Verify Courtesy Callback ECC Variables for Microapp



user.microapp.FromExtVXML configuration:

Figure 74: Courtesy Callback Configure FromExtVXML



Step 9 Verify that you have at least one available route and skill group to map to the route and skill group in the example script.

Step 10 Save the script, then associate the call type and schedule the script.

Note: For an example of scheduling the script refer to *Getting Started with Cisco Unified Customer Voice Portal*, the *Create a Call Type Manager Entity Routing Script and Call Schedule* topic.

Multicast Music on Hold (MMoH)

Multicasting may be used for Music On Hold with supplementary services on Unified CM as an alternative to the unicast MoH.

There are two ways to deploy with this feature:

- With the Unified CM multicasting the packets on the local LAN

- With the branch gateway(s) multicasting on their local LAN(s)

The latter is used when SRST (survivable remote site telephony) is configured on the gateway, and allows the deployment to utilize MOH locally and avoid MOH streaming over the WAN link.

For configuration details access the following links:

- **Configuring Music on Hold** (http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmemoh.html#wp1020115)
- **Integrating Cisco CallManager and Cisco SRST to Use Cisco SRST as a Multicast MoH Resource** (http://www.cisco.com/en/US/products/sw/iosswrel/ps5207/products_feature_guide09186a00802d1c31.html#wp1099756)

Note: The SIP Trunk for Unified CVP (configured on Unified Communications Manager) must be associated with a Media Resource Group List (MRGL) that supports MMOH resources. Details on creating the MRGL are available from the links above.

Post Call Survey for SIP

A Post Call Survey is a survey that takes place after normal call treatment and is typically used to determine whether a customer was satisfied with their call center experience. This feature enables you to configure a call flow that, after the agent disconnects from the caller, the call is optionally sent to a DNIS configured for a Post Call Survey.

The caller can be prompted during IVR treatment as to whether they would like to participate in a Post Call Survey. If they choose to do so, they are automatically transferred to the Post Call Survey after the normal call flow completes, for example, after the agent ends the conversation.

The post call survey call works just like a regular call from the Unified Contact Center Enterprise point of view. Scripts can be invoked and the customer can use the keypad on a touch tone phone and/or voice with ASR/TTS to respond to questions asked during the survey. During Post Call Survey, the call context information is retrieved from the original customer call.

The ICM script can enable and disable Post Call Survey on a per-call basis, by setting the ECC variable **user.microapp.isPostCallSurvey** to *y* or *n*.

Note:

- For reporting purposes, the Post Call Survey call has the same CallGUID and call context as the original inbound call.
- The call context for the post call survey includes all context up to the point where the call is transferred to the agent. Context that the agent creates after the transfer is not included in the post call survey context.

If you wish to use the Post Call Survey feature through Unified CVP, you must configure it on the Call Server. Also, you can configure the Unified ICM script to toggle the use of Post Call Survey off and on. The two configuration topics that follow, explain these methods.

Note: For the H.323 discussion, refer to Restart to Alternate DNIS for H.323 (page 428)

Configuring the Unified CVP Call Server for Post Call Survey

To configure Post Call Survey on the Call Server:

Step 1 Log in to the Operations Console and select **Device Management > Unified CVP Call Server**.

The *Find, Add, Delete, Edit Call Servers* page displays.

Step 2 Click the Call Server for which you want to configure Post Call Survey.

The *Edit CVP Call Server Configuration* page displays.

Step 3 Click the **SIP** tab. Verify the **Override System Dialed Number Pattern Configuration** is not checked.

Step 4 Click **Save** and **Deploy** to deploy the Unified CVP Call Server device..

Step 5 Select **System > Dialed Number Pattern**.

The Dialed Number Pattern page displays.

Step 6 Click **Add New**.

Step 7 Fill in the appropriate configuration settings to associate incoming dialed numbers with survey numbers:

Table 43: Dialed Number Pattern Configuration Settings

Property	Description	Default	Value	
General Configuration				
Dialed Number Pattern	The incoming Dialed Number for calls that are to be directed to a Post Call Survey Dialed Number. This is the Dialed Number you want to redirect to the survey.	None	Must be unique Maximum length of 24 characters Can contain alphanumeric characters, wildcard characters such as exclamation	

Property	Description	Default	Value
			point (!) or asterisk (*),single digit matches such as the letter X or period (.) Can end with an optional greater than (>) wildcard character
Enable Post Call Survey for Incoming Calls	Check to enable post call survey for incoming calls. • **Survey Dialed Number Pattern -** Enter the dialed number of the Post Call Survey. This is the dialed number to which the calls should be transferred to after the normal call flow completes.	Disabled none	n/a Maximum length of 24 characters Can contain alphanumeric characters, wildcard characters such as exclamation point (!) or asterisk (*), single digit matches such as period (.) or X, and can end with an optional greater than (>)wildcard character

Step 8 Click **Save** to save the Dialed Number Pattern.

You are returned to the **Dialed Number Pattern** page. To deploy the Dialed Number Pattern configuration, click **Deploy** to deploy the configuration to all Unified CVP Call Server devices.

Configuring ICM for Post Call Survey

Configuration is not required on Unified ICM to use Post Call Survey, however, you can turn the feature off (and then on again) within an ICM script by using the ECC variable **user.microapp.isPostCallSurvey** and a value of *n* or *y* (value is case insensitive) to disable and re-enable the feature.

Configure the ECC variable to a value of *n* or *y* before the label node or before the Queue to Skillgroup node. This sends the correct value to Unified CVP before the agent transfer. This ECC var is not needed to initiate a Post Call Survey call, but you can use it to control the feature once Post Call Survey is configured using the Operations Console (as described inConfiguring the Unified CVP Call Server for Post Call Survey (page 460)). As long as a DN is mapped in the Operations Console for Post Call Survey, the call will be automatically transferred to the configured Post Call Survey DN.

Note:

- The Post Call Survey DN is called if the CVP has received at least one CONNECT message from ICM, either from the VRU leg or from the Agent leg. Use the END node in your Unified ICM script if the "Post Call Survey" is not required for the calls disconnected from the IVR.
- If Router Requery is configured incorrectly and the Ring-No-Answer timeout expires, the caller is still transferred to the Post Call Survey DN. This can occur if a Queue node is used and **Enable target requery** is not checked.

Step 1 On the Unified ICM Administration Workstation, using configuration manager, select the **Expanded Call Variable List** tool.

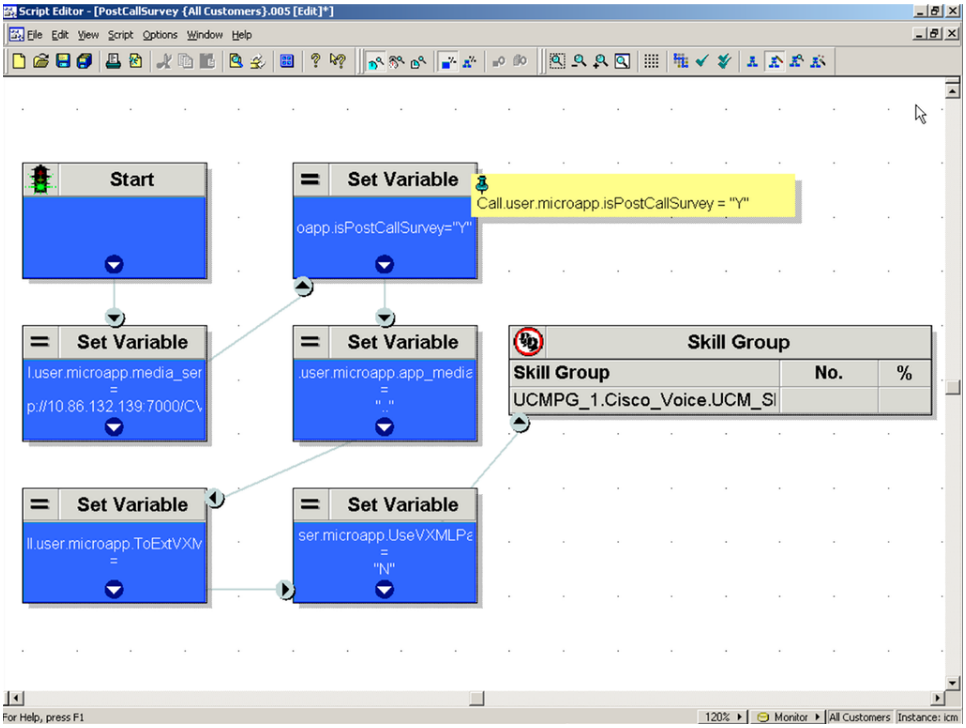
Step 2 Create a new ECC Variable with **Name:** `user.microapp.isPostCallSurvey`.

Step 3 Set **Maximum Length:** to 1.

Step 4 Check the **Enabled** checkbox then click **Save**.

Step 5 In your Unified ICM scripts, remember that, at script start, the default behavior of Post Call Survey equals **enabled**, even if **user.microapp.isPostCallSurvey** has not yet been set in the script. You can turn **off** Post Call Survey in the script by setting **user.microapp.isPostCallSurvey** to *n*. You can later re-enable Post Call Survey in the same path of the script by setting this variable to *y*.

Figure 75: Post Call Survey Script Example



Chapter 13

Configuring the H.323 Devices and VoIP

This chapter describes how to configure the H.323 devices. Once these devices have been configured, you can add them to the Operations Console network control panel. Once added, you can execute a subset of IOS Gateway and Gatekeeper commands on the Gateway and Gatekeeper from the Operations Console, along with adding the Unified ICME Server to a device pool (refer to "Configuring a Gateway (page 484)," "Configuring a Gatekeeper (page 486)," and "Configuring a Unified ICME Server (page 487)").

This chapter contains the following topics:

About Call Routing

Inbound call routing is the method by which an H.323 originating endpoint (for example, an IOS gateway or Cisco Unified Communications Manager) finds a Unified CVP H.323 Service to which to send an H.323 call. (In other words, calls that are considered *inbound* from Unified CVP's point of view.)

Outbound call routing is the method by which Unified CVP finds an H.323 endpoint as the destination for a call transfer. (In other words, calls that are considered *outbound* from Unified CVP's point of view.)

Note: For outbound call routing, a gatekeeper **must** exist in the call flow configuration; for inbound call routing, a gatekeeper is optional.

You can accomplish call routing using the tech-prefix method, the zone prefix method, or a combination of both. The sections that follow describe the tech-prefix and zone-prefix methods.

Note: There are variations to the routing methods described below, for example, the branch office model.

Call Routing (tech-prefix method)

One method for performing call routing with Unified CVP is the *tech-prefix* method. With this method, a specific class of H.323 devices registers with a unique tech-prefix. For example, Unified CVP endpoints could register to the gatekeeper with a tech-prefix of **2#**, VoiceXML gateways could register with **3#**, and Cisco Unified Communications Manager endpoints could register with **1#**. Additionally, with certain geographical distributions of devices, it might be desirable to further subdivide the classification. For example, assuming there are two data centers, you might want to register the Unified CVP devices in Data Center 1 with **98#** and Unified CVP devices in Data Center 2 with**99#**. This is strictly a matter of customer preference and topological deployment needs.

With the tech-prefix method, a RAS Admission Request to the gatekeeper contains a DNIS prefixed with the tech-prefix (for example, **2#8005551212**). When the gatekeeper receives such a request, it will first look to see which endpoints are registered with that tech-prefix and then perform an approximate round-robin selection among those endpoints. It will return the selected IP address and port to the requesting endpoint in the RAS Admission Confirm message. The requesting endpoint will then route the call to the selected server. Each type of endpoint has a mechanism for stripping the tech-prefix, if desired, the details of which are described below.

Note: The gatekeeper command **show gatekeeper gw-type-prefix** can be very useful in determining how the gatekeeper will route calls based on tech-prefix.

In this first example, we will assume that:

- Unified CVP registers with **2#**.
- Ingress gateway registers with **3#**.
- VoiceXML gateway registers with **4#** (assume ingress andVoiceXML gateways are separate).
- Egress gateway registers with **5#**. In this case, the Egress gateway is front-ending a legacy ACD or IVR.
- Cisco Unified Communications Manager registers with **1#** (note that Cisco Unified Communications Manager is registering with the default tech-prefix).
- Default technology prefix in the gatekeeper is **1#**.

Gatekeeper Configuration for Call Routing (tech-prefix method)

The gatekeeper configuration only requires that you specify a default technology prefix; it is not necessary to explicitly define zone prefixes.

Configure the default tech-prefix as follows:

```
gatekeeper
    gw-type-prefix 1#* default-technology
```

Ingress Gateway Configuration for Call Routing (tech-prefix method)

The ingress gateway needs to locate a Unified CVP server to which to send the call (inbound call routing to Unified CVP).

Configure the Unified CVP VoIP dial-peer in the gateway as follows:

Note: Only items pertinent to tech-prefix routing are highlighted

```
interface Loopback0
   ip address 22.22.22.12 255.255.255.255
   h323-gateway voip interface
   h323-gateway voip id zoneA ipaddr 10.86.129.36 1719
   h323-gateway voip h323-id ingressgw1-zoneA
   * \ Next command is specific to tech-prefix routing
   h323-gateway voip tech-prefix 3#

dial-peer voice 5000 voip

* \This is the primary CVP dial-peer with highest preference
preference 0
destination-pattern 59..
voice-class codec 1

* \ Next two commands are specific to tech-prefix routing
tech-prefix 2#
session target ras
dtmf-relay rtp-nte h245-signal h245-alphanumeric
no vad
```

VoiceXML Gateway Configuration for Call Routing (tech-prefix method)

Unified CVP needs to be able to locate a VoiceXML gateway during the transfer to the IVR leg of the call in Comprehensive Model. Unified ICME passes Unified CVP this label as it was defined in Network VRU explorer. The Network VRU label as defined in Unified ICME must begin with **4#** for the current example. The **4#** is then stripped off by the VoiceXML gateway before initiating the IVR leg back to the Call Server. The Call Server 'Maximum Length of DNIS' under the ICM tab in Call Server Configuration should be set to the length of the label after the **4#** has been removed.

```
interface Loopback0
   ip address 22.22.22.13 255.255.255.255
   h323-gateway voip interface
   h323-gateway voip id zoneA ipaddr 10.86.129.36 1719
   h323-gateway voip h323-id vxmlgw1-zoneA
```

```
   * \ Next command is specific to tech-prefix routing
   h323-gateway voip tech-prefix 4#

* \ Next command is specific to tech-prefix routing
voice translation-rule 1
  rule 1 /4#/ //

* \ Next command is specific to tech-prefix routing
voice translation-profile techprefix
  translate called 1

dial-peer voice 111 voip
service isnapp

* \ Next two commands are specific to tech-prefix routing
translation-profile incoming techprefix
incoming called-number 4#T
dtmf-relay h245-signal h245-alphanumeric
codec g711ulaw
no vad
```

Unified CM Configuration for Call Routing (tech-prefix method)

Unified CM sometimes needs to be able to locate a Unified CVP server to which to route a call (inbound call routing to Unified CVP). Also, Unified CVP needs to be able to locate a Unified CM to which to route a transferred call (outbound call routing from Unified CVP). To minimize tech-prefix configuration in Unified CM and Unified ICME device target labels, it is best to let Unified CM register to the gatekeeper with the default technology prefix (**1#**). All other devices (Unified CVP, VoiceXML gateways, Ingress gateways) register with tech-prefixes other than the default tech-prefix. Therefore, any DNIS which the gatekeeper is asked to route which is not prepended with a tech-prefix will route the call to Cisco Unified Communications Manager.

Do the following:

1. In Unified CM Administration, use **Device > Gatekeeper** to create the gatekeeper device.
2. Use **Device > Trunk** to create an H.225 Gatekeeper Controlled trunk. (This is the trunk to enable inbound call routing from Cisco Unified Communications Manager to Unified CVP.)
   - Under Gatekeeper Information:
     - — Select the gatekeeper that was defined in the previous step.
     - — Select **Gateway** as the Terminal Type.
     - — Set a non-existing tech-prefix (for example, 77#).
     - — Select a zone. (If left blank, Unified CM will register to the first local zone defined in the gatekeeper.)
3. Create a Route Pattern that will point the desired set of DNIS values to Unified CVP:

- Select **Call Routing > Route/Hunt > Route Pattern**.
- Enter the DNIS value that will route to Unified CVP.
- Under Gateway/Route List, select the Gatekeeper Trunk created in the previous step.
- In Prefix Digits (Outgoing Calls), enter the tech-prefix of Unified CVP (**2#**).

4. Use **Device > Trunk** to create a second H.225 Gatekeeper Controlled trunk. (This is the trunk to enable outbound call routing to agents from Unified CVP to Cisco Unified Communications Manager.)
   - Ensure that *Media Termination Point required* is **disabled**.
   - Under Gatekeeper Information:
     - — Select the gatekeeper that was defined in the previous step.
     - — Select **Gateway** as the Terminal Type.
     - — Set the technology prefix to **1#**.
     - — Select a zone. (If left blank, Cisco Unified Communications Manager will register to the first local zone defined in the gatekeeper.)

Unified CVP Configuration for Call Routing (tech-prefix method)

You do not need to perform specific configuration for the H.323 Service for inbound call routing when using the tech-prefix method unless you choose to change the default tech-prefix value of Unified CVP to some other value. You would do so using the **setTechPrefix** command. For example, if there are distinct Unified CVP server farms in various geographic locations, you might choose to route calls from one bank of gateways to a specific farm of Unified CVP servers. In this case, you could assign 2# to one farm, 3# to the next, and so on. Unified CVP by default will strip off the tech-prefix before passing the remainder of the DNIS to Unified ICME. The setTechPrefixRemoval VBAdmin command can be used to prevent stripping the tech-prefix, if so desired.

Note: Use the **setTechPrefixRemoval** VBAdmin command if you want to prevent stripping off the tech-prefix.

Call Routing (zone-prefix method)

Another method of performing inbound call routing to and from Unified CVP is to use the zone-prefix method. In this method, all H.323 devices register to the gatekeeper with the same tech-prefix. Call routing is then determined solely on the zone-prefixes defined in the gatekeeper.

Note: The gatekeeper command **show gatekeeper gw-type-prefix** can be very useful in determining how the gatekeeper will route calls based on tech-prefix.

Gatekeeper Configuration for Call Routing (zone-prefix method)

Do the following:

1. Configure the default tech-prefix:

   ```
   gatekeeper

   gw-type-prefix 1#* default-technology
   ```

2. Define the zone prefix(es) for all DNIS values that should be routed to Unified CVP.

   ```
   zone prefix zoneA 5900* gw-priority 10 h323-id-of-CVP1
   h323-id-of-CVP2
   ```

   – The H323-id is what must be specified in the zone-prefix command. Although by default the H323-id is the IP address of the Unfied CVP machine, it can be changed using the setH323ID VBAdmin command so care must be taken to make sure this matches. This is a very common mistake when configuring the gatekeeper.

   – If multiple Unified CVP devices are specified, the gatekeeper will perform an approximate round-robin selection algorithm when selecting a Unfied CVP endpoint to which to route the call.

   – It is best to use the wildcard (*) syntax for all zone prefixes. Mixing the dot syntax with * syntax can have unpredictable routing results

3. Define the zone prefix for the Network VRU label. This is how Unified CVP will locate a VoiceXML gateway to process the IVR leg of the call.

4. Define the zone prefix(es) for agent phone device target ids. Typically, the H.323-ids of all subscribers in the Cisco Unified Communications Manager cluster are listed here

5. Define alternate endpoints for all egress gateways (both VoiceXML gateways and ACD egress gateways, if present). In the event Unified CVP sends a call setup to an egress gateway which fails, Unified CVP will then try the all the alternate endpoints in turn, until it receives a successful response.

   ```
   endpoint alt-ep h323id <h323-id-of-failed-gateway>
   <ip-address-of-alternate-gateway>
   ```

Caution: Do not use outbound alternate endpoint functionality for agent routing. It should be used only for IVR routing. This will prevent the undesirable condition where the agent reserve timer expires before the call arrives, allowing a different call to reach the agent that does not match the call data previously sent to the agent by the Unfied ICME software. Router requery is the preferred method in the agent case.

Ingress Gateway Configuration for Call Routing (zone-prefix method)

The Unified CVP voip dial-peer in the gateway should be configured as follows. Note the items pertinent to tech-prefix routing are highlighted. Note the absence of the tech-prefix command.

```
interface Loopback0
   ip address 22.22.22.12 255.255.255.255
   h323-gateway voip interface
   h323-gateway voip id zoneA ipaddr 10.86.129.36 1719
   h323-gateway voip h323-id ingressgw1-zoneA
   * \ Next command is specific to tech-prefix routing
   h323-gateway voip tech-prefix 1#

dial-peer voice 5000 voip
destination-pattern 59..
voice-class codec 1

* \ Next command is specific to tech-prefix routing
session target ras
dtmf-relay rtp-nte h245-signal h245-alphanumeric
no vad
```

VoiceXML Gateway Configuration for Call Routing (zone-prefix method)

Unified CVP needs to be able to locate a VoiceXML gateway during the transfer to the IVR leg of the call in Comprehensive Model. Unified ICME passes Unified CVP this label as it was defined in Network VRU explorer. The Call Server 'Maximum Length of DNIS' under the ICM tab in Call Server Configuration should be set to the length of this label.

```
interface Loopback0
   ip address 22.22.22.13 255.255.255.255
   h323-gateway voip interface
   h323-gateway voip id zoneA ipaddr 10.86.129.36 1719
   h323-gateway voip h323-id vxmlgw1-zoneA
   * \ Next command is specific to tech-prefix routing
   h323-gateway voip tech-prefix 1#

dial-peer voice 111 voip
service isnapp

* \ Next command is specific to tech-prefix routing
incoming called-number 1111111T (assuming the Network VRU label was
1111111)
dtmf-relay h245-signal h245-alphanumeric
codec g711ulaw
no vad
```

Egress Gateway Configuration for Call Routing (zone-prefix method)

An egress gateway is typically used in Call Director Model to provide access to a call center ACD or third-party IVR. The ACD is connected to the egress gateway through T1/E1 trunks.

In this example, the ACD uses a numbering plan in the format **xxxxyyzzzz**, where:

- **xxxx** is a location code (8888 for the ACD in our example)
- **yy** is the destination trunk group on the ACD
- **zzzz** is the DNIS (Dialed Number Identification Service), identifying an ACD agent skillgroup, service, extension, et cetera.

Unified CVP initiates the transfer using all these digits:

- The gatekeeper uses the **xxxx** digits to determine a destination gateway.
- The gateway uses the **yy** digits to determine the correct ACD trunk group.
- The gateway outpulses the **zzzz** digits to the proper ACD trunk group, which uses them to connect the call to an agent.

```
interface Loopback0
   ip address 22.22.22.14 255.255.255.255
   h323-gateway voip interface
   h323-gateway voip id zoneA ipaddr 10.86.129.36 1719
   h323-gateway voip h323-id egressgw1-zoneA
   * \ Next command is specific to tech-prefix routing
   h323-gateway voip tech-prefix 1#

dial-peer voice 1 pots
* \ GW to route calls with leading digits 888800 to
* \ GWvoice port 0:1
* \ (T1/E1 controller # :D Channel), with highest
* \ preference (0)
destination-pattern 888800….
port 0:1
preference 0

dial-peer voice 2 pots
* \ Defines backup voice port (0:2) for same digits
destination-pattern 888800….
port 0:2
preference 1

dial-peer voice 3 pots
* \ Routes calls with different yy digits (01) to a
* \ different GW voice port (1:1).
* \ This is how the GW differentiates the ACD trunkgroups.
destination-pattern 888801….
```

```
port 1:1
preference 0

dial-peer voice 4 pots
* \ Defines backup voice port (1:2) for the same trunkgroup
destination-pattern 888801….
port 1:2
preference 1

dial-peer voice 5 pots
* \ Routes calls to a third trunk group
destination-pattern 888802….
port 2:1
preference 0

dial-peer voice 6 pots
* \ Backup voice port for third trunk group
destination-pattern 888802….
port 2:2
preference 1
```

Unified CM Configuration for Call Routing (zone-prefix method)

Cisco Unified Communications Manager sometimes needs to be able to locate a Unified CVP server to which to route a call (inbound call routing to Unified CVP). Also, Unified CVP needs to be able to locate a Cisco Unified Communications Manager to which to route a transferred call (outbound call routing from Unified CVP).

Do the following:

1. In Cisco Unified Communications Manager Admin, use **Device > Gatekeeper** to create the gatekeeper device.

2. Use **Device > Trunk** to create an H.225 Gatekeeper Controlled trunk. (This is the trunk to enable inbound call routing from Cisco Unified Communications Manager to Unified CVP.)

   – Enable **Media Termination Point required**. (This is necessary for calls that originate from Cisco Unified Communications Manager into Unified CVP because Unified CVP cannot process the H.323 Empty Capability Set feature for inbound calls.)

   – Under Gatekeeper Information:

     — Select the gatekeeper that was defined in the previous step.

     — Select **Gateway** as the Terminal Type.

     — Set technology prefix to **1#**.

     — Select a zone. (If left blank, Cisco Unified Communications Manager will register to the first local zone defined in the gatekeeper.)

3. Create a Route Pattern that will point the desired set of DNIS values to Unified CVP:

- Select **Call Routing > Route/Hunt > Route Pattern**.
- Enter the DNIS value that will route to Unified CVP.
- Under Gateway/Route List, select the gatekeeper trunk created in the previous step.

4. Use **Device > Trunk** to create a *second* H.225 Gatekeeper Controlled trunk. (This is the trunk to enable outbound call routing from Unified CVP to Cisco Unified Communications Manager.)
   - Make sure that *Media Termination Point required* is **disabled**.
   - Under Gatekeeper information:
     - — Select the gatekeeper that was defined in the previous step.
     - — Select **Gateway** as the Terminal Type.
     - — Set technology prefix to **1#**.
     - — Select a zone. (If left blank, Cisco Unified Communications Manager will register to the first local zone defined in the gatekeeper.)

Unified CVP Configuration for Call Routing (zone-prefix method)

In VBAdmin, enter the following command: **`setTechPrefix 1#`**.

Gatekeeper Redundancy

Host Standby Router Protocol (HSRP) Support

This section describes how to configure HSRP for Unified CVP.

H.323 Service

Depending on the registration interval configured in the gateways and Unified CM, there might be a period ranging from 30 seconds to a few minutes immediately after gatekeeper failure (when using HSRP) where you cannot transfer calls to the desired endpoint and callers lose connectivity.

You should follow these configuration steps below to ensure that this does not happen. The configuration specified below provides seamless HSRP failover behavior.

Note: Using this method, the caller can experience a 60 second wait period while the system stabilizes before being transferred to an agent.

Gatekeeper

Step 1 From the VBAdmin, set the Gatekeeper to the HSRP VIP of the Gatekeeper failover pair. If necessary, set the desired H.323 zone using the VBAdmin **`setGatekeeper`** command.

Step 2 Set up the Gatekeeper for HSRP failover. (Refer to your Gatekeeper and **Cisco IOS documentation** (http://www.cisco.com/en/US/products/sw/custcosw/ps1001/tsd_products_support_series_home.html).) This process should include putting the HSRP VIP in both a "**standby**" command on the FastEthernet interface and on the "**zone local**" command.

Step 3 Include all your IOS gateways in your zone as transfer candidates. You can prioritize them using the gw-priority on the "zone prefix" commands. By using all the gateways in the zone, the possibility of not finding a registered gateway immediately after HSRP failover are virtually eliminated, provided the gateway dial- peers are configured as defined in the following bullet.

Gateway

Step 1 Register your gateways with the VIP of the HSRP failover pair.

Step 2 Define a dial-peer of the highest preference (preference 0) that performs a gatekeeper search for an available Unified CVP H.323 Service (that is, "session target ras").

Step 3 Set the tech-prefix to the tech prefix of your Unified CVP H.323 Service.

Step 4 Define at least one other dial-peer with lower preference (preference 1) that uses a static Unified CVP H.323 Service address instead of a gatekeeper lookup (for example, session target ipv4:10.86.137.68).

Unified CM

Step 1 Under **Device > Gatekeeper**, set the gatekeeper name to the HSRP VIP of the gatekeeper failover pair.

Step 2 Set "Registration Request Time to Live" to 30 seconds. Set "Registration Retry Timeout" to 30 seconds.

Unified ICME

Step 1 Enable the router requery on all "Queue to Skill Group" nodes.

Step 2 Coming off the X path from the Queue to Skill Group node, add an "IF" node that checks the value of the variable Call.RequeryStatus. If the value is 1, play a 60 second prompt and go to

a second “Queue to Skill Group” node. If the value is not “1”, immediately go to the second “Queue to Skill Group Node.”

Alternate Gatekeeper Support

The Unified CVP H.323 Service can use an alternate gatekeeper in the event that the primary configured gatekeeper fails. Most customers want redundant gatekeepers, so Unified CVP has always supported paired gatekeepers in a Hot Standby Router Protocol (HSRP) configuration. HSRP provides a single (virtual) IP address behind which two gatekeepers can operate. However, HSRP is a relatively old protocol which has a number of shortcomings, most notably that the two gatekeepers in a pair must be co-located at the same site. This confounds customers' attempts to provide geographic redundancy, and requires such customers to purchase twice as many gatekeepers as they would otherwise need.

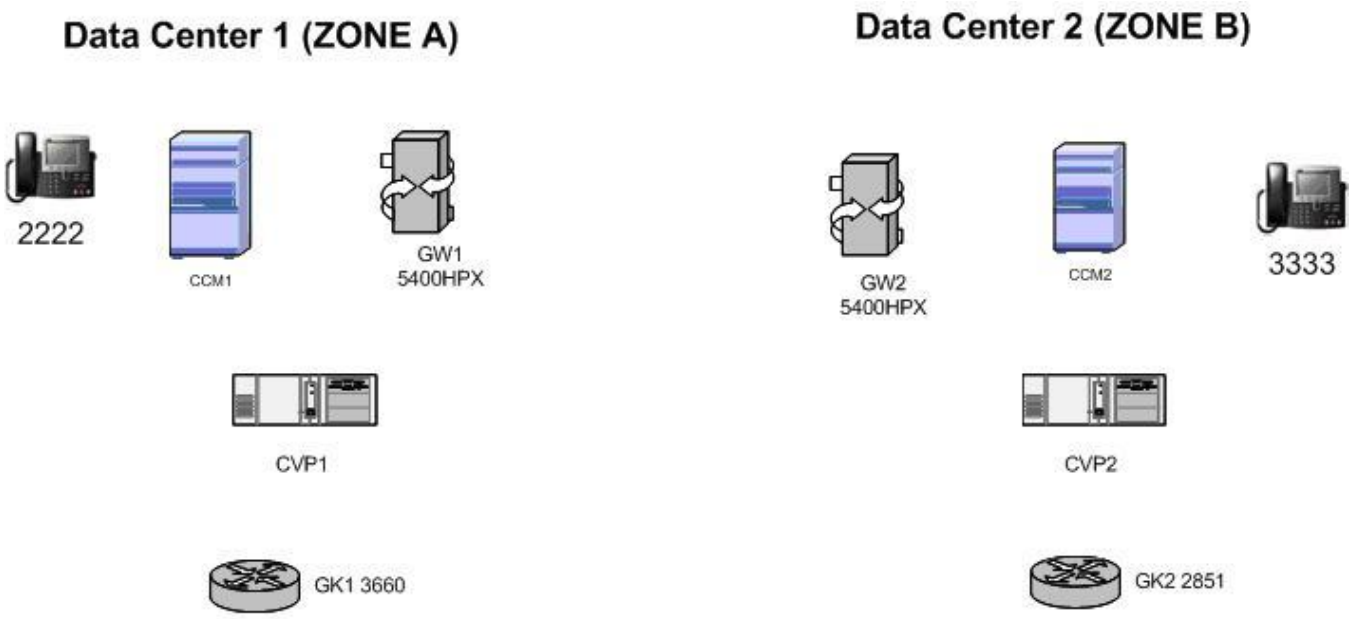
The term *alternate gatekeeper* has multiple meanings:

- The ability of an H.323 endpoint, such as a gateway, to statically configure two gatekeepers to provide a backup in the event the primary fails.
- The ability of an H.323 endpoint to register to a GUP cluster.

Best results are achieved when Unified CVP uses the GUP cluster method of alternate gatekeeper since solution tests have shown that proper load-balancing to Unified CVP devices is difficult to achieve with straight alternate gatekeeper methods. Although Unified CVP itself cannot process GUP messages, it can participate in the GUP cluster. It simply ignores GUP messages and functions as a straight alternate gatekeeper client as described below.

The following is a simple example of Unified CVP being used in a gatekeeper cluster.

Figure 76: Example of a Gatekeeper Cluster



VBAdmin assumption. Configured as follows:

```
setgk "10.86.129.118:zoneA_GK2, 10.86.129.36:zoneA"
```

GK1 assumption. 3660, running c3660-ix-mz.124-6.T, configured as follows:

```
gatekeeper
 zone local zoneA cisco.com 10.86.129.36
 zone local zoneB_GK1 cisco.com
 zone cluster local A_Cluster zoneA
  element zoneA_GK2 10.86.129.118 1719

 zone cluster local B_Cluster zoneB_GK1
  element zoneB 10.86.129.118 1719

 zone prefix zoneA 111111* gw-priority 10 gw1-zoneA
 zone prefix zoneB_GK1 112233* gw-priority 10 gw2-zoneB
 zone prefix zoneA 2222* gw-priority 10 ccm1-zoneA_1
 zone prefix zoneB_GK1 3333* gw-priority 10 ccm2-zoneB_1
 zone prefix zoneA 5900* gw-priority 10 cvp1-zoneA
 zone prefix zoneB_GK1 6900* gw-priority 10 cvp2-zoneB
 gw-type-prefix 1#* default-technology
 lrq forward-queries
 no shutdown
```

GK2 assumption. 851, running c2800nm-ipvoice_ivs-mz.124-6.T, configured as follows:

```
gatekeeper
 zone local zoneB cisco.com 10.86.129.118
 zone local zoneA_GK2 cisco.com
```

```
zone cluster local B_Cluster zoneB
 element zoneB_GK1 10.86.129.36 1719

zone cluster local A_Cluster zoneA_GK2
 element zoneA 10.86.129.36 1719

zone prefix zoneA_GK2 111111* gw-priority 10 gw1-zoneA
zone prefix zoneB 112233* gw-priority 10 gw2-zoneB
zone prefix zoneA_GK2 2222* gw-priority 10 ccm1-zoneA_1
zone prefix zoneB 3333* gw-priority 10 ccm2-zoneB_1
zone prefix zoneA_GK2 5900* gw-priority 10 cvp1-zoneA
zone prefix zoneB 6900* gw-priority 10 cvp2-zoneB
gw-type-prefix 1#* default-technology
lrq forward-queries
no shutdown
```

How It Works

The Unified CVP H.323 Service can be configured with a list of up to five gatekeepers. When the H.323 Service starts, it attempts to register to the first gatekeeper in the list. If the registration is not successful, it proceeds to sequentially try the remainder of the gatekeepers in the list until a successful registration occurs.

The H.323 Service stays registered to that gatekeeper until either:

- That gatekeeper experiences a failure.

  The Voice Browser recognizes a gatekeeper failure in the following ways:

  – The periodic RAS RRQ (registration request) to the gatekeeper times out or is rejected.

  – An ARQ (admission request) on a transfer times out.

- The gatekeeper proactively tells the H.323 Service to unregister, such as when the administrator does a shutdown on the gatekeeper configuration.

- The user executes another setGK from VBAdmin. This will cause the H.323 Service to register with the first gatekeeper in the list, if that gatekeeper is available; otherwise, it will once again do a sequential attempt.

The Unified CVP H.323 Service goes out of service when it detects a gatekeeper failure and stays out of service until it can successfully register to another gatekeeper. During the period that it is out of service, new calls will be rejected by the Unified CVP H.323 Service. The ingress gateway will need to take an alternate routing action for those incoming calls (for example, invoke Unified CVP survivability, or reject the call back to the PSTN network to take alternate call routing action). Calls that are in service will generally be unaffected. This is because either the call has already been transferred or by the time the call needs to be transferred (the only time that the gatekeeper is needed by the Unified CVP H.323 Service). The Unified CVP H.323 Service has most likely already registered to an alternate gatekeeper. However, there is a risk of a timing condition where one or both of the endpoints involved in the transfer (Unified CVP or the target H.323 endpoint) may not have had a chance to register to the alternate gatekeeper.

There is a possibility that these calls may be dropped unless survivability and Unified ICME scripting actions have been taken to compensate for the failure.

Note: Refer to "About the H.323 Service and the Gatekeeper (page 318)."

Although the Unified CVP H.323 Service does not support Gatekeeper Update Protocol (GUP) clustering, there is no reason that the gatekeepers cannot be defined as part of a GUP cluster. In this way, other H.323 endpoints that do support clustering (such as Cisco Unified Communications Manager and IOS gateways) can take advantage of the benefits of GUP.

Unified CVP will simply ignore GUP messages, such as when one of the gatekeepers in the cluster becomes overloaded. Unified CVP would use one or more of the gatekeepers in the cluster as the alternate gatekeepers in its list and detect failure according to the rules mentioned in the bullets above. It is important to note that you must explicitly itemize all the gatekeepers in the cluster using setGK in VBAdmin. If Unified CVP supported GUP clustering, it would only need to register to one of the gatekeepers and the gatekeeper would send Unified CVP the list of gatekeepers in the cluster upon registration.

How to Configure Alternate Gatekeeper Support in Unified CVP

Use the setGK command in VBAdmin to set a list of gatekeepers to which the Unified CVP H.323 Service will register.

Step 1 From Unified CVP VBAdmin, enter **`setGK 10.86.129.33`**.

This command sets up a single gatekeeper to which the H.323 Service will register.

Step 2 Enter: **`setGK 10.86.129.33:zonename1`**

This command sets up a single gatekeeper to which the H.323 Service will register to the zone in the gatekeeper named zonename1.

Step 3 Enter: **`setGK 10.86.129.33, 10.86.129.34, 10.86.129.35`**

This command sets up three gatekeepers to which the H.323 Service can register. In each case, the H.323 Service registers to the first local zone that is configured in that gatekeeper and uses the default RAS port 1719.

Step 4 Enter: **`setGK 10.86.129.33:zonename1:1718, 10.86.129.34`**

The H.323 Service first attempts to register to gatekeeper 10.86.129.33 with local zone zonename1 on port 1718. If that gatekeeper fails, the H.323 Service attempts to register to 10.86.129.34 on port 1719 with the first local zone defined on that gatekeeper.

Unified CVP in a GK Cluster Example

GK1 assumption. 3660, running c3660-ix-mz.124-6.T, configured as follows:

```
gatekeeper
 zone local zoneA cisco.com 10.86.129.36
 zone local zoneB_GK1 cisco.com
 zone cluster local A_Cluster zoneA
  element zoneA_GK2 10.86.129.118 1719

 zone cluster local B_Cluster zoneB_GK1
  element zoneB 10.86.129.118 1719

 zone prefix zoneA 111111* gw-priority 10 gw1-zoneA
 zone prefix zoneB_GK1 112233* gw-priority 10 gw2-zoneB
 zone prefix zoneA 2222* gw-priority 10 ccm1-zoneA_1
 zone prefix zoneB_GK1 3333* gw-priority 10 ccm2-zoneB_1
 zone prefix zoneA 5900* gw-priority 10 cvp1-zoneA
 zone prefix zoneB_GK1 6900* gw-priority 10 cvp2-zoneB
 gw-type-prefix 1#* default-technology
 lrq forward-queries
 no shutdown
```

GK2 assumption. 2851, running c2800nm-ipvoice_ivs-mz.124-6.T, configured as follows:

```
gatekeeper
 zone local zoneB cisco.com 10.86.129.118
 zone local zoneA_GK2 cisco.com
 zone cluster local B_Cluster zoneB
  element zoneB_GK1 10.86.129.36 1719

 zone cluster local A_Cluster zoneA_GK2
  element zoneA 10.86.129.36 1719

 zone prefix zoneA_GK2 111111* gw-priority 10 gw1-zoneA
 zone prefix zoneB 112233* gw-priority 10 gw2-zoneB
 zone prefix zoneA_GK2 2222* gw-priority 10 ccm1-zoneA_1
 zone prefix zoneB 3333* gw-priority 10 ccm2-zoneB_1
 zone prefix zoneA_GK2 5900* gw-priority 10 cvp1-zoneA
 zone prefix zoneB 6900* gw-priority 10 cvp2-zoneB
 gw-type-prefix 1#* default-technology
 lrq forward-queries
 no shutdown
```

Configuring Unified CVP to Propagate UUI to Unified ICME

The UUI extracted from the incoming call is passed to the Unified ICME scripting environment by the Unified CVP Call Server. This can be accomplished by populating the UUS parameter (often known as the UUI) in the IAM message of the GTD (Generic Transparency Descriptor) data that is sent to the gateway from the network in the Q.931 setup message. The gateway and Unified CVP can extract this data and send it to Unified ICME on a new call.

Additionally, other parameters in the GTD can also be extracted and sent to Unified ICME if the user chooses. Any parameter contained in the NSS IAM message can be extracted as long as the ingress IOS gateway also extracts it.

Note: For more information, refer to Delivering H.323 Incoming UUI to a Unified ICME Routing Script (page 335).

The examples in the sections that follow assume that the following GTD data arrived from the network:

```
6616807: *Jan 31 17:12:41.220: gtd msg = "IAM,
PRN,isdn*,,ATT5*,
USI,rate,c,s,c,1
USI,lay1,ulaw
TMR,00
CPN,00,,u,5900
CPC,09
FCI,,,,,,,y,
UUS,3,3132333435
GCI,87c0c79d91dd11daa9c4000bfda207f2
```

Example 1: VRU-Only Call Flow Model

Assume that you want to extract UUS.dat and convert it to ASCII before sending it to Unified ICME. Additionally, you also want to extract CPC.cpc.

Note: You can extract up to 20 GTD fields.

In the gateway, enter the following:

```
conf t
    application
    service <your-cvp-service-name>
    param gtd-attribute0 uus
    param gtd-field0 dat
    param gtd-format0 ascii
    param gtd-attribute1 CPC
    param gtd-field1 cpc
dial-peer voice 123 pots
    service <your-cvp-service-name>
```

The data that will appear in Unified ICME UserToUserInfo will look as follows:

```
uus.dat,12345;;cpc.cpc,09;;
```

Note: The NSS parameters are delimited by two semicolons; fields *within* the NSS parameter are delimited by a comma.

Example 2: Comprehensive Call Flow Model

Assume that you want to extract UUS.dat and convert it to ASCII before sending it to Unified ICME. Additionally, you also want to extract CPN data. Note that unlike with Advanced Speech, you cannot extract to field level; it is all-or-nothing for a parameter. In other words, if you wanted to extract the CPN **noa** field, you also will also extract the CPN **inn**, **noi**, and # fields as well.

On the Unified CVP H.323 Service, enter the following command in VBAdmin:

`setUUI "UUS:2 , CPN"`

Note: The ":2" means convert the second field in the UUS parameter string to ASCII before sending to Unified ICME.

The data that will appear in the Unified ICME UserToUserInfo will look as follows:

`UUS,3,12345;;CPN,00,,u,5900;;`

Note: The NSS parameters are delimited by two semicolons; fields *within* the NSS parameter are delimited by a comma.

If extracting OLI from the GTD, several things need to be done:

- OLI will arrive in the FDC NSS param.
- You need to enter the **`isdn ie oli hex value`** indicates the IE identifier of the OLI in the setup.

  Note: This step is necessary because different switches place OLI in different ISDN IE locations.
- The service provider might need to specifically configure the switch to pass OLI to the gateway.

Debugging Tips for VRU-Only Call Flow Model

On the gateway, enter these commands:

- **`debug voip application script`**
- debug gtd

In the gateway log, look for the following GTD values:

```
6616806: *Jan 31 17:12:41.220: cdapi_find_tsm: Found a gtd msg of
length 144:
6616807: *Jan 31 17:12:41.220: gtd msg = "IAM,
PRN,isdn*,,ATT5*,
USI,rate,c,s,c,1
USI,lay1,ulaw
TMR,00
CPN,00,,u,5900
CPC,09
FCI,,,,,,,y,
UUS,3,3132333435
12345
GCI,87c0c79d91dd11daa9c4000bfda207f2"
```

Look for the GTD values sent to the Unified ICME using the Unified CVP Call Server:

```
>>> CVP bootstrap.tcl: 87C0C79D.91DD11DA.A9C4000B.FDA207F2: UUI sent
to Unified ICME is uus.dat,12345;;cpc.cpc,09;;
```

Debugging Tips for Comprehensive Model

In VBAdmin, execute the following commands:

- **`setH323Trace on`**
- **`setIntTrace on`**

The H.323 Service log will show the GTD that arrives from the gateway and subsequently the CALL_UUI string that is passed to the Unified CVP Call Server. Note that because of URL-encoding, the string that is shown in the Unified CVP logs may not appear totally correct. The validity of the data should be checked in the Unified ICME script or TCD record.

GTD data arriving from the gateway:

```
10:13:25 VoiceBrowser-VB Trace: 00000009: H323: gtd = IAM,
PRN,isdn*,,ATT5*,
USI,rate,c,s,c,1
USI,lay1,ulaw
TMR,00
CPN,00,,u,5900
CPC,09
FCI,,,,,,,y,
GCI,ec0fbfed926911da8aa8ccd458cce346
10:13:28 VoiceBrowser-VB Trace: 00000009: INTF: Fetching VXML.
URL: chleblan-
mcs:8000/servlet/isn?MSG_TYPE=CALL_NEW&ERROR_CODE=0&CLIENT_TYPE=IS
N&CALL_ID=ec0fbfed-9269-11da-8aa8-
ccd458cce346&CALL_DNIS=5900&CALL_UUI=CPC,09
GCI,ec0fbfed926911da8aa8ccd458cce346 &
```

Unified ICME Data Debugging

The UserToUser column in the Termination_Call_Detail table database record in the data base should contain the value of the GTD that got passed. It can also be accessed from the Call.UserToUserInfo call variable in Unified ICME Script Editor.

Unified CVP Endpoint Limitations

The following table lists endpoint limitations of the Unified CVP H.323 Service.

Note: CVP does not currently support the Early Media feature. Early Media is defined when media begins to flow before the call is officially connected. Early Media is the ability for two

user agents to communicate before a call is actually establishedAt present, an H.245 connection is opened only after the called party has accepted the connection (answered the call).

Table 44: Unified CVP Endpoint Limitations

Limitation	Description
Gatekeeper single point failure.	If a Gatekeeper fails and no hot standby is configured, the following features of call routing will be affected: • The H.323 Service will go out of service. • No switch transfers will succeed.
Outbound Alternate Endpoint functionality not fully supported when transferring to agents reserved by Unified ICME.	The agent reserve timer can expire before the call arrives, allowing a different call to reach the agent that does not match the call data previously sent to the agent by the Unified ICME software. Use the router requery feature instead.
G711ulaw is required codec when doing ASR/TTS.	The ASR/TTS vendors currently only support g711ulaw for voice recognition and synthesis.
In Unified CVP Standalone model or VRU-only model, calls from IP phones require an IPIPGW IOS if transferring to another IP endpoint.	When an IP-originated call arrives to an IOS gateway for VoiceXML treatment in either Unified CVP Standalone or VRU-only model, and that call is to be transferred to an IP endpoint, it is necessary to enable the following IOS command: `voice service voip Allow-connections XXX to YYY` where **xxx** = sip or h323, depending on the protocol of the calling party and **yyy** = sip or h323, depending on the protocol of the called party.
H323 Fast Start is not supported for calls originating from or being sent to Unified CM from the Unified CVP H.323 Service.	Disable Cisco Unified Communications Manager Fast Start by changing "H323 FastStart Inbound" flag to false in Cisco Unified Communications Manager service parameters (for Cisco Unified Communications Manager 4.0 and earlier) or by unchecking the "Enable Inbound Fast Start" on the Unified CVP H.323 Service Gateway Device (Cisco Unified Communications Manager 4.1 and later).

Configuring a Gateway

From the Device Management menu, Gateway option, you can add a pre-configured IOS Gateway to the Operations Console. Once added, you can execute a subset of IOS Gateway commands on the Gateway from the Operations Console.

The Ingress Gateway is the point at which an incoming call enters the Unified CVP solution. It terminates TDM phone lines on one side and implements VoIP on the other side. It also provides for sophisticated call switching capabilities at the command of other Unified solution components. It works with either H.323 or SIP protocols, and also supports MGCP for use with Unified CM.

The Ingress Gateway may be deployed separately from the VoiceXML Gateway, but in most implementations they are one and the same: one gateway performs both functions. Gateways are often deployed in farms, for centralized deployment models. In Branch deployment models, one combined gateway is usually located at each branch office.

The VoiceXML Gateway hosts the IOS voice browser, the component which interprets VoiceXML pages from either the Unified CVP IVR service or the VXML Server, plays .wav files and Text-to-Speech (TTS), inputs voice and DTMF, and sends results back to theVoiceXML requestor. It also mediates between Media Servers, VXML Servers, ASR and TTS Servers, and the IVR service.

Unless it is combined with the Ingress Gateway, the VoiceXML Gateway does not require any TDM hardware. All its interfaces are VoIP on one side and HTTP (carrying VXML or **`.wav`** files) and MRCP (carrying ASR and TTS traffic) on the other side. As with Ingress Gateways, VXML Gateways are often deployed in farms for centralized deployment models, or one per office in Branch deployments.

Voice Codec class configuration should be symmetrical on the Ingress gateway and the Egress gateway. Use the voice class codec command to define the list of codecs available and the order in which they should be tried.

For example, if G.729 is the preferred codec, the codec class should be configured to prefer G.729 and then G.711 on both gateways as shown in the example below. Codec class 2 is used for Codec negotiation.

```
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw


voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
```

Refer to the Operations Console online help topics under *Managing Devices > Configuring a Gateway* to perform the following tasks:

- Adding a Gateway
- Editing a Gateway
- Deleting a Gateway
- Gateway Configuration Settings
- Executing Gateway Commands
- Getting Gateway Statistics
- Transferring a File to a Gateway
- Finding a Gateway

Transferring a File to a Gateway

You can transfer a single file at a time from the Operations Server to one or more gateways.

If you want to transfer multiple files at a time, use Bulk Administration File Transfer. Refer to the Operations Console online help topic *Bulk Administration > Transferring Scripts and Media Files*.

Procedure

To transfer files between the Operations Server and a gateway:

Step 1 Choose **Device Management > Gateway**.

The Find, Add, Delete, Edit Gateway window lists any gateways that have been added to the Operations Console.

Step 2 Select a gateway by clicking on the link in its name field or by clicking the radio button preceding it and then clicking **Edit**.

The Edit Gateway Configuration window opens.

Step 3 Select **File Transfer > Scripts** from the Gateway configuration toolbar.

The File Transfer window opens.

Step 4 Select a file to transfer to the gateway.

a. If the script is located on your local machine, click **Select a script file from your local PC**, then click **Browse** and select the script file to transfer to the Operations Server.

b. If the script is located on the Operations Server, click **Select from available script files**.

Step 5 When you have selected the file to transfer, click **Transfer** to copy the selected file to the Operations Console.

Configuring a Gatekeeper

From the Device Management menu, Gatekeeper option, you can add a pre-configured IOS Gatekeeper to the Unified CVP Operations Console. Once added, you can execute a subset of IOS Gatekeeper commands on the Gatekeeper from the Operations Console.

Refer to the Operations Console online help topics under *Managing Devices > Configuring a Gatekeeper* for information on the following tasks:

- Adding a Gatekeeper

- Finding a Gatekeeper
- Deleting a Gatekeeper
- Editing a Gatekeeper
- Executing Gatekeeper Commands
- Gatekeeper Configuration Settings
- IOS Gatekeeper Commands
- Getting Gatekeeper Statistics
- Transferring a File to a Gatekeeper

Configuring an ICM Server

Unified CVP provides Voice over IP (VoIP) routing services for the Cisco Unified Intelligent Contact Management Enterprise (Unified ICME) and Cisco Unified Contact Center Express products. ICME provides the services necessary to determine where calls should be routed, whether to ACDs, specific agents, or to VRUs, but the routing services themselves must be provided by an external routing client.

An ICM Server is required in Unified CVP Comprehensive, Call Director, and VRU-Only call flow models.

Refer to the Operations Console online help topic *Configuring a Unified ICM Server* to complete the following tasks:

- Adding an ICM Server

  You must configure the ICM Server before adding it to the Unfied CVP Operations Console. Once added, you can add the ICM Server to a device pool.

  Record the IP address of the ICM Server before adding it to the Operations Console.

- Deleting an ICM Server

  Deleting an ICM Server deletes the configuration of the selected ICM Server in the Operations Console database and removes the ICM Server from displayed list of ICM Servers.

- Editing an ICM Server (change an existing server configuration)

- Finding an ICM Server within the list of configured servers

  You can find the record for a specific server by scrolling through a list of configured servers, or by searching for the server name. Advanced search is also available in the Operations Console.

The Operations Console online help topic *Adding a Unified ICM Server* provides a table of server parameters that can be identified and configured. In addition to basic fields such as IP address, the help describes new fields available in Unified CVP release 8.0(1) and greater.

Chapter 14

Configuring High Availability for Unified CVP

This chapter provides information about how to accommodate load balancing and redundancy in Unified CVP deployments.

This chapter contains the following topics:

Using Cisco Content Services Switch (CSS) for Load Balancing in Unified CVP

A Content Services Switch (CSS) provides load-balancing services for HTTP and MRCP trafic, but not for call control signaling SIP or H.323 messages. You can add a CSS to the network map that is displayed in the control center Once added, you can add a CSS to a device pool and execute a subset of IOS commands on the CSS from the Operations Console and transfer files between the CSS and the Unified CVP Operations Server (see the "Configuring a Content Services Switch (CSS) (page510)" and the "Configuring a Speech Server (page 517)" sections).

In this application of CSS, the engine is used primarily to direct initial session requests for a particular type of service. There are four types of services:

- http prompts
- ASR/TTS

- Unified CVP Call Server
- VXML Server

Note: CSS doesn't work with Helix Streaming Server.

The following general approach will apply to configuring each Unified CVP component type for use with CSS. Specific component-type configuration is covered in the following sections.

Note: For the following topics, refer to **CSS configuration documentation** (http://www.cisco.com/en/US/products/hw/contnetw/ps792/products_installation_and_configuration_guides_list.html).

Services—One CSS Service is configured for each Unified CVP component that needs CSS load balancing.

Rules—A content rule should be established for each Unified CVP component type. In order to define a content rule, a content owner must be defined on the Content Switch. (Refer to *CSS Documentation*.) Media Servers, Call Servers, ASR/TTS Servers and VXML Servers might each have their own content rules defined.

Groups—It may be necessary to define source groups for each service. Group configuration will vary depending upon whether a single VLAN or multiple VLANs are used for the CSS' inbound and outbound traffic from the voice gateway and the Unified CVP component servers. If the servers and the voice gateway clients are on the same VLAN, then destination services will need to be specified. This configuration is sometimes referred to as a'one-arm' configuration because all traffic passes through one interface. Refer to **CSS Load Balancing Using One Interface Configuration Example** (http://www.cisco.com/en/US/products/hw/contnetw/ps789/products_configuration_example09186a0080093dff.shtml)f.

Also, refer to **CSS configuration documentation** (http://www.cisco.com/en/US/products/hw/contnetw/ps792/products_installation_and_configuration_guides_list.html) for help determining whether your setup will require destination services.The examples in the next section incorporate the use of destination services.

Keepalives—Each Unified CVP component type will also have a varying type of keepalive defined. The CSS keepalive definitions will allow for an appropriate method of determining whether the component is functional and able to participate in receiving requests.

Server Groups

A Server Group is a dynamic routing feature that enables the originating endpoint to have knowledge of the status of the destination address before attempting to send the SIP INVITE. Whether the destination is unreachable over the network, or is out of service at the application layer, the originating SIP user agent can have fore-knowledge of the status through a heartbeat mechanism.

Although, there was already an H.323 endpoint registration mechanism ("local SRV"), the Server Groups features adds a heartbeat mechanism with endpoints for SIP. This feature enables faster failover on call control by eliminating delays due to failed endpoints.

The following list is a summary of important configuration items:

- Server Groups are not automatically added to your configuration. You must explicitly configure Server Groups for their deployment and turn on this feature.
- If you have already configured the "local SRV" feature and therefore created a srv.xml file, you must run the **srvimport.bat** command before you configure Server Groups using the Operations Console. Otherwise, your existing definitions will be overwritten. This process is explained in the configuration details that follow.
- You define Server Groups using the Operations Console. You must always configure at least one Call Server first, because you will not be able to save the Server Groups configuration without assigning it to at least one Call Server.
- Using Server Groups with Proxy Servers. If you are using the CUP proxy, typically the SRV cluster name (such as "proxy-servers.cisco.com") will need to be defined in the service parameters section of the proxy configuration. Otherwise a *404 not found* rejection may result. The CUSP proxy has a similar configuration in CLI.

Configure Server Groups

Complete the following steps to configure Server Groups:

1. If you have previously created an srv.xml file, after you upgrade your Unified CVP installation, run the batch file **srvimport.bat** to transfer your prior configuration to the new Server Groups feature.

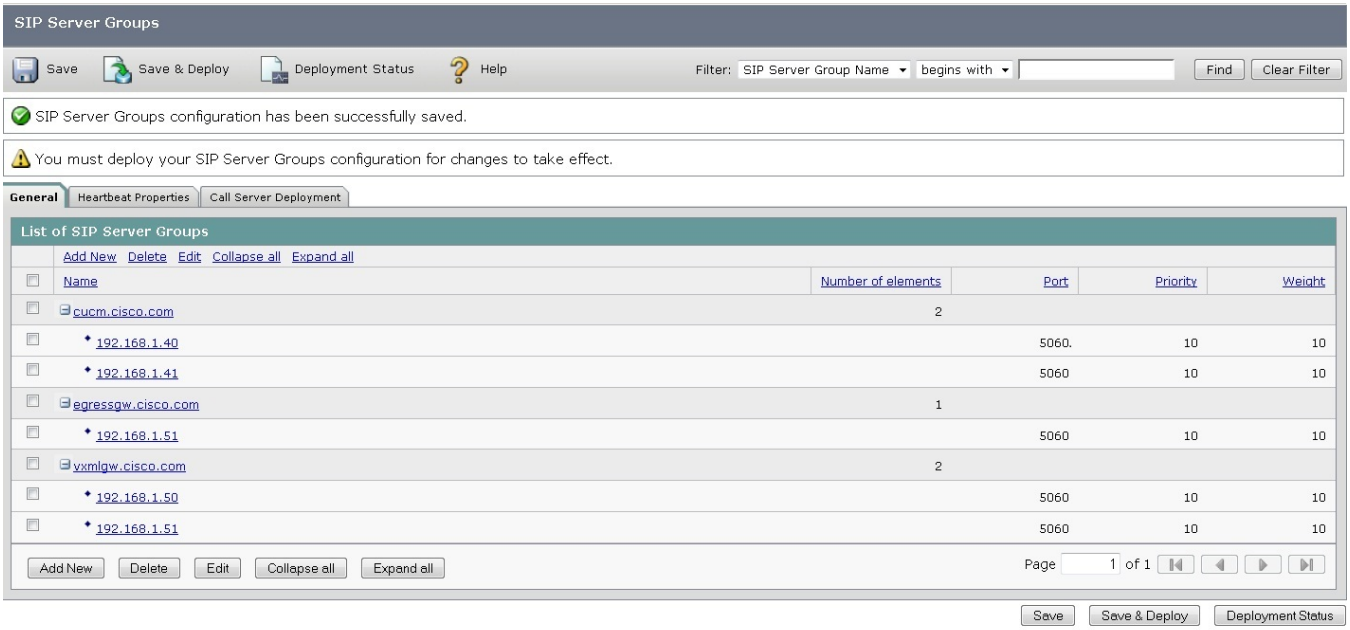
   The srvimport.bat file is located in the **CVP bin directory**. This batch file takes your **srv.xml** file as an argument. Copy this file from your Call Server configuration directory. Running srvimport.bat brings this configuration data into the Operations Console.

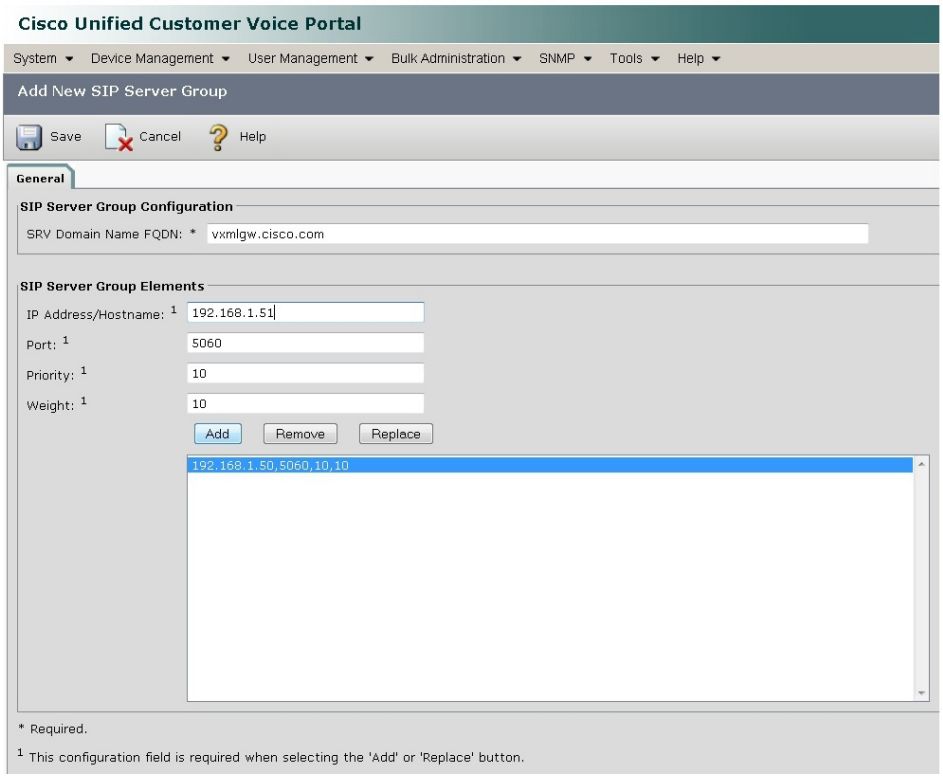
   Note: You must **stop** the OAMP (Operations Console) service before you run the .bat file.

2. If you have not defined a Call Server using the Operations Console, refer to *Configuring a Call Server* in the Operations Console online help.

3. Use the following window in the Operations Console to configure Server Groups. Access this window by selecting **System > SIP Server Groups**.

Figure 77: Server Groups Configuration Using Operations Console





4. A Server Group consists of one or more destination addresses (endpoints) and is idenfied by a Server Group domain name. This domain name is also known as the SRV cluster name, or Fully Qualified Domain Name (FQDN). Define the FQDN and add it to the list. Refer to Configuring Server Groups in the Operations Console online help.

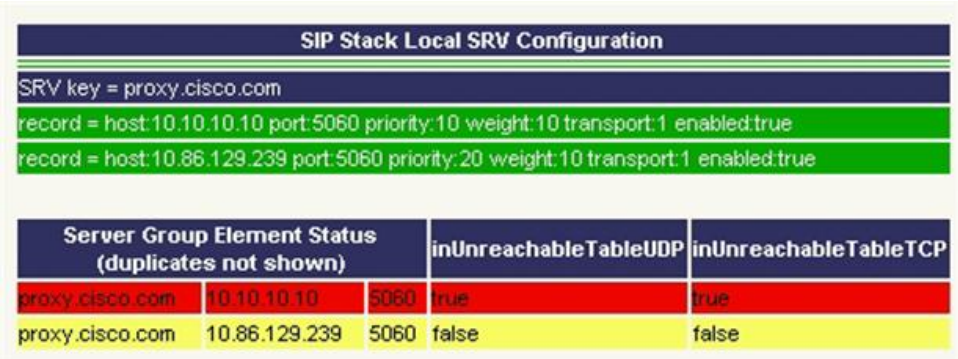
5. Refer to SIP Server Group Configuration Settings in the Operation Console online help to complete the Server Group configuration.

6. Click the **Call Server Deployment** tab and select the Call Server(s) that you want to associate with the Server Group(s). The click **Save & Deploy**.

Server Groups Diagnostics

The CVP log file has traces which show endpoint status events. From the diagnostic servlet, click on the link for *dump SIP state machine* to display information like in the following example:

Figure 78: Server Group CVP Log File Endpoint Status Events



SIP Stack Local SRV Configuration

SRV key = proxy.cisco.com

record = host:10.10.10.10 port:5060 priority:10 weight:10 transport:1 enabled:true

record = host:10.86.129.239 port:5060 priority:20 weight:10 transport:1 enabled:true

Server Group Element Status (duplicates not shown)			inUnreachableTableUDP	inUnreachableTableTCP
proxy.cisco.com	10.10.10.10	5060	true	true
proxy.cisco.com	10.86.129.239	5060	false	false

CSS Configuration for Media Servers

Media Servers are standard web servers that are responsible for serving Unified CVP prompt files to the voice gateway.

Note: The following sample configuration applies only to Comprehensive configuration (Type 7), when the Voice Gateway is making the request for the media file. Do not use media servers in Type 5 configurations where the Unified CVP H.323 Service is requesting media files. The Unified CVP H.323 Service does not support redirect and all media requests would be channeled through the content server. This would degrade performance considerably.

CSS Configuration for Media Servers: Service

Configure one CSS Service per Media Server.

Note: Services must be activated before they are available to be used.

The following three services represent three Media Servers that can be accessed for Unified CVP prompt playing.

```
Example of Configuring Services:

service mediaserver1
  ip address 10.1.1.1
  port 80
  domain 10.1.1.1
  keepalive type http
```

```
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive uri "/index.html"

service mediaserver2
  ip address 10.1.1.2
  port 80
  domain 10.1.1.2
  keepalive type http
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive uri "/index.html"

service mediaserver3
  ip address 10.1.1.3
  port 80
  domain 10.1.1.3
  keepalive type http
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive uri   "/index.html"
```

CSS Configuration for Media Servers: Content Routing Rule

Using the content owner you have defined, define a content routing rule for the Unified CVP Media Servers. CSS offers many balancing types for choosing an active media server. This application uses the default defined round-robin as the balancing method. Additionally, a primarySorryServer is defined.

The following content rule implements a layer 5 http round-robin routing rule for two media servers, with the third server reserved for failover.

```
Example of Content Rule for Media Servers
content MEDIA
 vip address 10.1.1.4
 protocol any
 port 80
 url "/*"
 add service mediaserver1
 add service mediaserver2
 primarySorryServer mediaserver3
 active
```

CSS Configuration for Media Servers: Keepalive Method

The media servers are standard http servers. CSS supports varying methods of keepalives used to determine the availability of an individual service.

For media servers, use the http keepalive method. This will ensure that CSS determines the availability of a server at the application layer. In order to implement this keepalive method, the service must identify an available html page which the keepalive will use to determine http availability. For this reason, in addition to the Unified CVP prompt audio files, the server should

have an available html document defined. The example assumes an “index.html” document is available on the media servers. The http keepalive method will continuously check for the availability of this document as a means to determine that the server is alive and capable of serving documents.

It is essential that all media servers included in a Media Server Content rule contain all of the Unified CVP audio prompt files intended for use by that content rule to ensure that no matter which server CSS directs the request to, the prompt file will be available.

Use these CSS commands to setup the http keepalives while configuring each media server service:

```
keepalive type http
keepalive uri "/index.html"
```

CSS Configuration for Media Servers: Keepalive Times

Because these services are directly involved in processing voice calls, the shortest possible failover and re-activation times are desired. As such, set the following keepalive parameters:

```
keepalive retryperiod 2
keepalive maxfailure 1
```

CSS Configuration for Media Servers: Groups

Define a group for the participating media servers if your configuration requires source groups (see the discussion of Groups in the "General Approach (page 489)" section for details). Then:

- Activate services, group, and rules
- Activate each service and group of media-servers

```
group mediaservers
 add destination service mediaserver1
 add destination service mediaserver2
 add destination service mediaserver3
 vip address 10.1.1.4
```

CSS Configuration for Media Servers: Voice Gateway

The Unified CVP media server names are defined inside the VoiceXML documents, which are passed to the Voice Gateway for processing. The media server name is set within the Unified ICME routing script using an enterprise call variable. (Note that for use with the content switch, the default (not specified) media server name of “file:\\..\MediaServer” is an invalid option. You must specify a valid media-server name.) You must also ensure that the host name that Unified ICME will deliver to the voice gateway as the name of the media server resolves to the virtual IP address of the content rule for the media servers setup on CSS. For example, if the Unified ICME script sets a Media Server name as <<media>>, then the voice gateways to which the request will be delivered must resolve the name <<media>> to the VIP of the media-server

content rule. This example would then be configured within the voice gateway using the command:

```
ip host media 10.1.1.4
```

Note that when using CSS, the media server logic will generate VoiceXML documents that contain <<-backup>> default backup media-server names. A VoiceXML document returned to the gateway for prompt playback contains failover logic within the vxml. For example, if the media server is named <<media>>, this failover logic would include instructions for trying a server that would be named <<media-backup>>, as the example below reflects.

```
ip host media-backup 10.1.1.4
```

Although a properly configured, redundant CSS setup means that the gateway will never actually be required to access the <<-backup>> default backup media-server names. A VoiceXML document returned to the gateway for prompt playback contains failover logic within the VoiceXML. For example, if the media server is named <<media>>, this failover logic would include instructions for trying a server that would be named <<media-backup>>, as the example below reflects.

CSS Configuration for Media Servers: Multiple Media-Server Rules

You can apply the media-server configuration in a CSS to support a logical division of media based upon any reasonable grouping that the design might call for. As long as the Unified ICME routing script can decide and set the media server name appropriately, CSS can support multiple media server routing rules.

For example: If a Unified CVP system deployment includes multiple language prompts, English and Spanish, the company may elect to house all Spanish versions of media server prompts on 3 media servers, and all English prompts on three separate media servers.

Note: This choice would be independent of the virtual directory setting, which is controlled by the Unified CVP "language" parameter. While this parameter alters the path in the url of media, our example assumes that the company further wants to house different languages on different servers—perhaps for traffic volume purposes. For more information on the language parameter, see Chapter 16, "Configuring the Media Servers (page 531)."

In this example, the Unified ICME routing script that constructs the VRU call flow would identify and set the media-server name differently depending upon whether the caller initially has chosen English or Spanish. If they choose English, you might re-set the Media Server name to "media-english" and if Spanish is selected, "media-spanish."

To support this, six services would be configured within CSS, one service for each media server. Two (2) content rules would also be configured. Each content rule would map to a different vip address and contain only those services which match the language the rule is configured for. Finally, the voice gateways would have separate host entries for the English and Spanish media-server names (and their –backup version) as set within the Unified ICME scripts. After the content rules are applied, the following host entries would then appear on the voice gateways:

```
ip host media-english 10.2.1.1
```

```
ip host media-english-backup 10.2.1.1
ip host media-spanish 10.2.1.2
ip host media-spanish-backup 10.2.1.2
```

Here is what the content rules would look like:

```
content MEDIA-ENGLISH
 vip address 10.2.1.1
 protocol any
 port 80
 url "/*"
 add service mediaserver-english1
 add service mediaserver-english2
 primarySorryServer mediaserver-english3

content MEDIA-SPANISH
 vip address 10.2.1.2
 protocol any
 port 80
 url "/*"
 add service mediaserver-spanish1
 add service mediaserver-spanish2
 primarySorryServer mediaserver-spanish3
```

CSS Configuration for Call Servers

In Unified CVP Release 4.0 and later, CSS can only be used for load-balancing of Call Servers with a Unified CVP VRU-Only call flow model.

Note: The Unified CVP H.323 Service maintains a list of its own Call Servers used as an interface between the H.323 Service and the Unfied ICME for route requests. Retry logic within the H.323 Service controls requests to the H.323 Service's Call Servers for routing requests via the "CallServerList" H.323 Service definitions. If separate Call Servers are used for a Unified CVP H.323 Service's interface to Unified ICME for route requests, these Call Servers do not require configuration within CSS.

CSS Configuration for Call Servers: Service

Configure one Service per Unified CVP Call Server.

The following services represent Call Servers that will function as IVR Services, or as routing client interfaces to Unified ICME where a Unified CVP H.323 Service is making the call request.

Note: If no keepalive retry period is specified, the default value of 5 seconds (recommended value) is used.

```
service callserver1
```

```
protocol tcp
keepalive port 8000
keepalive type http
port 8000
ip address <ip address>
keepalive uri "/cvp/VBServlet?MSG_TYPE=HEARTBEAT&TIMEOUT=0"
keepalive maxfailure 1

service callserver2
protocol tcp
keepalive port 8000
keepalive type http
port 8000
ip address <ip address>
keepalive uri "/cvp/VBServlet?MSG_TYPE=HEARTBEAT&TIMEOUT=0"
keepalive maxfailure 1
```

CSS Configuration for Call Servers: Content Routing Rule

Using the content owner you have defined, define a content routing rule for the Call Servers. CSS offers many balancing types for choosing an active Call Server. This application uses the default defined round-robin as the balancing method. Additionally, a primarySorryServer is defined.

The AS content rule is implemented as a layer 4 routing rule.

```
content callserver
vip address <vip ip address>
add service callserver1
add service callserver2
protocol tcp
port 8000
```

CSS Configuration for Call Servers: Groups

Define a group for the participating Call Servers if your configuration requires source groups (see the discussion of Groups in the "General Approach (page 489)" section for details).

```
group CVPCallServers
vip address <vip ip address>
add destination service callserver1
add destination service callserver2
```

CSS Configuration for Call Servers: Activate Services, Group, and Rules

Activate each service and group of Call Servers.

CSS Configuration for Call Servers: Voice Gateway Configuration

The Unified CVP Call Server names are defined inside the VoiceXML documents which are passed to the Voice Gateway for processing. The names are hard-coded within the initial bootstrap.vxml VoiceXML document housed on the Gateway.

Note: For more information, see Chapter 2, "Configuration Overview (page 25), in Part 1.

For a CSS implementation, the host name embedded within the bootstrap document is "Customer Voice Portal-vxml". You must ensure that the host name isn-vxml is set to resolve to the virtual ip address of the content rule for the Call Servers setup on CSS. This is configured within each voice gateway using the command:

```
ip host cvpserver 10.86.130.90
```

where 10.86.130.90 is the VIP address servicing the Call Server content rule.

CSS Configuration for ASR/TTS Servers

If your Unified CVP deployment includes the use of ASR/TTS resources for call processing, the ASR/TTS servers may also be configured within CSS. If yourASR/TTS deployment utilizes a multi-tier architecture to separate MRCP requests from recognition and vocalizer functions (such as a multi-tier Nuance configuration), only the MRCP servers are load-balanced by CSS. The RTSP traffic generated as part of MRCP will be routed around CSS, rather than through it. While this is generally handled by default within MRCP, it may be necessary in some MRCP configurations to specifically instruct the server to pass the RTSP address. Consult yourASR/TTS documentation for specific details.

To route the voice stream around CSS rather than through it, you need to configure the Gateway using the “mrcp client rtpsetup enable” configuration command.

CSS Configuration for ASR/TTS Servers: Service

Configure one Service per ASR/TTS Server responsible for handling MRCP requests.

```
service asrtts1
  port 554
  protocol tcp
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive type tcp
  keepalive port 554
  ip address 10.1.1.9

service asrtts2
  port 554
  protocol tcp
  keepalive retryperiod 2
```

```
    keepalive maxfailure 1
    keepalive type tcp
    keepalive port 554
    ip address 10.1.1.10

service asrtts3
    port 554
    protocol tcp
    keepalive retryperiod 2
    keepalive maxfailure 1
    keepalive type tcp
    keepalive port 554
    ip address 10.1.1.11
```

CSS Configuration for ASR/TTS Servers: Content Rule

Using the content owner you have defined, define a content routing rule for the ASR/TTS Servers. The ASR/TTS content rule is implemented as a layer 4 routing rule.

```
content ASR
 add service asr1
 add service asr2
 primarySorryServer asr3
 protocol tcp
 port 554
 vip address 10.1.3.12

content TTS
 add service tts1
 add service tts2
 primarySorryServer tts3
 protocol tcp
 port 554
 vip address 10.1.3.13
```

CSS Configuration for ASR/TTS Servers: Keepalive Method

For the MRCP ASR/TTS servers, a socket-level keepalive is used. By implementing a socket keepalive, the CSS will make a connection to the MRCP port to validate that the MRCP server is running. The service is considered down if unable to connect to port 554 for MRCP traffic.

To configure the CSS MRCP Keepalive, use the following:

```
keepalive type tcp
keepalive port 554
```

Note: For detailed instructions on implementing keepalives for CSS, refer to **CSS configuration documentation** (http://www.cisco.com/en/US/products/hw/contnetw/ps792/products_installation_and_configuration_guides_list.html).

CSS Configuration for ASR/TTS Servers: Keepalive Times

Because these services are directly involved in processing voice calls, the shortest possible failover detection is desired. As such, set the following keepalive parameters:

```
keepalive retryperiod 2
keepalive maxfailure 1
```

CSS Configuration for ASR/TTS Servers: Groups

Define a group for the participating ASR/TTS Servers if your configuration requires source groups (see the discussion of Groups in the "General Approach (page 489)" section for details).

```
group asrtts
  add destination service asrtts1
  add destination service asrtts2
  add destination service asrtts3
  vip address 10.1.1.12
```

CSS Configuration for ASR/TTS Servers: Activate Services, Group, and Rules

Activate each service and group of ASR/TTS servers.

CSS Configuration for ASR/TTS Servers: Voice Gateway

As with media servers and Call Servers, the ASR / TTS names should resolve to the VIP address running the content rule for ASR and/or TTS on the CSS. This name resolution occurs at each voice gateway. Note that ASR and TTS requests each utilize separate names within the Unified CVP infrastructure, even if the same server(s) handle both ASR and TTS functions. The default English ASR and TTS server names are:

`sr-en-us`

and

`asr-en-us-backup`

For TTS:

`tts-en-us`

and

`tts-en-us-backup`

These host names should be set to resolve to the VIP address of the content rule that governs each service. (Note that in the CSS sample above, both ASR and TTS functions are handled by the same content rule and group of resources.)

And the gateway entries at the voice gateway would look like this:

```
ip host asr-en-us 10.1.3.12
ip host asr-en-us-backup 10.1.3.12
ip host tts-en-us 10.1.3.13
ip host tts-en-us-backup 10.1.3.13
```

If a network design contains separate MRCP servers for ASR and TTS functions, CSS would contain separate services, separate source groups (if needed), and two separate content rules / vip addresses—one for ASR and another for TTS resources.

In this case, the CSS content rules might look like this:

```
content ASR
  add service asr1
    add service asr2
      primarySorryServer asr3
    protocol tcp
  port 554
  vip address 10.1.3.13


content TTS
  add service tts1
    add service tts2
      primarySorryServer tts3
    protocol tcp
  port 554
  vip address 10.1.3.13
```

And the gateway entries at the voice gateway would look like this:

```
ip host asr-en-us-backup 10.1.3.12
ip host tts-en-us 10.1.3.13
ip host tts-en-us-backup 10.1.3.13
```

CSS Configuration for VXML Servers

VXML Servers are web servers that are responsible for serving VoiceXML documents to the voice gateway.

CSS Configuration for VXML Servers: Service

Configure one CSS service per VXML Server.

Note: Services must be activated before they are available to be used.

The following two services represent two VXML Servers that can be accessed for VoiceXML documents.

```
service CVPVoicexml1
  ip address 209.165.200.225
  port 7000
  keepalive type http
  keepalive type 7000
  keepalive uri "/index.jsp"
  protocol tcp
  active
```

CSS Configuration for VXML Servers: Content Rule

Using the content owner you have defined, define a content routing rule for theVXML Servers. CSS offers many load balancing types for choosing an active VXML Server. This application uses the default defined round-robin as the balancing method. Additionally, once the VXML Server is chosen for the call, the CSS will use an internally generated cookie to guarantee that all future VoiceXML requests in that call use the same VXML Server.

Example: Content Rule for VXML Servers

```
content CVPVoicexml2_content
  protocol tcp
  add service CVPVoicexml1
  add service CVPVoicexml2
  vip address 209.165.200.227
  port 7000
  url "/*"
  advanced-balance arrowpoint-cookie
  active
```

CSS Configuration for VXML Servers: Keepalive Method

For VXML Servers, use the http keepalive method. This ensures that the CSS determines the availability of a server at the application layer. In order to implement this keepalive method, the service must identify an available html or jsp page which the keepalive will use to determine http availability. For this reason. the server should have an available html/jsp document defined. This example assumes “index.jsp” is available on the VXML Server. The http keepalive method continuously checks for availability of this document as a means to determine that the server is capable of serving documents.

Use these CSS commands to set up the http keepalive parameters while configuring each VoiceXML Service. Do not specify any other keepalive parameters such as frequency or retry count. The CSS defaults of a 5-second frequency and 3 retries will be in effect. If the VXML Server is running on an Apache Tomcat Call Server, configure the following keepalive parameters:

```
keepalive type http
keepalive uri "/index.jsp"
```

CSS Configuration for VXML Servers: Groups

Define a group for the participating VXML Servers if your network configuration requires source groups (see the discussion of Groups in the "General Approach" section for details).

```
group vxml_group1
 vip address 10.1.3.14
 add destination service CVPVoicexml1
 add destination service CVPVoicexml2
 active
```

Redundancy and Failover for Unified CVP

This section describes redundancy and failover mechanisms for ASR, TTS, Media, and VXML Servers in the Unified CVP solution.

ASR, TTS, and Media Server Redundancy for VXML Server Applications (with CSS)

If you use a CSS for ASR or TTS servers, then the IP address specified in the following ivr commands should be the Virtual IP (VIP) address for the corresponding ASR or TTS service. VXML Server applications rely on the gateway's configured default for ASR and TTS servers, which allow only a single host name or IP address to be specified for each. This differs from the Unified CVP micro-applications based applications, which support automatic retries to specifically named backup ASR and TTS servers.

Use the following configuration on the gateway if you are using Nuance, Scansoft, or IBM WebSphere ASR/TTS servers:

```
ip host asr-en-us 10.10.10.1
ip host tts-en-us 10.10.10.2
```

Use the following configuration on the gateway if you are using Nuance, Scansoft, or IBM WebSphere ASR/TTS servers:

```
mrcp client rtpsetup enable
ivr asr-server rtsp://asr-en-us/recognizer
ivr tts-server rtsp://tts-en-us/synthesizer
http client cache memory pool 15000
http client cache memory file 500
ivr prompt memory 15000
ivr prompt streamed none
mrcp client timeout connect 5
mrcp client timeout message 5
rtsp client timeout connect 10
rtsp client timeout message 10
vxml tree memory 500
http client connection idle timeout 10
no http client connection persistent
```

The URL configured by the above ivr commands defines the gateway's default target for ASR and TTS services, and is in effect for all calls handled by that gateway. You can override it dynamically in your VXML Server application by populating the Cisco-proprietary VoiceXML properties "com.cisco.asr-server" or "com.cisco.tts-server."

Media servers require no special gateway configuration to work with a CSS for automatic failover.

ASR, TTS, and Media Server Redundancy for Micro-app-based Applications (without CSS)

If you *do not* use a CSS for ASR or TTS servers, the IVR Service plays a significant role in implementing a failover mechanism for Media Servers, ASR/TTS Servers and micro-app-based applications. Up to two of each such servers are supported, and the IVR Service orchestrates retries and failover between them.

Note:

- This redundancy mechanism is only available for Unified CVP micro-applications.
- For information about setting up the IVR Service to accommodate failover, refer to the Operations Console online help.

About the IVR Service Failover Mechanism

The IVR Service failover mechanism applies to:

- Connections between the IVR Service and the IOS Voice Browser, only.

  Note: Unified CVP H.323 Service has its own failover mechanism for Media Server and does not communicate with an ASR/TTS Server.

- All communication between the IOS Voice Browser and an ASR Server, TTS Server, or Media Server. For ASR/TTS Servers, the failure mechanism is only useful if a CSS *is not* configured for the gateway aliases: **`asr-<locale>`** and **`tts-<locale>`**.

- Media Server, when the ICM Script ECC variable, **user.microapp.media_server**, is set to **`mediaserver`**. When **user.microapp.media_server** is set to **`mediaserver`**, the IVR Service uses the IP Address defined on the gateway as:

  – ip host mediaserver 10.86.129.50

  – ip host mediaserver-backup 10.86.129.51

  Note: If **user.microapp.media_server** is configured as the hard-coded IP Address of the media server, then the IVR Service will not perform any failover for the media server.

If the IVR Service receives a Call Result error code value of **9** (MEDIA_FILE_NOT_FOUND), **33** (GENERAL_ASR_TTS), **31** (MEDIA_RESOURCE_ASR) or **32** (MEDIA_RESOURCE_TTS), it does the following:

- When attempting to connect to a *Media Server*, the IVR Service:
  - Resends the request the number of times defined in the IVR Service Configuration's **Media Server Retry Attempts** field.
  - If the connection is not successful after the specified number of attempts, and the IVR Service Configuration's **Use Backup Media Servers** field is set to **Yes** (the default), the IVR Service makes the same number of attempts to retrieve the media from a backup media server before failing and generating an error.

    Note: The backup media server is defined on the gateway as <mediaserver>-backup.
  - Passes the error in a Call State Event to the ICM Service, which then passes it to Unified ICME.
- When attempting to connect to an *ASR/TTS Server*, the IVR Service:
  - Resends the request the number of times defined in the IVR Service Configuration's **ASR/TTS Server Retry Attempts** field.
  - If the connection is not successful after the specified number of attempts, and the IVR Service Configuration's **Use Backup ASR/TTS Servers** field is set to **Yes** (the default), the IVR Service makes the same number of attempts to connect to a backup ASR/TTS server before failing and generating an error.

    Note: The backup ASR and TTS servers are defined on the gateway as asr-<locale>-backup and tts-<locale>-backup.
  - Passes the error in a Call State Event to the ICM Service, which then passes it to Unified ICME.

Each new call attempts to connect to the primary server. If failover occurs, the backup server is used for the duration of the call; the next new call will attempt to connect to the primary server.

Note: This failover mechanism differs from that used in prior releases of Unified CVP software. Legacy releases used a *sticky* connection. In a sticky connection, if failover occurs to a backup server, subsequent new calls automatically connect to the backup server, rather than attempt to connect with the primary server.

CSS Redundancy

CSS supports several types of failover mechanisms. The Virtual IP redundancy mechanism will allow for a rapid failover from a defined Master CSS to its backup using VRRP. This failover takes less than three seconds and might or might not additionally incorporate the use of ASR. Unified CVP transactions to servers do not require a statefull connection to the gateway. If the CSS fails during a prompt, the VoiceXML documents that drive Unified CVP will retry and re-stream the prompt from the new CSS access.

For this reason, general VIP redundancy is implemented. The sample below shows the redundant section for each of the two CSSs.

```
!************ CIRCUIT (CSS1)************
circuit VLAN1
  ip address 10.1.1.100 255.255.255.0
  ip virtual-router 1 priority 230 preempt
  ip redundant-interface 1 10.1.1.200
  ip redundant-vip 1 10.1.1.4
  ip redundant-vip 1 10.1.1.8
  ip redundant-vip 1 10.1.1.12

!************ CIRCUIT (CSS2) ************
circuit VLAN1
  ip address 10.1.1.101 255.255.255.0
  ip virtual-router 1 priority 230 preempt
  ip redundant-interface 1 10.1.1.200
  ip redundant-vip 1 10.1.1.4
  ip redundant-vip 1 10.1.1.8
  ip redundant-vip 1 10.1.1.12
```

See Also

For information and step-by step configuration guidelines on implementing CSS Redundancy using the VIP and VRRP mechanism, refer to *CSS Advanced Configuration Guide*.

Sample Configuration for CSS Redundancy

The following is a complete CSS sample configuration for each of two CSS Servers configured in a Virtual Router Redundancy configuration and implementing each of the Unified CVP Services described above.

CSS Sample Configuration

Note that in this configuration example, a "one-arm" configuration is used because the inbound traffic and the services exist on the same vlan.

```
*************** GLOBAL ****************
ip route 0.0.0.0 0.0.0.0 10.1.1.20 1

!************** INTERFACE **************
interface 2/6
  phy 100Mbits-FD
interface 2/16
  bridge vlan 2

!************** CIRCUIT ****************
circuit VLAN1
  ip address 10.1.1.100 255.255.255.0
  ip virtual-router 1 priority 230 preempt
  ip redundant-interface 1 10.1.1.200
  ip redundant-vip 1 10.1.1.4
  ip redundant-vip 1 10.1.1.8
  ip redundant-vip 1 10.1.1.12
```

```
!************* SERVICE *****************
service mediaserver1
  ip address 10.1.1.1
  port 80
  domain 10.1.1.1
  keepalive type http
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive uri "/index.html"
  active
service mediaserver2
  ip address 10.1.1.2
  port 80
  domain 10.1.1.2
  keepalive type http
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive uri "/index.html"
  active
service mediaserver3
  ip address 10.1.1.3
  port 80
  domain 10.1.1.3
  keepalive type http
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive uri "/index.html"
  active
service appserver1
  ip address 10.1.1.5
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive type script ap-kal-httpvxml "10.1.1.5"
  active
service appserver2
  ip address 10.1.1.6
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive type script ap-kal-httpvxml "10.1.1.6"
  active
service appserver3
  ip address 10.1.1.7
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive type script ap-kal-httpvxml "10.1.1.7"
  active
service asrtts1
  port 554
  protocol tcp
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive type tcp
  keepalive port 554
  ip address 10.1.1.9
  active
```

```
service asrtts2
  port 554
  protocol tcp
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive type tcp
  keepalive port 554
  ip address 10.1.1.10
  active
service asrtts3
  port 554
  protocol tcp
  keepalive retryperiod 2
  keepalive maxfailure 1
  keepalive type tcp
  keepalive port 554
  ip address 10.1.1.11
  active

!************** OWNER *****************
owner CVP

content MEDIA
  vip address 10.1.1.4
  protocol any
  port 80
  url "/*"
  add service mediaserver1
  add service mediaserver2
  primarySorryServer mediaserver3
  active
content APPSERVERS
  protocol tcp
  port 8000
  add service appserver1
  add service appserver2
  primarySorryServer appserver3
  vip address 10.1.1.8
  active
content ASRTTS
  add service asrtts1
  add service asrtts2
  primarySorryServer asrtts3
  protocol tcp
  port 554
  vip address 10.1.1.12
  active

!************** GROUP *****************
group mediaservers
  add destination service mediaserver1
  add destination service mediaserver2
  add destination service mediaserver3
  vip address 10.1.1.4
  active
group appservers
```

```
  add destination service mediaserver1
  add destination service mediaserver2
  add destination service mediaserver3
  vip address 10.1.1.8
  active
group asrtts
  add destination service mediaserver1
  add destination service mediaserver2
  add destination service mediaserver3
  vip address 10.1.1.12
  active
```

Gateway Configuration for Failover

In a failure scenario, where connectivity is lost between the IOS gateway and CVP, for quicker redirect of new incoming calls and quicker cleanup of existing calls, the following settings are required under the sip-ua section on the IOS gateway configuration:

```
sip-ua

    retry invite 2

    retry bye 2
```

Without these settings, the outbound legs to CVP for the terminated calls could stick around up to 60 seconds. These calls will continue to consume memory along with new incoming calls which could lead to a gateway crash due to gateway running out of memory.

Note: As of CVP 8.x, these settings are included in the standard CVP IOS gateway templates.

Configuring a Content Services Switch (CSS)

A Content Services Switch (CSS) provides load-balancing services for servers. You can add a CSS to the Operations Console. Once added, you can add a CSS to a device pool and execute a subset of IOS commands on the CSS from the Operations Console and transfer files between the CSS and the Unified CVP Operations Server.

The Content Services Switch (CSS) is the focus for high availability design in the TCP arena. The CSS can be placed between one (or more) VoiceXML Gateways, VXML Servers, Media Servers, and ASR/TTS Servers. Various mechanisms allow it to implement transparent load balancing and failover across these types of devices. CSS is an optional device, but it comes highly recommended. Without it, the CVP IVR Service implements a failover mechanism for up to two of each of the above components, but they are not load balanced, and various retries and delays are part of the algorithm, all of which is avoided if CSS is used. The CSS is normally deployed as a Virtual Router Redundancy Protocol (VRRP) pair. It is useful in all deployment models except for Call Director.

You can perform the following tasks:

- Adding a Content Services Switch (page 511)

Adding a Content Services Switch

Before You Begin

Collect the following information from the Content Services Switch before adding the switch to the Operations Console:

Information you will need:

- IP address
- Device type
- User name and password
- Enable password

Procedure

To add a Content Services Switch:

Step 1 Choose **Device Management > Content Services Switch**.

The Find, Add, Delete, Edit Content Services Switch window opens.

Step 2 Click **Add New.**

The Content Services Switch Configuration window opens.

Step 3 Fill in the appropriate configuration settings on the General tab as described in Content Services Switch Configuration Settings.

Step 4 Select the **Device Pool** tab and add the Content Services Switch to a device pool.

Step 5 When you finish configuring the Content Services Switch, click **Save** to save the configuration.

See Also

Content Services Switch Configuration Settings (page 514)

Editing a Content Services Switch

You can change an existing Content Services Switch configuration.

Procedure

To edit a Content Services Switch:

Step 1 Choose **Device Management > Content Services Switch**.

The Find, Add, Delete, Edit Content Services Switches window opens.

Step 2 From the list of matching records, choose the Content Services Switch that you want to edit.

Step 3 Click **Edit.**

The Content Services Switch Configuration window opens with the current settings displayed.

Step 4 Fill in the appropriate configuration settings on the General tab as described in Content Services Switch Configuration Settings.

Step 5 Optionally, select the **Device Pool** tab and add the Content Services Switch to a device pool.

Step 6 When you finish configuring the Content Services Switch, click**Save** to save the configuration.

See Also

Finding a Content Services Switch (page 513)

Content Services Switch Configuration Settings (page 514)

Deleting a Content Services Switch

Deleting a Content Services Switch deletes the configuration of the selected Content Services Switch in the Operations Console database and removes the gateway from displayed list of Content Service Switches.

Procedure

To delete a Content Services Switch:

Step 1 Choose **Device Management > Content Services Switch**.

The Find, Add, Delete, Edit Content Services Switches window displays.

Step 2 From the list of matching records, choose the Content Services Switch that you want to delete.

Step 3 Click **Delete**.

Step 4 When prompted to confirm the delete operation, click **OK** to delete or click **Cancel** to cancel the delete operation.

See Also

Finding a Content Services Switch (page 513)

Finding a Content Services Switch

You can locate a Content Services Switch on the basis of specific criteria. Use the following procedure to locate a Content Services Switch.

Procedure

To find a Content Services Switch:

Step 1 Choose **Device Management > Content Services Switch** from the Main menu.

The Find, Add, Delete, Edit Content Services Switches window lists the available Content Services Switch of the type you selected, sorted by name, 10 at a time.

Step 2 If the list is long, click **Next** to view the next group of available devices.

Step 3 If you know the host name of a particular Content Services Switch, enter its name in the Search text box and then click **Go.**

Step 4 To perform an advanced search, click **Advanced**. From the first Advanced Search window drop-down list box, choose one of the following criteria:

Search criteria:

- Hostname
- IP address
- Description

Step 5 From the second window drop-down list box, choose one of the following criteria:

Search criteria:

- begins with

- contains
- is exactly
- ends with
- is empty

Step 6 Specify the appropriate search text, if applicable, and click **Find**.

Content Services Switch Configuration Settings

You can change the settings described in the following table to configure a Content Services Switch.

Table 45: Content Services Switch Configuration Settings

Field	Description	Default	Range	Restart Required
General				
IP Address	The IP address of the Content Services Switch	None	Valid IP address	Yes
Hostname	The host name of the Content Services Switch	None	Valid DNS name, which can include letters in the alphabet, the numbers 0 through 9, and a dash	Yes
Device Type	The type of device	None	11xxxx	No
Description	The description of the Content Services Switch	None	Any string	No
Enable secure communication with the Ops console	Select to enable secure communications between the Operations Server and this component. The device is accessed using SSH and files are transferred using HTTPS. You must configure secure communications *before* you enable this option. See Chapter 6 in the *Configuration and Administration Guide for Cisco Unified Customer Voice Portal*.	None	On or Off	No
User Name and Passwords				
User Name	User name to access the device (telnet or ssh user name). This user name must be configured on the device.	None	None	No

Field	Description	Default	Range	Restart Required
User Password	Password to access the device (telnet or ssh password), needs to be configured on device.	None	None	Yes
Enable Password	Password to change to exec mode on device.	None	None	Yes

IOS Content Services Switch Commands

You can execute the following IOS commands on the Content Services Switch from the IOS Commands drop-down menu on the Content Services Switch Configuration window:

Table 46: IOS Content Services Switch Commands

Command	Description
Show running-config	Displays running-config
Show startup-config	Displays startup-config
Show version	Displays IOS version

See Also

Executing IOS Commands on the Content Services Switch (page 515)

Executing IOS Commands on the Content Services Switch

You can use a drop-down menu to select and execute a subset of available IOS commands when you are editing a Content Services Switch configuration.

See Also

IOS Content Services Switch Commands (page 515)

Transferring a Script File to a Content Services Switch

You can transfer files, one at a time or in bulk to a Content Services Switch or switches. Refer to the following Operations Console online help topics:

- For single files, or a few files to a single switch, see: **Managing Devices > Configuring a Content Services Switch (CSS) >Transferring a Script File to a Content Services Switch**
- For one or more files to one or more switches, see: **Bulk Administration > File Transfer**.

Disabling the Microsoft FTP Publishing Service

In order to successfully allow log file transfers, you need to disable the Microsoft FTP Publishing Service.

Step 1 Open the Microsoft Services application and right-click **FTP Publishing Service** in the Services (Local) area.

Step 2 Select **Properties** from the drop-down list. The FTP Publishing Service Properties (Local Computer) window displays.

Step 3 Select **Disabled** from the Startup type list box.

Step 4 Click **Apply**, then click **OK**.

Procedure

To transfer files from the Operations Console to a Content Service Switch:

Step 1 Choose **Device Management > Content Service Switch**.

The Find, Add, Delete, Edit Content Services Switches window lists any Content Services Switches that have been added to the Operations Console.

Step 2 Select a Content Services Switch by clicking on the link in its name field or by clicking the radio button preceding it and then clicking **Edit**.

You can also search for a Content Services Switch.

The Content Services Switch Configuration window opens.

Step 3 Select **File Transfer > Scripts** from the toolbar.

The Script File Transfer page opens, listing the IP address and host name of the currently selected Content Services Switch.

Step 4 Select a file to transfer to the Content Services Switch.

a. If the script is located on your local machine, click **Select a script file from your local PC**, then click **Browse** and select the script file to transfer to the Operations Console.

b. If the script is located on the Operations Console, click **Select from available script files**.

Step 5 When you have selected the file to transfer, click **Transfer** to copy the selected file to the Content Services Switch.

Configuring a Speech Server

A speech server provides speech recognition and synthesis services.You can add a pre-configured speech server to the Operations Console. Once added to the Operations Console, you can add a speech server to one or more device pools.

A speech server provides speech recognition services and text-to-speech services for aVoiceXML Gateway. For capacity and redundancy reasons, a Content Services Switch is usually used to mediate between a farm of such servers. If you do not use a Content Services Switch, Unified CVP can support a maximum of two speech servers.

Caution: The Operations Console can only manage speech servers installed on *Windows*, not on Linux. If the speech server is installed on Linux, the server cannot be managed.

The following list indicates some of the tasks you can perform on a speech server. Refer to the Operations Console online help for details on these tasks.

- Add a Speech Server
- Examine and change a Speech Server Configuration settings
- Editing a Speech Server Configuration
- Applying a License to a Speech Server
- Deleting a Speech Server
- Adding and Removing Speech Servers from a Device Pool
- Finding a Speech Server

Using Application Control Engine (ACE) for Load Balancing in Unified CVP

For configuration details, refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

You may use the Application Control Engine (ACE) as an alternative to the Content Services Switch (CSS) for server load balancing and failover. ACE provides load-balancing services for HTTP, MRCP and RTSP traffic, but not for call control signaling SIP or H.323 messages. As a load-balancing device, ACE determines which server in a set of load-balanced servers, should receive the client request for service. Load balancing helps fulfill the client request without overloading either the server or the server farm as a whole. Also, by monitoring the state of each server and transferring a server's load to a working server during a server failure, ACE provides high availability support.

In this application of ACE, the engine is used primarily to direct initial session requests for a particular type of service. There are four types of services:

- http prompts
- ASR/TTS
- Unified CVP Call Server
- Unified CVP VXML Server

The following general approach applies to configuring each Unified CVP component type for use with ACE.

- **Real Servers** – One ACE real server is configured for each group of Unified CVP components (Call Servers, VXML Servers, etc.) that need ACE Load balancing. Refer to the ACE component diagram (page 519). For general step-by-step guidelines for configuring Real Servers, refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

- **Server Farms** – Typically, in data centers, servers are organized into related groups called server farms. Servers within server farms often contain identical content (referred to as mirrored content) so that if one server becomes inoperative, another server can take its place immediately. After you create and name a server farm, you can add existing real servers to it and configure other server-farm parameters, such as the load-balancing predictor, server weight, backup server, health probe, and so on. For general step-by-step guidelines for configuring server farms, refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

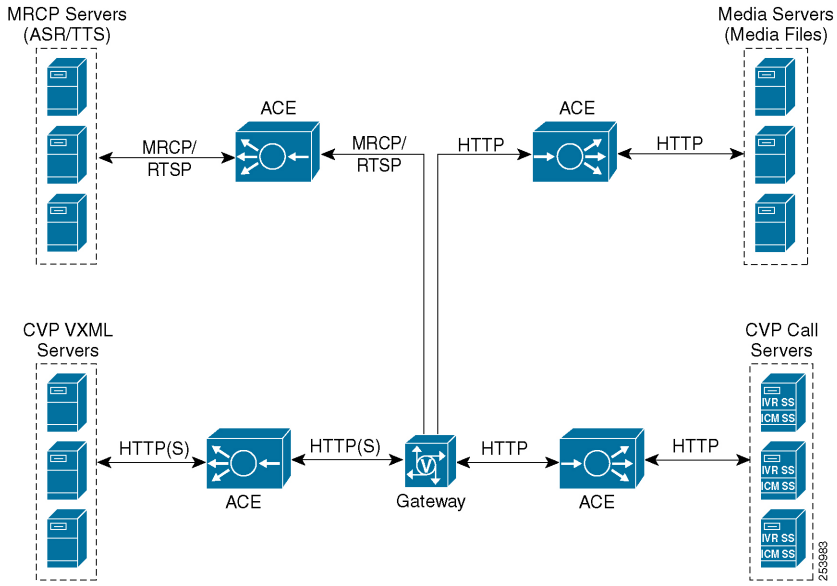
- **Health Monitoring** - You can instruct the ACE servers to check the health of servers and server farms by configuring health probes (sometimes referred to as keepalives). After you create a probe, you assign it to a real server or a server farm. A probe can be one of many types, including TCP, ICMP, Telnet, or HTTP. The ACE server sends out probes periodically to determine the status of a load-balanced server, verifies the server response, and checks for other network problems that may prevent a client from reaching a server. For general step-by-step guidelines for configuring probes, refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

- **Class-Map and Policy Map** - The ACE server uses several configuration elements to filter traffic and then to perform various actions on that traffic before making the load-balancing decision. These filtering elements and subsequent actions form the basis of a traffic policy for server load balancing. For general step-by-step guidelines for configuring traffic policies, refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

Specific component-type configuration is covered in the following sections.

In this section, you will configure the probes and other configuration needed for the ACE server to ensure that each server in each server farm is operating properly, so that the ACE server can load balance between all the servers of each type that are usable at any given moment.

Figure 79: Overview of ACE Load Balancing



General Probes

In your ACE unit's configuration, create an ICMP probe to check for server connectivity. In the subtopics that follow you associate this probe with each of your real servers.

```
probe icmp PROBE_SERVICE_ICMP
   interval 5
   receive 3
   faildetect 1
   passdetect interval 5
   passdetect count 1
```

Unified CVP Media Servers

Media Servers are standard web servers that are responsible for serving Unified CVP prompt files to the voice gateway.

Create an HTTP Probe

The probe below is used to determine whether the Media Server is operating properly. A simple http request is sent to the Media Server and the probe does a check for HTTP return code 200.

The Media Server probe sends an HTTP request to **/index.html**. The request is sent to the default HTTP port (80) and the IP address of the real server associated with the probe.

In the probe below, the following parameters are set. These are only example values; set the actual values according to your own requirements. Refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

To create the HTTP probe for the media servers, place the following code in the configuration for the ACE server.

```
probe http PROBE_HTTP
   interval 5
   receive 3
   faildetect 1
   passdetect interval 5
   passdetect count 1
   request method get url /index.html
   expect status 200 200
   open 1
```

Configure the Physical Servers

Create a real server for every physical media server you would like to load balance. Associate the ICMP probe with each server by creating a section, like the following, for each media server in the server farm.

```
 rserver host mediaServer1
   ip address 10.1.1.1
   probe PROBE_SERVICE_ICMP
   inservice
rserver host mediaServer2
   ip address 10.1.1.2
   probe PROBE_SERVICE_ICMP
   inservice
```

Group Your Physical (Media) Servers

In the ACE configuration file, create a server farm and associate servers with this farm. The following example applies the HTTP Probe to the server farm and the ACE server probes each media server in the server farm. However, you can also associate this probe with the physical server.

Note: By specifying the port, only connections on this port will be accepted by this server farm.

```
serverfarm host media_server_FARM
  description Media Server Farm
  probe PROBE_HTTP
  rserver mediaServer1 80
     inservice
  rserver mediaServer2 80
     inservice
```

Class Map Configuration

The configuration below defines a Layer 3 and a Layer 7 class-map.

- The Layer 3 class-map is used to define a Virtual IP and the allowed traffic port. This class map gets applied to the Layer 3/4 policy-map. Traffic sent to the virtual IP is directed by the ACE server to real media servers based on the load balancing policy.
- The Layer 7 class-map is used to filter traffic based on the URL pattern specified. This class-map is associated with a Layer 7 policy-map, which contains information about which servers to load balance.

When traffic entering the ACE server matches the class-map L3_Media_Server_VIP, the ACE server applies the actions specified in Media_Server_L7SLB, which is defined below.

```
class-map match-all L3_Media_Server_VIP
   2 match virtual-address 10.1.1.3 tcp eq www

class-map type http loadbalance match-all L7_HTTP_CLASS
   2 match http url .*
```

Policy Map Configuration

Note: In the code below, the layer 7 Class map gets associated with the layer 7 policy map.

```
 policy-map type loadbalance first-match Media_Server_L7SLB
  class L7_HTTP_CLASS
     serverfarm media_server_FARM

policy-map multi-match POLICY
   class L3_Media_Server_VIP
      loadbalance vip inservice
      loadbalance policy Media_Server_L7SLB
      loadbalance vip icmp-reply active
```

ASR/TTS Servers

Probe

The probe below is used to determine whether the MRCP ASR/ TTS Server is up. The ACE server makes a connection to the MRCP port to validate that the ASR/TTS server is running. In the configuration below, a TCP probe is used. The probe waits for the configured 3 seconds to receive information from the server. The ASR/TTS service is considered down if the ACE server is unable to connect to port 554 for MRCP traffic.

In the probe below, the following parameters are set. These are only example values; set the actual values according to your own requirements. Refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

The following configuration example is part of the ACE server's configuration.

```
probe tcp PROBE_ASR_TTS
   port 554
   interval 5
   receive 3
   faildetect 1
   passdetect interval 5
   passdetect count 1
   open 1
```

Real Server and Server Farm Configuration

The following code defines your physical servers and associates them with the ICMP probe.

```
rserver host asrtts1
   ip address 10.1.1.12
   probe PROBE_SERVICE_ICMP
   inservice
rserver host asrtts2
   ip address 10.1.1.13
   probe PROBE_SERVICE_ICMP
   inservice
```

The following code defines your server farms and associates them with the PROBE_ASR_TTS probe. The servers in the server farm only accept connections on port 554.

```
serverfarm host ASR
   description ASR Farm
   probe PROBE_ASR_TTS
   rserver asrtts1 554
      inservice
   rserver asrtts2 554
      inservice
serverfarm host TTS
   description TTS Farm
   probe PROBE_ASR_TTS
   rserver asrtts1 554
      inservice
   rserver asrtts2 554
      inservice
```

Class-map Configuration

Create a class-map that accepts connections only on port 554. (By default, rtsp maps to port 554.)

```
class-map match-all ASR_CLASS_L3
   2 match virtual-address 10.1.1.14 tcp eq rtsp


 class-map match-all TTS_CLASS_L3
   2 match virtual-address 10.1.1.18 tcp eq rtsp
```

Policy-map Configuration

```
policy-map type loadbalance first-match ASR_POLICY_L7
   class class-default
      serverfarm ASR

   policy-map type loadbalance first-match TTS_POLICY_L7
    class class-default
        serverfarm TTS
   policy-map multi-match POLICY
    class ASR_CLASS_L3
       loadbalance vip inservice
       loadbalance policy ASR_POLICY_L7
       loadbalance vip icmp-reply active
    class TTS_CLASS_L3
       loadbalance vip inservice
       loadbalance policy TTS_POLICY_L7
       loadbalance vip icmp-reply active
       inspect rtsp
```

Unified CVP Call Servers

Note: Call Server load balancing is only supported on *IVR only* deployments.

Probes

The probe below is used to determine whether the Call Server is up and in service. The probe passes only if the Call Server is *In Service*. This probe is an HTTP probe using the ACE server.

The ACE server Call Server probe sends an HTTP request to **/cvp/VBServlet?MSG_TYPE=HEARTBEAT&TIMEOUT=0**. This probe takes a little more than 4 seconds to send back a response. If the Call Server is *In Service*, the HTTP 200 OK response returns.

Refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

To create the Call Server HTTP probe, place the following lines in the configuration for the ACE server:

```
probe http PROBE_CALLSERVER_HTTP
  port 8000
   interval 6
   faildetect 1
   passdetect interval 6
   passdetect count 1
   receive 5
   request method get url /cvp/VBServlet?MSG_TYPE=HEARTBEAT&TIMEOUT=0
   open 1
   expect status 200 200
```

Unified CVP VXML Servers

Real Servers

Configure the Physical Servers

Create a real server for every physical VXML Server you would like to load balance. Associate the probe with each server by creating a section, like the following, for each VXML server in the server farm.

```
rserver host vxml1
   probe PROBE_SERVICE_ICMP
   ip address 10.1.1.15
   inservice
rserver host vxml2
   probe PROBE_SERVICE_ICMP
   ip address 10.1.1.16
   inservice
```

HTTP Probe Configuration

The probe below is used to determine whether the VXML Server is up and in service. The probe passes only if the VXML Server is *In Service*. To create the VXML Server HTTP probe, place the following lines in the configuration for the ACE server.

The VXML Server probe sends a HTTP request to /CVP/Server?probe=true. If the VXML Server is up and inservice, HTTP 200 OK is returned. In the HTTP probe below, the http request is made to the port specified in the probe and the IP of the real server that this probe is associated with.

```
probe http PROBE_VXMLSERVER_HTTP
   port 7000
   interval 5
   receive 3
   faildetect 1
   passdetect interval 5
   passdetect count 1
   request method get url /CVP/Server?probe=true
   expect status 200 200
   open 1
```

Note: In order to get the "?", press CTRL-V before pressing the question mark.

Server Farm Configuration

```
serverfarm host vxmlserver
   probe PROBE_VXMLSERVER_HTTP
   rserver vxml1 7000
      inservice
   rserver vxml2 7000
      inservice
```

Sticky Server farm

For a VXML Server to preserve HTTP session information, you must ensure that, once the ACE server has chosen a particular VXML Server from the list of servers in a server farm, the ACE server continues to send all traffic for that session to the same VXML Server. To accomplish this, use a *sticky group*.

Refer to the **Cisco ACE 4700 Series Appliance Server Load-Balancing Configuration Guide** (http://www.cisco.com/en/US/docs/app_ntwk_services/data_center_app_services/ace_appliances/vA3_1_0/configuration/slb/guide/slbgd.html).

The following definitions apply to the settings shown below:

- **http-cookie:** Sticky method being used. In this method, when the ACE server examines a request for content, and determines through policy matching that the content is sticky, the ACE server examines any cookie or URL present in the content request. The ACE server uses the information in the cookie or URL to direct the content request to the appropriate server.
- **Cookie insert:** The ACE server inserts the cookie on behalf of the VXML Server upon the return request, so that the ACE server can perform cookie stickiness even when the VXML servers are not configured to set cookies. The cookie contains information that the ACE server uses to ensure persistence to a specific real server.

The following ACE server configuration code accomplishes the *sticky* function.

```
sticky http-cookie ACE_COOKIE VXMLServer_HTTP_STICKY
   cookie insert
   serverfarm vxmlserver
```

Class-map Configuration

```
class-map match-all vxmlserver_HTTP_CLASS_L3
   2 match virtual-address 10.1.1.17 tcp eq 7000
```

Policy-map Configuration

```
policy-map type loadbalance first-match vxmlserver_HTTP_POLICY_L7
   class L7_HTTP_CLASS
      sticky-serverfarm VXMLServer_HTTP_STICKY

policy-map multi-match POLICY
   class vxmlserver_HTTP_CLASS_L3
      loadbalance vip inservice
      loadbalance policy vxmlserver_HTTP_POLICY_L7
      loadbalance vip icmp-reply active
```

ACE Sample Configuration

The following sample configuration does require some additional ACE configuration. Some of the modifications include interface, resource, access-list, Admin and additional context configuration changes.

```
*************PROBE CONFIGURATION***************

probe icmp PROBE_SERVICE_ICMP
    interval 5
    receive 3
    faildetect 1
    passdetect interval 5
    passdetect count 1

probe http PROBE_HTTP
    interval 5
    receive 3
    faildetect 1
    passdetect interval 5
    passdetect count 1
    request method get url /index.html
    expect status 200 200
    open 1

probe tcp PROBE_ASR_TTS
    port 554
    interval 5
    receive 3
    faildetect 1
    passdetect interval 5
    passdetect count 1
    open 1

probe http PROBE_CALLSERVER_HTTP
    port 8000
    interval 6
    faildetect 1
    passdetect interval 6
    passdetect count 1
    receive 5
   request method get url /cvp/VBServlet?MSG_TYPE=HEARTBEAT&TIMEOUT=0

    open 1
    expect status 200 200

probe http PROBE_VXMLSERVER_HTTP
    port 7000
    interval 5
    receive 3
    faildetect 1
    passdetect interval 5
    passdetect count 1
    request method get url /CVP/Server?probe=true
    expect status 200 200
    open 1


*************REAL SERVER CONFIGURATION*****************

rserver host mediaServer1
    ip address 10.1.1.1
```

```
    probe PROBE_SERVICE_ICMP
    inservice

rserver host mediaServer2
    ip address 10.1.1.2
    probe PROBE_SERVICE_ICMP
    inservice

rserver host asrtts1
    ip address 10.1.1.12
    probe PROBE_SERVICE_ICMP
    inservice

rserver host asrtts2
    ip address 10.1.1.13
    probe PROBE_SERVICE_ICMP
    inservice

rserver host callserver1
    ip address 10.1.1.19
    probe PROBE_SERVICE_ICMP
    inservice

rserver host callserver2
    ip address 10.1.1.20
    probe PROBE_SERVICE_ICMP
    inservice

rserver host vxml1
    probe PROBE_SERVICE_ICMP
    ip address 10.1.1.15
    inservice

rserver host vxml2
    probe PROBE_SERVICE_ICMP
    ip address 10.1.1.16
    inservice


*************SERVERFARM CONFIGURATION***************

serverfarm host media_server_FARM
    description Media Server Farm
    probe PROBE_HTTP
    rserver mediaServer1 80
        inservice
    rserver mediaServer2 80
        inservice

 serverfarm host ASR
     description ASR Farm
     probe PROBE_ASR_TTS
     rserver asrtts1 554
         inservice
     rserver asrtts2 554
         inservice
```

```
 serverfarm host TTS
     description TTS Farm
     probe PROBE_ASR_TTS
     rserver asrtts1 554
         inservice
     rserver asrtts2 554
         inservice

serverfarm host callserver_farm
    description Call Server Farm
    probe PROBE_CALLSERVER_HTTP
    rserver callserver1 8000
        inservice
    rserver callserver2 8000
        inservice

serverfarm host vxmlserver
    probe PROBE_VXMLSERVER_HTTP
    rserver vxml1 7000
        inservice
    rserver vxml2 7000
        inservice


*************STICKY-SERVERFARM CONFIGURATION****************

sticky http-cookie ACE_COOKIE VXMLServer_HTTP_STICKY
    cookie insert
    serverfarm vxmlserver


*************CLASS-MAP CONFIGURATION**************

class-map match-all L3_Media_Server_VIP
    2 match virtual-address 10.1.1.3 tcp eq www

class-map type http loadbalance match-all L7_HTTP_CLASS
    2 match http url .*

class-map match-all ASR_CLASS_L3
    2 match virtual-address 10.1.1.14 tcp eq rtsp

class-map match-all TTS_CLASS_L3
    2 match virtual-address 10.1.1.18 tcp eq rtsp

class-map match-all L3_CallServer_VIP
    2 match virtual-address 10.1.1.21 tcp eq 8000

class-map match-all vxmlserver_HTTP_CLASS_L3
    2 match virtual-address 10.1.1.17 tcp eq 7000


*************POLICY-MAP CONFIGURATION*************

policy-map type loadbalance first-match Media_Server_L7SLB
    class L7_HTTP_CLASS
```

```
        serverfarm media_server_FARM

policy-map type loadbalance first-match ASR_POLICY_L7
    class class-default
        serverfarm ASR

policy-map type loadbalance first-match TTS_POLICY_L7
    class class-default
        serverfarm TTS

policy-map type loadbalance first-match CallServer_L7SLB
    class L7_HTTP_CLASS
        serverfarm callserver_farm

policy-map type loadbalance first-match vxmlserver_HTTP_POLICY_L7
    class L7_HTTP_CLASS
        sticky-serverfarm VXMLServer_HTTP_STICKY

policy-map multi-match POLICY
    class L3_Media_Server_VIP
        loadbalance vip inservice
        loadbalance policy Media_Server_L7SLB
        loadbalance vip icmp-reply active
    class ASR_CLASS_L3
        loadbalance vip inservice
        loadbalance policy ASR_POLICY_L7
        loadbalance vip icmp-reply active
    class TTS_CLASS_L3
        loadbalance vip inservice
        loadbalance policy TTS_POLICY_L7
        loadbalance vip icmp-reply active
        inspect rtsp
   class L3_CallServer_VIP
        loadbalance vip inservice
        loadbalance policy CallServer_L7SLB
        loadbalance vip icmp-reply active
  class vxmlserver_HTTP_CLASS_L3
      loadbalance vip inservice
      loadbalance policy vxmlserver_HTTP_POLICY_L7
      loadbalance vip icmp-reply active
      nat dynamic 1 vlan 192

interface vlan 192
  description "Client-Server VLAN"
  ip address 192.168.150.5 255.255.255.0
  alias 192.168.150.193 255.255.255.0
  peer ip address 192.168.150.6 255.255.255.0
  no normalization
  access-group input ALL
  nat-pool 5 192.168.150.107 192.168.150.107 netmask 255.255.255.0
  nat-pool 5 192.168.150.158 192.168.150.158 netmask 255.255.255.0
  nat-pool 5 192.168.150.184 192.168.150.184 netmask 255.255.255.0
  nat-pool 5 192.168.150.197 192.168.150.198 netmask 255.255.255.0
  nat-pool 5 192.168.150.224 192.168.150.229 netmask 255.255.255.0
  nat-pool 5 192.168.150.190 192.168.150.190 netmask 255.255.255.0
  service-policy input remote_mgmt_allow_policy
```

```
service-policy input CLIENT-VIPS
no shutdown
```

Chapter 15

Configuring the Media Servers

A Media Server administers the media files that contain messages and prompts callers hear. You can add a pre-configured Media Server to the Operations Console Control Center. Once added, you can add a Media Server to one or more device pools (refer to "Configuring a Media Server (page 552)").

This chapter contains the following topics:

Media File Overview

This section presents a brief overview of how Unified CVP performs media file handling.

It includes information about:

- What the Media Server is.
- The media file names and types Unified CVP supports.
- How to specify the address of a media file.
- Locale syntax backward compatibility.

Media Server

In Unified CVP, the Media Server is a computer or set of computers, which “serve” the media files that contain messages and prompts that callers will hear. Media files can be installed on an individual server or colocated with Unified CVP CallServer or VXML Server.

There is no artificial limit on the number of prompts; prompts will be limited only by system capacity.

Note: To maximize Unified CVP performance, do not install the HTTP Media Server on the same machine as the Unified CVP H.323 Service.

Tools for prompt creation are off-the-shelf, such as Audition by Adobe (formerly known as Cool Edit Pro by Syntrillium Software Corporation), and Vox Studio (**http://www.xentec.be**).

Note: It is the customer's responsibility to select the tool, select a voice talent, record the system and application media files in the supported locales, format and encoding, and contact the person who is responsible for the media files on the Media Server(s).

Media File Names and Types

A *media file name* is specified through Unified ICME Network VRU Script Configuration and used in the Run VRU Script request for the Play Media, Play Data, Get Digits, Menu, and Get Speech (in non-TTS applications) micro-applications.The media file naming convention allows alpha-numeric characters with the underbar character as a separator (Spaces or hyphens are not allowed.) This naming convention provides a mechanism for an "understandable" naming convention as opposed to numeric media file names typically used by stand-alone VRUs.

Caution: The Unified Customer Voice Portal includes a library of media files/prompts for individual digits, months (referenced internally by Unified Customer Voice Portal software for a Play Data script type request), and default error messages, et cetera. Creation of a full set of media/prompts for each locale referenced by the Unified CVP customer is the responsibility of the customer's Media Administrator. For specific comments on the H.323 Service critical media file, refer to "System Media File Error Messages (page 550)."

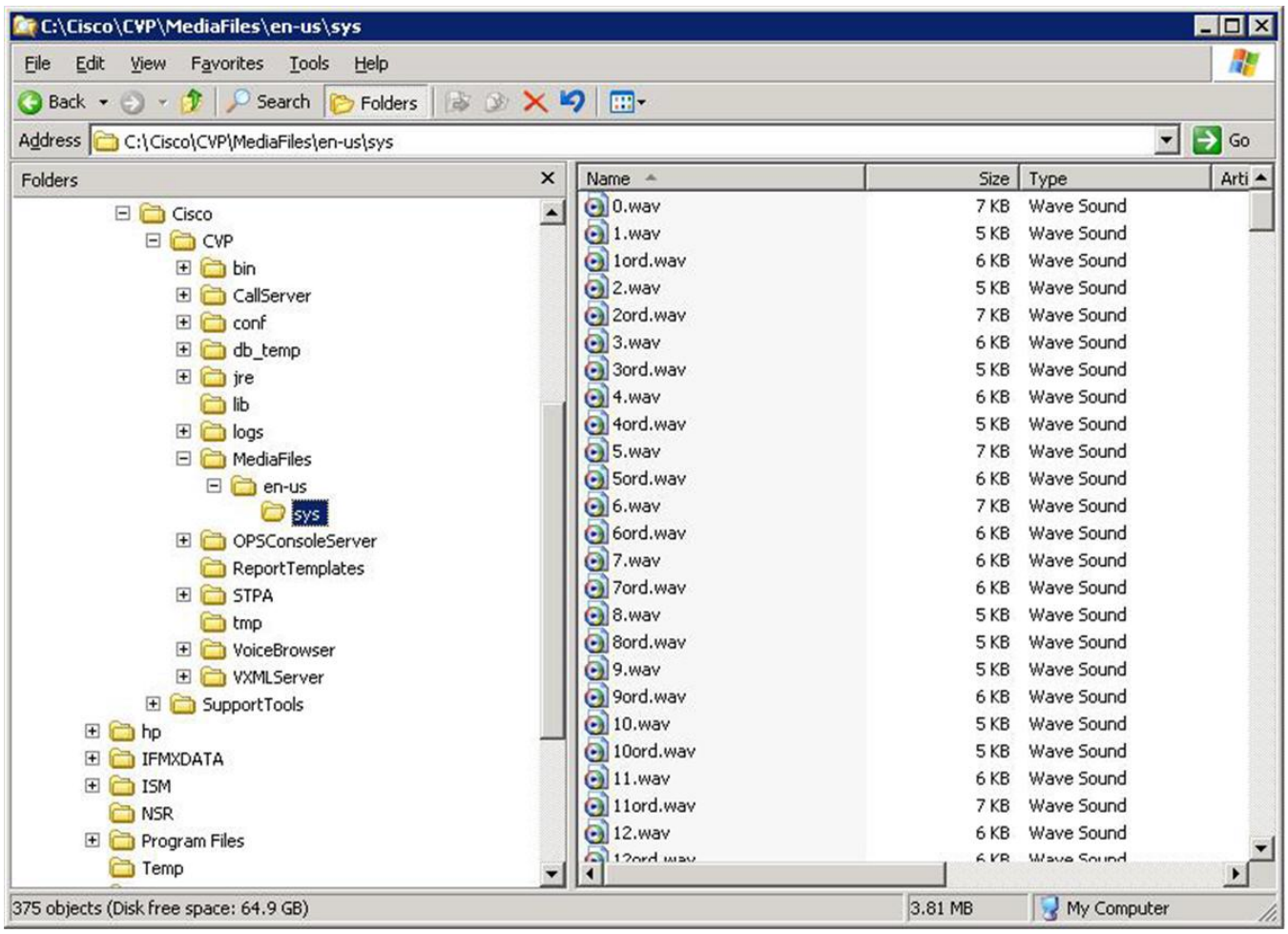
The *media file types* Unified CVP supports are µ-Law 8-bit .wav files and A-law 8-bit .wav files. Media files specified with an extension will be used "as is," for example, hello.xxx. (The default file extension is .wav.)

Caution: Any unexpected (and unsupported) type of media file encountered will generate the logging of an error and a result code of False will be returned to Unified ICME along with the ECC user.microapp.error_code set appropriately. From the caller's perspective, nothing was played, however it is the Script Editor developer's responsibility to write the script to handle this error condition.

Location of Media Files

The figure below displays the location of the media files if you choose to install System Media Files during Unified CVP installation.

Figure 80: Media File Location



Media File Address

The address for media files that reside on the Media Server(s) is generated by the Unified CVP. Unified ICME provides information about the file location or base URL address in the Unified ICME/IVR messages it passes when the Run VRU Script node is executed. The Unified ICME/IVR messages include ECC variables for: locale, media server set address, as well as optional system and application library name overrides. (For details about the Unified ICME/IVR messages passed to Unified CVP, refer to "Writing Scripts for Unified CVP (page 141)."

The table below summarizes the data that combines to form the address of the media file:

Table 47: Media File Address Components

Parameter	Location of Data	Description	Examples
Media Server Set	ECC variable: user.microapp.media_server	File location or base URL for the Media Server. When the Media Server URL is the DNS name and the DNS Server is configured to	Base URL example: **http://www.machine1.com/dir1/dirs/cust1** **Note:** By convention, the service provider may include their

Parameter	Location of Data	Description	Examples
		return multiple IP addresses for a host name, the Unified Customer Voice Portal will attempt to get the media files from each Media Server IP address in sequence with the priority given to those on the subnet. **Note:** Unified Customer Voice Portal supports playing prompts from flash on the GW. In order to do this, set the media_server to "flash:" instead of the hostname or IP address of the media server. When using the Media Server set for external grammars or external VXML, if the Media Server URL is the DNS name with multiple IP addresses for the hostname, it is the ASR Engine's responsibility to decide which machine to retrieve the grammar file from.	customers' name at the end of the Media Server set.
Locale	ECC variable: user.microapp.locale **Default:** en-us	This field is a combination of language and country with a default of en-us for English spoken in the United States.	en-us
Media Library Type	The Media Library Type value passed from the VRU Script Name field. Valid options are: **A** - Application prompt library. **S** - System prompt library. **V** - External VXML. **Default:** A	The media library (directory) for the prompt is either the application prompt library defined by ECC variable user.microapp.app_media_lib (default "app") or the system prompt library defined by ECC variable user.microapp.sys_media_lib (default "sys"). **Note:** • When the Media Library Type is V (external VXML), the VXML file will reside in the	**A** (user.microapp.app_media_lib= app_banking)

Parameter	Location of Data	Description	Examples
		Application Prompt Library. • When the Media Library Type is A (Application prompt library), you must create the directory specified by this variable. For example, if you use the default "app" directory, you must create an app directory in ./MediaFiles/en-us	
Media File Name	The Media File Name value passed from the VRU Script Name field. Valid options are the name of the .wav file to be played, or external VXML file name, or <blank>, which translates to playing no media. This file name will be ignored if TTS is being used (that is, if the **user.microapp.inline_tts** ECC variable contains a value.) **Default:** none	Name of media file or external VXML file to be played.	Main_menu
Media File Name Type	If not given as part of the Media File Name, the type is .wav	Type of media file to be played.	.wav

Based on the examples shown in the table above, a valid address for the Media File might be:

- http://www.machine1.com/dir1/dirs/cust1/en-us/app_banking/main_menu.wav

Locale Backward Compatibility

The locale string values are compatible with current industry naming schemes:

- **en_US** has changed to **en-us**. That is, "**en** underscore **US**" (upper case) has changed to "**en** hyphen **u**s" (lower case).
- **en_GB** has changed to **en-gb**. That is, "**en** underscore **GB**" (upper case) has change to "**en** hyphen **gb**" (lower case).

Existing scripts from previous versions of Unified CVP will continue to work with the current version of Unified CVP:

- **en_US** and **en-us** both map to U.S. English in the Application Server for use by the Application Server's internal grammar
- **en_GB** and **en-gb** both map to U.K. English in the Application Server for use by the Application Server's internal grammar.
- The base URL for media prompts will use the locale that is specified, without making modifications. For example, if the locale is set to**EN_US**, the base URL will contain**EN_US**. if the locale is set to **XX**, the base URL will contain **XX**.

To use the Unified CVP Version 1.1 default locale directory (for example, **en_US**), you must explicitly set it. When you upgrade to the current version of Unified CVP, only the new files are installed under the Unified CVP default locale directory, **en-us**. You want to have all your system prompts under one directory and all your application prompts and, optionally, external VXML in another directory. Use the **user.microapp.locale** ECC variable to set the locale directory to use, such as **en_US**.

Note: You do not need to set the **user.microapp.locale** ECC variable if you used the default **en-us**. Also, remember that all locale values are case-sensitive.

System Media Files

The following tables describe the English System Media Files installed by Unified CVP. These system media files are intended as samples only. It is the Customer/Media Administrator's responsibility to record all the system prompts for all the locales.

The table that follows lists the System Media File information for cardinal numbers.

Table 48: System Media Files, Cardinal Numbers

Symbol (where applicable)	Decimal Value[6]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		point	point	Number
		minus	minus	Number
0	48	0	zero	All except DOW
1	49	1	one (masculine version), uno (es-mx and es-es)	All except DOW
2	50	2	two	All except DOW
3	51	3	three	All except DOW
4	52	4	four	All except DOW
5	53	5	five	All except DOW
6	54	6	six	All except DOW
7	55	7	seven	All except DOW

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Symbol (where applicable)	Decimal Value[6]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
8	56	8	eight	All except DOW
9	57	9	nine	All except DOW
		10	ten	Same for the rest of all the numbers
		11	eleven	
		12	twelve	
		13	thirteen	
		14	fourteen	
		15	fifteen	
		16	sixteen	
		17	seventeen	
		18	eighteen	
		19	nineteen	
		20	twenty	
		21	twenty-one	
		22	twenty-two	
		23	twenty-three	
		24	twenty-four	
		25	twenty-five	
		26	twenty-six	
		27	twenty-seven	
		28	twenty-eight	
		29	twenty-nine	
		30	thirty	
		31	thirty-one	
		32	thirty-two	
		33	thirty-three	
		34	thirty-four	
		35	thirty-five	
		36	thirty-six	
		37	thirty-seven	
		38	thirty-eight	
		39	thirty-nine	

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Symbol (where applicable)	Decimal Value[6]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		40	forty	
		41	forty-one	
		42	forty-two	
		43	forty-three	
		44	forty-four	
		45	forty-five	
		46	forty-six	
		47	forty-seven	
		48	forty-eight	
		49	fora>49	
		50	fifty	
		51	fifty-one	
		52	fifty-two	
		53	fifty-three	
		54	fifty-four	
		55	fifty-five	
		56	fifty-six	
		57	fifty-seven	
		58	fifty-eight	
		59	fifty-nine	
		60	sixty	
		61	sixty-one	
		62	sixty-two	
		63	sixty-three	
		64	sixty-four	
		65	sixty-five	
		66	sixty-six	
		67	sixty-seven	
		68	sixty-eight	
		69	sixty-nine	
		70	seventy	
		71	seventy-one	

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Symbol (where applicable)	Decimal Value[6]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		72	seventy-two	
		73	seventy-three	
		74	seventy-four	
		75	seventy-five	
		76	seventy-six	
		77	seventy-seven	
		78	seventy-eight	
		79	seventy-nine	
		80	eighty	
		81	eighty-one	
		82	eighty-two	
		83	eighty-three	
		84	eighty-four	
		85	eighty-five	
		86	eighty-six	
		87	eighty-seven	
		88	eighty-eight	
		89	eighty-nine	
		90	ninety	
		91	ninety-one	
		92	ninety-two	
		93	ninety-three	
		94	ninety-four	
		95	ninety-five	
		96	ninety-six	
		97	ninety-seven	
		98	ninety-eight	
		99	ninety-nine	
		oh	oh	24TOD, Date
		hundred	hundred	Number, 24TOD, Date, Currency
		thousand	thousand	Number, Date, Currency

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Symbol (where applicable)	Decimal Value[6]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		million	million	Number, Currency
		billion	billion	Number, Date, Currency
		trillion	trillion	Number, Currency

The table that follows lists the System Media File information for ordinal numbers.

Note: If ordinal system prompts are to be used in a script for a purpose other than dates, they should be recorded as application prompts with the true ordinal values.

Table 49: System Media Files, Ordinal Numbers

Symbol (where applicable)	Decimal Value[7]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		1ord	first	Date
		2ord	second	Date for all ordinal numbers
		3ord	third	
		4ord	fourth	
		5ord	fifth	
		6ord	sixth	
		7ord	seventh	
		8ord	eighth	
		9ord	nineth	
		10ord	tenth	
		11ord	eleventh	
		12ord	twelveth	
		13ord	thirteenth	
		14ord	fourteenth	
		15ord	fifteenth	
		16ord	sixteenth	
		17ord	seventeenth	
		18ord	eighteenth	
		19ord	nineteenth	

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Symbol (where applicable)	Decimal Value[7]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		20ord	twentieth	
		21ord	twenty-first	
		22ord	twenty-second	
		23ord	twenty-third	
		24ord	twenty-fourth	
		25ord	twenty-fifth	
		26ord	twenty-sixth	
		27ord	twenty-seventh	
		28ord	twenty-eight	
		29ord	twenty-nineth	
		30ord	thirtieth	
		31ord	thirty-first	

The table that follows lists the System Media File information for measurements.

Table 50: System Media Files, Measurements

Symbol (where applicable)	Decimal Value[8]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
½	189	one_half	one half	Char
¼	188	one_quarter	one quarter	Char
¾	190	three_quarters	three quarters	Char
A, a	65,97	a	A	Char
B,b	66,98	b	B	Char
C, c	67,99	c	C	Char
D, d	68,100	d	D	Char
E, e	69,101	e	E	Char
F, f	70,102	f	F	Char
G, g	71,103	g	G	Char
H, h	72,104	h	H	Char
I, I	73,105	I	I	Char
J, j	74,106	j	J	Char
K, k	75,107	k	K	Char
L, l	76,108	l	L	Char

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Symbol (where applicable)	Decimal Value[8]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
M, m	77,109	m	M	Char
N, n	78,110	n	N	Char
O, o	79,111	o	O	Char
P, p	80,112	p	P	Char
Q, q	81,113	q	Q	Char
R, r	82,114	r	R	Char
S, s	83,115	s	S	Char
T, t	84,116	t	T	Char
U, u	85,117	u	U	Char
V, v	86,118	v	V	Char
W, w	87,119	w	W	Char
X, x	88,120	x	X	Char
Y, y	89,121	y	Y	Char
Z, z	90,122	z	Z	Char
Œ, œ	140,156	oe_140_156	Ligature OE	Char
À,à	192,224	a_192_224	A grave	Char
Á,á	193,225	a_193_225	A acute	Char
Â,â	194,226	a_194_226	A circumflex	Char
Ã,ã	195,227	a_195_227	A tilde	Char
Ä,ä	196,228	a_196_228	A umlaut	Char
Å,å	197,229	a_197_229	A with ring above	Char
Æ,æ	198,230	ae_198_230	Ligature AE	Char
È,è	200,232	e_200_232	E grave	Char
É,é	201,233	e_201_233	E acute	Char
Ê,ê	202,234	e_202_234	E circumflex	Char
Ë,ë	203,235	e_203_235	E umlaut	
Ì,ì	204,236	i_204_236	I grave	Char
Í, í	205,237	i_205	I acute	Char
Î,î	206,238	i_206	I circumflex	Char
Ï,ï	207,239	i_207	I umlaut	Char
Ð	208	char_208	character 208	Char
ð	240	char_240	character 240	
Ò,ò	210,242	o_210_242	O grave	Char

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Symbol (where applicable)	Decimal Value[8]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
Ó,ó	211,243	o_211_243	O acute	Char
Ô,ô	212,244	o_212_244	O circumflex	Char
Õ,õ	213,245	o_213_245	O tilde	Char
Ö,ö	214,246	o_214_246	O umlaut	Char
x	215	multiply	multiplication sign	Char
Ø,ø	216,248	o_216_248	oh stroke	Char
Ù,ù	217,249	u_217_249	U grave	Char
Ú,ú	218,250	u_218_250	U acute	Char
Û,û	219,251	u_219_251	U circumflex	Char
Ü,ü	220,252	u_220_252	U umlaut	Char
Ý,ý	221,253	y_221_253	Y acute	Char
Þ	222	char_222	character 222	Char
ß	223	ss	double s	Char
÷	247	divide	division sign	Char
þ	254	char_254	character 254	Char
Ÿ,ÿ	159,255	y_159_255	character 159 or 255	Char

The table that follows lists the System Media File information for month values.

Table 51: System Media Files, Months

Symbol (where applicable)	Decimal Value[9]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		January	January	Date
		February	February	Date
		March	March	Date
		April	April	Date
		May	May	Date
		June	June	Date
		July	July	Date
		August	August	Date
		September	September	Date
		October	October	Date
		November	November	Date

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Symbol (where applicable)	Decimal Value[9]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		December	December	Date

The table that follows lists the System Media File information for month values.

Table 52: System Media Files, Days

Symbol (where applicable)	Decimal Value[10]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		Sunday	Sunday	DOW
		Monday	Monday	DOW
		Tuesday	Tuesday	DOW
		Wednesday	Wednesday	DOW
		Thursday	Thursday	DOW
		Friday	Friday	DOW
		Saturday	Saturday	DOW

The table that follows lists the System Media File information for month values.

Table 53: System Media Files, Time

Symbol (where applicable)	Decimal Value[11]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		hour	hour	Etime, 24TOD per locale, TOD per locale
		hours	hours	Etime,24TOD per locale,TOD per locale
		minute	minute	Etime
		minutes	minutes	Etime
		second	second	Etime,24TOD
		seconds	seconds	Etime,24TOD
		on	on	per locale(unused for en-us)
		at	at	per locale(unused for en-us)
		am	am	TOD

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Symbol (where applicable)	Decimal Value[11]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		pm	pm	TOD
		oclock	oclock	TOD

The table that follows lists the System Media File information for currency values.

Note: The customer's Media Administrator may prefer to replace the contents of "currency_minus" (for the negative amount) and "currency_and" (the latter can even be changed to contain silence).

Table 54: System Media Files, Currency

Symbol (where applicable)	Decimal Value[12]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		currency_ minus	minus	Currency
		currency_and	and	Currency
$	36	USD_dollar	dollar	Currency
		USD_dollars	dollars	Currency
			Note: Unified CVP uses the USD_dollar.wav and USD_dollars.wav media files; the dollar.wav and dollars.wav used by ISN Version 1.0 are no longer installed.	
$	36	CAD_dollar	dollar	Currency
		CAD_dollars	dollars	Currency
		HKD_dollar	dollar	Currency
		HKD_dollars	dollars	Currency
¢	162	cent	cent	Currency
		cents	cents	Currency
		euro	euro	Currency
£	163	GBP_pound	pound	Currency
		GBP_pounds	pounds	Currency
		penny	penny	Currency
		pence	pence	Currency
		MXN_peso	peso	Currency
		MXN_pesos	pesos	Currency
		centavo	centavo	Currency

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Symbol (where applicable)	Decimal Value[12]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		centavos	centavos	Currency

The table that follows lists the System Media File information for gaps of silence and miscellaneous phrases.

Table 55: System Media Files, Silence and Miscellaneous Phrases

Symbol (where applicable)	Decimal Value[13]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
		silence_.1_ sec	(.1 second of silence)	Used for pauses where needed
		silence_.25_ sec	(.25 second of silence)	Used for pauses where needed
		silence_.5_ sec	(.5 second of silence)	Used for pauses where needed
		silence_1_sec	(1 second of silence)	Used for pauses where needed
		and	and	Etime,TOD,25TOD

The table that follows lists the System Media File information for ANSI characters.

Table 56: System Media Files, ANSI Characters

Symbol (where applicable)	Decimal Value[14]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
	32	space	space	Char
!	33	exclamation_ mark	exclamation mark	Char
"	34	double_ quote	double quote	Char
#	35	pound	pound	Char
%	37	percent	percent	Char
&	38	ampersand	ampersand	Char
'	39	apostrophe	apostrophe	Char
(	40	open_ parenthesis	open parenthesis	Char
)	41	close_ parenthesis	close parenthesis	Char
*	42	asterisk	asterisk	Char
+	43	plus	plus	Char

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Symbol (where applicable)	Decimal Value[14]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
,	44	comma	comma	Char
-	45	hyphen	hyphen	Char
.	46	period	period	Char
/	47	slash	slash	Char
:	58	colon	colon	Char
;	59	semicolon	semicolon	Char
<	60	less_than	less than	Char
=	61	equal	equal	Char
>	62	greater_than	greater than	Char
?	63	question_ mark	question mark	Char
@	64	at_symbol	at	Char
[	91	left_square_bracket	left square bracket	Char
\	92	backslash	backslash	Char
]	93	right_square_bracket	right square bracket	Char
^	94	caret	caret	Char
_	95	underscore	underscore	Char
`	96	single_quote	single quote	Char
{	123	open_brace	open brace	Char
|	124	pipe	pipe	Char
}	125	close_brace	close brace	Char
~	126	tilde	tilde	Char
’	130	char_130	low single quote	Char
ƒ	131	char_131	F with hook	Char
”	132	low double quote	low double quote	Char
…	133	ellipsis	ellipsis	Char
†	134	char_134	character 134	Char
‡	135	char_135	character 135	Char
ˆ	136	char_136	character 136	Char
‰	137	per_mille	per mile	Char
Š	138	char_138	character 138	
<	139	left_pointing _angle	left pointing angle	Char
‘	145	left_single_ quote	left single quote	Char
’	146	right_single_ quote	right single quote	Char
“	147	left_double_ quote	left double quote	Char

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Symbol (where applicable)	Decimal Value[14]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
”	148	right_double _quote	right double quote	Char
·	149	bullet	bullet	Char
–	150	en_dash	en dash	Char
—	151	em_dash	em dash	
˜	152	small_tilde	small tilde	Char
™	153	trade_mark	trade mark	Char
š	154	char_154	character 154	Char
›	155	char_155	character 155	Char
¡	161	exclamation_ mark_ inverted	inverted exclamation mark	Char
¤	164	char_164	character 164	Char
¦	166	broken_pipe	broken pipe	Char
§	167	section	section	Char
¨	168	char_168	character 168	Char
©	169	copyright	copyright	Char
ª	170	char_170	character 170	Char
«	171	left_double_ angle_ quote	left double angle quote	Char
¬	172	not	not	Char
-	173	char_173	character 173	Char
®	174	registered	registered	Char
¯	175	char_175	character 175	Char
°	176	degree	degree	Char
±	177	plus_minus	plus or minus	Char
²	178	superscript_ 2	superscript two	Char
³	179	superscript_ 3	superscript three	Char
´	180	acute_accent	acute accent	Char
µ	181	micro	micro	Char
¶	182	paragraph	paragraph	Char
·	183	middle_dot	middle dot	Char
¸	184	cedilla	cedilla	Char
¹	185	superscript_ 1	superscript one	Char
º	186	char_186	character 186	Char
»	187	right_double _angle_ quote	right double angle quote	Char
¿	191	question_ mark_ inverted	inverted question mark	Char

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Miscellaneous Files

The table that follows lists files that are not used by Unfied CVP micro-applications; these files are included for use in customer scripts.

Table 57: Miscellaneous Media Files

Symbol (where applicable)	Decimal Value[15]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
Error	v	invalid_entry_error	Your entry is invalid.	Error message
	v	no_entry_error	Please make a selection.	Error message
	v	system_error	We are currently experiencing technical difficulties with this site. Please try again later when we can service you much better.	Error message
	v	critical_error	We are currently experiencing technical difficulties with this site. Please try again later when we can service you much better.	Error message
	v	critical_error_ULaw .	We are currently experiencing technical difficulties with this site. Please try again later when we can service you much better	Error message
	v	critical_error_ALaw	We are currently experiencing technical difficulties with this site. Please try again later when we can service you much better.	Error message
	v	440beep	<single beep tone>	Unused
	v	busy_tone	<single busy tone>	Unused
	v	busy_tone30	<busy tone 1 per second for 30 seconds>	Unused
	v	central	Central	Unused
	v	credit_of	Credit Of	Unused
	v	dash	dash	Unused

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Symbol (where applicable)	Decimal Value[15]	Media File Name	Media File Content	Data Play Back Types / When Media File Is Used
	v	daylight	daylight	Unused
	v	dialtone	<4 seconds of dial tone>	Unused
	v	dialtone2fastbusy60	<9 seconds of dialtone> followed by <30 seconds of fast busy tone>	Unused
	v	dot	dot	Unused
	v	eastern	Eastern	Unused
	v	ENTER_PHONE_NUMBER	Please enter the phone number.	Unused
	v	fastbusy	<a single fastbusy tone + silence (total of 1 second)>	Unused
	v	fastbusy60	30 seconds of <fastbusy tone>	Unused
	v	FINISHED	When you have finished, press	Unused
	v	goodbye	Goodbye	Unused
	v	Mountain	Mountain	Unused
	v	negative	negative	Unused
	v	of	of	Unused
	v	pmgr_sys	pmgr_sys	Unused
	v	pacific	Pacific	Unused
	v	positive	positive	Unused
	v	ringback	<ring back tone for 1 second followed by 2 seconds of silence>	Unused
	v	savings	savings	Unused
	v	standard	Standard	Unused
	v	Star	star	Unused
	v	thankyou	Thank you	Unused
	v	the	the	Unused
	v	time	time	Unused
	v	try_again	Please try again	Unused

System Media File Error Messages

Three error messages are included with the System Media files:

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- **Critical error.** Message played when system problem exists and the H.323 or SIP Service cannot process the call. (Example content for en-us:"We are currently experiencing technical difficulties with the site, please try again later and we can serve you much better.")

  Note: If you do not want an English spoken critical media, you need to copy the language specific files to the location specified in this section.

  Critical error messages are *not* located on the Media Server:

  – For **H.323 Service**, the critical_error_Alaw.wav and critical_error_Ulaw.wav media files are located in <*install path*>\VoiceBrowser (for example, C:\Cisco\CVP\VoiceBrowser).

  – For **SIP Service**, the critical_error.wav media file is located in <*install path*>\OpsConsoleServer\GWDownloads (for example, C:\Cisco\CVP\OpsConsoleServer\GWDownloads).

  – For **non-Unified CVP H.323 or SIP Service**, an error.wav media file is located in <*install path*>\CVP\audio (for example, C:\Cisco\VXMLServer\Tomcat\webapps\CVP\audio).

  Note: You can record "override" prompts to replace the critical media files. However, you must save them with their original hard-coded names and place them in their original locations.

- **no_entry_error**. Message played when the caller does not respond to a menu prompt. (Example content for en-us:"Please make a selection.") The original prompt is then repeated.

- **invalid_entry_error**. Message played when the caller enters an incorrect response to a menu prompt. (Example content for en-us: "Your entry is invalid.") The original prompt is then repeated.

Note: These files are shared by all applications.

If a dialogue needs to be altered for a specific Get Digits, Get Speech or Menu request in the Unified ICME script, override flags can be set in the Network VRU Script Configuration Parameters.

Note: Override flags are available for the Get Digits, Get Speech, and Menu micro-applications, only. Refer to "Writing Scripts for Unified CVP (page 141)."

You must record the "override" prompts, save them with the hard coded names <prompt name>_no_entry_error.wav and <prompt_name>_invalid_entry_error.wav, and place them with other application-specific media files in the Application Media library.

Note: This override will not work when there is not a specific file name used (for instance, when Unified CVP is using the TTS feature).

Configuring a Media Server

A Media Server administers the media files that contain messages and prompts callers hear. You can add a pre-configured media server to the Operations Console. Once added, you can add a media server to one or more device pools.

The media server is a simple web server with the sole purpose within Unified CVP is to store and serve **`.wav`** files to the VoiceXML gateway, as required in order to render VoiceXML pages. As with ASR/TTS Servers, media servers can be deployed alone, as a redundant pair, or with a Content Services Switch in a farm. The VoiceXML gateway caches the **`.wav`** files it retrieves from the media server. In most deployments, the media server encounters extremely low traffic from Unified CVP.

The Operations Console online help topic *Managing Devices > Configuring a Media Server* provides details for performing the following tasks:

- Adding a Media Server

  Collect the IP address from the media server before adding it to the Operations Console.

- Editing a Media Server Configuration
- Deleting a Media Server
- Adding and Removing a Media Servers from a Device Pool
- Finding a Media Server

Appendix A

Using the Helix Server

The Cisco IOS gateway supports µ-Law 8-bit wav file format. All files in your playlist must be of the µ-Law wav file format to work correctly.

Creating a Broadcast Stream Using the Helix™ Server

Procedure

To create a broadcast stream using the Helix™ Server:

Step 1 Install Helix Server.

The default installation and configuration of Helix server is all that is required for use with Unified CVP. Refer to *Helix Server Administration Guide* for information about installing and configuring Helix Server.

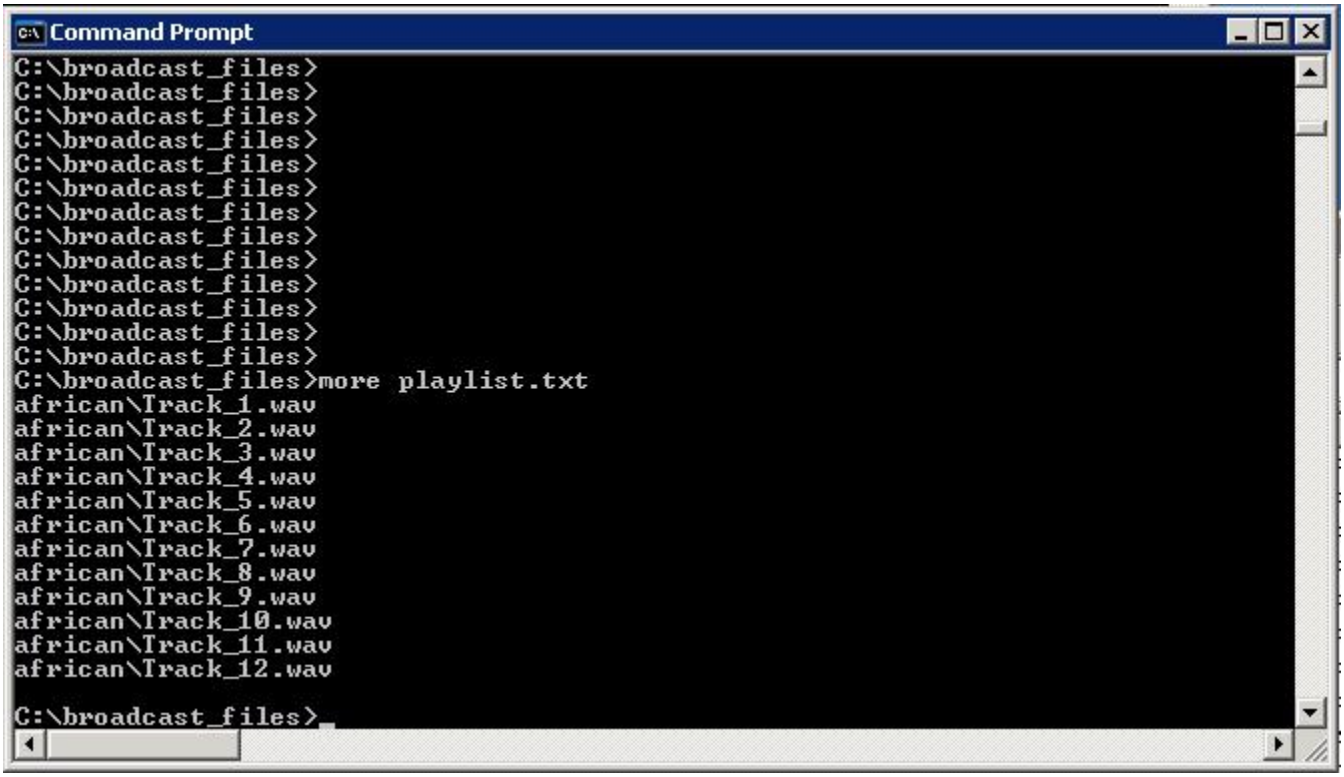
Step 2 Use Simulated Live Transport Agent (SLTA) of Helix Server to create a broadcast stream.

Helix provides many ways to generate broadcast streams. Use the SLTA utility to create a broadcast stream of wav files that are already encoded in the µ-Law format.

Step 3 Create a text file playlist that contains the path andfilename of each file you intend to broadcast.

All files in the playlist must be encoded in the µ-Law wav format. Using a mix of other file formats in the playlist is not supported by Helix Server and may cause a broadcast stream to terminate.

Figure 81: Create Playlist Text File

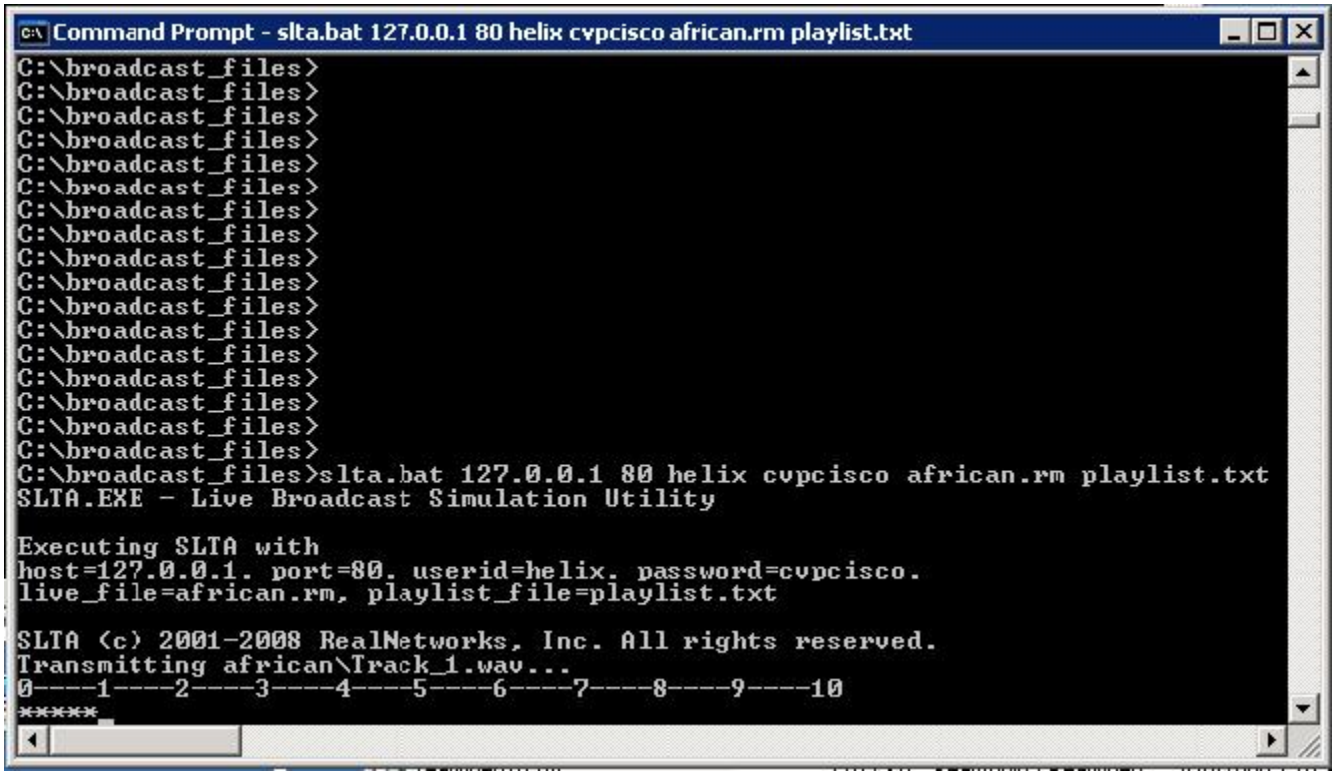


Step 4 Using the SLTA utility, run the **slta.bat script. slta.bat <ip-address> <port> <helix-username> <helix-password> <stream-name> <playlist>** to start the broadcast stream.

The helix userid and password will match the user name and password that you entered when you installed Helix Server.

In the following illustration you see the creation of the RTSP broadcast stream for african.rm in which the files specified in playlist.txt will be continuously looped.

Figure 82: Create the Broadcast Stream



See Also

Configuring Custom Streaming Ringtones (page 363)

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