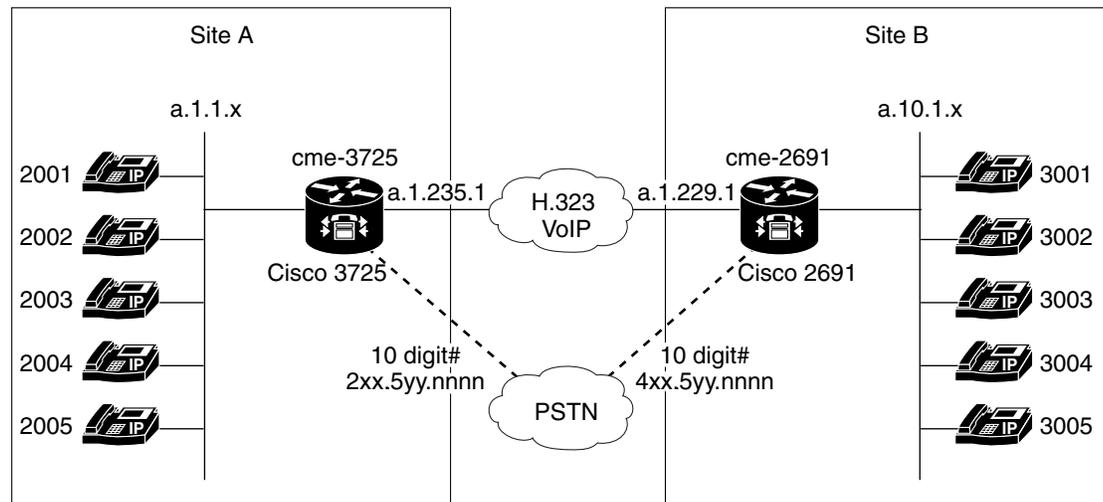




Cisco IPC Express System Configuration Example

This application note provides a step-by-step configuration summary to help you set up your Cisco IP Communications (IPC) Express system. This application note uses a two-site sample network, as shown in [Figure 31](#). This application note walks through the setup and configuration of Site A, the primary site. Site B has a very similar basic configuration and is used to illustrate additional configuration required to connect two Cisco IPC Express sites.

Figure 31 Cisco IPC Express Network with Two Sites



Site A uses the following hardware components:

- PSTN—VWIC-2MFT-T1 and AIM-ATM-VOICE-30 (WIC slot 0)
- Router—Cisco 3725 router
- NM-CUE—Cisco Unity Express 2.0 (NM Slot 1)
- PSTN interface—NM-HD-2V and VIC2-2FXO (NM slot 2)
- Cisco IOS image—Cisco IOS Release 12.3(11)T2 IP Voice
- IP phones—Two real Cisco 7960 IP Phones and three virtual phones

Site B uses the following hardware components:

- Router—Cisco 2691 router

- NM-CUE—Cisco Unity Express 2.0 (slot 1)
- Cisco IOS image—Cisco IOS Release 12.3(11)T2 IP Voice
- IP phones—Two real Cisco 7960 IP Phones and three virtual phones

The Site A numbering plan starts with two, and Site B's extensions start with three. Both sites use a four-digit extension numbering plan.

The following sections summarize the process of configuring Cisco IPC Express system:

- [Step 1—Planning and Offline Staging, page 88](#)
- [Step 2—Basic Router Setup, page 90](#)
- [Step 3—Initial Cisco CME System Setup, page 92](#)
- [Step 4—Configuring Extensions and Phones, page 96](#)
- [Step 5—Configuring the PSTN Interface, page 99](#)
- [Step 6—Configuring Cisco Unity Express AA and Voice Mail, page 102](#)
- [Step 7: Configuring Cisco CME Call Processing Features, page 113](#)
- [Step 8—Interconnecting Multiple Cisco IPC Express Systems, page 126](#)

The “[Sample System Configurations](#)” section on [page 128](#) provides the full configurations of Sites A and B built during these steps.

Step 1—Planning and Offline Staging

Before building a configuration for a Cisco IPC Express system, note the parameters of the configuration you want to put in place. This includes such information as the names of the employees in your office, the extension numbering scheme you want to use, the Public Switched Telephone Network (PSTN) numbers your office has assigned, where PSTN calls will be routed, what type of phone each employee and room will have, what the IP addressing of your office is, and where the Trivial File Transfer Protocol (TFTP) server is located.

You can proceed from here in two ways, depending on whether you have the equipment ready in your office or lab:

- If you have already ordered the equipment and it has been delivered, or if you are reusing existing routers and IP phones, move on to Step 2, Basic Router Setup.
- If you have no equipment readily available, but you want to start building a configuration for staging purposes, read about offline staging in this section before moving on. The rest of this application note cannot be executed without access to the equipment.

A Cisco CME Installation Configuration Tool (ICT) is an offline HTML tool available as a shareware application provided for Cisco Partners and Resellers. You can download the tool from the Cisco.com software center for Cisco CME, and use it to set up the basic telephony-service, IP phone, and voice mail configuration for all Cisco CME supported platforms.

You fill in basic fields about the system's desired configuration and scan in the phones' Media Access Control (MAC) addresses (if they are available). The tool output provides the router configuration (command-line interface [CLI]) as well as the Cisco Unity Express configuration, which you can cut and paste into the equipment when that arrives.

Bring up a Microsoft Internet Explorer (IE) browser with the `expresso.htm` file. Note that `expresso.htm` is the only file that serves as an entrance to the tool. Figure 32 shows this tool's main window, where you enter the number of phones, the IP addressing, whether you have voice mail, and, if so, the pilot numbers for AA and voice mail.

Figure 32 Cisco CME ICT System Parameters

Cisco CME Initial Configuration Tool
Version 1.3.1

General System Information

Company Name:

CCME Feature License: CCME Version:

Administrator UserID: Administrator Password:

IOS Image to Boot on Flash:

Number of CME phones: First Phone Number: dual-line

Time Zone: Daylight Saving

Network Parameters

DHCP Network IP Address: Subnet Mask:

DHCP Excluded Address: to

CME IP Address: Subnet Mask:

Ethernet Type: Port Number:

NTP Server 1 IP Address: NTP Server 2 IP Address:

PSTN Connectivity Parameters

PSTN Connection

PSTN Connection Type: Starting Port Number:

DID Available First DID Number: Total DID:

EtherSwitch Module Parameters

EtherSwitch Module

Voicemail Parameters

Voice Mail System

146208

Click the **Phone Parameters** tab on the left panel to go to the next window, shown in Figure 33, where you can enter or scan in the phones' MAC addresses. Using a barcode scanner is an effective method to add phones to a configuration with the ICT tool. The Flic scanner from <http://www.flicscanner.com> has been tested with the ICT tool.

Click the **show cli** button on the left panel after filling in the phone parameters to generate both the Cisco CME and Cisco Unity Express CLI configurations for the given parameters. This is a flat text file that you can cut and paste into a document and ultimately into the console or a Telnet session into the router CLI to enter the configuration into the system.

Figure 33 Cisco CME ICT Phone Parameters

The screenshot shows the Cisco CME Initial Configuration Tool interface. On the left, there are navigation buttons for 'System Parameters', 'Phone Parameters', and 'Show CLI'. The main area displays a table with 10 rows of phone parameters. Each row includes a line number, Mac Address, Phone number, Dual Line checkbox, Ext. Number, DID Number, Name, UserID, PW, VM checkbox, CFB, CFNA, and TO. The 'Dual Line' checkbox is checked for all lines. The 'VM' checkbox is also checked for all lines. The 'Name' field contains 'First Last Name' for all entries. The 'UserID' field contains 'user1' through 'user10'. The 'PW' field contains '2001' through '2010'. The 'CFB' and 'CFNA' fields both contain '2105'. The 'TO' field contains '10' for all entries.

	Mac Address	Phone	Dual Line	Ext. Number	DID Number	Name	UserID	PW	VM	CFB	CFNA	TO
1.	0000.0000.0001	7940	<input checked="" type="checkbox"/>	2001	yyy2000	First Last Name	user1	2001	<input checked="" type="checkbox"/>	2105	2105	10
2.	0000.0000.0002	7940	<input checked="" type="checkbox"/>	2002	yyy2001	First Last Name	use2	2002	<input checked="" type="checkbox"/>	2105	2105	10
3.	0000.0000.0003	7940	<input checked="" type="checkbox"/>	2003	yyy2002	First Last Name	use3	2003	<input checked="" type="checkbox"/>	2105	2105	10
4.	0000.0000.0004	7940	<input checked="" type="checkbox"/>	2004	yyy2003	First Last Name	use4	2004	<input checked="" type="checkbox"/>	2105	2105	10
5.	0000.0000.0005	7940	<input checked="" type="checkbox"/>	2005	yyy2004	First Last Name	use5	2005	<input checked="" type="checkbox"/>	2105	2105	10
6.	0000.0000.0006	7940	<input checked="" type="checkbox"/>	2006	yyy2005	First Last Name	use6	2006	<input checked="" type="checkbox"/>	2105	2105	10
7.	0000.0000.0007	7940	<input checked="" type="checkbox"/>	2007	yyy2006	First Last Name	use7	2007	<input checked="" type="checkbox"/>	2105	2105	10
8.	0000.0000.0008	7940	<input checked="" type="checkbox"/>	2008	yyy2007	First Last Name	use8	2008	<input checked="" type="checkbox"/>	2105	2105	10
9.	0000.0000.0009	7940	<input checked="" type="checkbox"/>	2009	yyy2008	First Last Name	use9	2009	<input checked="" type="checkbox"/>	2105	2105	10
10.	0000.0000.0010	7940	<input checked="" type="checkbox"/>	2010	yyy2009	First Last Name	user10	2010	<input checked="" type="checkbox"/>	2105	2105	10

Step 2—Basic Router Setup

As soon as you have the equipment on hand, the configuration of the real system can start. You can do this either by copying and pasting the staging configuration you built in the preceding section or by working through the following sections to set up the basic router and then the Cisco CME and Cisco Unity Express parameters. Even if you use the staging configuration, it is still recommended that you scan the steps in this application note. This will help you make sure that the parameters in the staged configuration are, in fact, the values you want for your real system configuration. It also helps you make any adjustments along the way to tailor the system for your use.

The following basic setup requirements are summarized in this section:

- [Installing Hardware and Software, page 90](#)
- [Configuring Router IP Addressing, page 91](#)
- [Setting the Router Clock, page 91](#)
- [Setting Up the LAN Switch, page 91](#)
- [Connecting Phones, page 92](#)
- [Connecting the TFTP Server, page 92](#)
- [Downloading and Extracting Cisco CME Files, page 92](#)

Installing Hardware and Software

Cisco routers and Cisco Unity Express ship from the factory preinstalled with the hardware and software you ordered. Unless you must upgrade the software, there is no need to do an installation when you unpack your equipment. If you must make changes from what was ordered, or if you're reusing older equipment, power down the router, insert the hardware components, and power the router back up. Refer to Cisco.com for instructions for all types of hardware.

Ensure that you have an IP Voice or greater image running on your router. Install the releases of Cisco CME and Cisco Unity Express that have the feature complement you're interested in. Install at least Cisco CME 3.2 (12.3.11T) or Cisco Unity Express 2.0 to get the features described in this application note.

Configuring Router IP Addressing

Connect to your router's console port, and enter an IP address for the Ethernet interface so that you can connect the router to your network:

```
interface FastEthernet0/0
 ip address a.1.235.1 255.255.0.0
```

Also, set your router's host name to a descriptive string. For Site A, the name is cme-3725:

```
hostname cme-3725
```

Setting the Router Clock

Ensure that the clock is set correctly on your router, for example:

```
Cme-3725# clock set 10:00:00 19 Aug 2004
```

You might also set the router clock using Network Time Protocol (NTP). The details are given in the [“Configuring NTP” section on page 95](#).

Setting Up the LAN Switch

You can use either an external LAN switch or an internal EtherSwitch housed inside the router chassis. Define virtual LAN (VLAN) for both voice and data traffic.

External LAN Switch

For an external LAN switch, to enable separate VLANs for voice and data on a single router port, configure a trunk between the Cisco CME router and the LAN switch. The Cisco Catalyst 3550 and Cisco Catalyst 3560 support autodetection of the VLAN type for IP phones (voice VLAN) and PCs (data VLAN). VLAN configuration varies between LAN switch types. Consult the appropriate documentation for the LAN switch model you are using.

Internal EtherSwitch

You can use several internal EtherSwitch network modules on a Cisco CME router, such as the NM-16ESW-PWR. If you need only one voice VLAN, the simplest configuration is to use the default VLAN 1 for all the IP phones connected to the same EtherSwitch, as shown in the following example. The configuration in this example shows an internal EtherSwitch module, but the actual Site A configuration in the full configurations being built in this application note uses an external LAN switch.

```
ip dhcp excluded-address b.168.1.1
!
ip dhcp pool ipphone
 network b.168.1.0 255.255.255.0
```

```

    option 150 ip b.168.1.1
    default-router b.168.1.251
    dns-server b.70.168.183
!
ip dhcp-server b.168.1.1
!
interface Vlan1
 ip address b.168.1.1 255.255.255.0
 no shutdown
!
interface FastEthernet4/0
 no ip address
!
interface FastEthernet4/1
 no ip address

interface FastEthernet4/2
 no ip address

interface FastEthernet4/3
 no ip address

interface FastEthernet4/4
 no ip address
!output omitted for brevity

```

Connecting Phones

Connect your IP phones to the LAN switch, and verify that they are powered. Confirm that there is connectivity between the router and the LAN switch.

Connecting the TFTP Server

Confirm that your TFTP server can be reached across the network by doing a **ping** from the router to the server's IP address.

Downloading and Extracting Cisco CME Files

Download the Cisco CME GUI files from <http://www.cisco.com/cgi-bin/tablebuild.pl/ip-iostsp>, and extract them into the router Flash memory.

Step 3—Initial Cisco CME System Setup

You have now set up the basic routing and LAN switching. The next task to accomplish is to start setting up the basic Cisco CME configuration. You can accomplish this in several ways:

- Use the Cisco CME Setup Utility
- Copy and paste the configuration, or segments thereof, from the staging configuration (if you used that tool)
- Configure the commands individually on Cisco CME

In the next section, you use the Cisco CME Setup Utility to do the basic Cisco CME configuration. If you are not using this utility, check the resulting CLI from the sections following the description of the Setup Utility, and compare that to the CLI that exists in your configuration to ensure that all the different parameters are set to the desired values.

The following initial Cisco CME setup requirements are summarized in this section:

- [Running the Cisco CME Setup Utility, page 93](#)
- [Configuring Router Parameters for Cisco CME, page 93](#)
- [Configuring Cisco CME GUI Administrators, page 95](#)
- [Adjusting Basic Cisco CME Parameters, page 95](#)

Running the Cisco CME Setup Utility

The following configuration example provides a log of running the Cisco CME Setup Utility to build the Site A basic configuration.

```
Cme-3725(config)# telephony-service setup
--- Cisco IOS Telephony Services Setup ---
Do you want to setup DHCP service for your IP Phones? [yes/no]: yes
Configuring DHCP Pool for Cisco IOS Telephony Services :
  IP network for telephony-service DHCP Pool:a.1.1.0
  Subnet mask for DHCP network :255.255.255.0
  TFTP Server IP address (Option 150) :a.1.1.100
  Default Router for DHCP Pool :a.1.1.100
Do you want to start telephony-service setup? [yes/no]: yes
Configuring Cisco IOS Telephony Services :
  Enter the IP source address for Cisco IOS Telephony Services :a.1.1.100
  Enter the Skinny Port for Cisco IOS Telephony Services : [2000]:
  How many IP phones do you want to configure : [0]: 5
  Do you want dual-line extensions assigned to phones? [yes/no]: yes
  What Language do you want on IP phones [0]: 0
  Which Call Progress tone set do you want on IP phones [0]:
  What is the first extension number you want to configure : 2001
Do you have Direct-Inward-Dial service for all your phones? [yes/no]: no
Do you want to forward calls to a voice message service? [yes/no]: yes
  Enter extension or pilot number of the voice message service: 2105
  Call forward No Answer Timeout : [18]: 10
Do you wish to change any of the above information? [yes/no]: no
---- Setup completed config ----
Cme-3725(config)#
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 1.1, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 1.2, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 2.1, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 2.2, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 3.1, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 3.2, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 4.1, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 4.2, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 5.1, changed state to up
2w3d: %LINK-3-UPDOWN: Interface ephone_dsp DN 5.2, changed state to up
```

Configuring Router Parameters for Cisco CME

You must set a number of options to configure the IP environment to ensure that Cisco CME has access to devices such as TFTP servers and IP phones:

- TFTP for phone firmware download
- Dynamic Host Configuration Protocol (DHCP) to provide automatic IP addresses for the IP phones
- NTP for clock synchronization
- HTTP for the GUI to function correctly

Configuring TFTP

Cisco CME phone loads must be available for TFTP download, as shown in the following example. Note that the **tftp-server** command parameters are case-sensitive; specifying filenames in lowercase does not work. The **tftp-server** commands must be configured manually for phone types, including the Cisco ATA, 7905, 7912, 7935, 7936, 7920, and 7970 IP Phones, even if the Cisco CME Setup Utility is used. The Cisco 7970 IP Phone requires multiple phone load files to be served by **tftp-server** commands, whereas other phones require only one phone load.

```
tftp-server flash:P00303020214.bin
tftp-server flash:P00303020209.bin
tftp-server flash:P00305000200.bin
tftp-server flash:P00305000300.bin
```

Configuring DHCP

Basic DHCP was configured during the Cisco CME Setup Utility. If you chose not to configure DHCP there and now you want to add it, or you want to make changes to the basic DHCP configuration, follow the information in this section.

You must define a DHCP pool for IP address assignment to the IP phones. If you use separate VLANs for voice and data, you need two DHCP pools to assign IP addresses dynamically to the IP phones and the data devices. On the voice DHCP pool, Option 150 will be the LAN interface address. You can also define a DHCP pool to map IP addresses statically to phone MAC addresses. If existing devices use static IP addresses in the same range defined in the DHCP pools, these addresses must be excluded from the DHCP pool to avoid addressing conflicts. You can use the **show ip dhcp binding** command to verify the addresses assigned to the IP phones. The following example shows a sample two-VLAN configuration. Note that you can use the **ip dhcp exclude-address** command to exclude individual or multiple addresses from the DHCP pool.

```
Router# show running-config
ip dhcp excluded-address a.0.0.1
ip dhcp excluded-address b.0.0.1
!
ip dhcp pool data
  network a.0.0.0 255.255.255.0
  default-router a.0.0.1
!
ip dhcp pool voice
  network b.0.0.0 255.255.255.0
  option 150 ip b.0.0.1
  default-router b.0.0.1
!
ip dhcp-server a.0.0.1
ip dhcp-server b.0.0.1
```

```
Router# show ip dhcp binding
Bindings from all pools not associated with VRF:
IP address      Hardware address      Lease expiration      Type
b.0.0.2         0100.3094.xxxx.xx     Mar 02 1993 06:58 AM  Automatic
b.0.0.3         0100.3094.xxxx.xx     Mar 02 1993 06:27 AM  Automatic
```

Configuring NTP

You must configure NTP (even if the Cisco CME Setup Utility is used) to ensure that clocks are synchronized and voice mail time stamps are correct. If you have a Cisco Unity Express module in your system, be sure to reboot it at some point after the NTP configuration is complete so that it can pick up the correct time stamps before you start day-to-day operation on your system.

Configuring HTTP

The Cisco CME router must be set to be an HTTP server (even if the Cisco CME Setup Utility is used) for the Cisco IPC Express GUI to function. Enter the following statements in configure terminal mode on the router:

```
ip http server
ip http path flash
```

Configuring Cisco CME GUI Administrators

If you plan to use Cisco CME GUI access, configure a system administration account and password with the **web admin** command, as shown in the following example. (Replace *admin* and *cisco* with your choices.) Also set the **dn-webedit** and **time-webedit** commands to make sure you can access directory numbers (DNs), and set the router clock from the GUI.

```
telephony-service
web admin system name admin password cisco
dn-webedit
time-webedit
```

If you are a Cisco Partner or Reseller and you plan to customize GUI access for your end customer, enter a customer administrator account, as shown in the following example (replace *custlogin* and *custpswd* in the web admin command with your choices):

```
telephony-service
web admin customer name custlogin password custpswd
web customize load sample.xml
```

You can now point your browser to the following URL to access the Cisco CME GUI, and log in with your configured username and password: <http://ip-address-of-cme/ccme.html>

Adjusting Basic Cisco CME Parameters

The following parameters control system Cisco CME operation:

- Source address and TCP port
- Maximum phone and maximum DN parameters
- Phone loads

The first two parameters were set during the Cisco CME Setup Utility, but you might want to adjust the values.

The source address is the IP address associated with Cisco CME. You typically use the IP address of the interface where the local IP phones are connected. Port 2000 is the default TCP port. Use the following commands to set these values:

```
telephony-service
ip source-address a.1.1.100 port 2000
```

The maximum phone and maximum DN parameters were set to the default values during the Cisco CME Setup Utility. This setting might not have the values you want for your system. Reset these values if needed to numbers that are comfortably more than the actual number of phones and DNs you foresee using, but that are less than or equal to the Cisco CME seat license you purchased. Use the following commands to set these values:

```
telephony-service
max-ephones 20
max-dn 120
```

In addition to the **tftp-server** commands given earlier, you must specify the phone load to use for the IP phones. Only specify the loads that are actually used. Do not use the .bin or .sbn extensions in the **load** command. The create **cnf-files** command generates an XML phone configuration file. Use commands similar to this one to specify phone loads for particular IP phone types:

```
load 7960G-7940G P00305000301
create cnf-files
```

After you've completed the configuration up to this point, IP phones that are powered on and that have received an IP address via DHCP automatically register with Cisco CME and download firmware from the router. It can take up to 5 minutes for the phones to register. All Cisco 7960G IP Phones registered with Cisco CME show "Cisco CME" on the lower portion of the phone display.

At this point, your router and basic Cisco CME system are operational. But you cannot yet make calls between IP phones, because no extensions have been defined or assigned to any buttons on the phones, nor do the phones have dial tone.

Step 4—Configuring Extensions and Phones

On Cisco CME you normally configure DNs (ephone-dns) before configuring the phones (ephones). This is different from Cisco CallManager, where phones must register before you configure extensions. On Cisco CME, you can configure extensions regardless of whether the phones are registered. The following sections walk you through setting up phones and extensions on this application note's sample system.

The following configuration steps are summarized in this section:

- [Defining Extensions, page 97](#)
- [Assigning Extensions to IP Phones, page 97](#)
- [Resetting or Restarting Phones, page 98](#)
- [Making Calls Between IP Phones, page 99](#)

Defining Extensions

The Cisco CME Setup Utility has already created basic DN and phone definitions for the number of phones you specified. For the Site A setup, five phones were specified.

One of the Setup Utility questions that inquires about dual-line phones is, “Do you want dual-line extensions assigned to phones?” For Site A, the reply was yes. A dual-line configuration is needed for features such as call transfer, call waiting, and call conferencing. A dual-line configuration uses one line with two channels so that a second call to the same line can be put in call waiting mode, or a call transfer or conference call can be initiated using the second channel.

You cannot change a nondual-line ephone-dn to dual-line mode. You must insert the ephone-dn with this mode. If you chose dual-line mode in the Setup Utility, you are set. If not, you might be required to delete the ephone-dns created by the utility. (Use the command **no ephone-dn x**, where *x* is the number of the ephone-dns.) Reenter them in dual-line mode using the **ephone-dn x dual-line** command.

The following example shows the first two of the five ephone-dns created by the Setup Utility log. Because the replies in the utility also specified that a voice mail system existed and that the pilot number was 2105, the call forward CLI has already been entered for the ephone-dns.

```
ephone-dn 1 dual-line
 number 2001
 call-forward busy 2105
 call-forward noan 2105 timeout 10
!
ephone-dn 2 dual-line
 number 2002
 call-forward busy 2105
 call-forward noan 2105 timeout 10
```

You can create additional ephone-dns using the CLI, or you can log in to the GUI at this point and add extensions via the **Configuration > Extensions** window.

Many types of DNs and different ephone-dn features exist. The configuration done so far in this section provides only the basic extension (DN) configuration necessary to make calls, but you have not yet configured the phones. The following section shows you how to complete some basic phone configuration.

Assigning Extensions to IP Phones

The Cisco CME Setup Utility has already entered ephone definitions into the configuration. They show up as a single line only in the CLI, as shown here, until the phones are powered, connected, and registered with Cisco CME:

```
ephone 1
ephone 2
```

As soon as the phones are powered and registered (setup steps completed in the [“Connecting Phones” section on page 92](#) and [“Adjusting Basic Cisco CME Parameters” section on page 95](#)), the MAC addresses of the IP phones are known to Cisco CME and are automatically populated into the configuration, as shown in the following example.

```
ephone 1
 mac-address 0003.6BAA.xxxx
 type 7960
!
ephone 2
 mac-address 0003.6BAA.xxxx
 type 7960
```

Next, you must assign extensions (ephone-dns) to the phones by using the **button** CLI command, as shown in the following example, or by going to the **Configuration > Phones** GUI window and configuring the extensions on each phone's buttons. The number following the colon in the button command refers to the ephone-dn (the extension) attached to this button.

```
ephone 1
 mac-address 0003.6BAA.xxxx
 type 7960
 button 1:1
!
ephone 2
 mac-address 0003.6BAA.xxxx
 type 7960
 button 1:2
```

You can set up a Cisco CME system in Key System mode or in private branch exchange (PBX) mode. In Key System mode, you normally configure two or more lines per phone, so buttons 1 and 2 (and more) on each phone are each mapped to an individual extension. In PBX mode, there is usually only one line per phone, so you configure only button 1 with an extension on each phone.

If you did not use the Cisco CME Setup Utility to create initial ephone definitions, you can discover the MAC addresses of the phones connected to Cisco CME by using the **show ephone** command, as shown in the following example, and then use the MAC addresses shown in the output to configure your ephone definitions.

```
router#show ephone
ephone-1 Mac:0008.218C.xxxx TCP socket:[-1] activeLine:0 UNREGISTERED
mediaActive:0 offhook:0 ringing:0 reset:0 reset_sent:0 paging 0 debug:0
IP:0.0.0.0 0 Unknown 0 keepalive 0 max_line 0
```

Resetting or Restarting Phones

Configuration changes on an IP phone or its associated DN's usually require a phone restart or reset to take effect. Note that when a PC is connected to the LAN from the access port on an IP phone, the PC temporarily loses network connectivity while the phone resets. It can take up to 5 minutes for the PC to regain network connectivity.

- The **restart** command performs a softphone reboot without contacting the DHCP and TFTP servers. If you have a PC connected to the IP phone being restarted, this PC doesn't lose network connectivity during the time the phone is being restarted.
- The **reset** command performs a complete phone reboot that includes contacting the DHCP and TFTP servers for the latest configuration information.

You can **reset** or **restart** an individual IP phone or globally **reset all** or **restart all** the IP phones connected to the same Cisco CME system.

The **restart all** or **reset all** commands cause the router to pause for 15 seconds between the resetting of each successive phone. These commands are illustrated in the following examples:

```
cme-3725(config-telephony)#reset all ?
 <0-60> time interval in seconds between each phone reset
cme32-3745(config-telephony)#restart all ?
cme32-3745(config-telephony)#reset sequence-all
```

The **sequence-all** option on the **reset** command causes the router to wait until one phone's reset is complete before resetting the next phone.

The following example shows how to restart or reset phone 1 or globally reset or restart all the phones connected to the Cisco CME system:

```
telephony-service
ephone 1
restart
```

```
telephony-service
ephone 1
reset
```

```
telephony-service
reset all
```

```
telephony-service
restart all
```

Making Calls Between IP Phones

At this point in the configuration, you can make calls from one IP phone to another. The extension mapped to each of the phone buttons appears on the phone's display. Calls can also be transferred and conferenced (because of the dual-line configuration used earlier) between IP phones.

You can use the following **debug** commands for troubleshooting if your system does not allow you to make calls or if your phones have not registered correctly:

- **debug ephone error**
- **debug ephone register**
- **debug ephone detail**
- **debug tftp events**
- **debug dhcp detail**
- **debug ip dhcp server event**

Step 5—Configuring the PSTN Interface

The next step in Cisco CME system configuration is to route PSTN calls into your office to the extensions on the IP phones, and to allow IP phones to make outgoing PSTN calls. The following sections cover voice port and PSTN trunk configurations necessary to route PSTN calls to IP phones.

The following PSTN interface configuration steps are summarized in this section:

- [Configuring Voice Ports, page 99](#)
- [Routing PSTN Calls to IP Phones, page 100](#)

Configuring Voice Ports

The following sections build examples of basic PSTN connectivity for both an analog Foreign Exchange Office (FXO) and a digital T1 Primary Rate Interface (PRI) trunk. It is likely that your office will use one or the other, but not both.

If analog hardware is present in your Cisco CME system, ports show up automatically in the router configuration as follows:

```
voice-port 2/0/0
voice-port 2/0/1
```

To add caller ID to analog FXS or FXO ports, use the **caller-id** command, as shown in the following example:

```
voice-port 2/0/0
  caller-id enable
!
voice-port 2/0/1
  caller-id enable
```

Digital ports do not show up in the configuration simply because the hardware is present in your router. All you see by default is the **controller** statement alerting you that T1 or E1 port hardware is present. On the newer controller cards that allow software configuration for either T1 or E1 operation on the same hardware, the controller doesn't show up in the configuration until you configure the port type to be either T1 or E1 operation. Use the **card type** command to accomplish this.

The following example shows the configuration for a T1 PRI trunk type with the 5ESS switch type. Which switch type you use depends on the central office you connect to and varies between geographic locations, as well as for T1 compared to E1 ports. The voice port, 0/0:23 in the preceding example, is created automatically by the **pri-group** configuration, as is the D channel interface (interface Serial0/0:23 in the preceding example). If you have T1 Channel Associated Signaling (CAS) or E1 R2 connectivity to the PSTN, you use the **ds0-group** command instead of the **pri-group** command.

```
network-clock-participate wic 0
network-clock-participate aim 0
isdn switch-type primary-5ess
!
controller T1 0/0
  pri-group timeslots 1-24
!
interface Serial0/0:23
  no ip address
  isdn switch-type primary-5ess
  isdn incoming-voice voice
  no cdp enable
!
voice-port 0/0:23
```

You can use various hardware cards on the router to provide a digital T1 or E1 connection to the PSTN. The configuration shown in the preceding example, with the exception of the **network-clock-participate** commands, is generic to all T1/E1 trunks and does not vary based on which hardware you are using.

Routing PSTN Calls to IP Phones

FXO analog lines deliver no dial-in digits, so it is necessary to configure an autoterminate destination for these PSTN calls. Direct these calls to the AA, but because you have not set up the AA for Site A yet, the configuration in this section terminates the calls on extension 2001 for the time being. This is sufficient to test that PSTN calls into your Cisco CME system work properly. In the [“Configuring the AA” section on page 109](#), the configuration changes to terminate the PSTN calls onto the AA.

The **connection plar opx** option does not provide answer supervision (connect) to the PSTN if the Cisco CME IP phone does not answer the call. Thus, it does not generate billing until the call is answered. The **connection plar** option, on the other hand, generates answer supervision to the PSTN (and therefore starts billing) the moment the router accepts the call, whether or not the call is answered by an IP phone.

The autoterminate destination for calls on the FXO port is configured under the **voice-port**, as shown in the following example. All calls arriving on voice port 2/0/0 automatically start ringing on extension 2001.

```
voice-port 2/0/0
 connection plar opx 2001
```

PSTN calls arriving on the FXO port can now terminate on extension 2001, but calls arriving on the PRI trunk cannot yet ring any phone. All trunks other than FXO provide dialed digits, so the router can switch the calls based on the digits received from the PSTN. However, the digits delivered by the PSTN do not yet match any Cisco CME extension, so the calls receive overflow tone. Some digit manipulation is required. The current ephone-dn for extension 2001 is as follows:

```
ephone-dn 1 dual-line
 number 2001
```

The PSTN number for dialing this phone is 2xx.5yy.2001, so you need to change the longer PSTN (E.164) number to 2001 so that it can match the ephone-dn configuration. Digit manipulation can be done in various ways. The most straightforward way is simply to configure a secondary number associated with the ephone-dn so that calls to 2001 and calls to 2xx.5yy.2001 terminate on the same phone. This is shown in the following example.

```
ephone-dn 1 dual-line
 number 2001 secondary 2xx5yy2001
```

Calls coming in from the PSTN to extension 2001 now ring the IP phone and can be answered. You can also use dial plan patterns to accomplish this.

Routing IP Phone Calls to the PSTN

Your employees most likely are used to dialing an access code to get a PSTN line. Assuming that this access code is 9, the dial plan entered into the configuration will direct calls dialed with a leading 9 to the PSTN trunks. Because the Site A system currently has both a PRI and analog FXO trunks to the PSTN, you likely want to give the PRI preference, and use the FXO as a backup only if no timeslots are available. You can achieve this by putting preferences on the dial peers.

The dial plan must also take care of local PSTN calls (9 + seven digits) and long-distance calls (9 + 11 digits). You might also want to add more dial peers to allow (or disallow) international PSTN dialing. The dial plan (and dial peers supporting it) can become very sophisticated. The following example shows just the basic plain old telephone service (POTS) dial peers necessary on the voice ports to route seven-digit and 11-digit PSTN calls to the PRI trunk first and to the FXO trunk second. The **forward-digits** command instructs the router to deliver a certain number of digits to the PSTN. In this example, it suppresses the “9” access code and forwards the rest of the digits the IP phone user dialed. You can add any number of digits to or delete any number of digits from the original string, or send completely different digits to the PSTN from what was dialed by the IP phone.

```
dial-peer voice 1000 pots
 destination-pattern 91.....
 port 0/0:23
 forward-digits 11
!
dial-peer voice 1001 pots
 preference 1
```

```

destination-pattern 91.....
port 2/0/0
!
dial-peer voice 1002 pots
destination-pattern 9[2-9].....
port 0/0:23
forward-digits 7
!
dial-peer voice 1003 pots
preference 1
destination-pattern 9[2-9].....
port 2/0/0

```

As soon as you enter this configuration, IP phones can call 914xx5yy1212 (11-digit) or 95yy1212 (seven-digit) and have these calls routed to the PSTN.

Step 6—Configuring Cisco Unity Express AA and Voice Mail

If your system does not have Cisco Unity Express installed, you can skip this section and proceed to the [“Step 7: Configuring Cisco CME Call Processing Features”](#) section on page 113. The following sections step you through setting up the basic Cisco Unity Express configuration necessary to use the AA and voice mail on your system.

The following Cisco Unity Express AA and voice-mail configuration steps are summarized in this section:

- [Routing IP Phone Calls to the PSTN, page 101](#)
- [Configuring Basic Cisco Unity Express, page 104](#)
- [Configuring Voice Mail, page 106](#)
- [Configuring the AA, page 109](#)

Setting Up the Router for Cisco Unity Express

Before you can configure and use Cisco Unity Express applications such as AA and voice mail, you must set up the following basic IP connectivity parameters to ensure that the Cisco Unity Express software can communicate with its environment:

- [IP Addressing, page 102](#)
- [Call Routing to Cisco Unity Express, page 103](#)
- [H.323-to-SIP Call Routing, page 103](#)
- [Message Waiting Indicator, page 103](#)

IP Addressing

The Cisco Unity Express hardware module must be configured with an IP address on the router. The configuration for the Cisco Unity Express module for Site A is shown in the following example:

```

interface Service-Engine1/0
ip unnumbered FastEthernet0/0
service-module ip address a.1.235.128 255.255.0.0
service-module ip default-gateway a.1.235.1
ip route a.1.235.128 255.255.255.255 Service-Engine1/0

```

You can show the status of the Cisco Unity Express module by using the command shown in the following example. You can also **ping** the IP interface (a.1.229.128 in this example) to ensure that IP connectivity is established.

```
Cme-3725# service-module service-engine 1/0 status
Service Module is Cisco Service-Engine1/0
Service Module supports session via TTY line 33
Service Module is in Steady state
Getting status from the Service Module, please wait..
cisco service engine 1.1
```

Call Routing to Cisco Unity Express

Cisco CME and Cisco Unity Express communicate using a SIP interface. Therefore, Cisco CME uses SIP dial peers to determine which calls must be routed to Cisco Unity Express. The AA pilot number for Site A is 2100, and the voice mail pilot is 2105. The following example shows the SIP dial peers necessary to route calls to these pilot numbers from Cisco CME to Cisco Unity Express.

```
dial-peer voice 2100 voip
destination-pattern 21..
session protocol sipv2
session target ipv4:a.1.235.128
dtmf-relay sip-notify
codec g711ulaw
no vad
```

H.323-to-SIP Call Routing

If you have multiple sites to connect to each other, such as Site B in the sample topology used in this application note, calls between sites use H.323, and calls to Cisco Unity Express use SIP. This requires an H.323-to-SIP translation on the Cisco CME router. The following CLI enables this function (available with Cisco CME 3.2 and later):

```
voice service voip
allow-connections h323 to sip
```

Message Waiting Indicator

The application note entitled “[Cisco IPC Express Integrated Voice Mail](#)” explains the mechanism for turning MWI on and off from Cisco Unity Express voice mail to Cisco CME phones. The MWI definitions shown in The following example illustrates commands necessary to enable MWI for Site A.

```
ephone-dn 51
number 8000....
mwi on
!
ephone-dn 52
number 8001....
mwi off
```

You are now ready to point a browser to Cisco Unity Express (<http://a.1.235.128/> for Site A) to run through the Cisco Unity Express Initialization Wizard. This is covered in the next section.

Configuring Basic Cisco Unity Express

The Cisco Unity Express Initialization Wizard allows you to import the configuration (primarily phones and extensions) already done on your Cisco CME system into Cisco Unity Express, create mailboxes for the users, and define the pilot numbers for the AA and voice mail.

Importing Users from Cisco CME

Figure 34 shows how the six Site B users already defined on the router are imported into Cisco Unity Express in the first Initialization Wizard window.

Check the boxes in the Mailbox column for all six users to automatically create personal mailboxes for all six users at the end of the Initialization Wizard.

Figure 34 Importing Users



Setting System Defaults

In the System Defaults window, you set attributes such as the PIN and password generation policy for new accounts. Figure 35 shows the settings and selections made for the Site B configuration.

Figure 35 **Setting System Defaults**

The screenshot displays the 'Cisco Unity Express Initialization Wizard' interface. On the left, a 'Steps' sidebar lists: 1 CallManager Express Login, 2 Import CME Users, 3 Defaults (selected), 4 Call Handling, and 5 Commit. The main area is titled 'Defaults' and contains the following configuration options:

- System Default Language:** A dropdown menu set to 'English (United States)'.
- Password & PIN options:** Radio buttons for 'Generate random password' (selected) and 'Blank password', and 'Generate random PIN' (selected) and 'Blank PIN'.
- Mailbox Defaults:**
 - Mailbox Size *:** A text input field containing '5520' followed by 'seconds'.
 - Maximum Caller Message Size *:** A text input field containing '60' followed by 'seconds'.
 - Message Expiry Time *:** A text input field containing '60' followed by 'days'.

A note at the bottom states: '* indicates a mandatory field'. At the bottom right, there are buttons for 'Back', 'Next', 'Finish', 'Cancel', and 'Help'. A vertical ID '146211' is visible on the right edge of the screenshot.

Setting Call Handling Parameters

The pilot numbers for the AA (3101), voice mail (3105), and the administration TUI (3106) are set in the Call Handling window, shown in [Figure 36](#). Also, the MWI DNs imported from the Cisco CME definitions entered earlier (see example configuration in [“Message Waiting Indicator”](#) section on [page 103](#)) are imported into this window.

Figure 36 Setting Call Handling Parameters

Cisco CallManager Express
> Powered by Cisco IOS®

CISCO SYSTEMS

Cisco Unity Express Initialization Wizard

Steps

- 1 Call Manager Express Login
- 2 Import CCME Users
- 3 Defaults
- 4 Call Handling
- 5 Commit

Call Handling

Enter the Call in Numbers for Voice Mail, Auto Attendant and the Greeting Management System.

Voice Mail Number *: 3105

Voice Mail Operator Extension *: 0

Auto Attendant Access Number: 3101

Auto Attendant Operator Extension: 0

Greeting Management Number: 3106

MWI on Number : 8000....

MWI off Number : 8001....

* indicates a mandatory field

Back Next Finish Cancel Help

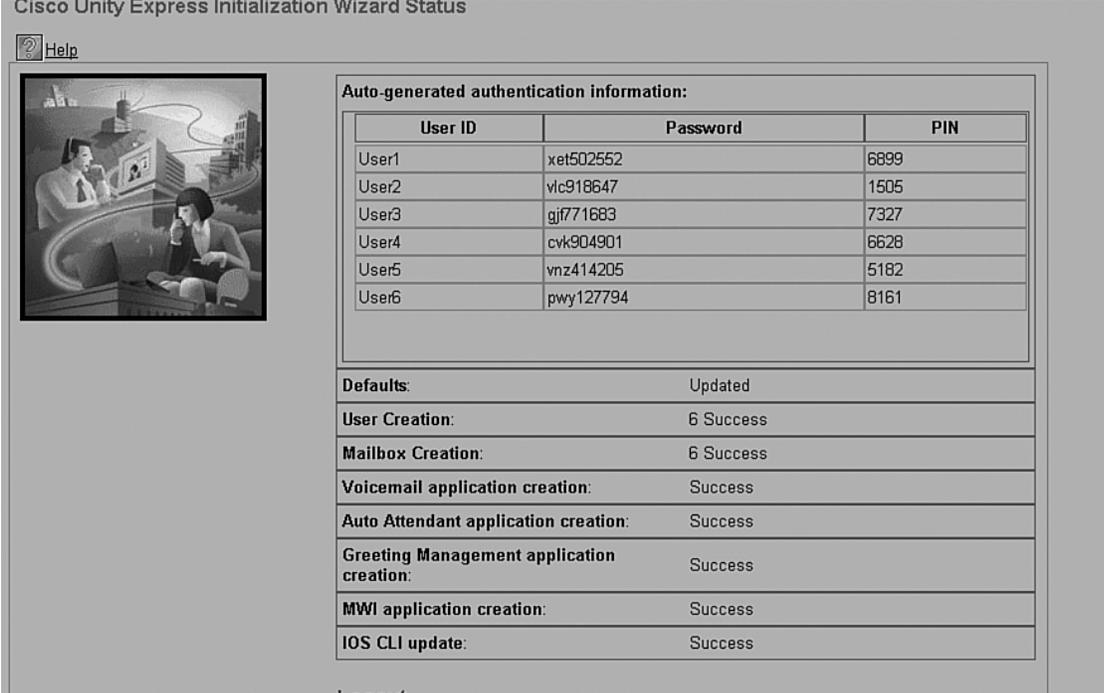
146212

At this point in the system setup, you can call into the system AA and voice mail.

Configuring Voice Mail

Figure 37 shows the final window of the Initialization Wizard. It summarizes the auto-generated system passwords and PINs for all the user profiles and mailboxes created by the Initialization Wizard. This is a handy window to preserve as a windowshot or printout to help you let each user know his default password and PIN for first-time login to his mailbox.

Figure 37 Cisco Unity Express User Passwords and PINs



Cisco Unity Express Initialization Wizard Status

Help

Auto-generated authentication information:

User ID	Password	PIN
User1	xet502552	6899
User2	vlc918647	1505
User3	gjf771683	7327
User4	cvk904901	6628
User5	vnz414205	5182
User6	pwyl27794	8161

Defaults:	Updated
User Creation:	6 Success
Mailbox Creation:	6 Success
Voicemail application creation:	Success
Auto Attendant application creation:	Success
Greeting Management application creation:	Success
MWI application creation:	Success
IOS CLI update:	Success

Logout

146213

All users have mailboxes with the tutorial set to yes. When you notify users of their default system-assigned PINs, they can log in to their mailboxes and work through the setup tutorial. The tutorial helps them record a spoken name and an outgoing greeting and forces them to change their PIN to a private setting not known to you as the system administrator.

Calls to your employees' phones automatically forward into voice mail, where the caller hears the standard system greetings until your employees have logged in to customize their mailbox greetings. Call forwarding was set up in the “[Step 3—Initial Cisco CME System Setup](#)” section on page 92, earlier in this application note, by the Cisco CME Setup Utility. The following example shows the call forwarding setup for ephone-dns. If you want to change this call forward destination, go to the **Configure > Extensions** GUI window or use the **ephone-dn** command in the CLI.

```
ephone-dn 1 dual-line
number 2001 secondary 2xx5yy2001
call-forward busy 2105
call-forward noan 2105 timeout 10
```

In the current configuration, PSTN calls cannot get into voice mail. Instead, callers hear, “Sorry, there is no mailbox associated with this extension,” even though internal calls from other IP phones get into voice mail correctly. One way to address this is to add the direct inward dial (DID) number associated with the extension (and therefore the mailbox) to the Primary E.164 Number field in the **Configure > Users** window, as shown in [Figure 38](#). Now PSTN calls also work into voice mail. Alternatively, you can use Cisco IOS translation rules or Cisco CME dial plan patterns to translate the DID numbers to extensions before the calls enter voice mail.

Figure 38 Configuring the DID Number for a Mailbox

The screenshot shows the 'User Profile - User5 U5' configuration window in Microsoft Internet Explorer. The 'Mailboxes' tab is active. The configuration fields are as follows:

- User ID: user5
- First Name *: User5
- Last Name *: U5
- Nick Name *: User5 U5
- Display Name*: User5 U5
- Primary E.164 Number: xxxyyy2001
- Associated Phone: 1111.2222.3344 (with Add/Edit and Remove buttons)
- Primary Extension: None (selected), Other: 2001
- Language: System Default
- Password options: Password specified below
- Password: [masked]
- Confirm Password: [masked]
- PIN options: PIN specified below
- PIN: [masked]
- Confirm PIN: [masked]

* indicates a mandatory field

If you did not add mailboxes in the Initialization Wizard, go to the **Configuration > Users** GUI window to add user definitions and mailboxes for the employees on your system.

At this point, basic voice mail is set up and working. The following example summarizes the mailboxes defined on the system and the time used in each.

Cue-3725# **show voicemail mailboxes**

OWNER	MSGS	NEW	SAVED	MSGTIME	MBXSIZE	USED
"xxxuser1"	1	0	1	13	5520	1 %
"xxxuser2"	1	0	1	9	5520	1 %
"xxxuser4"	0	0	0	5	5520	1 %
"xxxuser5"	0	0	0	5	5520	1 %
"xxxuser3"	0	0	0	5	5520	1 %

Configuring the AA

Cisco Unity Express ships with a system AA that is very easy to set up. It offers callers dial-by-extension, dial-by-name, and transfer-to-the-operator choices. You can also install a fully customized AA into Cisco Unity Express so that you can tailor the AA menus and choices to your own business needs.

Setting Up the System AA

If your office intends to use only the Cisco Unity Express system AA, follow the steps in this section; otherwise, proceed to the next section.

The system AA is set up by default and is working already with the AA pilot number you assigned during the Cisco Unity Express Initialization Wizard. Dial-by-extension from the system AA works without any further setup, but dial-by-name requires that you configure the names of your employees in their user profiles. In the **Configure > Users** GUI window, click each user to see his or her profile, and fill in the First Name and Last Name fields with the strings you want to be used for matching in the AA dial-by-name feature.

The last remaining field to set up in the system AA is to customize your company's AA welcome greeting. Record this greeting offline on a PC, and upload the .wav file to Cisco Unity Express. You also can record the greeting by using the administrative TUI (extension 2106 for Site A or 3106 for Site B) on Cisco Unity Express. Associate the .wav file with the system AA.

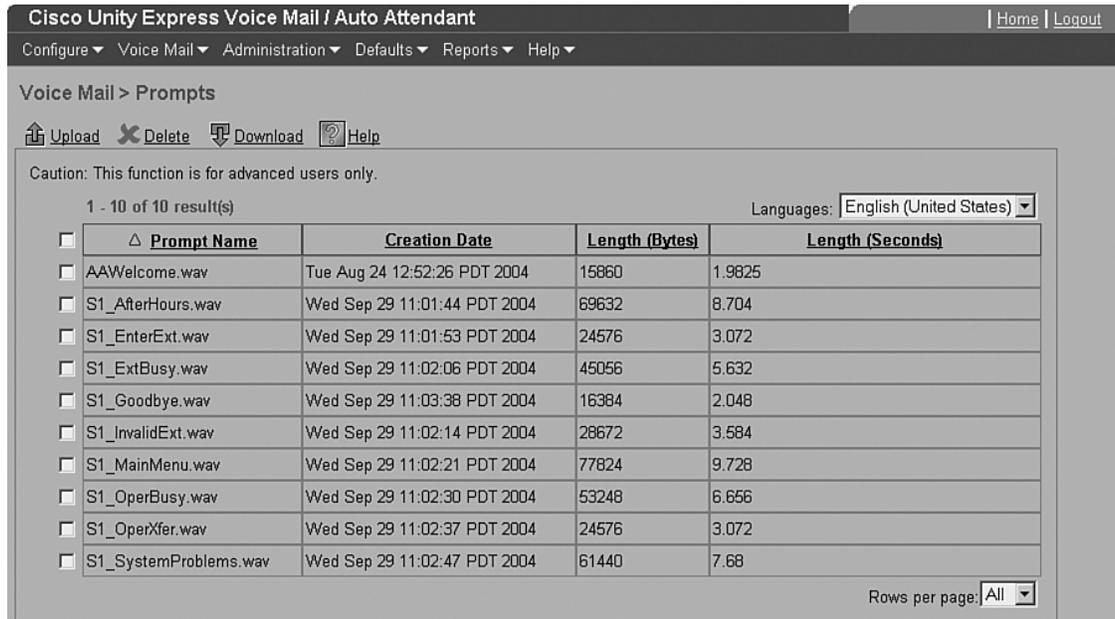
The sample system setup in this application note uses Cisco Unity Express 2.0, and you have now configured all the system AA parameters for this release. If you are using Cisco Unity Express 2.1 (or later), a few more parameters need to be customized.

Setting Up a Custom AA

If your business requires a more sophisticated AA than the Cisco Unity Express system AA, use the guidelines in this section to set up your custom AA. The application note entitled "[Cisco IPC Express Automated Attendant Options](#)" explains how to customize an AA script on Cisco Unity Express. Assume at Site B that you require only the system AA, whereas a custom AA is necessary at Site A.

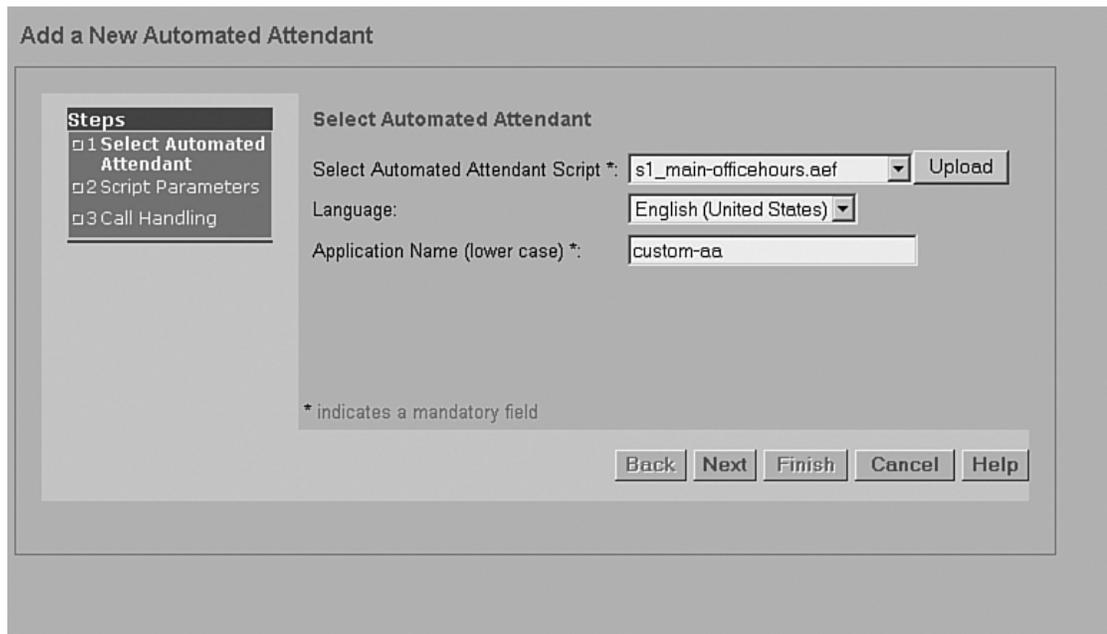
For example, the script written for Site A is named S1_Main-OfficeHours.aef, and it calls two subflows: S1_DialbyExtension.aef and S1_XfertoOper.aef. You can download these scripts and the prompts they use from Cisco.com by going to the Software Center for Cisco Unity Express. The S1_Main-OfficeHours script includes nine different prompts that you record on a PC or in a studio as a .wav file. Upload all three scripts (.aef files) and nine prompts (.wav files) from your PC to Cisco Unity Express, as shown in [Figure 39](#).

Figure 39 Uploading Scripts and Prompts to Cisco Unity Express



As soon as the scripts and prompt files are available on the Cisco Unity Express system, you must add a custom AA to the system. Start this activity by going to the **Voice mail > Auto Attendant** GUI window, and choose **Add**. Select the S1_Main-OfficeHours.aef script for the AA, as shown in Figure 40. Insert a name for the AA (the example uses **custom-aa**).

Figure 40 Selecting a Script for the Custom AA



Assign a pilot number for the AA, as shown in Figure 41. Because your real AA pilot number, extension 2100, is already assigned to the system AA, you cannot choose that. For the time being, choose 2101.

Figure 41 Choosing a Pilot Number for the Custom AA

The screenshot shows a configuration window titled "Add a New Automated Attendant". On the left, a "Steps" pane lists three steps: "1 Select Automated Attendant", "2 Script Parameters", and "3 Call Handling", with "3 Call Handling" selected. The main area is titled "Call Handling" and contains the following fields and options:

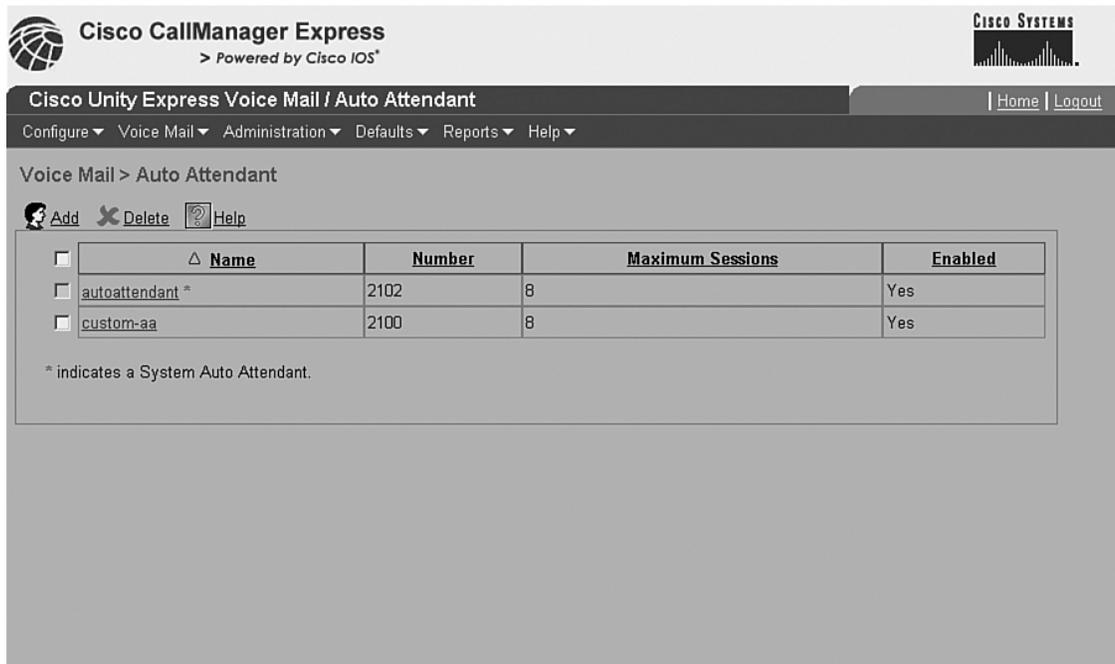
- Call-in Number:** A text input field containing the value "2101".
- Maximum Sessions *:** A text input field containing the value "8".
- Enabled:** Radio buttons for "Yes" (which is selected) and "No".

Below these fields, a note states: "* indicates a mandatory field". At the bottom of the window, there are five buttons: "Back", "Next", "Finish", "Cancel", and "Help".

After you add the AA, you see custom-aa show up in the **Voice mail > Auto Attendant** GUI window. To switch around the pilot numbers, assuming that 2100 is the actual AA pilot number your business wants to use, click the system AA (**autoattendant**), and change its pilot number to a different extension (for example, 2102). Click **custom-aa** and change its pilot number to 2100. The resulting configuration is shown in [Figure 42](#). Your custom AA is now operational.

At this point, internal calls from the IP phones to the AA (extension 2100) work, but PSTN DID calls to the AA don't work. PSTN calls arrive at 2xx.5yy.2001, and this DID number must be mapped to the AA. Insert the Cisco IOS translation rule (or its equivalent), shown in the following example, into the Cisco CME router, and attach it to the dial peer in Cisco Unity Express that matches PSTN trunk calls. This same translation rule includes translations to allow calls into your voice mail (2105) and administrative TUI (2106) pilot numbers from the PSTN.

Figure 42 Correcting the Pilot Number for the Custom AA



The screenshot shows the Cisco CallManager Express web interface for configuring Auto Attendants. The page title is "Cisco Unity Express Voice Mail / Auto Attendant". Below the navigation menu, there are buttons for "Add", "Delete", and "Help". A table displays the configuration for two auto attendants:

<input type="checkbox"/>	△ Name	Number	Maximum Sessions	Enabled
<input type="checkbox"/>	autoattendant *	2102	8	Yes
<input type="checkbox"/>	custom-aa	2100	8	Yes

* indicates a System Auto Attendant.

```

voice translation-rule 10
 rule 1 /2xx5yy2100/ /2100/
 rule 2 /2xx5yy2105/ /2105/
 rule 3 /2xx5yy2106/ /2106/
!
voice translation-profile to_cue
 translate called 10
!
dial-peer voice 2100 voip
 destination-pattern 21..
 session protocol sipv2
 session target ipv4:a.1.235.128
 dtmf-relay sip-notify
 codec g711ulaw
 no vad
!
dial-peer voice 2101 voip
 description VM-AA-PSTN
 translation-profile outgoing to_cue
 destination-pattern 2xx5yy21..
 session protocol sipv2
 session target ipv4:a.1.235.128
 dtmf-relay sip-notify
 codec g711ulaw
 no vad

```

In the “Routing PSTN Calls to IP Phones” section on page 100, the FXO PSTN trunk was set up to ring employee extension 2001, because the AA was not yet implemented. Now that the AA is fully configured, the routing of the FXO calls will be changed to the AA (extension 2100) instead of IP phone 2001:

```

voice-port 2/0/1
 connection plar opx 2100

```

Step 7: Configuring Cisco CME Call Processing Features

Your Cisco IPC Express system is fully operational after the completion of Step 6 for all basic features. However, you probably want to configure numerous additional Cisco CME call processing features to better tailor the system to your business needs. The sections that follow describe configuring the following:

- [Configuring Phone and User Features, page 113](#)
- [Configuring System Features, page 120](#)
- [Configuring Conference Call, Call Transfer, and Call Forward, page 123](#)
- [Enabling Applications, page 125](#)

Configuring Phone and User Features

This sections summarize the following configuration activities:

- [Caller ID Name Display, page 113](#)
- [Phone Name Display, page 114](#)
- [Phone Button Label Customization, page 114](#)
- [Shared Lines, page 114](#)
- [Hunt Group, page 115](#)
- [Local Directory, page 116](#)
- [Speed Dial, page 116](#)
- [Local Speed Dial, page 117](#)
- [Personal Speed Dial, page 117](#)
- [Localization, page 118](#)
- [Autoline Selection, page 118](#)
- [IP Phone Softkey Customization, page 118](#)
- [Direct FXO Trunk Line Select, page 119](#)
- [Overlay DN, page 119](#)

Caller ID Name Display

Caller ID is the name associated with the calling party's extension when a call is ringing on an IP phone. To customize this field, set the Name field on the ephone-dn, as shown in the following example. The calling name appears only on IP phone-to-IP phone calls.

```
ephone-dn 1 dual-line
  number 2001 secondary 2xx5yy2001
  name xxxxx1
!
ephone-dn 2 dual-line
  number 2002 secondary 2xx5yy2002
  name xxxxx2
```

Phone Name Display

The top line of an idle IP phone display can show the name of the extension or the phone's owner. To show a name in this field, set the Description field on the ephone-dn, as shown in the following example. Note that top-line display customization is supported only on Cisco 7960 and Cisco 7940 IP Phones.

```
ephone-dn 1 dual-line
number 2001 secondary 2xx5yy2001
description xxxx1 name1
```

Phone Button Label Customization

The IP phone displays the extension configured on each button next to the button. For example, on xxxx1's phone, 2001 is displayed by default next to button 1. To customize this display to a different string, set the Label field on the ephone-dn, as shown in the following example:

```
ephone-dn 1 dual-line
number 2001 secondary 2xx5yy2001
label xxxx1
```

Shared Lines

A shared line is an extension that appears on multiple phones. There are different ways of defining shared lines, depending on the operation you want.

To define a shared line that rings on several phones simultaneously, define a single ephone-dn, and place it as a button appearance on multiple phones.

Another way to define a shared line is to have the extensions ring in succession. For example, assume that two employees, User xxxx4 (2004) and xxxx5 (2005), work in your office as receptionists. Reception is extension 2060. User xxxx4 is the main receptionist, and xxxx5 fills in for xxxx4 while she is not at her desk.

The following example shows how to configure this. Two ephone-dns are defined with the same extension number (2060). The first has no **huntstop** configured, and the second has **preference 1**. A call to extension 2060 is first routed to ephone-dn 6, which appears on button 1 of xxxx4's phone. If xxxx4 (ephone-dn 6) is busy, the call is routed to xxxx5's phone (ephone-dn 7 on button 1 of ephone 5). If xxxx5 (ephone-dn 7) is also busy, the calling party hears busy tone. Note that preference 0 is the default configuration on an ephone-dn and does not show in the CLI.

```
ephone-dn 6
number 2060
label Reception
name Reception
no huntstop
!
ephone-dn 7
number 2060
label Reception
name Reception
preference 1
!
ephone 4
mac-address 0003.AAAA.0004
button 1:6
!
```

```
ephone 5
  mac-address 0003.AAAA.0005
  button 1:7
```

Hunt Group

The hunt group feature redirects incoming calls to a hunt pilot number on busy or no answer from one ephone-dn to another based on a list defined in the ephone-hunt group. The calling party's number is displayed on IP phones with Cisco CME 3.2 or later. There are three types of hunt groups:

1. *Sequential*—The ephone-dns ring from first to last in the list specified. If the call is not answered, the call is redirected to the final destination configured.
2. *Peer*—In this configuration, the list of ephone-dns in the hunt group are rung in a round-robin manner. The next ephone-dn to ring is the number in the list to the right (as you read the CLI configuration from left to right) of the last ephone-dn that rang when the hunt pilot number was last called. Ringing proceeds in a circular manner through the list, first to last, for the number of hops specified in the hunt group. If the call is not answered in the specified number of hops, it is redirected to the final destination configured.
3. *Longest-idle*—A new incoming call is directed to the ephone-dn that has been idle for the longest time. The longest-idle time is determined from the last time a phone registered, reregistered, or went on-hook.

Site B has two employees (User2 at extension 3002 and User3 at 3003) who take customer service calls. The number called internally for customer service is 3050 and 4xx.5yy.3050 from the PSTN. To distribute calls equitably across the two employees, a longest-idle hunt group is defined as shown in the following example:

```
ephone-hunt 1 longest-idle
  pilot 3050 secondary 4xx5yy3050
  list 3002, 3003
  final 3105
```

The final destination is defined as the voice mail pilot number (3105) so that a caller can leave a message if neither of the customer service employees is available. User2 and User3 are members of a customer service group defined on Cisco Unity Express. A general delivery mailbox (GDM) is associated with extension 3050, as shown in [Figure 43](#).

Figure 43 GDM for Customer Service

The screenshot shows a configuration window titled "Group Profile - customer-service". At the top, there are buttons for "Apply", "Cancel", and "Help". Below these are four tabs: "Profile", "Owners/Members", "Owner/Member of Groups", and "Mailboxes". The "Profile" tab is selected. The fields are as follows:

- Group ID: customer-service
- Full name *: customer-service
- Description: customer-service group
- Primary Extension: 3050
- Primary E.164 Number: 4445553050

Under the "Capabilities" section, there are two checkboxes:

- Super Users:
- Greeting Manager:

A vertical number "146219" is visible on the right side of the window.

Local Directory

A local directory of names and phone numbers is automatically built using the Name and Number fields under the ephone-dn configuration, as shown in the following example. You can configure the local directory to show directory information in first name, last name format or last name, first name format. Phones must be reset for directory changes to take effect.



Note

The directory command does not automatically reorder names entered under the ephone-dn. If you change the directory format, you must manually change all the names under the ephone-dns to match the directory format.

```
telephony-service
  directory last-name-first
!
ephone-dn 1 dual-line
  number 2001 secondary 2xx5yy2001
  name xxxx1 name1
!
ephone-dn 2 dual-line
  number 2002 secondary 2xx5yy2002
  name xxxx2 name2
```

Speed Dial

You can define a number of speed-dial buttons for each IP phone, limited by the number of buttons supported on the phone. This configuration is shown in the following example. If more speed-dial buttons are configured than the number of available buttons on the phone, the extra speed-dial configurations are ignored. If speed dials are mapped to buttons with DN's assigned, speed-dial

configurations are ignored. Speed dials cannot appear between buttons with DNs assigned. For example, on a Cisco 7960 IP Phone with six buttons, if button 1 and button 6 have an ephone-dn defined, the entire speed-dial configuration is ignored, even if buttons 2 to 5 are unassigned.

You can configure up to four speed-dial buttons on a Cisco 7960G IP Phone. The following example shows speed dials on buttons 2 and 3.

```
ephone 1
 mac-address 0003.6BAA.D1F8
 speed-dial 1 2005 label "xxxx5"
 speed-dial 2 3050 label "Customer Svc"
 type 7960
 button 1:1
```

Local Speed Dial

To work around the number of speed-dial entries limited by the number of buttons per phone, you can create a speeddial.xml file, as shown in the following example to list all the speed-dial entries. The local speed-dial Cisco CME feature provides a system-wide list of frequently called numbers (up to 32) accessible from the directory button on the phone.

```
<CiscoIPPhoneDirectory>
<Title>Speed Dials</Title>
<Prompt>Record 1 to 2 of 2 </Prompt>
  <DirectoryEntry>
    <Name>Directory Assistance</Name>
    <Telephone>95yy1212</Telephone>
  </DirectoryEntry>
  <DirectoryEntry>
    <Name>Your friends at Cisco</Name>
    <Telephone>4085264000</Telephone>
  </DirectoryEntry>
</CiscoIPPhoneDirectory>
```

After you have created the XML file, place the speeddial.xml file in the Cisco CME router's Flash memory by using the **copy tftp flash** command:

```
Router(config)#copy tftp://ip address/speeddial.xml flash:speeddial.xml
```

The local speed-dial menu appears when you select **Local Speed Dial** from the phone's **Directory** menu.

Personal Speed Dial

The Personal Speed Dial feature adds personal speed-dial entries, on a per-phone basis, to the system-wide speed-dial directory (described in the preceding section).

You create personal speed-dial entries by using the **fastdial** command under an ephone definition, as shown in the following example. Phone users access personal speed-dial numbers through the **Directories > Local Services > Personal Speed Dial** menu. Personal speed-dial numbers appear in the order in which they are entered into the configuration.



Note

The fastdial command is supported only on the Cisco 7940 and Cisco 7960 IP Phones.

```
ephone 2
 username "xxxuser2"
 mac-address 0003.6BAA.D362
 fastdial 1 3001 name User1
```

```

fastdial 2 3002 name User2
fastdial 3 2001 name xxxx1
fastdial 4 2005 name xxxx5
speed-dial 1 2004 label "xxxx4"
type 7960
button 1:2

```

Localization

Cisco CME supports country-specific language displays and call progress tones for IP phones. All phones on a Cisco CME system must use the same language settings. You can't configure multiple languages on a single CME router. The default language is set to English/U.S. The following example shows how to configure a system for Italian. (Both Sites A and B in the sample configuration are configured for U.S. English.)

```

telephony-service
user-locale IT
network-locale IT
reset all

```

Autoline Selection

Up to Cisco CME 3.1, the first available line is automatically selected to make or answer a call on an IP phone. Since release 3.1, Cisco CME supports four different modes for selecting the line to answer on an incoming call or to select an extension (by pressing a button) to make an outgoing call:

- Autoline
- No autoline
- Autoline incoming
- Autoline button number

You can find details on these modes in the Cisco CME Command Reference on Cisco.com.

For most of the phones at Site A and Site B, the default (autoline) is sufficient. The exception is the two receptionists, where you want incoming calls to select a line automatically, but you want outgoing calls to explicitly press a button (choosing between their own extension or the receptionist's extension) before making the call. The following example shows this configuration for xxxx4's phone.

```

ephone 4
mac-address 0003.AAAA.0004
auto-line incoming
button 1:6 2:4

```

IP Phone Softkey Customization

You can change or disable the display order of softkeys on the Cisco 7960, Cisco 7940, Cisco 7905, and Cisco 7912 IP Phones using the **ephone-template** command. The template can specify softkey order settings for phone states including alerting, connected, idle, and seized, as shown in the following example. You must restart an IP phone to make the changes in the template take effect. You can define up to five templates.

```

ephone-template 1
softkeys idle Redial Pickup Dnd Login Gpickup
softkeys seized Pickup Redial Endcall Gpickup

```

```
!
ephone 1
  ephone-template 1
```

Direct FXO Trunk Line Select

Cisco CME can automatically seize an FXO trunk line when you press a line button or lift the handset. This allows you to hear stutter dial tone from the central office (CO) when you lift the handset if you get voice mail service from your PSTN provider. The **destination-pattern** configured in the trunk's dial peer must match the trunk code defined on the ephone-dn.

User xxxx2, at extension 2002, is the manager at your Site A office, and you want him to have direct access to a CO line for emergency 911 access. Button 2 on his phone will be labeled "911." When pressed, this button must select dial tone directly from the CO for outside dialing. The configuration shown in the following example achieves this by making an automatic connection between button 2 on user xxxx2's phone and the second FXO line (port 2/0/1) on the Cisco CME system. When user xxxx2 goes off-hook on button 2, his phone gets dial tone from the CO. The following things happen behind the scenes:

- User xxxx2 goes off-hook on button 2, which automatically activates ephone-dn 10.
- Ephone-dn 10 has a **trunk** statement that automatically originates a call to the 911 trunk access code.
- The 911 access code dialed matches POTS dial peer 1004, which connects the outgoing call to FXO port 2/0/1.
- FXO port 2/0/1 goes off-hook and draws dial tone from the CO.

```
voice-port 2/0/1
!
dial-peer voice 1004 pots
  destination-pattern 911
  port 2/0/1
  forward-digits all
!
ephone-dn 10
  number 2999
  label 911
  trunk 911 timeout 5
!
ephone 2
  type 7960
  button 1:2 2:10
```

Overlay DN

To overcome the button limitation on certain phone types and to support more extensions on a phone, you can use overlay DN's to associate multiple ephone-dn's to a single button on a phone. That way the phone can receive calls to multiple extensions on the same button.

As the manager of Site A, user xxxx2 has a special extension (2020) where you can reach him directly, but other employees in the office do not know this number. Because he seldom receives calls on this number, user xxxx2 does not want to dedicate a button on this phone just for this number. Instead, he wants these calls to come in on button 1, where his regular calls to extension 2002 also ring. The following example shows this configuration using overlay DN's.

```
ephone-dn 30
```

```

number 2020
!
ephone 2
  button 102,30 2:10 3:21

```

Configuring System Features

This section summarizes configuration of the following general system features:

- [Music on Hold, page 120](#)
- [On-Hold Call Notification, page 120](#)
- [Interdigit Timeout, page 121](#)
- [Intercom, page 121](#)
- [Paging, page 122](#)
- [Ringing Timeout, page 122](#)
- [Call Park, page 122](#)
- [Time and Date Format, page 123](#)

Music on Hold

The music on hold (MOH) feature supports the .au and .wav file formats. MOH applies only to G.711 intersite VoIP calls and PSTN calls. All other calls, including local calls between Cisco CME phones on the same system, hear tone on hold.

A sample MOH file, music-on-hold.au, is included in Cisco CME as a .zip or .tar file. It also can be downloaded from the Cisco.com Software Center. You normally download the MOH file to a TFTP server in the network, and then copy the file into Flash from the Cisco CME system CLI by using the following:

```
copy tftp://ip address/music-on-hold.au flash:
```

where ip address is the TFTP server's address. Be sure to enter **n** when prompted to erase Flash. The following example shows the content of the Flash and the configuration necessary to enable MOH.

```

Cme-3725# show flash
-#- --length-- -----date/time----- path
...
11      496521 Apr 02 2002 11:27:06 -08:00 music-on-hold.au

Cme-3725# show running-config
telephony-service
  moh music-on-hold.au

```

On-Hold Call Notification

The on-hold call notification feature sends an audible notification on Cisco CME phones to alert the user that a call is on hold. This feature is typically used when multiple lines are configured and the user might forget that a call on a secondary line is still on hold. Three types of on-hold notifications can be configured:

- Idle

- Originator
- Shared

The following example shows a configuration of an on-hold call notification on user xxxx3's phone (extension 2003).

```
ephone-dn 3 dual-line
number 2003 secondary 2xx5yy2003
hold-alert 15 originator
```

Interdigit Timeout

The interdigit timeout specifies how many seconds the system waits between the initial and subsequent digit presses on the phone keypad when a caller dials a call. This is shown in the following example. If the timeout expires before the destination is identified, a fast-busy is given, and the call ends.

```
telephony-service
timeouts interdigit 20
```

The initial timeout, which specifies how many seconds Cisco CME waits before the caller enters the initial digit, cannot be configured and is set to 10 seconds. Phones must be reset to make the configuration take effect.

Intercom

The intercom feature allows one-way, autoanswer voice connections. Specially configured speed-dial buttons allow a call to be placed to the selected extension. On the destination phone, the call is automatically answered in speakerphone mode with mute enabled. To respond to the intercom call and open two-way voice, the recipient can deactivate the mute button (or, in the case of a Cisco 7910 IP Phone, lift the handset). Intercom lines cannot be used in shared-line configurations. There are three types of intercom features:

- Dedicated intercom
- Intercom to PBX phone
- Dialable intercom

At Site A, user xxxx1 is user xxxx2's assistant. Their phones must be linked via the intercom feature so that xxxx1 can announce a visitor's arrival to user xxxx2, or user xxxx2 can easily ask user xxxx1 something without having to dial a call. The configuration shown in the following example builds a dedicated intercom facility between button 3 on each of the two phones.

```
ephone-dn 20
number A2222
name Intercom from xxxx1
intercom A2223
!
ephone-dn 21
number A2223
name Intercom from xxxx2
intercom A2222
!
ephone 1
button 1:1 3:20
!
ephone 2
button 1:2 2:10 3:21
```

Paging

The paging feature operates in a similar fashion to intercom, but it provides only one-way voice broadcast. Only idle phones receive paging announcements. Paging defines an extension that can be called to broadcast an audio page to a group of idle Cisco CME phones participating in the paging group.

The paging mechanism uses audio distribution using IP Multicast, replicated unicast, and a mixture of both. Therefore, multicast is used where possible, and unicast is allowed on specific phones that cannot be reached through multicast.

The recommended configuration of paging groups is to use a multicast address. If multicast is not configured, IP phones are paged individually using IP Unicast (to a maximum of ten IP phones). When multiple paging extensions are configured, each extension must use a unique IP Multicast address.

At Site A, you want to configure paging to all five employees' phones so that they can make announcements to the entire office staff by dialing extension 2010. This configuration is shown in the following example.

```
ephone-dn 25
  number 2010
  name All Office
  paging ip b.0.1.20 port 2000
!
ephone 1
  paging-dn 25
```

Ringling Timeout

You can define how long a phone can ring with no answer before returning disconnect tone to the caller. This timeout is used only for extensions that do not have call-forward-no-answer (CFNA) configured. The ringing timeout, shown here, keeps calls from ringing forever over interfaces such as FXO that do not have forward-disconnect supervision:

```
telephony-service
  timeouts ringing 120
```

Call Park

After you answer a call, you can press the **call park** softkey on the phone to park a call to one of the following two types of park slots:

- A slot with the same last two digits as the extension the call is on
- Randomly to any configured or designated park slot

The phone display shows where the call is parked so that you, or another employee, can pick up the call again. You can create a call park slot reserved for use by one extension by assigning that slot a number whose last two digits are the same as the last two digits of the extension number. When an extension starts to park a call, the system first searches for a call park slot that shares the same final two digits as the extension. If no such call park slot is found, the system then chooses any other available call park slot.

The following example defines a park slot for each of the five extensions at Site A so that each employee has a dedicated park slot. Note that in the call park ephone-dns 101, 102 to 105 map to ephone-dns 1, 2 to 5 that have extensions with the same last two digits 01, 02 to 05. Each park slot 101, 102 to 105 is reserved for use by ephone-dns 1, 2 to 5, respectively.

```
ephone-dn 1
```

```
number 2001
ephone-dn 2
  number 2002
ephone-dn 3
  number 2003
ephone-dn 4
  number 2004
ephone-dn 5
  number 2005
ephone-dn 101
  number 7401
  park-slot timeout 30 limit 10
!
ephone-dn 102
  number 7402
  park-slot timeout 30 limit 10
!
ephone-dn 103
  number 7403
  park-slot timeout 30 limit 10
!
ephone-dn 104
  number 7404
  park-slot timeout 30 limit 10
!
ephone-dn 105
  number 7405
  park-slot timeout 30 limit 10
```

To pick up a call from the same phone where the call was parked, simply press the **pickup** softkey and dial *. To pick up the call from any other phone, press the **pickup** softkey, and dial the extension of the call park slot shown on the phone display.

Time and Date Format

Cisco CME phones use time and date information from the router, which in turn is set up using NTP. You can customize the format in which the date and time are shown as follows:

```
telephony-service
  time-format 24
  date-format dd-mm-yy
```

Configuring Conference Call, Call Transfer, and Call Forward

This section summarizes configuration of the following call transfer and conference-related features:

- [G.711 Conferencing, page 124](#)
- [Call Transfer, page 124](#)
- [Call Forward, page 125](#)
- [Call Forward All Restrictions, page 125](#)

G.711 Conferencing

G.711 conferencing is the default operation on Cisco CME and requires no configuration. All parties in a conference call must use either the G.711 u-law codec or G.711 a-law codec. No other codec is supported for conferencing. G.729A endpoints can be supported via transcoding to G.711. Conferencing requires at least two lines configured on the phone initiating the conference. The dual-line configuration entered on the extensions in the “[Defining Extensions](#)” section on page 97 is sufficient to enable conferencing.

Cisco CME also supports conference cascading so that up to a maximum of eight or 16 conference sessions can be supported based on the platform you are using. The maximum number of G.711 conferences supported by Cisco CME varies by platform. If you want to adjust the maximum number of conference sessions, you can use the following CLI:

```
telephony-service
max-conferences 8
```

Call Transfer

The various releases of Cisco CME have different functionality with respect to the transfer operation. As of Cisco CME 3.2, the following methods of transfer are supported (command listed):

- **transfer-system blind**—Performs blind call transfers (without consultation) with a single phone line using a Cisco-proprietary method.
- **transfer-system full-blind**—Performs call transfers without consultation using the H.450.2 or SIP REFER standard methods.
- **transfer-system full-consult**—Performs H.450.2 or SIP call transfers with consultation using a second phone line if available. This method falls back to full blind if a second line is unavailable. This is the recommended mode for most systems. Also use the **supplementary-service** command under the **voice service voip** and **dial-peer** commands for call transfer between multiple Cisco CME and non-H.323 endpoints.
- **transfer-system local-consult**—Performs Cisco-proprietary call transfers with local consultation using a second phone line if available. This method falls back to blind for nonlocal transfer targets.

The following example shows Site A set up for full-consult transfers between Sites A and B. If transfers to PSTN destinations must also be allowed, you must also define transfer patterns matching PSTN dial plan patterns.

```
telephony-service
transfer-system full-consult
transfer-pattern 3...
transfer-pattern 2...
```

Cisco CME allows you to configure a transfer mode for each ephone-dn to override the global transfer mode set for all phones.

Transferring an incoming PSTN call to another PSTN destination can cause FXO ports to remain connected after both call parties disconnect. For you to avoid this problem, your PSTN provider must support one of the following disconnect methods on analog lines:

- Battery reversal
- Ground start signaling
- Power denial
- Supervisory tone disconnect

Call Forward

You can configure forwarding calls using the **call-forward busy**, **call-forward noan**, or **call-forward-all** commands for an ephone-dn. Cisco CME can forward calls using either a proprietary method or an H.450.3 standard method. If a **forward-pattern** is configured, as shown here, calls from the pattern (such as 2001, the calling number, not the called number) are forwarded using H.450.3, and all other calling parties are forwarded using the Cisco CME-proprietary forwarding method:

```
telephony-services
  forward-pattern 2...
```

Call Forward All Restrictions

You can restrict the maximum number of digits that can be entered by using the **cfwdall** softkey, as shown in the following example. Note that call forward restrictions apply only to destinations entered from the phone keypad, not to destinations entered using the CLI or GUI.

```
ephone-dn 1 dual-line
  number 2001 secondary 2xx5yy2001
  call-forward max-length 4
!
ephone-dn 2 dual-line
  number 2002 secondary 2xx5yy2002
  call-forward max-length 8
```

Enabling Applications

This section summarizes configuration of the following features used to enable applications:

- [Idle URL, page 125](#)
- [XML Services, page 125](#)

Idle URL

The Idle URL feature lets you access a URL, and display its content on idle IP phones. As a general rule, the Idle URL page should be hosted on an external web server. Cisco CME does not support the Idle URL for files stored in router Flash. Use the configuration shown in the following example to turn on this feature.

```
telephony-service
  url idle http://1.1.1.1/idle.asp
```

XML Services

Each Cisco CME system allows you to configure a single URL for services hosted from a separate server:

```
telephony-service
  url services http://a.10.10.4/CCMUser/123456/urltest.xml
```

However, the referenced service page urltest.xml can itself contain multiple URLs pointing to other services. The following example shows a sample urltest.xml file.

```
<CiscoIPPhoneMenu>
```

```

<Title>Services</Title>
<Prompt>Select an option</Prompt>
<MenuItem>
  <Name>App1</Name>
  <URL>http://a.10.10.4/CCMUser/123456/app.asp1</URL>
</MenuItem>
<MenuItem>
  <Name>App2</Name>
  <URL>http://a.10.10.4/CCMUser/123456/app2.asp</URL>
</MenuItem>
</CiscoIPPhoneMenu>

```

Step 8—Interconnecting Multiple Cisco IPC Express Systems

In the preceding seven steps, you set up the entire Cisco IPC Express system at Site A with examples of the most commonly deployed features you will require for your office. Although the Site B configuration was not described step by step, its configuration is very similar to that for Site A, with extensions starting at 3001 instead of 2001.

You cannot yet make calls between Sites A and B, because there is no dial plan to route calls between the sites. To achieve this, first ensure IP routing between the sites. In the sample configuration being built in this application note, Site A has an IP address of a.1.235.1 (with a netmask of 255.255.0.0), and Site B has an IP address of a.1.229.1 (with a netmask of 255.255.0.0), so these systems can easily reach each other. If your sites' IP addressing is more sophisticated, do the necessary configuration to achieve IP routing between your sites. Ensure that you see routes between the IP addresses of your sites with a **show ip route** command on each site's router. You will be able to **ping** one site from the other.

The sections that follow describe the following considerations:

- [Interconnecting Sites Via H.323, page 126](#)
- [Transcoding, page 127](#)
- [SIP RFC 2833 DTMF Relay, page 128](#)

Interconnecting Sites Via H.323

As soon as you have IP connectivity between the sites, the next step is to add dial peers to route calls between the sites. From Site A, if someone dials an extension that starts with 3, the call must be routed to Site B. Similarly, if someone at Site B dials an extension starting with 2, the call must be routed to Site A. If your dialing plan is less uniform than the sample network in this application note, you might need multiple dial peers to route all calls. Also, if only one site has PSTN access, and DID numbers for both the 2xxx and 3xxx ranges arrive on one PSTN trunk, more dial peers are needed to route all calls correctly.

For the sample network, the dial peers to route calls between the sites are shown in the following example.

```

! Site A (2xxx extension) dial-peers to direct calls to Site B (3xxx extensions)
dial-peer voice 3000 voip
destination-pattern 3...
session target ipv4:a.1.229.1
dtmf-relay h245-alphanumeric
codec g711ulaw
no vad
!
! Site B (3xxx extension) dial-peers to direct calls to Site A (2xxx extensions)

```

```
dial-peer voice 2000 voip
 destination-pattern 2...
 session target ipv4:a.1.235.1
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
```

At this point, the sites can call each other by simply dialing the extensions of the IP phones. One more configuration must be added to ensure that you can transfer calls between the sites. Add the **transfer-patterns** shown in the following example to both sites' configurations.

```
telephony-service
 transfer-system full-consult
 transfer-pattern 3...
 transfer-pattern 2...
```

In the preceding configuration setup, G.711 is used for all calls, including those between sites. To conserve bandwidth on the link between your sites, it is likely that you want to use G.729 on those calls instead. If so, remove the **codec g711ulaw** statement from the dial peers in preceding example for example configuration for Site A and Site B..

You can specify G.729 explicitly (by using the **codec g729r8** command), or you can simply delete the G.711 statement, because G.729 is the default codec for a VoIP dial peer. Whether the actual codec used is G.729 or G.729A depends on the codec complexity configuration of the PSTN trunk voice card. It doesn't matter for call connectivity, because G.729 and G.729A are fully compatible with each other.

Transcoding

Transcoding is required when part of a call must use the G.711 and another part of the same call must use G.729. When you use G.729 for calls between sites, and calls forward into voice mail, these calls currently fail on the configuration, because Cisco Unity Express voice mail supports only G.711. To fix this, configure transcoding resources on both sites to terminate G.729 calls, and transcode them locally to G.711 before they enter voice mail.

The following example gives a sample transcoding configuration. Ensure that you have enough digital signal processor (DSP) resources on the voice cards in your system to support this. If you don't, add more DSPs.

```
voice-card 2
 dsp services dspfarm
 !
 interface Loopback1
 ip address a.32.153.45 255.255.255.252
 h323-gateway voip interface
 h323-gateway voip bind srcaddr a.32.153.45
 !
 sccp local Loopback1
 sccp ccm a.32.153.45 identifier 1
 sccp
 sccp ccm group 1
 bind interface Loopback1
 associate ccm 1 priority 1
 associate profile 1 register MTP000e833595e0
 keepalive retries 5
 dspfarm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 codec gsmfr
```

```

codec g729br8
codec g729r8
maximum sessions 10
associate application SCCP

telephony-service
ip source-address a.32.153.45 port 2000
sdspfarm units 1
sdspfarm transcode sessions 30
sdspfarm tag 1 MTP000e833595e0

```

Transcoding is required to support the following call flows:

- Conference with one or more G.729 participants
- Call transfer or forward of a G.729 call into Cisco Unity Express AA or voice mail
- Playing MOH streams to G.729 calls for call on hold, call park, and consult transfer

SIP RFC 2833 DTMF Relay

You need SIP DTMF relay if you are using SIP trunking between sites. If you have Cisco Unity Express integrated on your sites, as in the sample configurations built in this application note, you must use H.323 trunking between the sites. SIP trunking is not yet supported with Cisco Unity Express 2.1. Note, however, that a SIP dial peer is required to route IP phone and PSTN calls to Cisco Unity Express.

If you are using AA and voice mail solutions other than Cisco Unity Express with Cisco CME, or a future Cisco Unity Express software release that might support this feature, you can use SIP trunking between sites. In a SIP trunking configuration, the out-of-band DTMF relay to the SCCP IP phones must be converted to in-band RFC 2833 DTMF relay on the SIP trunk. This is done using the configuration sample shown in the following example.

```

dial-peer voice 2000 voip
destination-pattern 8005yy1212
session protocol sip2
session target ipv4:1.1.1.2
dtmf-relay rtp-nte

```

Sample System Configurations

This section provides the full configurations of Site A and Site B built during the steps in this application note. The configuration descriptions are split into the following sections:

- [Site A Cisco CME Router Configurations, page 128](#)
- [Site A Cisco Unity Express AA and Voice Mail Configurations, page 135](#)
- [Site B Cisco CME Router Configurations, page 138](#)
- [Site B Cisco Unity Express AA and Voice Mail Configurations, page 142](#)

Site A Cisco CME Router Configurations

Site A is a Cisco 3725 router with extensions in the 2xxx range. [Figure 31](#), at the beginning of the application note, summarized the site layout. This section provides the Cisco CME configuration.

The show version Output

The following example provides the **show version** output for the Site A Cisco CME router.

```
cme-3745# show version
Cisco IOS Software, 3700 Software (C3745-IPVOICE-M), Version 12.3(11)T2, RELEASE
SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2004 by Cisco Systems, Inc.
Compiled Fri 29-Oct-04 06:38 by cmong
ROM: System Bootstrap, Version 12.2(8r)T2, RELEASE SOFTWARE (fc1)
cme32-3745 uptime is 2 weeks, 2 days, 4 hours, 39 minutes
System returned to ROM by reload at 15:40:35 PST Wed Dec 8 2004
System restarted at 14:05:42 PST Wed Dec 8 2004
System image file is "flash:c3745-ipvoice-mz.123-11.T2"

Cisco 3725 (R7000) processor (revision 0.1) with 111616K/19456K bytes of memory.
Processor board ID JAB0606800M
R7000 CPU at 240MHz, Implementation 39, Rev 3.3, 256KB L2 Cache
2 FastEthernet interfaces
31 Serial interfaces
2 terminal lines
2 Channelized T1/PRI ports
1 ATM AIM
2 Voice FXS interfaces
2 cisco service engine(s)
DRAM configuration is 64 bits wide with parity disabled.
55K bytes of NVRAM.
31360K bytes of ATA System CompactFlash (Read/Write)
```

The show running-config Output

The following example provides the **show running-config** output for the Site A Cisco CME router.

```
Cme-3725# show running-config
Building configuration...
Current configuration : 8014 bytes
!
Last configuration change at 22:13:14 PST Sat Oct 2 2004
!
NVRAM config last updated at 21:35:27 PST Sat Oct 2 2004
!
version 12.3
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname cme-3725
!
boot-start-marker
boot system flash: c3745-ipvoice-mz.123-11.T2

boot-end-marker
!
enable secret 5 $1$4dpj$Ta3tjuFSq/pehqZy0VAd.1
enable password cisco
!
memory-size iomem 15
clock timezone PST -8
network-clock-participate wic 0
network-clock-participate aim 0
voice-card 3
dspfarm
!
```

```

no aaa new-model
ip subnet-zero
ip cef
!
ip dhcp pool ITS
    network a.1.1.0 255.255.255.0
    option 150 ip a.1.1.100
    default-router a.1.1.100
!
no ip domain lookup
no ftp-server write-enable
isdn switch-type primary-5ess
!
voice service voip
    allow-connections h323 to sip
!
voice translation-rule 10
    rule 1 /2xx5yy2100/ /2100/
    rule 2 /2xx5yy2105/ /2105/
    rule 3 /2xx5yy2106/ /2106/
!
voice translation-profile to_cue
    translate called 10
!
controller T1 0/0
    pri-group timeslots 1-24
!
controller T1 0/1
!
interface FastEthernet0/0
    ip address a.1.235.1 255.255.0.0
    duplex auto
    speed auto
    no cdp enable
!
interface Serial0/0:23
    no ip address
    isdn switch-type primary-5ess
    isdn incoming-voice voice
    no cdp enable
!
interface FastEthernet0/1
    ip address a.1.1.100 255.255.0.0
    speed auto
    half-duplex
    no cdp enable
!
interface Service-Engine1/0
    ip unnumbered FastEthernet0/0
    service-module ip address a.1.235.128 255.255.0.0
    service-module ip default-gateway a.1.235.1
    hold-queue 60 out
!
router ospf 100
    log-adjacency-changes
    network a.1.0.0 0.0.255.255 area 0
!
ip classless
ip route a.1.235.128 255.255.255.255 Service-Engine1/0
ip route a.1.235.228 255.255.255.255 Service-Engine0/0
ip route b.107.0.0 255.255.0.0 FastEthernet0/0
ip route c.255.254.0 255.255.255.0 a.1.0.1
ip route c.255.254.0 255.255.255.0 FastEthernet0/0
!

```

```
ip http server
ip http path flash:
ip pim bidir-enable
!
tftp-server flash:P00303020214.bin
tftp-server flash:P00303020209.bin
tftp-server flash:P00305000200.bin
tftp-server flash:P00305000300.bin
!
voice-port 0/0:23
!
voice-port 2/0/0
  connection plar opx 2100
  caller-id enable
!
voice-port 2/0/1
  connection plar opx 2100
  caller-id enable
!
dial-peer cor custom
!
dial-peer voice 2100 voip
  destination-pattern 21..
  session protocol sipv2
  session target ipv4:a.1.235.128
  dtmf-relay sip-notify
  codec g711ulaw
  no vad
!
dial-peer voice 1000 pots
  destination-pattern 91.....
  port 0/0:15
  forward-digits 11
!
dial-peer voice 1001 pots
  preference 1
  destination-pattern 91.....
  port 2/0/0
!
dial-peer voice 1002 pots
  destination-pattern 9[2-9].....
  port 0/0:15
  forward-digits 7
!
dial-peer voice 1003 pots
  preference 1
  destination-pattern 9[2-9].....
  port 2/0/0
!
dial-peer voice 2101 voip
  description VM-AA-PSTN
  translation-profile outgoing to_cue
  destination-pattern 2xx5yy21..
  session protocol sipv2
  session target ipv4:a.1.235.128
  dtmf-relay sip-notify
  codec g711ulaw
  no vad
!
dial-peer voice 3000 voip
  destination-pattern 3...
  session target ipv4:a.1.229.1
  dtmf-relay h245-alphanumeric
  codec g711ulaw
```

```

no vad
!
dial-peer voice 1004 pots
 destination-pattern 911
 port 2/0/1
!
telephony-service
 load 7960-7940 P00303020214
 max-ephones 50
 max-dn 120
 ip source-address a.10.1.1.100 port 2000
 time-format 24
 date-format dd-mm-yy
 auto assign 1 to 5
 timeouts interdigit 20
 timeouts ringing 120
 create cnf-files version-stamp 7960 Sep 27 2004 12:31:02
 voicemail 2105
 max-conferences 8
 moh music-on-hold.au
 web admin system name admin password cisco
 web admin customer name custlogin password custpswd
 dn-webedit
 time-webedit
 transfer-system full-consult
 transfer-pattern 3...
 transfer-pattern 2...
 directory last-name-first
!
ephone-template 1
 softkeys idle Redial Pickup Dnd Login Gpickup
 softkeys seized Pickup Redial Endcall Gpickup
!
ephone-dn 1 dual-line
 number 2001 secondary 2xx5yy2001
 label xxxx1
 description xxxx1 user1
 name xxxx1 user1
 call-forward max-length 4
 call-forward busy 2105
 call-forward noan 2105 timeout 10
!
ephone-dn 2 dual-line
 number 2002 secondary 2xx5yy2002
 label xxxx2
 description xxxx2 user2
 name xxxx2 user2
 call-forward max-length 8
 call-forward busy 2105
 call-forward noan 2105 timeout 10
!
ephone-dn 3 dual-line
 number 2003 secondary 2xx5yy2003
 label xxxx3
 description xxxx3 user3
 name xxxx3 user3
 call-forward max-length 4
 call-forward busy 2105
 call-forward noan 2105 timeout 10
 hold-alert 15 originator
!
ephone-dn 4 dual-line
 number 2004 secondary 2xx5yy2004
 label xxxx4

```

```
description xxxx4 user4
name xxxx4 user4
call-forward max-length 4
call-forward busy 2105
call-forward noan 2105 timeout 10
!
ephone-dn 5 dual-line
number 2005 secondary 2xx5yy2005
label xxxx5
description xxxx5 user5
name xxxx5 user5
call-forward max-length 4
call-forward busy 2105
call-forward noan 2105 timeout 10
!
ephone-dn 6
number 2060
label Reception
name Reception
no huntstop
!
ephone-dn 7
number 2060
label Reception
name Reception
preference 1
!
ephone-dn 10
number 2999
label 911
trunk 911 timeout 5
!
ephone-dn 20
number A2222
name Intercom from xxxx1
intercom A2223
!
ephone-dn 21
number A2223
name Intercom from xxxx2
intercom A2222
!
ephone-dn 25
number 2010
name All Office
paging ip b.0.1.20 port 2000
!
ephone-dn 30
number 2020
!
ephone-dn 51
number 8000....
mwi on
!
ephone-dn 52
number 8001....
mwi off
!
ephone-dn 101
number 7401
park-slot timeout 30 limit 10
!
ephone-dn 102
number 7402
```

```

    park-slot timeout 30 limit 10
    !
    ephone-dn 103
    number 7403
    park-slot timeout 30 limit 10
    !
    ephone-dn 104
    number 7404
    park-slot timeout 30 limit 10
    !
    ephone-dn 105
    number 7405
    park-slot timeout 30 limit 10
    !
    ephone 1
    ephone-template 1
    username "xxxuser1"
    mac-address 0003.6BAA.D1F8
    fastdial 1 3001 name User1
    fastdial 2 2060 name Reception
    fastdial 3 2002 name xxxx2
    speed-dial 1 2005 label "xxxx5"
    speed-dial 2 3050 label "Customer Svc"
    paging-dn 25
    type 7960
    button 1:1 3:20
    !
    ephone 2
    ephone-template 1
    username "xxxuser2"
    mac-address 0003.6BAA.D362
    fastdial 1 3001 name User1
    fastdial 2 3002 name User2
    fastdial 3 2001 name xxxx1
    fastdial 4 2005 name xxxx5
    speed-dial 1 2004 label "xxxx4"
    paging-dn 25
    type 7960
    button 1o2,30 2:10 3:21
    !
    ephone 3
    ephone-template 1
    username "xxxuser3"
    mac-address 0003.AAAA.0003
    speed-dial 2 3050 label "Customer Svc"
    paging-dn 25
    button 1:3
    !
    ephone 4
    ephone-template 1
    username "xxxuser4"
    mac-address 0003.AAAA.0004
    speed-dial 2 3050 label "Customer Svc"
    paging-dn 25
    auto-line incoming
    button 1:6 2:4
    !
    ephone 5
    ephone-template 1
    username "xxxuser5"
    mac-address 0003.AAAA.0005
    speed-dial 2 3050 label "Customer Svc"
    paging-dn 25
    auto-line incoming

```

```

button 1:7 2:5
!
line con 0
  exec-timeout 0 0
  password xxxxx
line 33
  no activation-character
  no exec
  password xxxxx
  login
line aux 0
line vty 0
  exec-timeout 0 0
  password xxxxx
  login
!
ntp clock-period 17181154
ntp master
ntp server a.1.100.1

```

Site A Cisco Unity Express AA and Voice Mail Configurations

This section provides the configuration for Cisco Unity Express at Site A.

The show software version and show software licenses Output

The following example provides the **show software version** and **show software licenses** output for the Site A Cisco Unity Express.

```

Cue-3725# show software version
Installed Packages:
- Core 2.0.1
- Auto Attendant 2.0.1
- Global 2.0.1
- Voice Mail 2.0.1

Installed Languages:
- US English 2.0.0

Cue-3725# show software licenses
Core:
- application mode: CCME
- total usable system ports: 8
Voicemail/Auto Attendant:
- max system mailbox capacity time: 6000
- max general delivery mailboxes: 15
- max personal mailboxes: 50
Languages:
- max installed languages: 1
- max enabled languages: 1

```

The show running-config Output

The following example provides the **show running-config** output for the Site A Cisco Unity Express system.

```

Cue-3725# show running-config

```

```

Generating configuration:
clock timezone America/Los_Angeles
hostname cue-3725
ip domain-name localdomain
ntp server a.1.100.1
software download server url "ftp://a.1.231.201/ftp" credentials hidden
  "6u/dKTN/hsEuSAEfw40XlF2eFhnZfyUTSd8ZZNgd+Y9J3x1k2B35j0nfgWTYHfmPSd8ZZNgd+Y9J3
  xlk2B35jwAAAAA="
!
groupname Administrators create
groupname Broadcasters create
groupname Sales create
username xxxuser1 create
username xxxuser2 create
username xxxuser3 create
username xxxuser4 create
username xxxuser5 create
username admin create
!
groupname Sales phonenummerE164 "2xx5yy2050"
groupname Sales phonenummer "2050"
!
username xxxuser1 phonenummerE164 "2xx5yy2001"
username xxxuser2 phonenummerE164 "2xx5yy2002"
username xxxuser3 phonenummerE164 "2xx5yy2003"
username xxxuser4 phonenummerE164 "2xx5yy2004"
username xxxuser5 phonenummerE164 "2xx5yy2005"
username xxxuser1 phonenummer "2001"
username xxxuser2 phonenummer "2002"
username xxxuser3 phonenummer "2003"
username xxxuser4 phonenummer "2004"
username xxxuser5 phonenummer "2005"
!
groupname Administrators member admin
groupname Sales member xxxuser3
groupname Sales member xxxuser2
groupname Administrators privilege superuser
groupname Administrators privilege ManagePrompts
groupname Broadcasters privilege broadcast
!
backup server url "ftp://a.1.231.201/pod12nm_27jul04/" credentials hidden
  "xxOaioWv/uC5WSZLs/L2XY/frZzvmJ2MSd8ZZNgd+Y9J3x1k2B35j0nfgWTYHfmPSd8ZZNgd+Y9J3
  xlk2B35jwAAAAA="
!
ccn application autoattendant
  description "autoattendant"
  enabled
  maxsessions 8
  script "aa.aef"
  parameter "MaxRetry" "3"
  parameter "operExtn" "0"
  parameter "welcomePrompt" "AAWelcome.wav"
  end application
!
ccn application ciscoawiapplication
  description "ciscoawiapplication"
  enabled
  maxsessions 8
  script "setmwi.aef"
  parameter "strMWI_OFF_DN" "8001"
  parameter "strMWI_ON_DN" "8000"
  parameter "CallControlGroupID" "0"
  end application
!

```

```
ccn application custom-aa
description "custom-aa"
enabled
maxsessions 8
script "s1_main-officehours.aef"
parameter "MainOperExt" "2001"
end application
!
ccn application promptmgmt
description "promptmgmt"
enabled
maxsessions 1
script "promptmgmt.aef"
end application
!
ccn application voicemail
description "voicemail"
enabled
maxsessions 8
script "voicebrowser.aef"
parameter "logoutUri" "http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
parameter "uri" "http://localhost/voicemail/vxmlscripts/login.vxml"
end application
!
ccn subsystem sip
gateway address "a.1.235.1"
end subsystem
!
ccn trigger sip phonenumber 2100
application "custom-aa"
enabled
locale "en_US"
maxsessions 8
end trigger
!
ccn trigger sip phonenumber 2102
application "autoattendant"
enabled
locale "en_US"
maxsessions 8
end trigger
!
ccn trigger sip phonenumber 2105
application "voicemail"
enabled
idletimeout 5000
locale "en_US"
maxsessions 8
end trigger
!
ccn trigger sip phonenumber 2106
application "promptmgmt"
enabled
idletimeout 5000
locale "en_US"
maxsessions 1
end trigger
!
voicemail default broadcast expiration time 30
voicemail default expiration time 30
voicemail default language en_US
voicemail default mailboxsize 5520
voicemail recording time 900
voicemail default messagesize 60
```

```

voicemail operator telephone 2001
voicemail capacity time 6000
voicemail mailbox owner "Sales" size 5520
  description "Sales mailbox"
  zerooutnumber "2100"
end mailbox
!
voicemail mailbox owner "xxxuser3" size 5520
  description "xxxuser3 mailbox"
  zerooutnumber "2100"
end mailbox
!
voicemail mailbox owner "xxxuser1" size 5520
  description "xxxuser1 mailbox"
  zerooutnumber "2100"
end mailbox
!
voicemail mailbox owner "xxxuser5" size 5520
  description "xxxuser5 mailbox"
  zerooutnumber "2100"
end mailbox
!
voicemail mailbox owner "xxxuser2" size 5520
  description "xxxuser2 mailbox"
  zerooutnumber "2100"
end mailbox
!
voicemail mailbox owner "xxxuser4" size 5520
  description "xxxuser4 mailbox"
  zerooutnumber "2100"
end mailbox

```

Site B Cisco CME Router Configurations

Site B is a Cisco 2691 router with extensions in the 3xxx range. [Figure 31](#), at the beginning of this application note, summarized the site layout. This section provides the Cisco CME configuration.

The show version Output

The following example provides the **show version** output for the Site B Cisco CME router.

```

Cme-2691# show version
Cisco IOS Software, 2600 Software (C2691-IPVOICE-M), Version 12.3(11)T2
  RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2004 by Cisco Systems, Inc.
Compiled Fri 29-Oct-04 06:38 by cmong
ROM: System Bootstrap, Version 12.2(8r)T2, RELEASE SOFTWARE (fc1)
cme-2691 uptime is 1 day, 16 hours, 27 minutes
System returned to ROM by reload at 16:00:52 PST Fri Dec 1 2004
System restarted at 16:00:44 PST Fri Dec 1 2004
System image file is "flash:c2691-ipvoice-mz.123-11.T2"
Cisco 2691 (R7000) processor (revision 0.1) with 111616K/19456K bytes of memory.
Processor board ID JMX0635L1MF
R7000 CPU at 160MHz, Implementation 39, Rev 3.3, 256KB L2 Cache
2 FastEthernet interfaces
1 terminal line
1 cisco service engine(s)
DRAM configuration is 64 bits wide with parity disabled.

```

```
55K bytes of NVRAM.  
31360K bytes of ATA System CompactFlash (Read/Write)
```

The show running-config Output

The following example provides the **show running-config** output for the Site B Cisco CME router.

```
Cme-2691# show running-config  
Building configuration...  
Current configuration : 4712 bytes  
! Last configuration change at 08:27:56 PST Sun Oct 3 2004  
! NVRAM config last updated at 08:27:57 PST Sun Oct 3 2004  
!  
version 12.3  
service timestamps debug uptime  
service timestamps log uptime  
no service password-encryption  
!  
hostname cme-2691  
!  
boot-start-marker  
boot system flash:c2691-ipvoice-mz.123-11.T2  
boot-end-marker  
!  
no logging console  
enable secret 5 $1$/UXs$yVeHaiJawV557ni62uZqb.  
!  
memory-size iomem 15  
clock timezone PST -8  
no aaa new-model  
ip subnet-zero  
ip cef  
!  
ip dhcp excluded-address a.10.1.100 a.10.1.200  
!  
ip dhcp pool 1  
    network a.10.1.0 255.255.255.0  
    option 150 ip a.10.1.100  
    default-router a.10.1.100  
!  
no ip domain lookup  
!  
voice service voip  
    allow-connections h323 to sip  
!  
voice translation-rule 10  
    rule 1 /4xx5yy3100/ /3100/  
    rule 2 /4xx5yy3105/ /3105/  
    rule 3 /4xx5yy3106/ /3106/  
!  
voice translation-profile to_cue  
    translate called 10  
!  
interface FastEthernet0/0  
    ip address a.1.10.1.229.1 255.255.0.0  
    no ip mroute-cache  
    load-interval 30  
    duplex auto  
    speed auto  
!  
interface FastEthernet0/1  
    ip address a.10.1.100 255.255.0.0
```

```

no ip mroute-cache
duplex auto
speed auto
!
interface Service-Engine1/0
ip unnumbered FastEthernet0/0
service-module ip address a.1.10.1.229.128 255.255.0.0
service-module ip default-gateway a.1.10.1.229.1
!
ip classless
ip route a.1.10.1.229.128 255.255.255.255 Service-Engine1/0
!
ip http server
ip http path flash:
!
tftp-server flash:P00303020214.bin
!
dial-peer cor custom
!
dial-peer voice 3100 voip
description VM-AA
destination-pattern 31..
session protocol sipv2
session target ipv4:a.1.229.128
dtmf-relay sip-notify
codec g711ulaw
no vad
!
dial-peer voice 3101 voip
description VM-AA-PSTN
translation-profile outgoing to_cue
destination-pattern 4xx5yy31..
session protocol sipv2
session target ipv4:a.1.229.128
dtmf-relay sip-notify
codec g711ulaw
no vad
!
dial-peer voice 2000 voip
destination-pattern 2...
session target ipv4:a.1.235.1
dtmf-relay h245-alphanumeric
codec g711ulaw
no vad
!
telephony-service
load 7960-7940 P00303020214
max-ephones 48
max-dn 192
ip source-address a.10.1.100 port 2000
system message CUE System 2691
create cnf-files version-stamp 7960 Jul 15 2003 13:48:12
voicemail 3105
max-conferences 8
web admin system name admin password cisco
dn-webedit
time-webedit
transfer-system full-consult
transfer-pattern 3...
transfer-pattern 2...
!
ephone-dn 1 dual-line
number 3001
description User1

```

```
name User1
call-forward busy 3105
call-forward noan 3105 timeout 10
!
ephone-dn 2 dual-line
number 3002
description User2
name User2
call-forward busy 3105
call-forward noan 3105 timeout 10
!
ephone-dn 3
number 3003
description User3
name User3
call-forward busy 3105
call-forward noan 3105 timeout 10
!
ephone-dn 4
number 3004
description User4
name User4
call-forward busy 3105
call-forward noan 3105 timeout 10
!
ephone-dn 5
number 3005
description User5
name User5
call-forward busy 3105
call-forward noan 3105 timeout 10
!
ephone-dn 6
number 3006
description User6
name User6
call-forward busy 3105
call-forward noan 3105 timeout 10
!
ephone-dn 51
number 8000....
mwi on
!
ephone-dn 52
number 8001....
mwi off
!
ephone 1
username "User1" password null
mac-address 0009.B7F7.5793
speed-dial 1 2001 label "xxxx1"
speed-dial 2 3050 label "Customer Svc"
speed-dial 4 3100 label "AA"
button 1:1
!
ephone 2
username "User2" password null
mac-address 0002.FD06.D959
speed-dial 1 2003 label "xxxx3"
speed-dial 2 2005 label "xxxx5"
button 1:2
!
ephone 3
username "User3" password null
```

```

mac-address 0002.FFFF.1000
speed-dial 4 3100 label "AA"
button 1:3,5
!
ephone 4
username "User4" password null
mac-address 0002.FFFF.1004
speed-dial 4 3100 label "AA"
button 1:4
!
ephone 5
username "User5" password null
mac-address 0002.FFFF.1005
speed-dial 4 3100 label "AA"
button 1:5
!
ephone 6
username "User6" password null
mac-address 0002.FFFF.1006
speed-dial 4 3100 label "AA"
button 1:6
!
ephone-hunt 1 longest-idle
pilot 3050 secondary 4xx5yy3050
list 3002, 3003
final 3105
!
line con 0
exec-timeout 0 0
line 33
exec-timeout 0 0
password cisco
login
no activation-character
no exec
transport preferred none
transport input all
transport output all
line aux 0
line vty 0 4
exec-timeout 0 0
password cisco
login
!
ntp master
ntp server a.1.100.1

```

Site B Cisco Unity Express AA and Voice Mail Configurations

This section provides the configuration for Cisco Unity Express at Site B.

The show software version and show software licenses Output

The following example provides the **show software version** and **show software licenses** output for the Site B Cisco Unity Express.

```

Cue-2691# show software version
Installed Packages:
- Core 2.0.1

```

```

- Auto Attendant 2.0.1
- Bootloader (Primary) 1.0.18
- Bootloader (Secondary) 2.0.0
- Global 2.0.1
- Voice Mail 2.0.1
Installed Languages:
- US English 2.0.0

```

```

Cue-2691# show software licenses
Core:
- application mode: CCME
- total usable system ports: 8
Voicemail/Auto Attendant:
- max system mailbox capacity time: 6000
- max general delivery mailboxes: 15
- max personal mailboxes: 50
Languages:
- max installed languages: 1
- max enabled languages: 1

```

The show running-config Output

The following example provides the show running-config output for the Site B Cisco Unity Express system.

```

Cue-2691# show running-config
Generating configuration:
clock timezone America/Los_Angeles
hostname cue-2691
ip domain-name localdomain
ntp server a.1.100.1
software download server url "ftp://127.0.0.1/ftp" credentials hidden "6u/dKTN/
hsEuSAEfw40XlF2eFHnZfyUTSd8ZZNgd+Y9J3x1k2B35jOnfGWTYHfmPSd8ZZNgd+Y9J3x1k2B35jw
AAAAA="
groupname Administrators create
groupname customer-service create
username admin create
username User1 create
username User2 create
username User3 create
username User4 create
username User5 create
username User6 create
!
groupname customer-service phonenumberE164 "4xx5yy3050"
groupname customer-service phonenumber "3050"

username User1 phonenumber "3001"
username User2 phonenumber "3002"
username User3 phonenumber "3003"
username User4 phonenumber "3004"
username User5 phonenumber "3005"
username User6 phonenumber "3006"
!
groupname Administrators member admin
groupname customer-service member User2
groupname customer-service member User3
groupname Administrators privilege superuser
groupname Administrators privilege ManagePrompts
!

```

```

backup server url "ftp://a.1.231.201/SiteA" credentials hidden "EWlTygcMhYmjazX
  hE/VNXHCkplVV4KjescbDaLa4fl4WLSPFvv1rWUnfgWTYHfmPSd8ZZNgd+Y9J3x1k2B35jwAAAAA="
!
ccn application autoattendant
  description "autoattendant"
  enabled
  maxsessions 8
  script "aa.aef"
  parameter "MaxRetry" "3"
  parameter "operExtn" "0"
  parameter "welcomePrompt" "AAgreeting.wav"
  end application
!
ccn application ciscoMWIapplication
  description "ciscoMWIapplication"
  enabled
  maxsessions 8
  script "setmwi.aef"
  parameter "strMWI_OFF_DN" "8001"
  parameter "strMWI_ON_DN" "8000"
  parameter "CallControlGroupID" "0"
  end application
!
ccn application promptmgmt
  description "promptmgmt"
  enabled
  maxsessions 1
  script "promptmgmt.aef"
  end application
!
ccn application voicemail
  description "voicemail"
  enabled
  maxsessions 8
  script "voicebrowser.aef"
  parameter "logoutUri" "http://localhost/voicemail/vxmlscripts/mbxLogout.jsp"
  parameter "uri" "http://localhost/voicemail/vxmlscripts/login.vxml"
  end application
!
ccn subsystem sip
  gateway address "a.1.229.1"
  end subsystem
!
ccn trigger sip phonenumber 3100
  application "autoattendant"
  enabled
  maxsessions 8
  end trigger
!
ccn trigger sip phonenumber 3105
  application "voicemail"
  enabled
  maxsessions 8
  end trigger
!
ccn trigger sip phonenumber 3106
  application "promptmgmt"
  enabled
  maxsessions 1
  end trigger
!
voicemail default expiration time 60
voicemail default language en_US
voicemail default mailboxsize 5520

```

```
voicemail recording time 900
voicemail default messagesize 60
voicemail operator telephone 0
voicemail capacity time 6000
voicemail mailbox owner "User1" size 5520
  end mailbox
!
voicemail mailbox owner "User2" size 5520
  end mailbox
!
voicemail mailbox owner "User3" size 5520
  end mailbox
!
voicemail mailbox owner "User4" size 5520
  end mailbox
!
voicemail mailbox owner "User5" size 5520
  end mailbox
!
voicemail mailbox owner "User6" size 5520
  end mailbox
!
voicemail mailbox owner "customer-service" size 5520
  description "customer-service mailbox"
  zerooutnumber "3101"
  end mailbox
```

