CHAPTER 4

Dial Plan Management

CWVM allows you to manage various types of dial plans and voice ports, as explained in the following topics:

- POTS Dial Plans, page 4-1
- Network Dial Plans, page 4-7
- Voice Port Configuration, page 4-16

Note that this release of CWVM does not support CME or SRST dial-plan management. No dial-plan management options are displayed in CME or SRST views.

POTS Dial Plans

You can create and modify POTS dial plans (also called local dial plans) with CWVM. A POTS dial plan is created when you assign a telephone number to a voice port. CWVM automatically propagates the dial-plan information to all other gateways in the group to which the gateway belongs. If you use a Cisco UBE gatekeeper, CWVM automatically propagates the dial-plan information to other gateways under the Cisco UBE gatekeeper.

By design, the POTS dial plan configuration windows are populated with the information listed in a device's running configuration (obtained by entering the **`sh run`** command). As a result, if the default values for any of the available options are not provided in the running configuration, then these values will not appear in any of the POTS dial plan configuration windows.

Note Because CWVM supports a wide variety of Cisco routers, only basic information about dial plans is contained in this document. For detailed information about dial plans, see the documentation provided with your Cisco router.

The following topics explain how to manage POTS dial plans:

- Creating a POTS Dial Plan (Phone Number Assignment), page 4-2
- Modifying a POTS Dial Plan, page 4-3
- Deleting a POTS Dial Plan, page 4-5
- Searching for a Phone Number, page 4-6
- Moving a Phone Number, page 4-6
- Deleting a Phone Number, page 4-7

Creating a POTS Dial Plan (Phone Number Assignment)

You create a POTS dial plan when you assign a telephone number to a voice port on a voice-enabled gateway.

Note Because CWVM supports a wide variety of Cisco routers, only basic information about dial plans is contained in this document. For detailed information about dial plans, see the documentation provided with your Cisco router.

Procedure

Step 1 From the tree, right-click the gateway for which you want to create a local dial plan. A popup menu appears.

Step 2 Select **Dial Plan > Phone Number Assignment**. The Phone Number window appears.

Step 3 Select the voice port to which you want to add a telephone number.

Step 4 Enter a destination pattern in the Phone Number field. You can enter individual numbers, or a range of numbers. Below are examples:

- 555;685;559 (Use the semicolon (;) as a delimiter when entering multiple phone numbers.)
- 300-400 (In this example, CWVM creates 101 POTS dial plans starting from 300, 301, 302...400.)

Step 5 Click **Next**. The Propagate Phone popup window appears:

- Click **Yes** to have CWVM automatically propagate the dial-plan information to the other gateways in the group or under the Cisco UBE gatekeeper.
- Click **No** to keep the dial-plan information from being propagated to other gateways in the group.

Step 6 The Schedule Operation dialog box appears:

- Click **Do Now** to create the POTS dial plan immediately.
- Click **Schedule** to schedule the operation to execute at a later time. See Scheduling a Task, page 3-46 for more information.

Step 7 Click **Finish**. The POTS dial plan is created.

Note When you add a phone number to a gateway belonging to a Cisco UBE gatekeeper, the number is propagated to other gateways (belonging to the same Cisco UBE gatekeeper) as a VoIP network dial peer with a RAS session target. For example, GatekeeperA has a local zone called GKzoneA with a prefix of 408. There are two gateways (GW1 and GW2) that are registered to GKzoneA. When you add a phone number to GW1, the number is propagated to GW2 as a VoIP network dial peer with RAS session target.

Related Topics

- Searching for a Phone Number, page 4-6
- Moving a Phone Number, page 4-6

Modifying a POTS Dial Plan

Procedure

Step 1 From the tree, right-click the gateway for which you want to configure a local dial plan. A popup menu appears.

Step 2 Select **Dial Plan > POTS Dial Plans...**. The Administer POTS Dial Plan window appears.

Step 3 Select the dial plan you want to modify.

Step 4 Click **Modify**. The Modify POTS Dial Plan window appears with the Main Settings tab active.

Step 5 Modify any of the options in the Main Settings tab. For more information on each field, see Interpreting Main Dial Plan Options, page 4-4.

Step 6 Click the POTS tab. The POTS options appear.

Table 4-1 POTS Dial Plan Options

Field	Description
Prefix	Prefix of the target number the dial plan is dialing. This number is usually an area code or an extension.
Forward Digits	Number of dialed digits that are forwarded as part of the call setup message.
Port Name	Name of the port the dial plan is configured on.
Session Target	Network-specific address for a specified dial peer.
Trunk Group	Enter the dial plan Trunk Group IDs and set a Preference for each Group ID. Click Add to add a new Group ID; highlight an existing ID and click Delete to delete it.
Register E.164	Telephone number assigned to this port that is registered with the gateway's associated Cisco UBE gatekeeper.
Digit-Strip	Digit-strip option for the POTS digits replacement.
Direct Inward Dial	Assigned telephone number used as the final destination number.

For more information, see the specific Cisco IOS documentation or visit http://www.cisco.com/web/psa/products/index.html or http://www.cisco.com/cgi-bin/Support/Cmdlookup/home.pl (Cisco login required).

Step 7 Click **Finish** to exit the screen.

Step 8 If applicable, you can modify other dial plans. When all modifications are done, click **Finish**. The POTS dial plan is modified.

Related Topics

- POTS Dial Plans, page 4-1
- Deleting a POTS Dial Plan, page 4-5

Interpreting Main Dial Plan Options

The following table contains descriptions of common Cisco IOS commands. If the command you are looking for is not here, see the specific Cisco IOS release documentation or visit http://www.cisco.com/web/psa/products/index.html or http://www.cisco.com/cgi-bin/Support/Cmdlookup/home.pl (Cisco login required).

Table 4-2 Main Dial Plan Options

Field	Description
Dial Peer Type	Indicates whether a VoIP, VoATM, or VoFR dial peer is configured for a dial plan.
Huntstop	If the dial peer selected is a match to an incoming number and the associated device is not reachable, do not continue further matches. The default is No.
Shutdown	Changes the administrative state of this peer to down.
Dial Peer Tag	Value (ranging from 1 to 2147483647) that uniquely identifies a dial peer.
Maximum Connections	Maximum allowed connections to the dial plan. A value of -1 represents no limits.
Destination Pattern	A full E.164 telephone number prefix.
Application	Application that handles the incoming call after the dial plan is selected. If no application name is specified, the default session application handles the incoming call.
Answer Address	Call destination number.
Out-Bound	The selected application for outbound calls.
Incoming	Call pattern for incoming calls. This address can be used to identify the peer. If this address is either unknown or identical to the address of the Outgoing Call Pattern, the value of this parameter is zero.
Information-Type	Information type for dial peer.
Preference	Peer selection order when multiple peers match the selection criteria. A value of 0 gives the highest priority for peer selection.
Numbering-Type	Calling or called party numbering type.
Permission	Set the call orig/term permission of this dial peer.
Cor List fields	Class of restriction (COR) lists to be used by incoming and outgoing dial peers.
Translate Outgoing fields	Translation rule to be applied to outbound calling and called party numbers.

Table 4-2 Main Dial Plan Options (continued)

Field	Description
VAD	• VAD: When selected, voice activity detection (VAD) is enabled for the call. • Aggressive: When selected, the VAD noise threshold is reduced from -78 to -62 dBm. Note the following: – This option is available only when session protocol multicast is configured. – Noise that falls below the -62 dBm threshold is considered to be silence and is not sent over the network. – Unknown packets are considered to be silence and are discarded.
Extra Commands	Dial peer commands not recognized by CWVM. Enter a CLI command, if applicable. **Note** CWVM does not validate these commands.

Related Topics

- Modifying a POTS Dial Plan, page 4-3
- Modifying a Network Dial Plan, page 4-14

Deleting a POTS Dial Plan

Procedure

Step 1 From the tree, right-click the gateway for which you want to delete a local dial plan. A popup menu appears.

Step 2 Select **Dial Plan > Phone Number Assignment...**. The Phone Number Assignment window appears.

Step 3 In the Phone # column, select the phone number that you want to delete.

Step 4 Clear the entry.

Step 5 Click **Next**. The Propagate Phone popup window appears.

a. Click **Yes** to have CWVM automatically propagate the dial-plan information to the other gateways in the group or to the Cisco UBE gatekeeper of the gateway.

b. Click **No** to stop the dial-plan information from being propagated to other gateways in the group.

Step 6 The Schedule Operation dialog box appears:

a. Click **Do Now** to assign the telephone number immediately.

b. Click **Schedule** to schedule the operation to execute at a later time. See Scheduling a Task, page 3-46.

Step 7 Click **Finish**.

Related Topics

- Creating a POTS Dial Plan (Phone Number Assignment), page 4-2

- Searching for a Phone Number, page 4-6
- Moving a Phone Number, page 4-6

Searching for a Phone Number

CWVM searches phone numbers that have an assigned port only.

Procedure

Step 1 From the tree, right-click the server (or network, if you are in the CWVM tree view) for which you want to find the phone number. A popup menu appears.

Step 2 Select **Phones > Search Phones...**. The Search Phone Number window appears.

Step 3 Enter the phone number; for example, 1234567.

Step 4 Click **Search**. The IP address of the gateway appears.

Related Topics

- Moving a Phone Number, page 4-6
- Deleting a Phone Number, page 4-7

Moving a Phone Number

You can move phone numbers within a single network only.

Procedure

Step 1 From the tree, right-click the server in the network view (or network in the CWVM view) that has the phone number you want to move. A popup menu appears.

Step 2 Select **Phones > Move Phones...**. The Move Phone Number window appears.

Step 3 Click **New Entry**.

Step 4 Enter the phone number you want to move in the Phone Number field.

Step 5 From the Destination Gateway drop-down list, select the gateway address to which you want the phone number moved.

Note CWVM displays all the gateways that have ports within the network. If a Cisco UBE gatekeeper or group has been moved, underlying gateways may still be in the originating network. See Moving a Group, page 3-2, or Moving a Cisco UBE Gatekeeper, page 3-6, for more information.

Step 6 From the Port drop-down list, select the port to which you want the phone number assigned.

Step 7 Click **Next**. The Schedule Operation window appears.

Step 8 Schedule the operation. See Scheduling a Task, page 3-46 for more information.

Step 9 Click **Finish**. The phone number is moved.

Note To remove an entry from this window, select the row and click **Delete Entry**.

Related Topics

- Creating a POTS Dial Plan (Phone Number Assignment), page 4-2
- Searching for a Phone Number, page 4-6
- Deleting a Phone Number, page 4-7

Deleting a Phone Number

Removing the telephone number assigned to a voice port on a gateway deletes the dial plan. See Deleting a POTS Dial Plan, page 4-5 for more information.

Related Topics

- POTS Dial Plans, page 4-1
- Creating a POTS Dial Plan (Phone Number Assignment), page 4-2
- Modifying a POTS Dial Plan, page 4-3

Network Dial Plans

CWVM gives you the ability to create and manage VoIP, VoFR, or VoATM network dial plans. By design, the dial plan configuration windows are populated with the information listed in a device's running configuration (obtained by entering the **`sh run`** command). As a result, if the default values for any of the available options are not provided in the running configuration, then these values will not appear in any of the dial plan configuration windows.

Note Because CWVM supports a wide variety of Cisco routers, only basic information about dial plans is contained in this document. For detailed information about dial plans, see the documentation provided with your Cisco router.

The following topics explain how to manage network dial plans:

- Creating a VoIP Network Dial Plan, page 4-8
- Creating a VoATM Network Dial Plan, page 4-11
- Creating a VoFR Network Dial Plan, page 4-13
- Modifying a Network Dial Plan, page 4-14
- Deleting a Network Dial Plan, page 4-15
- Verifying a Network Dial Plan, page 4-15
- Showing and Modifying Network Connections, page 4-15

Creating a VoIP Network Dial Plan

Procedure

Step 1 From the tree, right-click the gateway for which you want to configure a network dial plan. A popup menu appears.

Step 2 Select **Dial Plan > Network Dial Plans...**. The Administer Network Dial Plan window appears.

Step 3 Click **New Dial Plan**. The Create Network Dial Plan window appears.

Step 4 From the DialPeer Type drop-down list, select VoIP. The VoIP tab on the Create Network Dial Plan window is now active.

Step 5 Complete the relevant fields on the Main Settings tab. For more information, see Interpreting Main Dial Plan Options, page 4-4.

Step 6 Click the VoIP tab on the Create Network Dial Plan window. The VoIP General subtab appears, with additional QOS and Voice-Class subtabs.

Step 7 Complete the following optional fields on the General subtab as needed to reflect the design of your dial plan:

Table 4-3 VoIP Network Dial Plan: General Options

Field	Description
Session Protocol	Session protocol to be used between local and remote routers. **Note** *sipv2* appears if the device is SIP-enabled.
Session Target	Session target used for the call.
Max-Redirects	Maximum number of redirects for this dial peer.
Session Transport	Session transport for this dial peer.
Fax Rate	Rate at which faxes are received.
Fax Rate Bytes	Number of fax data bytes per frame.
Signal Type	Specifies signal type.
Tech Prefix	Technology prefix of peer passed in ARQ, along with called-party address, to the Cisco UBE gatekeeper for called party address resolution during call setup.
Settle-call	If enabled, uses the settlement server.
Roaming	If enabled, uses the roaming server.
DTMF Relay	Whether DTMF relay is used to improve end-to-end transport of DTMF tones. **Note** The sip-notify option is available only when the value of Session Protocol is set to *sipv2*.
Digit-Drop	If enabled, prevents DTMF relay from sending both in-band and out-of band tones to the outgoing leg when sending IPIPGW calls in-band (rtp-nte) to out-of band (h245-alphanumeric). **Note** The digit-drop option is available only when the value of DTMF relay is set to rtp-nte.
Clid-Strip	Removes the calling-party number from calling line ID information.

Table 4-3 VoIP Network Dial Plan: General Options (continued)

Field	Description
Strip-Name	Removes the calling-party name from calling line ID information.
Substitute-Name	Copies the calling number into the display name if Progress Indicator allows it (and the calling name is empty).
Network-Number	Establishes the calling-party network number in the calling line ID information.
Second-Number	Removes a previously configured second network number from the calling link ID information.
Network-Provided	Allows you to set the screening indicator to reflect the number that was provided by the network.
Clid-Restrict	Prevents calling party number from being presented by calling line ID.
Pi-Restrict All	Causes removal of the calling-party names and numbers from the CLID when the Progress Indicator (PI) is restricted.
Incoming URI	Displays which dial peer is matched for a specific uniform resource identifier (URI) in an incoming voice call, and a specified field in the call message for an inbound dial peer. There are three field options: Calling, Called, and Request.

Step 8 Click the QOS subtab and complete the following optional fields as needed to reflect the design of your dial plan:

Table 4-4 VoIP Network Dial Plan: QOS Options

Field	Description
Acc QOS	Acceptable quality of service for the call.
Req QOS	Required quality of service for the call.
ICPIF	Specifies the call's Impairment/Calculated Planning Impairment Factor.
Expect Factor	User-requested Expectation Factor of voice quality for the call.
Poor QOV Trap	Whether poor quality of voice trap is enabled.
IP UDP Checksum	Whether the outgoing voice-related UDP packet includes checksum information.
IP QOS DSCP/Precedence	The Quality of Service (QoS) differentiated services code point (DSCP) of voice packets. • Media: Applies DSCP to media payload packets. • Signaling: Applies DSCP to signaling packets.

Step 9 Click the Voice-Class subtab and complete the following optional fields as needed to reflect the design of your dial plan:

Table 4-5 VoIP Network Dial Plan: Voice-Class Options

Field	Description
Codec	Assigns a previously configured codec selection preference list (codec voice class) to a VoIP dial peer.
H.323	Assigns an H.323 voice class to a VoIP dial peer.
Permanent	Assigns a previously configured voice class for a Cisco trunk or FRF.11 trunk to a dial peer.
Source	Assigns source interface parameters for a VoIP dial peer.
AAA	Applies the properties defined in the AAA voice class to a dial peer.
Sip Transport	Enables switching between UDP and TCP transport mechanisms for large SIP messages.
SIP URL	Configures URLs to either the Session Initiation Protocol (SIP), telephone (TEL), or system (SYSTEM) format for your dial-peer SIP calls.
SIP Rel1xx	If enabled, all SIP provisional responses (other than 100 Trying) are sent reliably to the remote SIP endpoint. There are four applicable values (default is System): • Supported: Reliable provisional responses are supported. If there is an applicable rel1xx header string, enter this value in the Supported/Require field. • Require: Reliable provisional responses are required. If there is an applicable rel1xx header string, enter this value in the Supported/Require field. • System: Uses the value configured in voice service mode. This is the default. • Disable: Disables the use of reliable provisional responses.

For additional information, see the specific Cisco IOS release documentation, or visit http://www.cisco.com/web/psa/products/index.html or http://www.cisco.com/cgi-bin/Support/Cmdlookup/home.pl (Cisco login required).

Step 10 Click the Codec subtab and complete the following optional fields as needed to reflect the design of your dial plan:

Table 4-6 VoIP Network Dial Plan: Codec Options

Field	Description
Codec	Voice coder rate of speech.
Codec Bytes	Number of voice data bytes per frame. Value depends on the codec selected.

Table 4-6 VoIP Network Dial Plan: Codec Options (continued)

Field	Description
ilbc mode	Specifies the iLBC operating frame mode that is encapsulated in each packet. The default mode is 20. **Note** The iLBC mode option is available only when the value of codec is set to ilbc.
Gsmamr-nb	• Encapsulation frame format for the GSMAMR-NB codec; one of these: – bandwidth-efficient – octet-aligned with CRC – octet-aligned with no CRC (the default) • Modes—The eight speech-encoding modes (bit rates between 4.75 and 12.2 kbps) available in the GSMAMR-NB codec. Valid values are from 0 to 7. The default is 0-7. You can select modes as a range (for example, 0-2), or individual modes (for example, 2,4,6), or a combination of the two (for example, 0-2,4,6-7). **Note** The GSMAMR-NB mode option is available only when the value of codec is set to gsmamr-nb. The GSMAMR-NB mode option is applicable only for AS5350XM and AS5400XM devices.

Step 11 Click **Finish.** The VoIP network dial plan is created.

Related Topics

- Network Dial Plans, page 4-7
- Modifying a Network Dial Plan, page 4-14
- Deleting a Network Dial Plan, page 4-15
- Verifying a Network Dial Plan, page 4-15

Creating a VoATM Network Dial Plan

Note Because CWVM supports a wide variety of Cisco routers, only basic information about dial plans is contained in this document. For detailed information about dial plans, see the documentation provided with your Cisco router.

Procedure

Step 1 From the tree, right-click the gateway for which you want to configure a network dial plan. A popup menu appears.

Step 2 Select **Dial Plan > Network Dial Plans...**. The Administer Network Dial Plan window appears.

Step 3 Click **New Dial Plan**. The Create Network Dial Plan window appears.

Step 4 From the DialPeer Type drop-down list, select VoATM. The VoATM tab on the Create Network Dial Plan window is now active.

Step 5 In the Main Settings tab, enter all other pertinent information. For more information, see Interpreting Main Dial Plan Options, page 4-4.

Step 6 On the Create Network Dial Plan window, click the VoATM tab. The VoATM options appear.

Step 7 Modify any the following optional fields to reflect the design of your dial plan.

Table 4-7 VoATM Network Dial Plan Options

Field	Description
Session Protocol	Session protocol to be used between local and remote gateways **Note** SIPV2 appears if device is SIP enabled.
Session Target	Session target used for the call. Enter the following information: • Interface: Serial interface and interface number (slot number and port number) associated with this dial peer. • PVC: Specific ATM permanent virtual circuit (PVC) for this dial peer. • Sub Channel ID: ATM network sub-channel identifier (CID) of this PVC.
Signal Type	Specifies signal type
Called Number	Enables an incoming VoATM call leg to get bridged to the correct POTS call leg when a static FRF.11 trunk connection is used.
Fax Rate	Rate at which faxes are received
Fax Rate Bytes	Number of fax data bytes per frame
Sequence Numbers	Select this check box to enable the generation of sequence numbers in each frame generated by the digital signal processor (DSP) for VoATM applications.
Voice Class	Dial peer voice class control parameters.
Codec	Voice coder rate of speech
Codec Bytes	Number of voice data bytes per frame
DTMF Relay	• DTMF Relay check box: When selected, DTMF relay is used to improve end-to-end transport of DTMF tones. • Voaal2: DTMF relay parameter for AAL2 over Voice.

Step 8 Click **Finish** to exit the screen.

Step 9 If applicable, you can modify other dial plans. When all modifications are done, click **Finish**.

Related Topics

- Network Dial Plans, page 4-7
- Modifying a Network Dial Plan, page 4-14
- Deleting a Network Dial Plan, page 4-15
- Verifying a Network Dial Plan, page 4-15

Creating a VoFR Network Dial Plan

Note Because CWVM supports a wide variety of Cisco routers, only basic information about dial plans is contained in this document. For detailed information about dial plans, see the documentation provided with your Cisco router.

Procedure

Step 1 From the tree, right-click the gateway for which you want to configure a network dial plan. A popup menu appears.

Step 2 Select **Dial Plan > Network Dial Plans...**. The Administer Network Dial Plan window appears.

Step 3 Click **New Dial Plan**. The Create Network Dial Plan window appears.

Step 4 From the DialPeer Type drop-down list, select VoFR. The VoFR tab on the Create Network Dial Plan window is now active.

Step 5 In the Main Settings tab, enter all other pertinent information. For more information, see Interpreting Main Dial Plan Options, page 4-4.

Step 6 Click the VoFR tab on the Create Network Dial Plan window. The VoFR options appear.

Step 7 Modify any the following optional fields to reflect the design of your dial plan.

Table 4-8 VoFR Network Dial Plan Options

Field	Description
Session Protocol	Session protocol to be used between local and remote router.
Session Target	Session target used for the call. Enter the following information: • Interface: Interface (Serial, BRI, or Multilink Frame Relay bundle) and interface number (slot number and port number) associated with this dial peer. • Dlci: Data link connection identifier (DLCI) for this dial peer. Valid values range from 16 to 1007. • Sub Channel ID: DLCI subchannel to be used for data on FRF.11 calls. This field is active only when the session protocol is set to *frf11-trunk*. **Note** When *BRI* is the specified Interface value and the BRI interface is not configured as a leased line, an invalid DLCI value may be displayed in the Extra Commands field.
Signal Type	Specifies signal type.
Called Number	Enables an incoming Voice over Frame Relay (VoFR) call leg to get bridged to the correct plain old telephone service (POTS) call leg when a static FRF.11 trunk connection is used.
Fax Rate	Rate at which faxes are received.
Fax Rate Bytes	Number of fax data bytes per frame.
Codec	Voice coder rate of speech.
Codec Bytes	Number of voice data bytes per frame.

Table 4-8 VoFR Network Dial Plan Options (continued)

Field	Description
DTMF Relay	When selected, DTMF relay is used to improve end-to-end transport of DTMF tones.
Sequence Number	Select this check box to enable the generation of sequence numbers in each frame generated by the digital signal processor (DSP) for Voice over Frame Relay (VoFR) applications.
Voice Class	Dial peer voice class control parameters.

Step 8 Click **Finish** to exit the screen.

Step 9 If applicable, you can modify other dial plans. When all modifications are done, click **Finish**.

Related Topics

- Network Dial Plans, page 4-7
- Modifying a Network Dial Plan, page 4-14
- Deleting a Network Dial Plan, page 4-15
- Verifying a Network Dial Plan, page 4-15

Modifying a Network Dial Plan

You can modify a network dial plan to make changes to the existing configuration of the dial plan. For example, you might modify a VoIP dial plan to change the signal type.

Note All network dial plans are modified by the same procedure.

Procedure

Step 1 From the tree view, right-click the gateway for which you want to modify a network dial plan. A popup menu appears.

Step 2 Select **Dial Plan > Network Dial Plan...**. The Administer Network Dial Plan window appears.

Step 3 Select the dial plan you want to modify.

Step 4 Click **Modify**. The Main Settings tab and the corresponding Network Dial Plan Type tab are active.

Step 5 You can modify any of the following parameters on the Main Settings tab. For more information, see Interpreting Main Dial Plan Options, page 4-4.

Step 6 Click the VoIP, VoATM, or VoFR tab on the Create Network Dial Plan window. The options corresponding to the selection you have made appear. You can modify any of the optional fields to reflect the design of your specific dial plan.

Step 7 Click **Finish** to exit the screen.

Step 8 If applicable, you can modify other dial plans. When all modifications are done, click **Finish**.

Deleting a Network Dial Plan

Deleting a local dial plan removes the telephone number that is assigned to the IP address (VoIP) or the circuit (VoFR or VoATM).

Procedure

Step 1 From the tree view, right-click the gateway with the network dial plan you want to delete. A popup menu appears.

Step 2 Select **Dial Plan > Network Dial Plan...**. The Administer Network Dial Plan window appears.

Step 3 Select the dial plan you want to delete.

Step 4 Click **Delete**. The network dial plan is deleted.

Step 5 Click **Finish** to exit the screen.

Verifying a Network Dial Plan

Dial plan verification confirms that a dial plan exists between two telephone numbers.

Procedure

Step 1 From the tree view, right-click the server (or network, if you are in the CWVM tree view) that has the dial plan \ you want to verify. A popup menu appears.

Step 2 Select **Phones > Verify Dial Plan...**. The Verify Dial Plan window appears.

Step 3 Enter the source phone number in the Source Phone Number field.

Step 4 Enter the destination phone number in the Destination Number field.

Step 5 Click **Verify**. The dial-plan information for the two numbers appears in the text window.

Step 6 Click **Finish**. The Verify Dial Plan window closes.

Showing and Modifying Network Connections

This procedure is necessary to show the VoFR and VoATM connections so that you can configure each gateway correctly.

For example, if you have four gateways that are connected by either VoATM or VoFR, you will need to perform this procedure to find where each type of connection (VoATM or VoFR) is located. When the gateway is configured for the applicable connection, the information is used for dial peer propagation. This means that when you assign a phone number and propagate information for one of these gateways, the following cases apply:

- If the gateway is connected via VoATM, CWVM creates a VoATM and VoIP dial peer.
- If the gateway is connected via VoFR, CWVM creates a VoFR and VoIP dial peer.

- If the gateway is connected via VoFR and VoATM, CWVM creates a VoFR, VoATM, and VoIP dial peer.

This procedure also allows the user to set the destination IP address of the connection if it has not been set by the Cisco IOS software running on the gateway.

Procedure

Step 1 From a VoFR/ATM tab view, right-click on a gateway. A popup menu appears.

Step 2 Select **Dial Plan > Show Network Connection...**. The Show Network Connection window appears.

The following are displayed:

- Connection Name—Cisco IOS name associated with the connections.
- Connection Type—VoFR or VoATM.
- Destination Gateway—IP address where the Frame Relay or ATM connection terminates. If the IP address is not filled in, you must enter it.

Step 3 Click **Finish**.

Voice Port Configuration

You can use CWVM to modify the configuration of voice ports that have been initially configured through the CLI on a router. For example, you can use CWVM to modify the configuration of an FXO voice port on a voice-enabled Cisco 3600 series router. However, the initial configuration of the FXO voice port must be done through the CLI.

The following topics explain how to configure voice ports:

- Configuring an FXO Voice Port, page 4-16
- Configuring an FXS Voice Port, page 4-19
- Configuring an E&M Voice Port, page 4-19
- Configuring an ISDN Voice Port, page 4-20

Note Because CWVM supports a wide variety of Cisco routers, only basic information about the specific values of parameters on voice ports is contained in this document. For detailed information about voice ports, see the documentation provided with your Cisco router.

Configuring an FXO Voice Port

CWVM automatically detects the types and configurations of voice ports enabled on a router when you add the router to CWVM. You can use CWVM to modify the configuration of a voice port.

Procedure

Step 1 From the tree view, right-click the gateway for which you want to configure FXO voice ports. A popup menu appears.

Step 2 Select **Dial Plan > Configure Voice Ports...**. The Configure Voice Ports—Step 1 window appears.

Step 3 Select the FXO voice port you want to configure:

- To configure only the FXO voice port you have selected, click **Configure Port**. The Configure Voice Ports—Step 2 window appears.
- To specify additional FXO voice ports that you also want configured, click **Batch Configure**. The Configure Voice Ports—Step 2 window appears.

Note The additional ports that are configured depend on the number entered in the Number of Ports field. For example, if you enter 2, the first two FXO voice ports listed under the selected port are configured.

Step 4 You can modify any of the parameters on the Main tab of the Configure Voice Ports—Step 2 window. See Interpreting Main Voice Port Configuration Settings, page 4-17 for more information.

Step 5 Click the FXO tab. The FXO options appear.

Step 6 You can modify either of the following parameters on the FXO tab of the Configure Voice Ports—Step 2 window:

- Signal Type—Signal types you can assign to the FXO interface. Valid values are LoopStart (the default) or GroundStart.
- Number of Rings—Number of rings detected before the loop is closed. The values range from 1 to 10, and the default is 1.
- Dial Type—Outgoing dial type for the interface. Valid values are either DTMF (specifying a touch-tone dialer), Pulse (specifying a pulse dialer), or MF (specifying a multifrequency tone dialer). The default is DTMF.
- Supervisory Disconnect—Assigns an FXO supervisory disconnect tone voice class to the voice port. Valid values are:
  - signal—Power denial recognition
  - anytone—Configures the voice port to disconnect on receipt of anytone
  - dualtoneMidcall—Specifies tone detection for the duration of the call
  - dualtonePreconnect—Specifies tone detection only during call setup

  By default, the supervisory disconnect command is enabled with the value signal.

Step 7 Click **Finish.** The FXO voice port is modified according to the changes you have made and the Configure Voice Ports window closes.

Interpreting Main Voice Port Configuration Settings

The following table lists available voice port configuration settings.

Table 4-9 Voice Port Configuration Settings

Field	Description
Port No.	Numerical identifier of the port(s) to be configured.
Number of Ports	Number of ports to be configured.

Table 4-9 Voice Port Configuration Settings (continued)

Field	Description
Type	Type of port to be configured.
Background Noise Enabled	Whether the background noise should be played to fill silence gaps if VAD is activated. Default is enabled.
Echo Cancel Enable	Enables/disables echo cancellation for the interface. Default is enabled.
Nonlinear Processing Enable	Enables/disables nonlinear processing for the interface. Enabling usually improves performance, but some users may perceive truncation of consonants. Default is enabled.
Music On Hold Threshold	Music-on-Hold threshold for the interface, in decibels. The values range from -70 to -30 dBm. Default is -38 dBm.
Input Gain	Level of input gain (in decibels) inserted at receiver side of interface. Values range from -6 to 14 dB. Default is 0 dB.
Output Attenuation	Output attenuation inserted (in decibels) at transmit side of interface. Values range from -6 to 14 dB. Default is 3 dB.
Echo Cancel Coverage	Coverage size of the echo canceller (EC). Default is 64 milliseconds. **Caution** If you specify a value that is not supported by your device, the CLI will fail.
Echo Cancel ERL Worst-Case	Specifies the worst-case Echo Return Loss (ERL) in decibels. The default is 6dB.
Connection Mode	Specifies the interface connection mode. Values are normal, trunking, and PLAR (the default). **Note** If you select trunking mode, a dialog box prompts you to confirm the change. Click **Yes** to restart the port so that the change takes effect.
Connection Number	Full E.164 telephone number (maximum 32 characters), used to establish connection with trunking or PLAR modes. Note the following: • If Connection Mode is set to *normal*, this parameter does not apply and defaults to null. • If Connection Mode is set to *trunking* and you change the current value for this field, a dialog box prompts you to confirm the change. Click **Yes** to restart the port so that the change takes effect.
Initial Digit Timeout	Specifies the amount of time the managed system waits for the initial input digit from a caller. Values range from 0 to 120 seconds. Default is 10 seconds.
Inter-Digit Timeout	Specifies the number of seconds the managed system waits for a subsequent input digit from a caller. Values range from 0 to 120 seconds. Default is 10.
Region Tone	Specifies region-specific standards for default tone, ring, and cadence settings.

Configuring an FXS Voice Port

CWVM automatically detects the types and configurations of voice ports enabled on a router when you add the router to CWVM. You can use CWVM to modify the configuration of a voice port.

Note Because CWVM supports a wide variety of Cisco routers, only basic information about the specific values of parameters on voice ports contained in this document. For detailed information about voice ports, see the documentation provided with your Cisco router.

Procedure

Step 1 From the tree view, right-click the gateway for which you want to configure FXS voice ports. A popup menu appears.

Step 2 Select **Dial Plan > Configure Voice Ports...**. The Configure Voice Ports—Step 1 window appears.

Step 3 Select the FXS voice port you want to configure:

- To configure only the FXS voice port you have selected, click **Configure Port**. The Configure Voice Ports—Step 2 window appears.
- To specify additional FXS voice ports that you also want configured, click **Batch Configure**. The Configure Voice Ports—Step 2 window appears.

Note The additional ports that are configured depend on the number entered in the Number of Ports field. For example, if you enter 2, the first two FXS voice ports listed under the selected port are configured.

Step 4 You can modify any of the parameters on the FXS tab of the Configure Voice Ports—Step 2 window. See the Interpreting Main Voice Port Configuration Settings, page 4-17 for more information.

Step 5 Click the FXS tab. The FXS options appear.

Step 6 You can modify either of the following parameters on the Main tab of the Configure Voice Ports—Step 2 window:

- Signal Type—Signal type assigned to the interface. Valid values are LoopStart (the default) or GroundStart.
- Ring Frequency—Ring frequency (in Hertz) to be used. For a list of valid values, see the documentation provided with your Cisco router.

Step 7 Click **Finish.** The FXS voice port is modified according to the changes you have made and the Configure Voice Ports window closes.

Configuring an E&M Voice Port

CWVM automatically detects the types and configurations of voice ports enabled on a router when you add the router to CWVM. You can use CWVM to modify the configuration of a voice port.

Note Because CWVM supports a wide variety of Cisco routers, only basic information about the specific values of parameters on voice ports is contained in this document. For detailed information about voice ports, see the documentation provided with your Cisco router.

Procedure

Step 1 From the tree view, right-click the gateway for which you want to configure E&M voice ports. A popup menu appears.

Step 2 Select **Dial Plan > Configure Voice Ports...**. The Configure Voice Ports—Step 1 window appears.

Step 3 Select the E&M Voice Port you want to configure:

- To configure only the E&M voice port you have selected, click **Configure Port**. The Configure Voice Ports—Step 2 window appears.
- To specify additional E&M voice ports that you want configured, click **Batch Configure**. The Configure Voice Ports—Step 2 window appears.

Note The additional ports that are configured depend on the number entered in the Number of Ports field. For example, if you enter 2, the first two E&M voice ports listed under the selected port are configured.

Step 4 You can modify any of the parameters on the Main tab of the Configure Voice Ports—Step 2 window. See Interpreting Main Voice Port Configuration Settings, page 4-17, for more information.

Step 5 Click the E&M tab. The E&M options appear.

Step 6 You can modify either of the following parameters on the E&M tab of the Configure Voice Ports—Step 2 window:

- Signal Type—Signal type assigned to the interface. Valid values are wink start, immediate dial, delay dial, and lmr. Default is wink start.
- Operation—Whether the operation of the interface is 2-wire or 4-wire. Default is 2-wire.
- Interface Type—Valid values are: Type I, Type II, Type III, Type IV, or Type V.
- Dial Type—Outgoing dial type for the interface. Valid values are either DTMF (specifying a touch-tone dialer), Pulse (specifying a pulse dialer), or MF (specifying a multifrequency tone dialer). The default is DTMF.

Step 7 Click **Finish**. The E&M voice port is modified according to the changes you have made and the Configure Voice Ports window closes.

Configuring an ISDN Voice Port

CWVM automatically detects the types and configurations of voice ports enabled on a router when you add the router to CWVM. You can use CWVM to modify the configuration of a voice port.

Note Because CWVM supports a wide variety of Cisco routers, only basic information about the specific values of parameters on voice ports is contained in this document. For detailed information about voice ports, see the documentation provided with your Cisco router.

Procedure

Step 1 From the tree view, right-click the gateway for which you want to configure ISDN voice ports. A popup menu appears.

Step 2 Select **Dial Plan > Configure Voice Ports...**. The Configure Voice Ports—Step 1 window appears.

Step 3 Select the ISDN voice port you want to configure:

- To configure only the ISDN voice port you have selected, click **Configure Port**. The Configure Voice Ports—Step 2 window appears.
- To specify additional ISDN voice ports that you want configured, click **Batch Configure**. The Configure Voice Ports—Step 2 window appears.

Note The additional ports that are configured depend on the number entered in the Number of Ports field. For example, if you enter 2, the first two ISDN voice ports listed under the selected port are configured.

Step 4 You can modify any of the parameters on the Main tab of the Configure Voice Ports—Step 2 window. See Interpreting Main Voice Port Configuration Settings, page 4-17 for more information.

Step 5 Click **Finish**. The ISDN voice port is modified according to the changes you have made, and the Configure Voice Ports window closes.