Cisco Unified Communications Manager Voice Gateways Overview

This chapter provides information about Cisco Unified Communications gateways which enable Cisco Unified Communications Manager to communicate with non-IP telecommunications devices. Cisco Unified Communications Manager supports several types of voice gateways.

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Set Up Gateway

Gateways can be configured with many different features and functions. Network engineers must understand gateway functions, be able to choose when to use which, and be able to configure their gateways. Engineers need a thorough understanding of dialpeer function and configuration to ensure proper call flow. In addition, they must address issues of security and redundancy.

The basic function of a gateway is to translate between different types of networks. In the data environment, a gateway might translate between a Frame Relay network and an Ethernet network, for example. In a VoIP environment, voice gateways are the interface between a VoIP network and the public switched telephone network (PSTN), a private branch exchange (PBX), or analog devices such as fax machines. In its simplest form, a voice gateway has an IP interface and a legacy telephone interface, and it handles the many tasks involved in translating between transmission formats and protocols.
Gateways enable Cisco Unified Communications Manager to communicate with non-IP telecommunications devices and with Internet Service Providers over SIP. In addition, when gateways are properly configured, many can take over for a Cisco Unified Communications Manager when it is unreachable. The following are some generic steps which are required to configure gateways in Cisco Unified Communications Manager:

**Procedure**

**Step 1** Install and configure the voice gateway in the network so that IP connectivity is established.

**Step 2** Gather the information regarding protocol (MGCP/H.323/SIP/SCCP) that you need to configure the gateway to operate with Cisco Unified Communications Manager.

**Step 3** On the gateway, perform any required configuration steps corresponding to the protocol selected.

**Step 4** Add and configure the gateway in Cisco Unified Communications Manager Administration.

**Step 5** Add and configure ports on the gateway if protocol chosen is MGCP or SCCP.

**Step 6** For FXS ports, add directory numbers, if appropriate.

**Step 7** Configure the dial plan for the gateway for routing calls out to the PSTN or other destinations in Cisco Unified Communications Manager. This configuration can include setting up a route group, route list, and route pattern for the Gateway in Cisco Unified Communications Manager or, for protocols like H.323 and SIP, configuring the dial plan on the gateway itself.

**Step 8** Reset the gateway to apply the configuration settings.

**Tip** To get to the default web pages for many gateway devices, you can use the IP address of that gateway. Make your hyperlink url = http://x.x.x.x/, where x.x.x.x is the dot-form IP address of the device. The web page for each gateway contains device information and the real-time status of the gateway.

### Set Up ISDN PRI Gateway

The flow for an ISDN call is different from an analog one. PRI and BRI circuits carry Q.921 and Q.931 signaling messages in the D channel. An MGCP gateway responds to Layer 2 Q.921 signals but does not try to interpret Q.931 call control signals. Instead, when a call needs to be set up, gateways send the Q.931 Layer 3 messages to their Cisco Unified Communications Manager. The gateway uses MGCP Backhaul to send these signals to Cisco Unified Communications Manager over TCP port 2428.

As an example, follow these steps to configure an ISDN-PRI gateway in Cisco Unified Communications Manager.
**Procedure**

**Step 1** Use the Cisco Software Advisor Tool to make sure that the platform and version of Cisco IOS software or Cisco Catalyst Operating system is compatible with MGCP for Unified Communications Manager.

**Step 2** Install and configure the voice gateway in the network such that IP connectivity is established.

**Step 3** In Cisco Unified Communications Manager Administration, choose **Device > Gateway**, and then click **Add New**.

**Step 4** Choose the appropriate MGCP gateway corresponding to the router model. Click **Next**.

**Step 5** Choose MGCP from the protocol drop down menu. Click **Next**.

**Step 6** Configure relevant details in MGCP Gateway Configuration. Select relevant Slots, VICs and endpoints.

**Step 7** Configure relevant details in MGCP Endpoint Configuration. Select relevant Device Pool, Location, PRI Protocol Type, Protocol side, Channel selection order, and PCM Type.

**Step 8** For Cisco IOS Configuration, only the two following commands are required to configure an MGCP gateway. Assume the TFTP server IP address is 10.10.10.10. (PRI/BRI configuration steps are not included)

```
ccm-manager config
ccm-manager config server 10.10.10.10
```

**Step 9** Apply the configuration and reset the gateway to apply the configuration settings.

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**Cisco Voice Gateways**

Cisco Unified Communications Manager supports several types of Cisco Unified Communications gateways. Gateways use call-control protocols to communicate with the PSTN and other non-IP telecommunications devices, such as private branch exchanges (PBXs), key systems, analog phones, fax machines, and modems.

Trunk interfaces specify how the gateway communicates with the PSTN or other external devices by using time-division-multiplexing (TDM) signaling. Cisco Unified Communications Manager and Cisco gateways use a variety of TDM interfaces, but supported TDM interfaces vary by gateway model.

The following list provides available interfaces that Cisco Unified Communications Manager supports with MGCP gateways:

- Foreign Exchange Office (FXO)
- Foreign Exchange Station (FXS)
- T1 Channel Associated Signaling (CAS) receive and transmit ear and mouth (E&M)
- Basic Rate Interface (BRI) Q.931
- T1 PRI-North American ISDN Primary Rate Interface (PRI)
- E1 PRI-European ISDN Primary Rate Interface (PRI)

The following list provides available interfaces that Cisco Unified Communications Manager supports with H.323 gateways:

- FXO
• FXS
• E&M
• Analog Direct Inward Dialing (DID)
• Centralized Automatic Message Accounting (CAMA)
• BRI Q.931
• BRI QSIG-Q signaling protocol that is based on ISDN standards
• T1 CAS FXS, FXO, and E&M
• T1 FGD
• T1/E1 PRI
• T1 PRI NFAS
• T1/E1 QSIG
• J1

The following list provides available interfaces that Cisco Unified Communications Manager supports with SCCP gateways:

• FXS

Cisco Unified Communications Manager can use H.323 gateways that support E1 CAS, but you must configure the E1 CAS interface on the gateway.

The following list provides available interfaces that Cisco Unified Communications Manager supports with Integrated Services Route (ISR) 44XX series gateways:

• T1 CAS/PRI and E1/PRI signaling using MGCP
• T1/PRI and PRI using SIP or H.323
• Analog FXS, FXO and BRI using MGCP
• Analog FXS and BRI using SCCP
• Analog FXS, FXO and BRI using SIP or H.323

The following list provides available interfaces that Cisco Unified Communications Manager supports with Integrated Services Route (ISR) 43XX series gateways:

• T1 CAS/PRI and E1/PRI signaling using MGCP
• T1/PRI and E1/PRI using SIP or H.323
• Analog FXS, FXO and BRI using mgcp
• Analog FXS and BRI using sccp
• Analog FXS, FXO and BRI using SIP or H.323
Standalone Voice Gateways

This section describes these standalone, application-specific gateway models that are supported for use with Cisco Unified Communications Manager.

**Cisco VG224 Analog Phone Gateway**

The Cisco VG224 Analog Phone Gateway, which has a standalone, 17-inch rack-mounted chassis with 24-FXS ports, allows on-premise analog telephones, fax machines, modems, and speakerphones to register with Cisco Unified Communications Manager.

This gateway supports the H.323, MGCP, SCCP, SIP, and T.38 fax relay.

**Cisco Voice Gateway 200**

The Cisco Unified Communications Voice Gateway (VG200) provides a 10/100BaseT Ethernet port for connection to the data network. The following list gives available telephony connections:

- 1 to 4 FXO ports for connecting to a central office or PBX
- 1 to 4 FXS ports for connecting to POTS telephony devices
- 1 or 2 Digital Access T1 ports for connecting to the PSTN
- 1 or 2 Digital Access PRI ports for connecting to the PSTN
- MGCP or H.323 interface to Cisco Unified Communications Manager
  - MGCP mode supports T1/E1 PRI, T1 CAS, FXS, FXO. (Only the user side supports BRI.)
  - H.323 mode supports E1/T1 PRI, E1/T1 CAS, FXS, and FXO. H.323 mode supports E&M, fax relay, and G.711 modem.

The MGCP VG200 integration with legacy voice-messaging systems allows the Cisco Unified Communications Manager to associate a port with a voice mailbox and connection.

**MGCP BRI Call Connections**

Previously, gateways used H.323 signaling to Cisco Unified Communications Manager to provide interfaces to the public switched telephone network (PSTN) for BRI ISDN connections.

Now, Cisco Unified Communications Manager can use a Media Gateway Control Protocol (MGCP) gateway to handle BRI ISDN connections to the PSTN and to provide a centrally administered gateway interface. Cisco Unified Communications Manager uses logical connections to exchange MGCP and ISDN Q.931 messages with the gateway. This connection uses a User Datagram Protocol (UDP) logical connection for exchanging MGCP messages and a Transmission Control Protocol (TCP) connection for the backhaul ISDN Q.931 messages.

The following figure shows a typical scenario that centralizes call processing for remote-site BRI trunk gateways that connect to the PSTN. When a call arrives from or goes to the PSTN over the BRI trunk, the
Cisco Unified Communications Manager and the gateway (based on an IOS router) exchange ISDN Q.931 messages across the WAN.

*Figure 1: Topology Shows a Scenario by Using MGCP BRI Interfaces*

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**Note**

The BRI gateway supports MGCP BRI backhaul for BRI trunk only. It does not support BRI phone or station. The IOS gateway supports BRI phones that use Skinny Client Control Protocol.

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**Switch-Based Gateways**

Several telephony modules for the Cisco Catalyst 4000 and 6000 family switches act as telephony gateways. You can use existing Cisco Catalyst 4000 or 6000 family devices to implement IP telephony in your network by using the following voice gateway modules:

- Install Catalyst 6000 voice gateway modules that are line cards in any Cisco Catalyst 6000 or 6500 series switch.
- Install the Catalyst 4000 access gateway module in any Catalyst 4000 or 4500 series switch.

**Cisco Catalyst 6000 8-Port Voice T1/E1 and Services Module**

The Cisco Catalyst 6000 8-Port Voice T1/E1 and Services Modules provide the following features:

- 8 ports for providing
  - Digital T1/E1 connectivity to the PSTN (T1/E1 PRI or T1 CAS)
  - Digital signal processor (DSP) resources for transcoding and conferencing
- MGCP interface to Cisco Unified Communications Manager
- Connection to a voice-messaging system (using T1 CAS)

Users have the flexibility to use ports on a T1 module for T1 connections or as network resources for voice services. Similarly, the E1 module provides ports for E1 connections or as network resources. The ports can serve as T1/E1 interfaces, or the ports will support transcoding or conferencing.
Either module supports DSP features on any port, but T1 modules cannot be configured for E1 ports, and E1 modules cannot be configured for T1 ports.

**Note**

Similar to the Cisco MGCP-controlled gateways with FXS ports, the Cisco 6608 T1 CAS gateway supports hookflash transfer. Hookflash transfer defines a signaling procedure that allows a device, such as a voice-messaging system, to transfer to another destination. While the device is connected to Cisco Unified Communications Manager through a T1 CAS gateway, the device performs a hookflash procedure to transfer the call to another destination. Cisco Unified Communications Manager responds to the hookflash by using a blind transfer to move the call. When the call transfer completes, the voice channel that connected the original call to the device gets released.

**Note**

Only E&M T1 ports support hookflash transfer.

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**Cisco Catalyst 6000 24 Port FXS Analog Interface Module**

The Cisco Catalyst 6000 24 Port FXS Analog Interface Module provides the following features:

- 24 Port RJ-21 FXS module
- V.34/V.90 modem, voice-messaging system, IVR, POTS
- Cisco fax relay
- MGCP interface to Cisco Unified Communications Manager

The Catalyst 6000 24 Port FXS Analog Interface Module provides 24 FXS ports for connecting to analog phones, conference room speakerphones, and fax machines. You can also connect to legacy voice-messaging systems by using SMDI and by associating the ports with voice-messaging extensions. The FXS module provides legacy analog devices with connectivity into the IP network. Analog devices can use the IP network infrastructure for toll-bypass applications and to communicate with devices such as SCCP IP phones and H.323 end stations. The FXS module also supports fax relay, which enables compressed fax transmission over the IP WAN and preserves valuable WAN bandwidth for other data applications.

**Cisco Catalyst 4000 Access Gateway Module**

The Cisco Catalyst 4000 Access Gateway Module provides an MGCP or H.323 gateway interface to Cisco Unified Communications Manager. You can configure this module with the following interface and service modules:

- 6 ports for FXS and FXO
- 2 T1/E1 ports for Digital Access PRI and Digital Access T1

**Cisco Catalyst 4224 Voice Gateway Switch**

The Cisco Catalyst 4224 Voice Gateway Switch provides a single-box solution for small branch offices. The Catalyst 4224 provides switching, IP routing, and PSTN voice-gateway services by using onboard digital
signal processors (DSPs). The Catalyst 4224 has four slots that you can configure with multiflex voice and WAN interface cards to provide up to 24 ports. These ports can support the following voice capabilities:

- FXS ports for POTS telephony devices
- FXO ports for PSTN connections
- T1 or E1 ports for Digital Access PRI, and Digital Access T1 services

The Cisco Catalyst 4224 Access Gateway Switch provides an MGCP or H.323 interface to Cisco Unified Communications Manager.

**H.323 Gateways**

H.323 devices comply with the H.323 communications standards and enable video conferencing over LANs and other packet-switched networks. You can add third-party H.323 devices or other Cisco devices that support H.323 (such as the Cisco 2900 series, 3900 series, or VG350 series gateways).

**Cisco IOS H.323 Gateways**

Cisco IOS H.323 gateways such as the Cisco 2900, 3900, 2800, 3800, 4451, 1861, ASR1000, VG200 and VG350 provide full-featured routing capabilities. See the documentation for each of these gateway types for information about supported voice gateway features and configuration.

**Outbound FastStart Call Connections**

Calls that are placed from IP phones over large WAN topologies can experience voice clipping when the called party goes off hook to answer the call. When H.323 trunks or gateways are separated from the Cisco Unified Communications Manager server, significant delays can occur because of the many H.245 messages that are exchanged when a call is set up.

With the FastStart feature, information that is required to complete a media connection between two parties gets exchanged during the H.225 portion of call setup, and this exchange eliminates the need for H.245 messages. The connection experiences one roundtrip WAN delay during call setup, and the calling party does not receive voice clipping when the called party answers the call.

Cisco Unified Communications Manager uses media termination points (MTP) for making an H.323 outbound FastStart call. Cisco Unified Communications Manager starts an outbound FastStart call by allocating an MTP and opening the receive channel. Next, the H.323 Fast Connect procedure sends the SETUP message with a FastStart element to the called endpoint. The FastStart element includes information about the receiving channel for the MTP.

The called endpoint accepts the H.323 Fast Connect procedure by sending a CALL PROCEEDING, PROGRESS, ALERT, or CONNECT message that contains a FastStart element. When Cisco Unified Communications Manager receives the FastStart element, it connects the media immediately and avoids the delays with the usual exchange of H.245 messages.

The called endpoint can refuse the H.323 Fast Connect procedure by not returning the FastStart element in any of the messages up to and including the CONNECT message. In this case, the Cisco Unified Communications Manager handles the call as a normal call and uses the MTP for subsequent media cut-through.

The Outbound FastStart feature requires an MTP. If an MTP is not available when the call is set up, the call continues without FastStart and with no supplementary services. If you want all calls to use FastStart only,
you can set the service parameter called "Fail call if MTP allocation fails," which is located in the Cluster Wide Parameters (Device-H323) portion of the service parameters for the Cisco Unified CallManager service. When you set this parameter to True, the system rejects calls when no MTP is available.

**MGCP Gateways**

MGCP separates the functions of call control and media translation into two separate devices:

- The voice gateway handles **media translation**
- A call agent handles **call control**.

This following text gives a brief overview of how MGCP functions and how you implement these functions using Cisco Unified Communications Manager as the call agent.

An MGCP gateway routes calls in response to instructions from the Cisco Unified Communications Manager. These calls can be to or from a telephone on the PSTN, or across a WAN to an IP or analog phone at a remote site. The gateway does not make call routing decisions. It needs to be able to reach a Cisco Unified Communications Manager before it can handle calls. When you are using an MGCP gateway, all the dial plan knowledge resides on the Cisco Unified Communications Manager. You do not need to configure dial peers, unlike H.323 and SIP. However, this leaves the gateway unable to route calls if it cannot reach a Cisco Unified Communications Manager.

Cisco Unified Communications Manager supports only version 0.1, but Cisco gateways can use version 1.0 with other call agents.

Cisco Unified Communications Manager is able to exercise per-port control of connections to the public switched telephone network (PSTN), legacy private branch exchanges (PBX), voice-mail systems, and plain old telephone service (POTS) phones. This allows complete control of the dial plan from Cisco Unified Communications Manager, which centralizes gateway management and provides scalable IP Telephony deployments.

One gateway can support multiple endpoints. Endpoint names have two components, a local identifier, and a gateway identifier. The entire name consists of the local identifier, followed by the @ symbol, and then the gateway identifier. For example:

```plaintext
local_identifier@gateway_identifier.domain_name
```

The gateway identifier is its configured hostname, such as MGCP-router. If the gateway is configured with a domain name, it is appended to the end of the hostname, such as `MGCP-router.cisco.com`.

The format of a local identifier varies depending on the type of interface. The local identifier for analog ports uses the following syntax:

```plaintext
Endpoint type/Slot #/Subunit #/Port #
```

MGCP was created for a centralized architecture, where most of the configuration and call-control intelligence resides on a call agent, such as Cisco Unified Communications Manager. The traditional role of an MGCP gateway is media translation. PSTN connections, such as Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), and PRI lines, typically terminate in the gateway. The gateway then translates between the PSTN and the IP network.
Voice Gateway Model Summary of Supported Protocols, Trunks, and Port Types

The following table summarizes Cisco voice gateways that Cisco Unified Communications Manager supports with information about the supported signaling protocols, trunk interfaces, and port types.

Table 1: Supported Voice Gateways, Protocols, Trunk Interfaces, and Port Types

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<tr>
<th>Gateway Model</th>
<th>Supported Signaling Protocols</th>
<th>Trunk Interfaces</th>
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</tr>
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<tr>
<td>T1 FGD</td>
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<td>E1 CAS</td>
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<tr>
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</tr>
<tr>
<td>T1/E1 PRI</td>
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<td>T1 PRI (Megacom/SDN)</td>
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<tr>
<td>Gateway Model</td>
<td>Supported Signaling Protocols</td>
<td>Trunk Interfaces</td>
<td>Port Types</td>
<td>Notes</td>
</tr>
<tr>
<td>-------------------------------------</td>
<td>-------------------------------</td>
<td>------------------</td>
<td>-----------------------------------</td>
<td>-----------------------------------</td>
</tr>
<tr>
<td>MGCP</td>
<td>FXS</td>
<td>Loopstart or groundstart</td>
<td>Basic calls only</td>
<td></td>
</tr>
<tr>
<td></td>
<td>FXO</td>
<td>Loopstart or groundstart</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>BRI</td>
<td>Loopstart or groundstart</td>
<td>User side only; no QSIG support</td>
<td></td>
</tr>
<tr>
<td></td>
<td>T1 CAS (E&amp;M)</td>
<td>Loopstart or groundstart</td>
<td>Supplementary Services</td>
<td></td>
</tr>
<tr>
<td></td>
<td>T1/E1 QSIG</td>
<td>Loopstart or groundstart</td>
<td>Supplementary Services</td>
<td></td>
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<td>T1/E1 PRI</td>
<td>Loopstart or groundstart</td>
<td>Per call</td>
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<td>FXS</td>
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<td>Basic calls only</td>
<td></td>
</tr>
<tr>
<td></td>
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<td>H.323 and SIP</td>
<td>Loopstart or groundstart</td>
<td>Per call</td>
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<tr>
<td>Cisco Analog Gateways</td>
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<tr>
<td>Cisco VG224 Analog Gateway</td>
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<td>Basic calls only</td>
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<td>SCCP</td>
<td>Loopstart or groundstart</td>
<td>Supplementary Services</td>
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<td>MGCP</td>
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<td>SCCP</td>
<td>FXS/BRI</td>
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<tr>
<td></td>
<td>H.323 and SIP</td>
<td>FXS/FXO/BRI</td>
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<tr>
<td>Cisco VG320</td>
<td>MGCP</td>
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<td>SCCP</td>
<td>FXS/BRI</td>
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<td>Gateway Model</td>
<td>Supported Signaling Protocols</td>
<td>Trunk Interfaces</td>
<td>Port Types</td>
<td>Notes</td>
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<tr>
<td></td>
<td>H.323 and SIP</td>
<td>T1/CAS/PRI</td>
<td>FXS/FXO/BRI</td>
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<td>E1/PRI</td>
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<td>Cisco VG350</td>
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<td></td>
<td>FXS/BRI</td>
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<tr>
<td></td>
<td>H.323 and SIP</td>
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<td>FXS/FXO/BRI</td>
<td></td>
</tr>
<tr>
<td>Cisco VG 202 and 204 Gateway</td>
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<td>FXS</td>
<td>Loopstart or groundstart</td>
<td>Basic calls only</td>
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<tr>
<td>SCCP</td>
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<td>Supplementary Services</td>
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<td>E1/PRI</td>
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<td>H.323 and SIP</td>
<td>T1/CAS/PRI</td>
<td>FXS/FXO/BRI</td>
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<tr>
<td></td>
<td></td>
<td>E1/PRI</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Gateways Dial Plans and Route Groups**

Gateways use dial plans to access or call out to the PSTN, route groups, and group-specific gateways. The different gateways that are used within Cisco Unified Communications Solutions have dial plans that are configured in different places:

- Configure dial plan information for both Skinny and MGCP gateways in the Cisco Unified Communications Manager.
- Configure dial plans in Cisco Unified Communications Manager to access the H.323-based Cisco IOS software gateways. Configure dial peers in the H.323-based gateways to pass the call out of the gateway.

The route group points to one or more gateways and can choose the gateways for call routing based on preference. The route group can serve as a trunk group by directing all calls to the primary device and then using the secondary devices when the primary is unavailable. One or more route lists can point to the same route group.

All devices in a given route group share the same characteristics such as path and digit manipulation. Cisco Unified Communications Manager restricts the gateways that you can include in the same route group and the route groups that you can include in the same route list.

Route groups can perform digit manipulation that will override what was performed in the route pattern. Configuration information that is associated with the gateway defines how the call is actually placed and can override what was configured in the route pattern.
You can configure H.323 trunks, not H.323 gateways, to be gatekeeper-controlled trunks. This means that before a call is placed to an H.323 device, it must successfully query the gatekeeper.

Multiple clusters for inbound and outbound calls can share H.323 trunks, but MGCP and Skinny-based gateways remain dedicated to a single Cisco Unified Communications Manager cluster.

**Related Topics**

- Configure Gatekeeper and Gatekeeper-Controlled Trunk
- Route Plan Overview

### Dependency Records for Gateways and Their Route Groups and Directory Numbers

To find route groups or directory numbers that a specific gateway or gateway port is using, click the Dependency Records link that is provided on the Cisco Unified Communications Manager Administration Gateway Configuration window. The Dependency Records Summary window displays information about route groups and directory numbers that are using the gateway or port. To find out more information about the route group or directory number, click the route group or directory number, and the Dependency Records Details window displays. If the dependency records are not enabled for the system, the dependency records summary window displays a message.

### Gateways and the Local Route Groups Feature

A special virtual Local Route Group can be bound to a real route group differently based on the Local Route Group device pool setting of the originating device. Devices, such as phones, from different locales can therefore use identical route lists and route patterns, but Cisco Unified Communications Manager selects the correct gateway(s) for their local end.

If the Local Route Group feature is in use, configuration of gateways changes, particularly with respect to configuration of the following gateway fields:

- Called Party Transformation CSS
- Use Device Pool Called Party Transformation CSS

### Gateways and the Calling Party Normalization Feature

In line with E.164 standards, calling party normalization enhances the dialing capabilities of some phones and improves call back functionality when a call is routed to multiple geographical locations; that is, the feature ensures that the called party can return a call without needing to modify the directory number in the call log directories on the phone. Additionally, calling party normalization allows you to globalize and localize phone numbers, so the appropriate calling number presentation displays on the phone.

Configuring calling party normalization alleviates issues with toll bypass where the call is routed to multiple locations over the IP WAN. In addition, it allows Cisco Unified Communications Manager to distinguish the origin of the call to globalize or localize the calling party number for the phone user.

SIP trunks and MGCP gateways can support sending the international escape character, +, for calls. H.323 gateways/trunks do not support the + because the H.323 protocol does not support the international escape.
character, +. For outgoing calls through a gateway that supports +, Cisco Unified Communications Manager can send the + with the dialed digits to the gateway/trunk. For outgoing calls through a gateway/trunk that does not support +, the international escape character + gets stripped when Cisco Unified Communications Manager sends the call information to the gateway/trunk.

SIP does not support the number type, so calls through SIP trunks only support the Incoming Calling Party Unknown Number (prefix and digits-to-strip) settings.

You can configure the international escape character, +, to globalize the calling party number. For information on the international escape character, +, see Use the International Escape Character.

### Apply the International Escape Character to Inbound Calls Over H.323 Trunks

The H.323 protocol does not support the international escape character, +. To ensure that correct prefixes, including the international escape character, +, get applied for inbound calls over H.323 gateways/trunks, you must configure the incoming called party settings in the service parameter, device pool, H.323 gateway, or H.323 trunk windows; that is, configuring the incoming called party settings ensures that when an inbound call comes from a H.323 gateway or trunk, Cisco Unified Communications Manager transforms the called party number back to the value that was originally sent over the trunk/gateway.

For example, to ensure that the correct DN patterns get used with SAF/call control discovery for inbound calls over H.323 gateways/trunks, you must configure the incoming called party settings in the service parameter, device pool, or H.323 (non-gatekeeper controlled) trunk window. See the following example for more information.

- A caller places a call to +19721230000 to Cisco Unified Communications Manager A.
- Cisco Unified Communications Manager A receives +19721230000 and transforms the number to 55519721230000 before sending the call to the H.323 trunk. In this case, your configuration indicates that the international escape character + should be stripped and 555 should be prepended for calls of International type.
- For this inbound call from the trunk, Cisco Unified Communications Manager B receives 55519721230000 and transforms the number back to +19721230000 so that digit analysis can use the value as it was sent by the caller. In this case, your configuration for the incoming called party settings indicates that you want 555 to be stripped and +1 to be prepended to called party numbers of International type.

The service parameters support the Cisco CallManager service. To configure the service parameters, click Advanced in the Service Parameter Configuration window for the Cisco CallManager service; then, locate the H.323 pane for the following parameters:

- Incoming Called Party National Number Prefix - H.323
- Incoming Called Party International Number Prefix - H.323
- Incoming Called Party Subscriber Number Prefix - H.323
- Incoming Called Party Unknown Number Prefix - H.323

These service parameters allow you to prefix digits to the called number based on the Type of Number field for the inbound offered call. You can also strip a specific number of leading digits before the prefix gets applied. To prefix and strip digits by configuring these parameter fields, use the following formula, x:y, where x represents the exact prefix that you want to add to called number and y represents the number of digits.
stripped; be aware that the colon separates the prefix and the number of stripped digits. For example, enter 91010:6 in the field, which means that you want to strip 6 digits and then add 901010 to the beginning of the called number. In this example, a national call of 2145551234 becomes 910101234. You can strip up to 24 digits and prefix/add up to than 16 digits.

**Gateway Failover and Fallback**

This section describes how these Cisco voice gateways handle Cisco Unified Communications Manager failover and fallback situations.

**MGCP Gateways**

To handle Cisco Unified Communications Manager failover situations, MGCP gateways receive a list of Cisco Unified Communications Managers that is arranged according to the Cisco Unified Communications Manager group and defined for the device pool that is assigned to the gateway. A Cisco Unified Communications Manager group can contain one, two, or three Cisco Unified Communications Managers that are listed in priority order for the gateway to use. If the primary Cisco Unified Communications Manager in the list fails, the secondary Cisco Unified Communications Manager gets used. If the primary and secondary Cisco Unified Communications Managers fail, the tertiary Cisco Unified Communications Manager gets used.

Fallback describes the process of recovering a higher priority Cisco Unified Communications Manager when a gateway fails over to a secondary or tertiary Cisco Unified Communications Manager. Cisco MGCP gateways periodically take status of higher priority Cisco Unified Communications Managers. When a higher priority Cisco Unified Communications Manager is ready, it gets marked as available again. The gateway reverts to the highest available Cisco Unified Communications Manager when all calls go idle or within 24 hours, whichever occurs first. The administrator can force a fallback either by stopping the lower priority Cisco Unified Communications Manager whereby calls get preserved, by restarting the gateway, which preserves calls, or by resetting Cisco Unified Communications Manager, which terminates calls.

**Note**

Skinny Client Control Protocol (SCCP) gateways handle Cisco Unified Communications Manager redundancy, failover, and fallback in the same way as MGCP gateways.

**IOS H.323 Gateways**

Cisco IOS gateways also handle Cisco Unified Communications Manager failover situations. By using several enhancements to the dial-peer and voice class commands in Cisco IOS Release 12.1(2)T, Cisco IOS gateways can support redundant Cisco Unified Communications Managers. The command, h225 tcp timeout seconds, specifies the time that it takes for the Cisco IOS gateway to establish an H.225 control connection for H.323 call setup. If the Cisco IOS gateway cannot establish an H.225 connection to the primary Cisco Unified Communications Manager, it tries a second Cisco Unified Communications Manager that is defined in another dial-peer statement. The Cisco IOS gateway shifts to the dial-peer statement with the next highest preference setting.

The following example shows the configuration for H.323 gateway failover:

```
interface FastEthernet0/0
ip address 10.1.1.10 255.255.255.0
dial-peer voice 101 voip
```
destination-pattern 1111
session target ipv4:10.1.1.101
preference 0
voice class h323 1
dial-peer voice 102 voip
destination-pattern 1111
session target ipv4:10.1.1.102
preference 1
voice class h323 1
voice class h323 1
h225 timeout tcp establish 3

To simplify troubleshooting and firewall configurations, Cisco recommends that you use the new
voip-gateway voip bind srcaddr command for forcing H.323 always to use a specific source IP address
in call setup. Without this command, the source address that is used in the setup might vary and depends
on protocol (RAS, H.225, H.245, or RTP).

Transfer Calls Between Gateways

Using Cisco Unified Communications Manager Administration, you can configure gateways as OnNet (internal)
gateways or OffNet (external) gateways by using Gateway Configuration or by setting a clusterwide service
parameter. Used in conjunction with the clusterwide service parameter, Block OffNet to OffNet Transfer, the
configuration determines whether calls can be transferred over a gateway.

To use the same gateway to route both OnNet and OffNet calls, associate the gateway with two different route
patterns. Make one gateway OnNet and the other OffNet with both having the Allow Device Override check
box unchecked.

Gateway Transfer Capabilities Configuration Settings

Using Cisco Unified Communications Manager Administration Gateway Configuration, you can configure
a gateway as OffNet or OnNet. The system considers the calls that come to the network through that gateway
OffNet or OnNet, respectively. Use the Gateway Configuration window field, Call Classification, to configure
the gateway as OffNet, OnNet, or Use System Default. See the table below for description of these settings.

The Route Pattern Configuration window provides a drop-down list box called Call Classification, which
allows you to configure a route pattern as OffNet or OnNet. When Call Classification is set to OffNet and the
Allow Device Override check box is unchecked, the system considers the outgoing calls that use this route
pattern as OffNet (if configured as OnNet and check box is unchecked, then outgoing calls are considered
OnNet).

You can use the same gateway to route both OnNet and OffNet calls by associating the gateway with two
different route patterns: one OnNet and the other OffNet, with both having the Allow Device Override check
box unchecked. For outgoing calls, the outgoing device setting classifies the call as either OnNet or OffNet
by determining whether the Allow Device Override check box is checked.

In route pattern configuration, if the Call Classification is set as OnNet, the Allow Device Override check
box is checked, and the route pattern is associated with an OffNet gateway, the system considers the outgoing
call OffNet.
Table 2: Gateway Configuration Call Classification Settings

<table>
<thead>
<tr>
<th>Setting Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>OffNet</td>
<td>This setting identifies the gateway as being an external gateway. When a call comes in from a gateway that is configured as OffNet, the outside ring gets sent to the destination device.</td>
</tr>
<tr>
<td>OnNet</td>
<td>This setting identifies the gateway as being an internal gateway. When a call comes in from a gateway that is configured as OnNet, the inside ring gets sent to the destination device.</td>
</tr>
<tr>
<td>Use System Default</td>
<td>This setting uses the Cisco Unified Communications Manager clusterwide service parameter Call Classification.</td>
</tr>
</tbody>
</table>

Set Up Transfer Capabilities by Using Call Classification Service Parameter

To configure all gateways to be OffNet (external) or OnNet (internal), perform the following two steps:

**Procedure**

**Step 1**
Use the Cisco Unified Communications Manager clusterwide service parameter Call Classification.

**Step 2**
Configure individual gateways to Use System Default in the Call Classification field that is on the Gateway Configuration window.

Block Transfer Capabilities by Using Service Parameters

Block transfer provides a way of restricting transfer between external devices, so fraudulent activity gets prevented. You can configure the following devices as OnNet (internal) or OffNet (external) to Cisco Unified Communications Manager:

- H.323 gateway
- MGCP FXO trunk
- MGCP T1/E1 trunk
- Intercluster trunk
- SIP trunk

If you do not want OffNet calls to be transferred to an external device (one that is configured as OffNet), set the Cisco Unified Communications Manager clusterwide service parameter, Block OffNet to OffNet Transfer, to True.

If a user tries to transfer a call on an OffNet gateway that is configured as blocked, a message displays on the user phone to indicate that the call cannot be transferred.
H.235 Support for Gateways

This feature allows Cisco Unified Communications Manager gateways to transparently pass through the shared secret (Diffie-Hellman key) and other H.235 data between two H.235 endpoints so that the two endpoints can establish a secure media channel.