Cisco Unified Communications Voice Services

Feature history for Cisco Unified Communications Voice Services

Table 1: Feature history

Feature Name	Release Information	Description
Cisco IP-based media services	Cisco IOS XE Catalyst SD-WAN Release 17.3.1a Cisco vManage Release 20.3.1	You can now enable Cisco IP-based media services using a feature template.

Feature Name	Release Information	Description
Cisco Unified Communications Voice Services	Cisco IOS XE Catalyst SD-WAN Release 17.2.1r	This feature enables you to configure Cisco Unified Communications (UC) voice services on supported routers using feature templates and voice policies. When enabled, these routers can process calls for various endpoints, including voice ports, POTS dial peers, SIP dial peers, and phone profiles in Cisco Unified SRST mode. You configure UC voice services from the Feature tab and the Voice Policy page for a supported device. To configure UC voice services, Cisco SD-WAN Manager must run Cisco vManage 20.1.1. This feature is supported on Cisco 4000 Series Integrated Services Routers.
DSP Farms for UC Voice Services with Workflow Library and Configuration Groups	Cisco IOS XE Catalyst SD-WAN Release 17.13.1a Cisco Catalyst SD-WAN Manager Release 20.13.1	You can now configure DSP farms for UC voice services using the Workflow Library and configuration groups.

Cisco Unified Communications Voice Services

Cisco Unified Communications (UC) voice services enable you to configure feature templates and voice policies for supported routers. These templates and policies configure parameters for Foreign Exchange Office (FXO), Foreign Exchange Station (FXS), and FXS/Direct Inward Dialing (DID) interfaces. Starting with Cisco IOS XE Catalyst SD-WAN Release 17.3.1a, you can also configure parameters for Primary Rate Interface (PRI) Integrated Services Digital Network (ISDN). Additionally, you can use the DSPFarm feature template to enable Cisco IP-based media services.

Capabilities

When enabled, UC voice services allow routers to process calls for various endpoints. These include voice ports for analog and digital interfaces, Plain Old Telephone Service (POTS) dial peers, Session Initiation Protocol (SIP) dial peers, and phone profiles in Cisco Unified Survivable Remote Site Telephony (SRST) mode.

Note Starting with Cisco IOS XE Catalyst SD-WAN Release 17.13.1a, you can also configure and enable UC voice services using the Workflow library or configuration groups. For more information, see Cisco Unified Communications Voice Profile.

System requirements

To configure UC voice services for Cisco Unified Communications, Cisco SD-WAN Manager must run Cisco SD-WAN Release 20.3.1 or later.

For detailed information about commands to configure and maintain Cisco IOS voice applications, see the Cisco IOS Master Command List.

Configure Unified Communications Voice Services

This task provides the comprehensive workflow to enable Cisco Unified Communications (UC) voice services for supported routers, allowing them to process calls for various endpoints including voice ports, POTS dial peers, SIP dial peers, and phone profiles in Cisco Unified SRST mode. This ensures that Cisco IOS XE Catalyst SD-WAN devices can process calls for various endpoints and integrate IP-based media services.

Follow these steps to configure Unified Communications Voice Services:

Before you begin

Ensure that Cisco Catalyst SD-WAN Manager runs Cisco vManage Release 20.3.1 or later.

Procedure

Step 1 Add a voice card feature template.

Configures analog and PRI ISDN digital interfaces for voice cards. See Add a voice card feature template, on page 4

Step 2 Add a call routing feature template.

Configures parameters for TDM-SIP trunking and dial plans. See Add call routing feature template, on page 14

Step 3 Add an SRST feature template.

Configures parameters for Cisco Unified Survivable Remote Site Telephony (SRST) for SIP. See Add an SRST feature template, on page 19.

Step 4 Add a DSPFarm Feature Template.

Sets up and provisions a Digital Signal Processor (DSP) farm for media services like transcoding and conferencing. See Add a DSPFarm feature template, on page 21

Step 5 Add a voice policy.

Defines how the system augments and manipulates calls for various endpoint types. See Add a voice policy, on page 33.

Step 6 Provision a device template for Unified Communications.

Selects UC-specific feature templates and sets up the voice policy to include with the device template. See Provision a device template for unified communications, on page 73.

After completing this supertask, UC voice services will be enabled on supported Cisco IOS XE Catalyst SD-WAN devices, allowing them to process calls for various endpoints.

What to do next

To monitor the real-time statuses of lines, calls, interfaces, and related items that a device processes, navigate to **Monitor** > **Devices** from the SD-WAN Manager menu.

Add a voice card feature template

A voice card feature template configures analog and PRI ISDN digital interfaces, which provide configuration settings for ports on voice cards in routers. Use this procedure to create and configure such a template.

When you add a voice card feature template, you configure the type of voice card, port information, and service provider parameters for analog interfaces. For digital interfaces, you configure the type of voice card, the T1 or E1 controller, and related parameters. SD-WAN Manager assists with module placement by displaying available slots and sub-slots based on the device model.

Follow these steps to add a voice card feature template:

Before you begin

Ensure you have selected the supported device to which you want to add voice services in Cisco SD-WAN Manager.

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Templates**.

Step 2 Click **Feature Templates**, and click **Add Template**.

In Cisco vManage Release 20.7.1 and earlier releases, **Feature Templates** is called **Feature**.

Step 3 Select the supported device to which you want to add voice services.

Step 4 Select **Voice Card** from the **Unified Communications** templates.

Step 5 In **Template Name**, enter a name for the template.

This field may contain uppercase and lowercase letters, digits 0 through 9, hyphens (-), and underscores (_).

Step 6 In **Description**, enter a description for the template.

This field can contain any characters and spaces.

Step 7 To configure an analog interface, click **New Analog Interface** and configure interface options as described in the Voice card analog interface configuration options, on page 5 topic.

From Cisco IOS XE Catalyst SD-WAN Release 17.3.1a, select **Analog Interface** in the Interface area to access **New Analog Interface**.

a) You can add as many analog interfaces as needed, based on the number of interfaces that your module supports. After you configure each analog interface, click **Add**.

If any analog interfaces are already configured, they appear in the interfaces table on this page. To edit an existing interface, click **…** and select its pencil icon to edit the options in the window that pops up as described in the Voice card analog interface configuration options, on page 5 topic, and click **Save Changes**. To delete an interface, click **…** and click the trash can icon.

Step 8 To configure a PRI ISDN digital interface, in the **Interface** area, click **Digital Interface**, click **New Digital Interface**, and configure interface options as described in the Voice card digital interface configuration options, on page 7 topic.

a) Based on the number of interfaces that your module supports, you can add as many PRI ISDN digital interfaces as needed. After you configure each PRI ISDN digital interface, click **Add**.

If any digital interfaces are already configured, they appear in the interfaces table on this page. To edit an existing interface, click **...** and select its pencil icon to edit the options in the window that pops up as described in the Voice card digital interface configuration options, on page 7 topic, and click **Save Changes**. To delete an interface, click **…**, and click its trash can icon.

Note
After you save the interface configuration, you cannot change the module type, interface type, slot or sub-slot, or time slots.

If you want to change time slots, you must delete the interface and create a new one.

If you want to change the module type, interface type, and slot or sub-slot, detach the template from the device, unmap the voice policies that are associated with the interfaces, and delete all interfaces that are associated with the module and slot or sub-slot. Next, push the template to the device, reload the device, and create new required interfaces. Finally, push the new template to the device, and reattach the template to the device.

Step 9 Click **Save**.

Step 10 (Optional) To configure additional analog or PRI ISDN digital interfaces for this template:

a) From the Cisco SD-WAN Manager menu, choose **Configuration** > **Templates**.
b) Click **Feature Templates**.

   In Cisco vManage Release 20.7.x and earlier releases, **Feature Templates** is called **Feature**.

c) Click **...** for the template you wish to configure, and click **Edit**.
d) Repeat Step 7 or Step 8 and Step 9.

A voice card feature template is successfully added and configured.

Example

What to do next

If you need to configure more analog or PRI ISDN digital interfaces for this template later, you can edit the template by navigating to **Configuration** > **Templates** > **Feature Templates**, clicking **...** next to the template name, and selecting **Edit**. You can then repeat the configuration steps for analog or digital interfaces.

Voice card analog interface configuration options

This reference topic describes the various options available for configuring analog interfaces on voice cards within Cisco Unified Communications Voice Services. These options allow you to define the type of voice module, port characteristics, and service provider parameters for analog voice ports.

Analog Interface Configuration Options

This table describes the options for configuring an analog interface.

Table 2: Analog Interface Options

Option	Description	Cisco IOS CLI Equivalent
Module	Select the type of voice module that is installed in the router.	—
Module Slot/Sub-slot	Enter the slot and sub-slot of the voice module.	**voice-card** *slot/subslot*
Use DSP	Enable this option if you want to use the built-in DSPs on the network interface module for TDM calls.	**no local-bypass**
Port Type	Select the type of ports on the voice module that you are configuring for this interface (**FXS** or **FXO**). You can select **All** to define the port type for all ports of the selected type, or **Port Range** to define the port type for a specified range of ports. Using Port Range, you can create analog interfaces as described later in this procedure to configure different ranges of ports.	—
Description	Enter a description of the selected port or ports. For example, fax machine or paging system.	**description** *string*
Secondary Dialtone	Available if you select FXO from the Port Type drop-down list. Set to **On** if you want the selected ports to generate a secondary dial tone when callers access an outside line.	**secondary dialtone**
Connection PLAR	Enter the Private Line Automatic Ringdown extension to which the selected ports forward inbound calls.	**connection plar** *digits*
OPX	Available if you select FXO from the Port Type drop-down list. Check this option if you want to enable Off-Premises Extension for the PLAR extension.	**connection plar opx** *digits*

Option	Description	Cisco IOS CLI Equivalent
Signal Type	Select the Signal Type that indicates an on-hook or off-hook condition for calls that the ports receive. Options are **Loopstart**, **Groundstart**, or **DID**. The DID option is available if you select FXS from the Port Type drop-down list.	**signal** {groundstart | loopstart} **signal did** {delay-dial | immediate | wink-start}
Caller-ID Enable	Available if you select a signal type of Loopstart or Groundstart. Set to **ON** if you want to enable caller ID information for inbound calls.	**caller-id enable**
DID Signal Mode	Available if you select a signal type of DID. Choose the mode for the DID signal type (**Delay Dial**, **Immediate**, or **Wink Start**). Default: Wink Start.	**signal did** {delay-dial | immediate | wink-start}
Shutdown	Set to **ON** if you want to shut down ports that are not being used. Default: Off.	**shutdown**

Voice card digital interface configuration options

This reference topic describes the various options available for configuring digital interfaces on voice cards within Cisco Unified Communications Voice Services. These options allow you to define the type of T1/E1 voice module, interface characteristics, clock sources, and other parameters for digital voice ports.

Digital Interface Configuration Options

This table describes the options for configuring a digital interface.

Table 3: Digital Interface Options

Option	Description	Cisco IOS CLI Equivalent
Digital Interface Tab Provides options for configuring parameters for a T1/E1 voice module and the clock source for the module ports. Before you configure these options, ensure that you have the appropriate DSP module installed for each T1/E1 voice module.		
Module	Select the type of T1/E1 voice module that is installed in the router.	—

Option	Description	Cisco IOS CLI Equivalent
Interface Type	Select the type of interface on the voice module: • **T1 PRI**—Specifies T1 connectivity of 1.544 Mbps through the telephone switching network, using AMI or B8ZS coding • **E1 PRI**—Specifies a wide-area digital transmission scheme used predominantly in Europe that carries data at a rate of 2.048 Mbps	**card type** {**t1** | **e1**} *slot sub-slot*
Slot/Sub-slot	Enter the slot and sub-slot of the voice module.	**voice-card** *slot*/*sub-slot*
Use DSP	Enable this option if you want to use the built-in DSPs on the network interface module for TDM calls.	**no local-bypass**

<table>
<tr><th>Option</th><th>Description</th><th>Cisco IOS CLI Equivalent</th></tr>
<tr><td>Interface</td><td>Perform these actions to configure the number of T1/E1 ports to be provisioned on the module, and the clock source for each port:
1. Click Add. The Port and Clock Selector window displays.
2. Check the check box that corresponds to each port that you want to configure. The number of ports that you can configure depends on the Module type that you select.
3. For each port, select the clock source:
• Line—Sets the line clock as the primary clock source. With this option, the port clocks its transmitted data from a clock that is recovered from the line receive data stream.
• Primary Clock—Sets the port to be a primary clock source.
• Secondary Clock—Sets the port to be a secondary clock source.
• Network—Sets the backplane clock or the system oscillator clock as the module clock source.
We recommend that you set one port to be the primary clock and set another port going to the same network as a secondary clock source to act as a backup.
4. Click Add.</td><td>controller {t1 | e1} slot/sub-slot/number
clock source {network | line | line primary | line secondary}</td></tr>
<tr><td>Network Participation</td><td>This check box displays after you add an interface.
Check this check box to configure the T1/E1 module to participate in the backplane clock.
Uncheck this check box to remove the clock synchronization with the backplane clock for the module.
By defult, this check box is checked.</td><td>network-clock synchronization participate slot/sub-slot</td></tr>
</table>

Option	Description	Cisco IOS CLI Equivalent
Shutdown	Perform these actions to disable or enable the controller, serial interface, or voice port that is associated with the interface port. 1. Click **Shutdown Selected**. The Shutdown window displays. 2. For each port, select the item or items that you want to enable (**Controller**, **Serial**, or **Voice Port**. If you do not select an item, it is enabled. 3. Click **Add**.	**controller e1/t1** *slot*/*sub-slot*/*port* **shutdown** **interface serial** *slot*/*sub-slot*/**port:**{**15** | **23**} **shutdown** **voice-port** *slot*/*sub-slot*/**port:**{**15** | **23**} **shutdown**
Time Slots	Select the number of time slots of the interface type. Valid ranges: • For T1 PRI—Time slots 1 through 24. The 24th time slot is the D channel. • For E1 PRI— Time slots 1 through 31. The 16th time slot is the D channel.	**controller e1/t1** *slot*/*sub-slot*/*port* **pri-group timeslots** *timeslot-range* [**voice-dsp**]
Framing	Select the frame type for the interface type. For a T1 PRI interface type, options are: • **esf**—Extended super frame (default) • **sf**—Super frame For an E1 PRI interface type, options are: • **crc4**—CRC4 framing type (default) • **no-crc4**—No CRC4 framing type	**controller t1** *slot*/*sub-slot*/*port* **framing** [**esf** | **sf**] **controller e1** *slot*/*sub-slot*/*port* **framing** [**crc4** | **no-crc4**] [**australia**]
Australia	This check box displays when you select E1 PRI for the interface type. Check this check box to use the **australia** framing type.	**controller e1** *slot*/*sub-slot*/*port* **framing** [**crc4** | **no-crc4**] **australia**

Option	Description	Cisco IOS CLI Equivalent
Line Code	Select the line code type for the interface type. For a T1 PRI interface type, options are: • **ami**—Use Alternate Mark Inversion as the line code type • **b8zs**—Use binary 8-zero substitution as the line code type (default) For an E1 PRI interface type, options are: • **ami**—Use Alternate Mark Inversion as the line code type • **hdb3**—Use high-density binary 3 as the line code type (default)	**controller t1** *slot*/*sub-slot*/*port* **linecode** [**ami** | **b8zs**] **controller e1** *slot*/*sub-slot*/*port* **linecode** [**ami** | **hdb3**]
Line Termination	This check box appears only for an Interface type of E1 PRI. Select the line termination type for the E1 controller: • **75-ohm**—75 ohm unbalanced termination • **120-ohm**—120 ohm balanced termination (default)	**controller e1** *slot*/*sub-slot*/*port* **line-termination** {**75-ohm** | **120-ohm**}
Cable Length Type	This check box appears only for an Interface type of T1 PRI. Select the cable length type for the T1 PRI interface type: • **long**—Long cable length • **short**—Short cable length	**controller t1** *slot*/*sub-slot*/*port* **cablelength** {**short** | **long**}
Cable Length	This check box appears only for an interface type of T1 PRI. Select the cable length for the T1 PRI interface type. Use this option to fine-tune the pulse of a signal at the receiver for a T1 cable. The default value is **0db**.	**controller t1** *slot*/*sub-slot*/*port* **cablelength** {**[short** [**110ft** | **220ft** | **330ft** | **440ft** | **550ft** | **660ft**]] [**long** [**-15db** | **-22.5db** | **-7.5db** | **0db**]]}

Option	Description	Cisco IOS CLI Equivalent
Network Side	Enable this option to have the device use the standard PRI network-side interface. By default, this option is disabled (set to **No**).	**interface serial** *slot*/*sub-slot*/**port:**{**15**| **23**} **isdn protocol-emulate** [**network**|**user**]
Switch Type	Select the ISDN switch type for this interface: • **primary-qsig**—Supports QSIG signaling according tot the Q.931 protocol. Network side functionality is assigned with the isdn protocol-emulate command. • **primary-net5**—NET5 ISDN PRI switch types for Asia, Australia, and New Zealand. ETSI-compliant switches for Euro-ISDN E-DSS1 signaling system. • **primary-ntt**—Japanese NTT ISDN PRI switches. • **primary-4ess**—Lucent (AT&T) 4ESS switch type for the United States. • **primary-5ess**—Lucent (AT&T) 5ESS switch type for the United States. • **primary-dms100**—Nortel DMS-100 switch type for the United States. • **primary-ni**—National ISDN switch type.	**interface serial** *slot*/*sub-slot*/**port:**{**15**| **23**} **isdn switch-type** [**primary-4ess** | **primary-5ess** |**primary-dms100** | **primary-net5** | **primary-ni** | **primary-ntt** | **primary-qsig**]

Option	Description	Cisco IOS CLI Equivalent
ISDN Timer	Perform these actions to configure the ISDN timers for the interface: 1. Click **Add**. The ISDN Timer window displays. 2. Configure the following timers as needed. The values are in milliseconds. • **T200**. Valid range: integers 400 through 400000. Default: 1000. • **T203**. Valid range: integers 400 through 400000. The default value is based on the switch type and network side configurations. • **T301**. Valid range: integers 180000 through 86400000. The default value is based on the switch type and network side configurations. • **T303**. Valid range: integers 400 through 86400000. The default value is based on the switch type and network side configurations. • **T306**. Valid range: integers 400 through 86400000. Default: 30000. • **T309**. Valid range: integers 0 through 86400000. The default value is based on the switch type and network side configurations. • **T310**. Valid range: integers 400 through 400000. The default value is based on the switch type and network side configurations. • **T321**. Valid range: Integers 0 through 86400000. The default value is based on the switch type and network side configurations. 3. Click **Add**.	**interface serial** *slot*/*sub-slot*/**port:**{**15**| **23**} **isdn timer T200** *value* **isdn timer T203** *value* **isdn timer T301** *value* **isdn timer T303** *value* **isdn timer T306** *value* **isdn timer T309** *value* **isdn timer T310** *value* **isdn timer T321** *value*
Delay Connect Timer	Select the duration, in milliseconds, to delay connect a PRI ISDN hairpin call. Valid range: integers 0 through 200. Default: 20.	**voice-port** *slot*/*sub-slot*/**port:**{**15**| **23**} **timing delay-connect** *value*

Option	Description	Cisco IOS CLI Equivalent
Clock Tab Use this tab to configure priority order for the primary and secondary clock sources that you selected for each module. This tab is vailable after you configure a PRI ISDN digital interface and click **Add**.		
Clock Priority Sorting	Configure the priority of up to six clock sources. The drop-down list displays the interface ports for which a primary or secondary clock source is defined and that is configured for network participation. Check a check box to select the port for inclusion in the priority list, and use the **Up arrow** next to a port to change its priority. The list displays the ports in order of priority, with the port with the highest priority at the top of the list. After you configure the priority, this field displays the selected ports in priority order. We recommend that all ports in the priority list be of the same type, either E1-PRI or T1-PRI.	**network-clock input-source** priority **controller** [**t1**| **e1**] *slot*/*sub-slot*/*port*
Automatically Sync	Select **Add** to enable network synchronization between all modules and the router. Default: **On**.	network-clock synchronization automatic
Wait to restore clock	Enter the amount of time, in milliseconds, that the router waits before including a primary clock source in the clock selection process. Valid range: 0 through 86400. Default: 300.	network-clock wait-to-restore *milliseconds*

Add call routing feature template

A call routing feature template configures parameters for TDM-SIP trunking, including trusted IP addresses for preventing toll fraud, and a dial plan. Use this procedure to add and configure such a template.

A dial plan, made up of dial peers, defines how a router routes traffic to and from voice ports to the PSTN or to another branch. This template allows for the definition of global SIP communication settings and detailed dial peer configurations.

To add a call routing feature template:

Before you begin

Ensure you have selected the supported device to which you want to add call routing features in Cisco SD-WAN Manager.

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Templates**.

Step 2 Click **Feature Templates**, and click **Add Template**.

In Cisco vManage Release 20.7.1 and earlier releases, **Feature Templates** is called **Feature**.

Step 3 Select the supported device to which you want to add voice services.

Step 4 Select **Voice Card** from the **Unified Communications** templates.

Step 5 In **Template Name**, enter a name for the template.

This field may contain uppercase and lowercase letters, digits 0 through 9, hyphens (-), and underscores (_).

Step 6 In **Description**, enter a description for the template.

This field can contain any characters and spaces.

Step 7 In **Global**, configure options as described in the Global call routing options, on page 15 topic.

Step 8 In **Dial Plan**, perform one of these actions:

To configure a dial peer directly, configure options as described in the Dial peer options, on page 16 topic.

To create or edit a dial peer CSV file, click **Download Dial Peer List** to download the system provided file named Dial-Peers.csv. The first time you download this file, it contains field names but no records. Update this file as needed by using an application such as Microsoft Excel. For more information about this file, see Dial peer CSV file, on page 76.

To import configuration information from a dial peer CSV file that you have created, click **Upload Dial Peer List**.

You can add as many dial peers as needed. Click **Add** after you configure each dial peer. If any dial peers already are configured, they appear in the dial peers table on this page. To edit a configured dial peer, click **...**, and click its pencil icon. Edit the options in the window that pops up as described in the table, and click **Save Changes**. To delete a dial peer, click **...**, and click its trash can icon.

Step 9 Click **Save**.

A call routing feature template is successfully added and configured.

What to do next

Proceed to add an SRST feature template if required for your Unified Communications deployment.

Global call routing options

This reference topic describes the global options available for configuring call routing parameters within a call routing feature template. These settings define how the router communicates through SIP and handles trusted IP addresses for call security.

Global call routing options

The following table describes global options for configuring call routing.

Table 4: Global Call Routing Options

Option	Description	Cisco IOS CLI Equivalent
Trusted IPv4 Prefix List	Enter the IPv4 addresses with which the router can communicate through SIP. Enter each IPv4 address in CIDR format. For example, 10.1.2.3/32. Separate each address with a comma (,). The router does not communicate with other IPv4 addresses, which prevents fraudulent calls being placed through the router. A Trusted IPv4 Prefix is required for TDM to IP calls.	**voice service voip** **ip address trusted list** **ipv4** *ipv4-address/ipv4-network-mask*
Trusted IPv6 Prefix List	Enter the IPv6 addresses with which the router can communicate through SIP. Separate each IPv6 address with a comma (,). The router does not communicate with other IPv6 addresses, which prevents fraudulent calls being placed through the router. A Trusted IPv6 Prefix is required for TDM to IP calls.	**voice service voip** **ip address trusted list** **ipv6** *ipv6-prefix//prefix-length*
Source Interface	Enter the name of the source interface from which the router initiates SIP control and media traffic. This information defines how the return/response to this traffic should be sent.	**voice service voip** **sip** **bind control source-interface** *interface-id* **bind media source-interface** *interface-id*

Dial peer options

This reference topic describes the various options available for configuring dial peers, which are essential components of a dial plan. Dial peers define how a router routes traffic for voice calls, including settings for incoming/outgoing calls and transport protocols.

Dial peer options

The following table describes options for configuring dial peers.

Table 5: Dial peer options

Option	Description	Cisco IOS CLI Equivalent
Voice Dial Peer Tag	Enter a number to be used to reference the dial peer.	**dial-peer voice** *number* {**pots** | **voip**}
Dial Peer Type	Select the type of dial peer that you are creating (**POTS** or **SIP**).	**dial-peer voice** *number* {**pots** | **voip**}
Direction	Select the direction for traffic on this dial peer (**Incoming** or **Outgoing**).	Incoming: dial-peer voice *number* {pots | voip} **incoming called-number** *string* Outgoing: dial-peer voice *number* {pots | voip} **destination-pattern** *string*
Description	Enter a description of this dial peer.	**description**
Numbering Pattern	Enter a string that the router uses to match incoming calls to the dial peer. Enter the string as an E.164 format regular expression in the form [0-9,A-F#*.?+%()-]*T?.	Incoming: dial-peer voice *number* {pots | voip} **incoming called-number***string* Outgoing: dial-peer voice *number* {pots | voip} **destination-pattern** *string*
Forward Digits Type	Available if you select the POTS dial peer type and the Outgoing direction. Select how the dial peer transmits digits in outgoing numbers: • **All**—The dial peer transmits all digits • **None**—The dial peer does not transmit digits that do not match the destination pattern • **Some**—The dial peer transmits the specified number of right-most digits Default: None.	All: dial-peer voice *number* pots **forward-digits all** None: dial-peer voice *number* pots **forward-digits 0** Some: dial-peer voice *number* pots **forward-digits** *number*

Option	Description	Cisco IOS CLI Equivalent
Forward Digits	Available if you select **Some** for Forward Digits Type. Enter the number of right-most digits in the outgoing number to transmit. For example, if you set this value to 7 and the outgoing number is 1112223333, the dial peer transmits 2223333.	dial-peer voice *number* pots **forward-digits** *number*
Prefix	Available if you select the POTS dial peer type and the Outgoing direction. Enter digits to be prepended to the dial string for outgoing calls.	dial-peer voice *number* pots **prefix** *string*
Transport Protocol	Available if you select SIP for the Dial Peer Type. Choose the transport protocol (**TCP** or **UDP**) for SIP control signaling.	dial-peer voice *number* voip **session transport** {**tcp** | **udp**}
Preference	Available if you select POTS or SIP for the Dial Peer Type. Select an integer from 0 to 10, where the lower the number, the higher the preference. If dial peers have the same match criteria, the system uses the one with the highest preference value. Default: 0 (highest preference).	dial-peer voice *number* voip **preference** *value* dial-peer voice *number* pots **preference** *value*
Voice Port	Available if you select the POTS dial peer type. Enter the voice port that the router uses to match calls to the dial peer. For an analog port, enter the port you want. For a digital **T1** PRI ISDN port, enter a port with the suffix:**23**. For a digital E1 PRI ISDN port, enter a port with the suffix :**15**. For an outgoing dial peer, the router sends calls that match the dial peer to this port. For an incoming dial peer, this port serves as an extra match criterion. The dial peers are matched only if a call comes in on this port.	dial-peer voice *number* pots For an analog port: **port** *slot*/*subslot*/*port* For a digital port: **port** *slot*/*subslot*/*port*:**15** **port** *slot*/*subslot*/*port*:**23**

Option	Description	Cisco IOS CLI Equivalent
Destination Address	Available if you select the SIP dial peer type and the Outgoing direction. Enter the network address of the remote voice gateway to which calls are sent after a local outgoing SIP dial peer is matched. Enter the address in one of these formats: • dns:*hostname.domain* • *sip-server* ipv4:*destination-addres* ipv6:*destination-address*	**session target** {**ipv4:***destination-address* | **ipv6:***destination-address*| **sip-server** | **dns:***hostname.domain*}

Add an SRST feature template

An SRST feature template configures parameters for Cisco Unified Survivable Remote Site Telephony (SRST) for SIP. With Cisco Unified SRST, if the WAN goes down or is degraded, SIP IP phones in a branch site can register to the local gateway so that they continue to function for emergency services. Use this procedure to add and configure such a template.

The SRST feature template allows you to define global SRST settings, such as system messages and maximum phone/directory numbers, as well as individual phone profiles for SIP phones.

•

Before you begin

Ensure you have selected the supported device to which you want to add Cisco Unified SRST features in Cisco SD-WAN Manager.

Follow these steps to add an SRST feature template:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Templates**.

Step 2 Click **Feature Templates**, and click **Add Template**.

In Cisco vManage Release 20.7.1 and earlier releases, **Feature Templates** is called **Feature**.

Step 3 Select the supported device to which you want to add Cisco Unified SRST features.

Step 4 Click **SRST** from the Unified Communications templates.

Step 5 In **Template Name**, enter a name for the template.

This field can contain uppercase and lowercase letters, digits 0 through 9, hyphens (-), and underscores (_).

Step 6 In **Description**, enter a description for the template.

This field can contain any characters and spaces.

Step 7 In **Global Settings**, configure options as described in the Global Cisco Unified SRST Options, on page 20 topic.

Step 8 From **Phone Profile**, click **New Phone Profile** to create a phone profile, and configure options as described in the SRST Phone profile options, on page 21 topic.

A phone profile provides pool tag and device network information for a SIP phone.

You can add as many phone profiles as needed. Click **Add** after you configure each phone profile.

If any phone profiles already are configured, they appear in the phone profiles table on this page. To edit a configured phone profile, click **...**, and click its pencil icon. Edit the options in the window that pops up as described in the table, and click **Save Changes**. To delete a phone profile, click **...**, and click its trash can icon.

Step 9 Click **Save**.

An SRST feature template is successfully added and configured.

What to do next

Proceed to add a DSPFarm Feature Template if required for your Unified Communications deployment.

Global Cisco Unified SRST Options

This reference topic describes the global options available for configuring Cisco Unified Survivable Remote Site Telephony (SRST) parameters. These settings define system-wide behaviors for SRST mode, including messages, phone capacity, and music on hold.

Table 6: Global Cisco Unified SRST Options

Option	Description	Cisco IOS CLI Equivalent
System Message	Enter a message that displays on endpoints when Cisco Unified SRST mode is in effect.	**voice register global** **system message** *string*
Max Phones	Enter the number of phones that the system can register to the local gateway when in Cisco Unified SRST mode. The available values and the maximum values that you can enter in this field depend on the device that you are configuring. Hover your mouse pointer over the Information icon next to this field to see maximum values for supported devices.	**voice register global** **max-pool** *max-voice-register-pools*
Max Directory Numbers	Enter the number of DNs that the gateway supports when in Cisco Unified SRST mode. The available values and the maximum values that you can enter in this field depend on the device that you are configuring. Hover your mouse pointer over the Information icon next to the Max phones to support field to see maximum values for supported devices.	**voice register global** **max-dn** *max-directory-numbers*

Option	Description	Cisco IOS CLI Equivalent
Music on Hold	Select **Yes** to play music on hold on endpoints when a caller is on hold when in Cisco Unified SRST mode. Otherwise, select **No**.	—
Music on Hold file	Enter the path and file name of the audio file for music on hold. The file must be in the system flash and must be in .au or .wav format. In addition, the file format must contain 8-bit 8-kHz data, for example, CCITT a-law or u-law data format.	**call-manager-fallback** **moh** *filename*

SRST Phone profile options

This reference topic describes the options available for configuring individual SRST phone profiles. These settings define the network identification for SIP phones registering to a local gateway in SRST mode.

Table 7: SRST Phone Profile Options

Option	Description	Cisco IOS CLI Equivalent
Voice Register Pool Tag	Enter the unique sequence number of the IP phone to be configured. The maximum value is defined by the Max phones to support option in the Global tab of the SRST feature template.	**voice register pool** *pool-tag*
Device Network IPv6 Prefix	Enter the IPv6 prefix of the network that contains the IP phone to support. For example, a.b.c.d/24.	**voice register pool** *pool-tag* **id** [**network** *address* **mask** *mask*]
Device Network IPv4 Prefix	Enter the IPv4 prefix of the network that contains the IP phone to support.	**voice register pool** *pool-tag* **id** [**network** *address* **mask** *mask*]

Add a DSPFarm feature template

A DSP farm is a pool of Digital Signal Processor (DSP) resources on a router used by Cisco Unified Communications Manager for controlled transcoding, conferencing (non-secure only), and media termination point (MTP) services. A DSPFarm feature template is used to set up and provision a DSP farm. Use this procedure to add and configure such a template.

When you add a DSPFarm feature template, you configure options for the following items:

- Media resource modules—DSP modules and their placement on a router. You determine and build DSP farm profiles based on media resource modules.
- DSP farm profiles—Each profile defines parameters for provisioning a specific DSP farm service type. A profile includes options for provisioning a group of DSP resources that is used for transcoding, conferencing (only non-secure conferencing is supported), or MTP services. A profile is registered to a

Cisco Unified Communications Manager so that the Cisco Unified Communications Manager can invoke the resources for a service as needed.

- SCCP config—Configures a local interface that is used to communicate with up to four Cisco Unified Communications Manager servers, and configures related information that is required to register the DSP farm profiles to Cisco Unified Communications Manager. Also configures one or more Cisco Unified Communications Manager groups, each of which includes up to four Cisco Unified Communications Manager servers that control the DSP farm services that, in turn, are associated with the servers.

Before you begin

Ensure you have selected the supported device to which you want to add a DSP farm in Cisco SD-WAN Manager.

Follow these steps to add a DSPFarm feature template:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Templates**.

Step 2 Click **Feature Templates**, and click **Add Template**.

Note
In Cisco vManage Release 20.7.1 and earlier releases, **Feature Templates** is called **Feature**.

Step 3 Select the supported device to which you want to add a DSP farm.

Step 4 Click **DSPFarm** from the **Unified Communications** templates.

Step 5 In **Template Name**, enter a name for the template.

This field can contain uppercase and lowercase letters, digits 0 through 9, hyphens (-), and underscores (_).

Step 6 In **Description**, enter a description for the template.

This field can contain any characters and spaces.

Step 7 Configure the DSPFarm parameters.

a) From **Media Resources Modules**, click **Add Media Resources**, and configure options as described in the Media resource options, on page 23 topic.

A media resource module is a DSP module that is used by DSP Farm profiles.

You can add as many media resources interfaces as needed.

Click **Add** after you configure each media resource. After you configure a media resource, you cannot modify or delete it because other configuration items are based on the module and its placement. If you need to change a media resource configuration, you must remove the DSPFarm feature template and create a new one.

If any media resources are already configured, they appear in the table in this tab. To edit a configured media resource, click **...**, and click its pencil icon. Edit the options in the window that pops up as described in the Media Resource Options table, and click **Save Changes**. To delete a media resource, click **...**, and click its trash can icon.

b) From **Profile**, click **Add New Profile** to add a profile for a DSP farm service on a router, and configure options for the profile as described in the DSPFarm service options, on page 23 topic.

Click **Add** after you configure a profile. You can add up to 10 DSP farm profiles for each feature template.

Before you create a profile, you must know the maximum number of sessions that can be configured with the DSP resources that are available on the router. These resources are based on the type of modules in the router. To determine these resources, you can use a DSP calculator.

After you add a profile, you can modify the List Codec, Maximum Sessions, Maximum Conference Participants, and Shutdown options. You cannot change the profile type. If you want to change the profile type, you must delete the profile and create a new one.

If any profiles are already configured, they appear in the table in this tab. To edit a configured profile, click **...**, and click its pencil icon. Edit the options in the window that pops up as described in the "DSP Farm Service Options" table, and click **Save Changes**. To delete a profile, click **...**, and click its trash can icon.

c) In **SCCP Config**, configure options as described in the SCCP options, on page 27 topic.

Step 8 Click **Save**.

A DSPFarm feature template is successfully added and configured.

What to do next

Proceed to add a voice policy if required for your Unified Communications deployment.

Media resource options

This reference topic describes the options available for configuring media resources within a DSPFarm feature template. These settings define the router resource modules that carry DSP resources and their physical placement (slot/sub-slot ID).

Table 8: Media Resource Options

Option	Description	Cisco IOS CLI Equivalent
Module	Select the router resource module to carry DSP resources that are used by DSPFarm profiles.	—
Slot/sub-slot ID	Select the slot and sub-slot in which the resource module that you selected resides.	**voice-card** *slot/subslot* **dsp service dspfarm**

DSPFarm service options

This reference topic describes the options available for configuring DSP farm services within a DSPFarm feature template. These settings define the type of DSP farm service (Transcoder, Conference, or MTP), codecs, maximum participants/sessions, and other service-specific parameters.

Table 9: DSP Farm Service Options

Option	Description	Cisco IOS CLI Equivalent
Profile Type	Select the type of DSP farm service that this profile is for. Options are **Transcoder**, **Conference**, and **MTP**	**dspfarm profile** *profile-identifier* {**conference** | **mtp** | **transcode**}
Profile ID	A system-generated unique identifier for the profile.	—
Universal	Available if you select Transcoder for the Profile Type When this check box is unchecked, transcoding is allowed only between the G.711 codec and other codecs. When this check box is checked, transcoding is allowed between codecs of any type.	**dspfarm profile** *profile-identifier* **transcode** [**universal**]

Option	Description	Cisco IOS CLI Equivalent

List Codec	Select the codecs that are available for the DSP farm service that this profile defines.
	
	The following codecs are supported. For MTP profile types, you can select one option, or you can select **pass-through** and one other option. If you want to change a codec, unselect the current codec before selecting a new one.
	
	• For the Transcoder profile type:
	• g711alaw, g711ulaw, g729abr8, g729ar8, g729br8, g729r8, g722-64, ilbc, iSAC
	• For the Conference profile type:
	  • g711alaw, g711ulaw,g722r-64, g729abr8, g729ar8, g729br8, g729r8
	• For the MTP profile type for software MTP only:
	  • g711ulaw, g711alaw, g722-64, g729abr8, g729ar8, g729br8, g729r8, ilbc, iSAC, pass-through
	• For the MTP profile type for hardware MTP only, or for hardware and software MTP:
	  • g711ulaw, g711alaw, pass-through
		codec *codec-name*

Conference Maximum Participants	Available if you select Conference for the Profile Type. Select the maximum number of parties that can participate in a conference bridge (**8**, **16**, or **32**).	**maximum conference-participants** *number*

Option	Description	Cisco IOS CLI Equivalent
Maximum Sessions	Available if you select Transcoder or Conference for the Profile Type. Enter the maximum number of sessions that this profile can support. This value depends on the maximum number sessions that can be configured with the DSP resources that are available on the router. These resources are based on the type of modules in the router. To determine these resources, you can use a DSP calculator.	**maximum sessions** *number*
MTP Type	Available if you select MTP for the Profile Type. Select the way in which the router performs minor MTP translations such as G.711alaw to G.711ulaw, and DTMF conversions. Options are: • **Hardware**—MTP translations and conversions are performed by the hardware DSP resources • **Software**—MTP translations and conversions are performed by the router CPU	**maximum session** {**hardware** | **software**}
MTP Maximum Hardware Sessions	Available if you select Hardware for the MTP type. Select the maximum number of hardware sessions that can be used for MTP translations and conversions. Maximum value: 4000	**maximum session hardware** *number*
MTP Maximum Software Sessions	Available if you select Software for the MTP type. Select the maximum number of CPU sessions that can be used for MTP translations and conversions. Maximum value: 6000	**maximum session software** *number*

Option	Description	Cisco IOS CLI Equivalent
Application	Select the type of application to which the DSP farm services that are provisioned on the device are associated.	**associate application sccp**
Shutdown	Enable this option to take this profile out of service.	**shutdown**

SCCP options

This reference topic describes the options available for configuring SCCP (Signaling Connection Control Part) within a DSPFarm feature template. These settings define the local interface for SCCP communication and the Cisco Unified Communications Manager servers and groups to which DSP farm profiles will register.

Table 10: SCCP Options

Option	Description	Cisco IOS CLI Equivalent
CUCM Tab Configure up to 12 Cisco Unified Communications Manager servers to which the profiles that you defined in the Profile tab register.		

Option	Description	Cisco IOS CLI Equivalent
Local Interface	Enter the local interface that DSP services that are associated with the SCCP application use to register with Cisco Unified Communications Manager. Enter the interface in this format: *interface-type*/*interface-numberport* where: • *interface-type*—Type of interface that the services use to register with Cisco Unified Communications Manager. The type can be a GigabitEthernet interface or a port channel interface. • *interface-number*—Interface number that the services use to register with Cisco Unified Communications Manager. • *port*—(Optional) Port on which the interface communicates with Cisco Unified Communications Manager. If you do not specify a port, the default value 2000 is used. For example: GigabitEthernet0/0/0.	**sccp local** *interface-type interface-number* [**port** *port-number*]

Option	Description	Cisco IOS CLI Equivalent
Server List - *x*	Designate a Cisco Unified Communications Manager server to which the profiles that you defined in the Profile tab register. In the first field, enter the IP address or DNS name of the Cisco Unified Communications Manager server. In the second field, enter a numerical identifier for the Cisco Unified Communications Manager server. Click the Plus Sign icon (+) to configure up to 11 additional servers. To remove a server, click its corresponding Minus Sign icon. (–).	**sccp ccm** {*ipv4-address* | *ipv6-address* | *dns*} **identifier** *identifier-number* **version 7.0+**
CUCM Groups Tab This tab is available when at least one Cisco Unified Communications Manager server is configured in the Cisco Unified Communications Manager tab. Configure a Cisco Unified Communications Manager group, which includes up to 4 Cisco Unified Communications Manager servers that control the DSP farm services that, in turn, are associated with the servers. If any Cisco Unified Communications Manager groups are already configured, they appear in the table in this tab. To edit a configured Cisco Unified Communications Manager group, click its pencil icon in the Action column, edit the options in the window that pops up as described in the following rows, and click **Save Changes**. To delete a Cisco Unified Communications Manager group, click its trash can icon in the Action column.		
Add New CUCM Group	Click to add a new Cisco Unified Communications Manager group.	**sccp ccm group** *group-id*

Option	Description	Cisco IOS CLI Equivalent
Server Groups Priority Order	Select the priority in which the Cisco Unified Communications Manager servers in this Cisco Unified Communications Manager group are used. To do so: **1.** Click this field to display a list of the Cisco Unified Communications Manager servers that you configured on the Cisco Unified Communications Manager tab. **2.** Select the server that you want to be the primary server. This server has the highest priority. **3.** Click the field again and select the server that you want to be the redundant server with the next highest priority. Repeat this step to select other redundant servers. The servers appear in this field in priority order. To remove a server from the group, click its X icon. To change the priority order of servers, remove the servers and add them back in the desired order.	**associate ccm** *cisco-unified-communications-manager-id* **priority** *priority*

Option	Description	Cisco IOS CLI Equivalent
CUCM Media Resource Name Profile to be Associated	In the Cisco Unified Communications Manager Media Resource Name field, enter a unique name that is used to register a DSP farm profile to the Cisco Unified Communications Manager servers. The name must contain from 6 to 15 characters. Characters can be letter, numbers, slashes (/), hyphens (-), and underscores (_). Space characters are not allowed. In the corresponding Profile to be Associated field, select a DSP farm profile to be registered to this Cisco Unified Communications Manager group using the name that you entered. To select a profile, click this field to display a list of the profile IDs that were configured on the Profile tab, and click the ID of the profile that you want. To add another Cisco Unified Communications Manager media resource name and profile, click the plus sign (+). You can add up to 4 Cisco Unified Communications Manager media resources and profiles. To remove a Cisco Unified Communications Manager media resource name and profile, click its corresponding minus sign (–).	**associate ccm** *profile-identifier* **register** *device-name*

Option	Description	Cisco IOS CLI Equivalent
CUCM Switchback	Select the switchback method that the Cisco Unified Communications Manager servers in this Cisco Unified Communications Manager group use to switch back after a failover: • **graceful**—Switchback occurs after all active sessions terminate gracefully. • **guard**—Switchback occurs either when active sessions are terminated gracefully or when the guard timer expires, whichever happens first. • **immediate**—Performs the Cisco Unified Communications Manager switchback to the higher priority Cisco Unified Communications Manager immediately when the timer expires, whether there is an active connection or not. Default: **graceful**.	**switchback method** {**graceful** | **guard** [*timeout-guard-value*] | **immediate**}
CUCM Switchover	Select the switchover method that Cisco Unified Communications Manager servers in this Cisco Unified Communications Manager use group when failing over: • **graceful**—Switchback occurs after all active sessions terminate gracefully. • **immediate**—Switchover occurs immediately, whether there is an active connection or not. Default: **graceful**.	**switchover method** {**graceful** | **immediate**}

Add a voice policy

A voice policy defines how the system augments and manipulates calls for various endpoint types. This supertask guides you through the process of creating a new voice policy and configuring its subpolicies for different communication endpoints.

Voice policies are crucial for tailoring call functionality for specific endpoints, including voice ports, POTS dial peers, SIP dial peers, and Cisco Unified SRST phone profiles.

Before you begin

Follow these steps to add a voice policy:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Unified Communications**.

Step 2 Click **Add Voice Policy**.

Step 3 In **Voice Policy Name**, enter a name for the policy.

Step 4 Configure the required subpolicies:

a) Configure voice ports for a voice policy.

Define how the system augments and manipulates calls for voice port endpoint types. See Configure voice ports for a voice policy, on page 33.

b) Configure POTS Dial Peers for a voice policy.

Define how the system augments and manipulates calls for POTS dial peer endpoint types. See Configure POTS dial peers for a voice policy, on page 49.

c) Configure SIP Dial Peers for a voice policy.

Define how the system augments and manipulates calls for SIP dial peer endpoint types. See Configure SIP Dial Peers for a voice policy, on page 58.

d) Configure SRST Phones for a voice policy.

Define how the system augments and manipulates calls for Cisco Unified SRST phone endpoint types. See Configure SRST phones for a voice policy, on page 71

Step 5 Click **Save Policy**.

A voice policy is successfully added and configured with its respective subpolicies.

What to do next

Proceed to provision a device template for Unified Communications to apply this voice policy to your devices.

Configure voice ports for a voice policy

This task defines how the system augments and manipulates calls for the voice port endpoint type within a voice policy. It allows you to set up various call functionality options such as trunk groups, translation rules, and line parameters.

When configuring voice ports for a voice policy, you define specific call functionality options depending on the type of voice card you are using (FXO, FXS, PRI ISDN, or FXS DID). These settings ensure calls are handled correctly based on your network requirements.

Before you begin

You must be in the process of adding a new voice policy or editing an existing one within Cisco SD-WAN Manager.

Follow these steps to configure voice ports for a voice policy:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Unified Communications**.

Step 2 Click **Add Voice Policy**, and choose **Voice Ports**.

Step 3 In **Voice Ports**, enter a name for the policy

From the **Add Voice Ports Policy Profile** drop-down list, select **Create New**.

Alternatively, you can select **Copy from Existing** to reuse settings from an existing policy profile.

Step 4 Select **FXO**, **FXS**, **PRI ISDN**, or **FXS DID** to specify the type of voice port that the policy is for.

Step 5 Select the types of call functionality policy options that you want to configure from the list of options that displays, and click **Next**.

Available options include **Trunk Group**, **Translation Profile**, **Station ID**, **Line Params**, **Tuning Params**, **Supervisory Disconnect**, and **DID Timers**

Step 6 Configure the selected call functionality options on the tabs that display: a.

a) For **Trunk Group** options, configure settings as described in the Voice Ports Trunk Group Options topic.

   Define how voice ports function as members of a trunk group. After saving, configure priority by double-clicking the Priority field (1-64). If any trunk groups are already configured, they appear in the table. To edit, click **...** and select its pencil icon; to delete, select its trash can icon.

b) For **Translation Profile** options, configure settings as described in the Voice ports translation profile options, on page 39 topic.

   Define translation rules for calling and called numbers. After saving, double-click the dash (-) in the **Direction** column to select **Incoming** or **Outgoing**. You can create up to two translation profiles for this endpoint.

   You can create up to two translation profiles for this endpoint.

c) For **Station ID** options, configure settings as described in the Voice ports station ID options, on page 42 topic.

   Define the name and number for caller ID display.

d) For **Line Params** options, configure settings as described in the Voice ports line parameters options, on page 42 topic.

   Define line parameters for voice quality.

e) For **Tuning Params** options, configure settings as described in the Voice Ports Tuning Parameters Options topic.

   Define parameters for signaling between voice ports and other instruments.

f) For **Supervisory Disconnect** options, configure settings as described in the Voice ports supervisory disconnect options, on page 46 topic.

Define parameters for supervisory disconnect events. You can configure as many supervisory disconnect events as needed.

g) For **DID Timers** options, configure settings as described in the Voice ports DID timers options, on page 48 topic.

Define timers for Direct Inward Dialing (DID) calls.

Step 7 Click **Next**.

Step 8 In **Policy Profile Name**, enter a name for this child policy.

Step 9 In **Policy Profile Description**, enter a description for this child policy.

Step 10 Click **Save**.

The voice ports for the voice policy are successfully configured.

What to do next

Return to the main Add a Voice Policy supertask to continue configuring other subpolicies (POTS Dial Peers, SIP Dial Peers, SRST Phones) as needed.

Voice ports trunk group options

This reference topic describes the options available for configuring trunk groups for voice ports within a voice policy. These settings define how voice ports function as members of a trunk group, including hunt schemes and call limits.

Table 11: Trunk group options

Option	Description	Cisco IOS CLI Equivalent
Add New Trunk Group	Click to add a trunk group for the selected card. You can add one trunk group for a voice port.	—
Copy from Existing	Click to copy an existing trunk group to a new trunk group. In the box that appears, change the name if desired, select a trunk group, and click **Copy**.	—
Name	Name of the trunk group. The name can contain up to 32 characters.	**trunk group** *name*

Option	Description	Cisco IOS CLI Equivalent
Hunt-Scheme		**trunk group** *name* **hunt-scheme least-idle** [**both** | **even** | **odd**] **hunt-scheme least-used** [**both** | **even** | **odd**] **hunt-scheme longest-idle** [**both** | **even** | **odd**] **hunt-scheme round-robin** [**both** | **even** | **odd**] **hunt-scheme sequential** [**both** | **even** | **odd**] **hunt-scheme random**

Option	Description	Cisco IOS CLI Equivalent
	Select the hunt scheme in the trunk group for outgoing calls: • **least-idle both**—Searches for an idle channel with the shortest idle time • **least-idle even**—Searches for an idle even-numbered channel with the shortest idle time • **least-idle odd**—Searches for an idle odd-numbered channel with the shortest idle time • **least-used both**—Searches for a trunk group member that has the highest number of available channels (applies only to PRI ISDN cards) • **least-used even**—Searches for a trunk group member that has the highest number of available even-numbered channels (applies only to PRI ISDN cards) • **least-used odd**—Searches for a trunk group member that has the highest number of available odd-numbered channels (applies only to PRI ISDN cards) • **longest-idle both**—Searches for an idle odd-numbered channel with the longest idle time • **longest-idle even**—Searches for an idle channel that has the highest number of available even-numbered channels • **longest-idle odd**—Searches for an idle channel that has the highest number of available odd-numbered channels • **round-robin both**—Searches trunk group members in turn for an idle channel, starting with the trunk group member that follows the last used member • **round-robin even**—Searches trunk group member in turn for an idle even-numbered channel, starting with the trunk group member that follows the last used member	

Option	Description	Cisco IOS CLI Equivalent
	• **round-robin odd**—Searches trunk group member in turn for an idle odd-numbered channel, starting with the trunk group member that follows the last used member • **sequential-both**—Searches for an idle channel, starting with the trunk group member with the highest preference within the trunk group • **sequential-even**—Searches for an idle even-numbered channel, starting with the trunk group member with the highest preference within the trunk group • **sequential-odd**—Searches for an idle odd-numbered channel, starting with the trunk group member with the highest preference within the trunk group • **random**—Searches for a trunk group member at random and selects a channel from the member at random Default: least-used both	
Max Calls	Enter the maximum number of calls that are allowed for the trunk group. If you do not enter a value, there is no limit on the number of calls. If the maximum number of calls is reached, the trunk group becomes unavailable for more calls. • In field—Enter the maximum number of incoming calls that are allowed for this trunk group • Out field— Enter the maximum number of outgoing calls that are allowed for this trunk group Valid range for both fields: integers 0 through 1000.	**trunk group** *name* **max-calls voice** *number-of-calls* **direction** [**in** | **out**]

Option	Description	Cisco IOS CLI Equivalent
Max-Retry	Select the maximum number of outgoing call attempts that the trunk group makes if an outgoing call fails. If you do not enter a value and a call fails, the system does not attempt to make the call again. Valid range: integers 1 through 5.	**trunk group** *name* **max-retry** *attempts*
Save Trunk Group	Click to save the Trunk Group that you configured.	—

Voice ports translation profile options

This reference topic describes the options available for configuring translation profiles for calling and called numbers for voice ports within a voice policy. These settings allow you to define rules for matching and replacing number strings.

Table 12: Translation profile options

Option	Description	Cisco IOS CLI Equivalent
Add New Translation Profile	Click to add a translation profile for the selected card. You can create up to two translation profiles for this endpoint.	**voice translation-profile** *name*
Copy from Existing	Click to copy an existing translation profile to a new translation profile. In the box that appears, change the name if desired, select a called translation rule and a calling translation rule, and click **Copy**.	—
Calling	Click to configure translation rules for the number that is calling in. The Translation Rules pane displays.	**translate calling** *translation-rule-number*
Called	Click to configure translation rules for the number that is being called. The Translation Rules pane displays.	**translate called** *translation-rule-number*

Option	Description	Cisco IOS CLI Equivalent
Translation Rules pane		**voice translation-rule** *number* Match and Replace Rule: **rule** *precedence /match-pattern/ / replace-pattern/* Reject Rule: **rule** *precedence* **reject** */match-pattern/*

Option	Description	Cisco IOS CLI Equivalent
	1. Click **Add New** to create a translation rule. Alternatively, you can click **Copy From Existing** to copy an existing translation rule to a new translation rule. In the box that appears, change the name if desired, select a called translation rule and a calling translation rule, and click **Copy**. 2. In the Translation Rule Number field, enter a unique number that designates the precedence for this rule. Valid range: integers 1 through 100. 3. (Optional) To copy existing translation rules from a CSV file, click **Import**. Continue to add rules or click **Finish**. For detailed information about this file, see Translation rules CSV file, on page 78. 4. Click **Add Rule**. 5. In the Match field, enter the string that you want the translation rule to affect. Enter the string in regular expression format beginning and ending with a slash (/). For example, /^9/. To include the backslash character (\) in a match string, precede the backslash with a backslash. 6. From the Action drop-down list, select the action that the system performs for calls that match the string in the Match field. The **Reject** option causes the system to reject the call. The **Replace** option causes the system to replace the match number with a value that you specify. 7. If you select the **Replace** action, in the Replace field that displays, enter the string to which to translate the matched string. Enter the number in regular expression format beginning and ending with a slash (/). For example, //, which indicates a replacement of no string.	

Option	Description	Cisco IOS CLI Equivalent
	To include the backslash character (\) in a replace string, precede the backslash with a backslash. As an example, if you specify a match string of /^9/ and a replace string of //, the system removes the leading 9 from calls with a number that begins with 9. In this case, the system translates 914085551212 to 14085551212. **8.** Click **Save**. **9.** Add more translation rules as needed. **10.** (Optional) Click **Export** to save the translation rules that you created in a CSV file. **11.** Click **Finish** at the bottom of the pane.	

Voice ports station ID options

This reference topic describes the options available for configuring station ID information for voice ports within a voice policy. These settings define the name and number displayed for caller ID.

Table 13: Station ID Options

Option	Description	Cisco IOS CLI Equivalent
Station Name	Enter the name of the station. The station name can contain up to 50 letters, numbers, and spaces, dashes (-), and underscores (_).	**station-id name** *name*
Station Number	Enter the phone number of the station in E.164 format. The station number can contain up to 15 numeric characters.	**station-id number** *number*

Voice ports line parameters options

This reference topic describes the options available for configuring line parameters for voice ports within a voice policy. These settings control voice quality aspects such as gain, attenuation, echo cancellation, and companding type.

Table 14: Line Parameters Options

Option	Description	Cisco IOS CLI Equivalent
Gain	Enter the gain, in dB, for voice input. Valid range: –6 through 14. Default: 0	**input gain** *decibels*
Attenuation	Enter the amount of attenuation, in dB, for transmitted voice output. Valid range: –6 through 14. Default: 3.	**output attenuation** *decibels*
Echo Canceller	Select **Enable** to apply echo cancellation to voice traffic. By default, this option is enabled.	**echo-cancel** *enable*
Voice Activity Detection (VAD)	Select **Enable** to apply VAD to voice traffic. By default, this option is enabled.	**vad**
Compand Type	Select the companding standard to be used to convert between analog and digital signals in PCM systems (**U-law** or **A-law**). Default: U-Law.	**compand-type** {**u-law** | **a-law**}
Impedance	This field does not apply to PRI ISDN cards. Select the terminating impedance for calls. Default: 600r.	**impedance** {**600c** | **600r 900c** | **900r** | **complex1** | **complex2** | **complex3** | **complex4** | **complex5** | **complex6**}
Call Progress Tone	Select the locale for call progress tones.	**cptone** *locale*

Voice ports tuning parameters options

This reference topic describes the options available for configuring tuning parameters for voice ports within a voice policy. These settings define signaling parameters between voice ports and other instruments, with specific options for FXO and FXS cards.

Table 15: Tuning Parameters Options

Option	Description	Cisco IOS CLI Equivalent
Tuning Params Options for FXO Cards		
Pre Dial Delay	Enter the delay, in seconds, of the delay on the FXO interface between the beginning of the off-hook state and the initiation of DTMF signaling. Valid range: 0 through 10. Default: 1.	**pre-dial-delay** *seconds*

Option	Description	Cisco IOS CLI Equivalent
Supervisory Disconnect	Select the type of tone that indicates that a call has been released and that a connection should be disconnected: • **Anytone**—Any tone indicates a supervisory disconnect • **Signal**—A disconnect signal indicates a supervisory disconnect • **Dualtone**—A dual-tone indicates a supervisory disconnect Default: Signal.	Anytone: supervisory disconnect **anytone** Signal: supervisory disconnect Dualtone: supervisory disconnect **dualtone** {**mid-call** | **pre-connect**}
Dial Type	Select the dialing method for outgoing calls: • **pulse**—Pulse dialer • **dtmf**—Dual-tone multifrequency dialer • **mf**—Multifrequency dialer Default: dtmf.	**dial-type** {**dtmf** | **pulse** | **mf**}
Timing Sup-Disconnect	Enter the minimum time, in milliseconds, that is required to ensure that an on-hook indication is intentional and not an electrical transient on the line before a supervisory disconnect occurs (based on power denial signaled by the PSTN or PBX). Valid range: 50 through 1500. Default: 350.	**timing sup-disconnect** *milliseconds*
Battery Reversal	Battery reversal reverses the battery polarity on a PBX when a call connects, then changes the battery polarity back to normal when the far-end disconnects. Select **Answer** to configure the port to support answer supervision by detection of battery reversal. Select **Detection Delay** to configure the delay time after which the card acknowledges a battery-reversal signal, then enter the delay time in milliseconds. Valid range: 0 through 800. Default: 0 (no delay). If an FXO port or its peer FXS port does not support battery reversal, do not configure battery reversal options to avoid unpredictable behavior.	**battery-reversal** [**answer**] **battery-reversal-detection-delay** *milliseconds*

Option	Description	Cisco IOS CLI Equivalent
Timing Hookflash out	Enter the duration, in milliseconds, of hookflash indications that the gateway generates on the FXO interface. Valid range: 50 through 1550. Default: 400.	**timing hookflash-out** *milliseconds*
Timing Guard out	Enter the number of milliseconds after a call disconnects before another outgoing call is allowed. Valid range: 300 through 3000. Default: 2000.	**timing guard-out** *milliseconds*
Tuning Params Options for FXS Cards		
Timing Hookflash In	Enter the minimum and maximum duration, in milliseconds, of an on-hook condition to be interpreted as a hookflash by the FXS card. Valid range for minimum duration: 0 through 400. Default minimum value: 50. Valid range for maximum duration: 50 through 1500. Default maximum value: 1000.	**timing hookflash-in** *maximum-milliseconds minimum-milliseconds*
Pulse Digit Detection	To enable pulse digit detection at the beginning of a call, select **Yes**. Default: Yes.	**pulse-digit-detection**
Loop Length	Select the length for signaling on FXS ports (**Long** or **Short**). Default: Short.	**loop-length** [**long** | **short**]
Ring	• **Frequency**—Select the frequency, in Hz, of the alternating current that, when applied, rings a connected device. Default: 25. • **DC Offset**—Applies only if Loop Length is set to Long. Select the voltage threshold below which a ring does not sound on devices. Valid values: 10-volts, 20-volts, 24-volts, 30-volts, and 35-volts.	**ring frequency** *number* **ring dc-offset** *number*
Ringer Equivalence Number (REN)	Select the REN for calls that this card processes. This number specifies the loading effect of a telephone ringer on a line. Valid range: 1 through 5. Default: 1.	**ren** *number*

Voice ports supervisory disconnect options

This reference topic describes the options available for configuring supervisory disconnect events for voice ports within a voice policy. These settings define how the system detects and handles call disconnections, particularly for FXO cards.

Table 16: Supervisory Disconnect Options

Option	Description	Cisco IOS CLI Equivalent
Add New Supervisory Disconnect	Click to add a supervisory disconnect event.	—
Mode	Choose the mode for the supervisory disconnect event: • **Custom CPTone**—Provides options for configuring cptone detection parameters for a supervisory disconnect event • **Dual Tone Detection Params**—Provides options for configuring dual-tone detection parameters for a supervisory disconnect event	**voice class custom-cptone** *cptone-name* **voice class dualtone-detect-params** *tag*
Supervisory Name	Applies to Custom CPTone mode. Enter a name for the supervisory disconnect event. The name can contain up to 32 characters. Valid characters are letters, numbers, dashes (-), and underscores (_).	**voice class custom-cptone** *cptone-name*
Dualtone	Applies to Custom CPTone mode. Select the type of dual-tone that causes a disconnect. Options are: • Busy • Disconnect • Number Unobtainable • Out of Service • Reorder • Ringback	**dualtone** {**ringback** |**busy** | **reorder** | **out-of-service** | **number-unobtainable** | **disconnect**}
Cadence	Applies to Custom CPTone mode. Enter the cadence interval, in milliseconds, of the dual-tones that cause a disconnect. Enter the cadence as an on/off value pair, separated with a space. You can enter up to 4 on/off value pairs, separated with a space.	**cadence** *cycle-1-on-time cycle-1-off-time* [*cycle-2-on-time cycle-2-off-time* [*cycle-3-on-time cycle-3-off-time* [*cycle-4-on-time cycle-4-off-time*]]]

Option	Description	Cisco IOS CLI Equivalent
Dualtone Frequency	Applies to Custom CPTone mode. Enter the frequency, in Hz, of each tone in the dual-tone. Valid range for each tone is 300 through 3600.	**frequency** *frequency-1* [*frequency-2*]
Supervisory Number	Applies to Custom Dual Tone Detection Params mode. Enter a unique number to identify dual-tone detection parameters. Valid range: 1 through 10000.	**voice class dualtone-detect-params** *tag-number*
Cadence-Variation	Applies to Custom Dual Tone Detection Params mode. Enter the maximum time, in milliseconds, by which the tone onset can vary from the onset time and still be detected. The system multiplies the value that you enter by 10. Valid range: 0 through 200 in units of 10. Default: 10.	**cadence-variation** *time*

Option	Description	Cisco IOS CLI Equivalent
Frequency	Applies to Custom Dual Tone Detection Params mode. • Max Delay—Enter the maximum delay, in milliseconds, before a supervisory disconnect is performed after the dual-tone is detected. The system multiplies the value that you enter by 10. Valid range: 0 through 100 in units of 10. Default: 10. • Max Deviation—Enter the maximum deviation, in Hz, by which each tone can deviate from configured frequencies and be detected. Valid range: 10 through 125. Default: 10. • Max Power—Enter the power of the dual-tone, in dBm0, above which a supervisory disconnect is no detected. Valid range: 0 through 20. Default: 10. • Min Power— Enter the power of the dual-tone, in dBm0, below which a supervisory disconnect is not detected. Valid range: 10 through 35. Default: 30. • Power Twist—Enter difference, in dBm0, between the minimum power and the maximum power of the dual-tone above which a supervisory disconnect is not detected. Valid range: 0 through 15. Default: 6.	**freq-max-delay** *time* **freq-max-deviation** *hertz* **freq-max-power** *dBm0* **freq-min-power** *dBm0* **freq-power-twist** *dBm0*
Save	Click to save the supervisory disconnect information that you configured.	—

Voice ports DID timers options

This reference topic describes the options available for configuring DID (Direct Inward Dialing) timers for voice ports within a voice policy.

Table 17: DID Timers Options

Option	Description	Cisco IOS CLI Equivalent
Wait Before Wink	Enter the amount of time, in milliseconds, that the card waits after receiving a call before sending a wink signal to notify the remote side that it can send DNIS information. Valid range: 100 through 6500. Default: 550.	**timing wait-wink** *milliseconds*
Wink Duration	Enter the maximum amount of time, in milliseconds, of the wink signal for the card. Valid range: 50 through 3000. Default: 200.	**timing wait-duration** *milliseconds*
Clear Wait	Enter the minimum amount of time, in milliseconds, between an inactive seizure signal and the call being cleared for the card. Valid range: 200 through 2000. Default: 400.	**timing clear-wait** *milliseconds*
Dial Pulse Min Delay	Enter the amount of time, in milliseconds, between wink-like pulses for the card. Valid range: 0 or 140 through 5000. Default: 140.	**timing dial-pulse min-delay** *milliseconds*
Answer Winkwidth	Enter the minimum delay time, in milliseconds, between the start of an incoming seizure and the wink signal. Valid range: 110 through 290. Default: 210.	**timing answer-winkwidth** *milliseconds*

Configure POTS dial peers for a voice policy

This task defines how the system augments and manipulates calls for the POTS dial peer endpoint type within a voice policy. It allows you to configure options such as trunk groups and translation profiles to manage call flow.

When configuring POTS dial peers for a voice policy, you define specific options that control how the system handles calls for this endpoint type. This includes settings for how calls are routed through trunk groups and how calling/called numbers are translated. Trunk Group configuration for POTS Dial Peers is available beginning with Cisco IOS XE Catalyst SD-WAN Release 17.3.1a.

Before you begin

You must be in the process of adding a new voice policy or editing an existing one within Cisco SD-WAN Manager.

Follow these steps to configure POTS dial peers for a voice policy:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Unified Communications**

Step 2 Click **Add Voice Policy**, and choose **POTS Dial Peer** in the left pane.

Step 3 From the **Add POTS Dial Peer Policy Profile** drop-down list, select **Create New**.

Alternatively, you can select **Copy from Existing** to reuse settings from an existing policy profile.

Step 4 Select the types of POTS dial peers that you that you want to configure from the list of options that displays, and click **Next**.

Options include **Trunk Group** (beginning with Cisco IOS XE Catalyst SD-WAN Release 17.3.1a) and **Translation Profile**.

Step 5 Configure the selected POTS dial peer options:

a) To configure trunk groups, configure options as described in the Voice policy POTS dial peer trunk group options, on page 51 topic.

Define how POTS dial peers function within trunk groups.

You can create up to 64 trunk groups for this endpoint. If any trunk groups are already configured, they appear in the trunk groups table on this page. To edit, click **...**, and select its pencil icon; to delete, select its trash can icon. After saving, configure the priority for a trunk group by double-clicking the Priority field (1-64) and entering a number. The number you enter is the priority of the POTS dial peer in the trunk group for incoming and outgoing calls.

b) To configure translation profiles, configure options as described in the Voice policy POTS dial peer translation profile options, on page 55 topic.

Define translation rules for calling and called numbers.

You can create up to two translation profiles for this endpoint. After saving, double-click the dash (-) in the **Direction** column to select **Incoming** or **Outgoing**.

The **Incoming** selection applies the corresponding translation rule to traffic that is incoming to this endpoint. The **Outgoing** selection applies the corresponding translation rule to traffic that is outgoing from this endpoint.

Step 6 Click **Next**.

Step 7 In **Policy Profile Name**, enter a name for this child policy.

Step 8 In **Policy Profile Name**, enter a name for this child policy.

Step 9 In **Policy Profile Description**, enter a description for this child policy.

Step 10 Click **Save**.

The POTS dial peers for the voice policy are successfully configured.

What to do next

Return to the main Add a Voice Policy supertask to continue configuring other subpolicies (SIP Dial Peers, SRST Phones) as needed.

Voice policy POTS dial peer trunk group options

This reference topic describes the options available for configuring trunk groups for POTS dial peers within a voice policy. These settings define how POTS dial peers are organized into trunk groups, including hunt schemes and maximum call limits.

Table 18: POTS dial peer trunk group options

Option	Description	Cisco IOS CLI Equivalent
Add New Trunk Group	Click to add a trunk group for the selected card. You can add one trunk group for a voice port.	—
Copy from Existing	Click to copy an existing trunk group to a new trunk group. In the box that appears, change the name if desired, select a trunk group, and click **Copy**. A trunk group name whose name is preceded with "{Master}" is already associated with this voice policy. When you copy a this type of trunk group, the system reuses the existing trunk group without creating another instance of the trunk group definition. In this case, you cannot change the name.	—
Name	Name of the trunk group. The name can contain up to 32 characters.	**trunk group** *name*

Option	Description	Cisco IOS CLI Equivalent
Hunt-Scheme		**trunk group** *name* **hunt-scheme least-idle** [**both** | **even** | **odd**] **hunt-scheme least-used** [**both** | **even** | **odd**] **hunt-scheme longest-idle** [**both** | **even** | **odd**] **hunt-scheme round-robin** [**both** | **even** | **odd**] **hunt-scheme sequential** [**both** | **even** | **odd**] **hunt-scheme random**

Option	Description	Cisco IOS CLI Equivalent
	Select the hunt scheme in the trunk group for outgoing calls: • **least-idle both**—Searches for an idle channel with the shortest idle time • **least-idle even**—Searches for an idle even-numbered channel with the shortest idle time • **least-idle odd**—Searches for an idle odd-numbered channel with the shortest idle time • **least-used both**—Searches for a trunk group member that has the highest number of available channels (applies to only PRI ISDN cards) • **least-used even**—Searches for a trunk group member that has the highest number of available even-numbered channels (applies only to PRI ISDN cards) • **least-used odd**—Searches for a trunk group member that has the highest number of available odd-numbered channels (applies only to PRI ISDN cards) • **longest-idle both**—Searches for an idle odd-numbered channel with the longest idle time • **longest-idle even**—Searches for an idle channel that has the highest number of available even-numbered channels • **longest-idle odd**—Searches for an idle channel that has the highest number of available odd-numbered channels • **round-robin both**—Searches trunk group members in turn for an idle channel, starting with the trunk group member that follows the last used member • **round-robin even**—Searches trunk group member in turn for an idle even-numbered channel, starting with the trunk group member that follows the last used member	

Option	Description	Cisco IOS CLI Equivalent
	• **round-robin odd**—Searches trunk group member in turn for an idle odd-numbered channel, starting with the trunk group member that follows the last used member • **sequential-both**—Searches for an idle channel, starting with the trunk group member with the highest preference within the trunk group • **sequential-even**—Searches for an idle even-numbered channel, starting with the trunk group member with the highest preference within the trunk group • **sequential-odd**—Searches for an idle odd-numbered channel, starting with the trunk group member with the highest preference within the trunk group • **random**—Searches for a trunk group member at random and selects a channel from the member at random Default: **least-used both**	
Max Calls	Enter the maximum number of calls that are allowed for the trunk group. If you do not enter a value, there is no limit on the number of calls. If the maximum number of calls is reached, the trunk group becomes unavailable for more calls. • **In field**—Enter the maximum number of incoming calls that are allowed for this trunk group. • **Out field**— Enter the maximum number of outgoing calls that are allowed for this trunk group. Valid range for both fields: integers 0 through 1000.	**trunk group** *name* **max-calls voice** *number-of-calls* **direction** [**in** | **out**]

Option	Description	Cisco IOS CLI Equivalent
Max-Retry	Select the maximum number of outgoing call attempts that the trunk group makes if an outgoing call fails. If you do not enter a value and a call fails, the system does not attempt to make the call again. Valid range: integers 1 through 5.	**trunk group** *name* **max-retry** *attempts*

Voice policy POTS dial peer translation profile options

This reference topic describes the options available for configuring translation profiles for POTS dial peers within a voice policy. These settings allow you to define rules for matching and replacing calling and called number strings for POTS dial peer calls.

Table 19: POTS dial peer translation profile options

Option	Description	Cisco IOS CLI Equivalent
Add New Translation Profile	Click to add a translation profile for the selected POTS dial peer. You can create up to two translation profiles for this endpoint.	—
Copy from Existing	Click to copy an existing translation profile to a new translation profile. In the box that appears, change the name if desired, select a called translation rule and a calling translation rule, and click **Copy**.	—
Name	Name of the translation profile. The name can contain up to 32 characters.	**voice translation-profile** *name*
Calling	Click to configure translation rules for the number that is calling in. The Translation Rules pane displays.	**translate calling** *translation-rule-number*
Called	Click to configure translation rules for the number that is being called. The Translation Rules pane displays.	**translate called** *translation-rule-number*

Option	Description	Cisco IOS CLI Equivalent
Translation Rules pane		**voice translation-rule** *number* Match and Replace Rule: **rule** *precedence /match-pattern/ / replace-pattern/* Reject Rule: **rule** *precedence* **reject** */match-pattern/*

Option	Description	Cisco IOS CLI Equivalent
	1. Click **Add New** to create a translation rule. Alternatively, you can click **Copy From Existing** to copy an existing translation rule to a new translation rule. In the box that appears, change the name if desired, select a called translation rule and a calling translation rule, and click **Copy**. **2.** In the Translation Rule Number field, enter a unique number that designates the precedence for this rule. Valid range: integers 1 through 100. **3.** (Optional) To copy existing translation rules from a CSV file, click **Import**. Continue to add rules or click **Finish**. For detailed information about this file, see Translation rules CSV file, on page 78. **4.** Click **Add Rule**. **5.** In the Match field, enter the string that you want the translation rule to affect. Enter the string in regular expression format beginning and ending with a slash (/). For example, /^9/. To include the backslash character (\) in a match string, precede the backslash with a backslash. **6.** From the **Action** drop-down list, select the action that the system performs for calls that match the string in the Match field. The **Reject** option causes the system to reject the call. The **Replace** option causes the system to replace the match number with a value that you specify. **7.** If you select the **Replace** action, in the Replace field that displays, enter the string to which to translate the matched string. Enter the number in regular expression format beginning and ending with a slash (/). For example, //, which indicates a replacement of no string.	

Option	Description	Cisco IOS CLI Equivalent
	To include the backslash character (\) in a replace string, precede the backslash with a backslash. As an example, if you specify a match string of /^9/ and a replace string of //, the system removes the leading 9 from calls with a number that begins with 9. In this case, the system translates 914085551212 to 14085551212. **8.** Click **Save**. **9.** Add more translation rules as needed. **10.** (Optional) Click **Export** to save the translation rules that you created in a CSV file. **11.** Click **Finish** at the bottom of the pane.	

Configure SIP Dial Peers for a voice policy

This task defines how the system augments and manipulates calls for the SIP dial peer endpoint type within a voice policy. It allows you to configure options such as translation profiles, media profiles, modem pass-through, and fax protocols.

When configuring SIP dial peers for a voice policy, you define specific options that control how the system handles calls for this endpoint type. This includes settings for how calling/called numbers are translated, codec preferences, DTMF relay options, and fax capabilities.

Before you begin

You must be in the process of adding a new voice policy or editing an existing one within Cisco SD-WAN Manager.

Follow these steps to configure SIP dial peers for a voice policy:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Unified Communications**.

Step 2 Click **SIP Dial Peer**.

Step 3 From the **Add SIP Dial Peer Policy Profile** drop-down list, choose **Create New**.

Alternatively, you can select **Copy from Existing** to reuse settings from an existing policy profile.

Step 4 Select the policy types that you want to create and click **Next**.

Step 5 In the page that displays, configure options in the tabs as needed:

a) For Translation Profile options, configure settings as described in the Voice policy SIP dial peer translation profile options, on page 59 topic.

Define translation rules for calling and called numbers.

After you click **Finish** when configuring a translation profile, you can add another translation profile (up to two for this endpoint). Click **Save Translation Profile**. For each translation profile, double-click the dash (-) in the **Direction** column to select **Incoming** or **Outgoing**.

b) For Media Profile options, configure settings as described in the Voice policy SIP Dial Peer Media profile options, on page 63 topic.

Define codecs and DTMF relay options for SIP calls.

c) For Modem Pass-through options, configure settings as described in the Voice Policy SIP Dial Peer Modem Pass-Through Options topic.

Configure the modem pass-through feature for this endpoint.

d) For Fax Protocol options, configure settings as described in the Voice policy SIP dial peer fax protocol options, on page 65 topic.

Configure the fax protocol capability for this endpoint.

Step 6 Click **Next**.

Step 7 In **Policy Profile Name**, enter a name for this child policy.

Step 8 In **Policy Profile Description**, enter a description for this child policy.

Step 9 Click **Save**.

The SIP dial peers for the voice policy are successfully configured.

What to do next

Return to the main Add a Voice Policy supertask to continue configuring other subpolicies (SRST Phones) as needed.

Voice policy SIP dial peer translation profile options

This reference topic describes the options available for configuring translation rules for calling and called numbers for SIP dial peers within a voice policy. These settings allow you to define rules for matching and replacing number strings for SIP dial peer calls.

Table 20: Voice policy SIP dial peer translation profile options

Option	Description	Cisco IOS CLI Equivalent
Add New Translation Profile	Click to add a translation profile for the selected SIP dial peer. You can create up to two translation profiles for this endpoint.	**voice translation-profile** *name*
Copy from Existing	Click to copy an existing translation profile to a new translation profile. In the box that appears, change the name if desired, select a called translation rule and a calling translation rule, and click **Copy**.	—

Option	Description	Cisco IOS CLI Equivalent
Calling	Click to configure translation rules for the number that is calling in. The Translation Rules pane displays.	**translate calling** *translation-rule-number*
Called	Click to configure translation rules for the number that is being called. The Translation Rules pane displays.	**translate called** *translation-rule-number*

Option	Description	Cisco IOS CLI Equivalent
Translation Rules pane		**voice translation-rule** *number* Match and Replace Rule: **rule** *precedence /match-pattern/ / replace-pattern/* Reject Rule: **rule** *precedence* **reject** */match-pattern/*

Option	Description	Cisco IOS CLI Equivalent
	1. Click **Add New** to create a translation rule. Alternatively, you can click **Copy From Existing** to copy an existing translation rule to a new translation rule. In the box that appears, change the name if desired, select a called translation rule and a calling translation rule, and click **Copy**. **2.** In the Translation Rule Number field, enter a unique number that designates the precedence for this rule. Valid range: integers 1 through 100. **3.** (Optional) To copy existing translation rules from a CSV file, click **Import**. Continue to add rules or click **Finish**. For detailed information about this file, see Translation rules CSV file, on page 78. **4.** Click **Add Rule**. **5.** In the Match field, enter the string that you want the translation rule to affect. Enter the string in regular expression format beginning and ending with a slash (/). For example, /^9/. To include the backslash character (\) in a match string, precede the backslash with a backslash. **6.** From the Action drop-down list, select the action that the system performs for calls that match the string in the Match field. The **Reject** option causes the system to reject the call. The **Replace** option causes the system to replace the match number with a value that you specify. **7.** If you select the **Replace** action, in the Replace field that displays, enter the string to which to translate the matched string. Enter the number in regular expression format beginning and ending with a slash (/). For example, //, which indicates a replacement of no string.	

Option	Description	Cisco IOS CLI Equivalent
	To include the backslash character (\) in a replace string, precede the backslash with a backslash. As an example, if you specify a match string of /^9/ and a replace string of //, the system removes the leading 9 from calls with a number that begins with 9. In this case, the system translates 914085551212 to 14085551212. **8.** Click **Save**. **9.** Add more translation rules as needed. **10.** (Optional) Click **Export** to save the translation rules that you created in a CSV file. **11.** Click **Finish** at the bottom of the pane.	

Voice policy SIP Dial Peer Media profile options

This reference topic describes the options available for configuring media profiles for SIP dial peers within a voice policy. These settings define the codecs available for SIP trunk communication and DTMF relay options for SIP calls.

Table 21: Voice policy SIP dial peer media profile options

Option	Description	Cisco IOS CLI Equivalent
Add New Media Profile	Click to add a translation profile for the dial peer.	—
Copy from Existing	Click to copy an existing media profile to a new media profile. In the box that appears, enter a media profile number for the profile, and click **Copy**.	—
Media Profile Number	Enter a number for this SIP media profile. Valid range: Integers 1 through 10000.	**voice class codec** *tag-number*
Codec	Move from the Source list to the Target list the codecs that you want to be made available for the SIP trunk to use when communicating with the remote dial peer. Codecs in the target list are in descending order of priority, with the highest priority at the top of the list. Drag and drop items in this list to rearrange them.	**voice class codec** *tag-number* **codec preference** *value codec-type*

Option	Description	Cisco IOS CLI Equivalent
DTMF	Move from the Source list to the Target list the DTMF relay options that you want the system to use for SIP calls. Items in the Target list are in descending order of priority, with the highest priority at the top of the list. Drag and drop items in this list to rearrange them. If you want to include the **Inband** option in the Target list, it can be the only option in that list. If you want to include other options in the Target list, move the Inband option to the Source list before saving the media profile.	**dtmf-relay** {[[**sip-notify**] [**sip-kpml**] [**rtp-nte**]]}
Save	Click to save the configuration settings that you made.	—

Voice policy SIP dial peer modem pass-through options

This reference topic describes the options available for configuring modem pass-through features for SIP dial peers within a voice policy. These settings define the protocol for modem pass-through functionality.

Table 22: Voice policy SIP dial peer modem pass-through options

Option	Description	Cisco IOS CLI Equivalent
Add New Modem Pass-through	Click to add a modem pass-through for this SIP dial peer endpoint.	—
Copy from Existing	Click to copy an existing modem pass-through to a new modem pass-through profile. In the box that appears, select an existing modem pass-through, enter new name if desired, and click **Copy**.	—
Name	Name of the modem pass-through. This name is used when you copy an existing modem pass-through profile to a new one.	—

Option	Description	Cisco IOS CLI Equivalent
Protocol	Select the protocol for the modem pass-through: • **None**—Modem pass-through is disabled on the device • **NSE G.711ulaw**—Uses named signaling events (NSEs) to communicate G.711ulaw codec switchover between gateways • **NSE G.711alaw**—Uses named signaling events (NSEs) to communicate G.711alaw codec switchover between gateways	None: **no modem passthrough** NSE G.711 ulaw: **modem passthrough nse codec g711ulaw** NSE G.711 alaw: **modem passthrough nse codec g711alaw**
Save Modem Pass-Through	Click to save the configuration settings that you made.	—

Voice policy SIP dial peer fax protocol options

This reference topic describes the options available for configuring fax protocol capabilities for SIP dial peers within a voice policy. These settings define the primary and fallback fax protocols, including various T.38 fax relay and pass-through options.

Table 23: Voice policy SIP dial peer fax protocol options

Option	Description	Cisco IOS CLI Equivalent
Add New Fax Protocol	Click to add a fax protocol for the dial peer.	—
Copy from Existing	Click to copy an existing fax protocol to a new fax protocol. In the box that appears, select an existing fax protocol, enter new name if desired, and click **Copy**.	—
Name	Name of the fax protocol. This name is used when you copy an existing fax profile to a new fax profile.	—

Option	Description	Cisco IOS CLI Equivalent

Primary	Select from a set of fax protocol options. Each option is a bundled set of related fax commands.
	
	For a detailed description of each bundle, see the "Primary Fax Protocol Command Bundles" table
	
	The descriptions of the bundles include the following components:
	• **nse**—Uses NSEs to switch to T.38 fax relay mode
	• **force**—Unconditionally uses Cisco Network Services Engines (NSE) to switch to T.38 fax relay
	• **version**—Specifies a version for configuring fax speed:
	  • **0**—Configures version 0, which uses T.38 version 0 (1998–G3 faxing)
	  • **3**—Configures version 3, which uses T.38 version 3 (2004–V.34 or SG3 faxing)
	• **none**—No fax pass-through or T.38 fax relay is attempted
	• **Pass-through**—The fax stream uses one of the following high-bandwidth codecs:
	  • **g711ulaw**—Uses the G.711 ulaw codec
	  • **g711alaw**—Uses the G.711 alaw codec
		fax protocol { **none** | **pass-through** {**g711ulaw** | **g711alaw**} [**fallback none**]| **t38** [**nse** [**force**]] [**version** {**0** | **3**}] [**ls-redundancy** *value* [**hs-redundancy** *value*]] [**fallback** {**none** | **pass-through** {**g711ulaw** | **g711alaw**}}]}

Fallback	Available when the primary protocol bundle name that you selected in the Primary field begins with "T.38" or with "Fax Pass-through." Select the fallback mode for fax transmissions. This fallback mode is used if the primary fax protocol cannot be negotiated between device endpoints. For a detailed description of each option, see the "Fallback Protocol Options" table.	**fax protocol** {none | pass-through {g711ulaw | g711alaw} [**fallback none**] | t38 [nse [force]] [version {0 | 3}] [ls-redundancy value [hs-redundancy value]] [**fallback** {**none** | **pass-through** {**g711ulaw** | **g711alaw**}}]}

Option	Description	Cisco IOS CLI Equivalent
Low Speed	Available when the primary protocol bundle name that you selected in the Primary field begins with “T.38.” Specifies the number of redundant T.38 fax packets to be sent for the low-speed V.21-based T.30 fax machine protocol. Range: varies from 0 (no redundancy) to 5. Default: 0.	**ls-redundancy** *value*
High Speed	Available when the primary protocol bundle name that you selected in the Primary field begins with “T.38.” Specifies the number of redundant T.38 fax packets to be sent for high-speed V.17, V.27, and V.29 T.4 or T.6 fax machine image data. Range: varies from 0 (no redundancy) to 2. Default: 0	**hs-redundancy** *value*
Save Fax Protocol	Click to save the configuration settings that you made.	—

The following table describes the bundled sets of fax commands that are available for the Primary option when you configure the fax protocol capability for a SIP dial peer endpoint.

For low speed (ls) redundancy, the range varies from 0 (no redundancy) to 5. For high speed (HS redundancy, the range varies from 0 (no redundancy) to 2.

Table 24: Primary fax protocol command bundles

Fax Command Protocol Bundle	Description	Cisco IOS CLI Equivalent
T.38 Fax Relay Version 3	Primary fax protocol is T.38 fax relay version 3. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 3 ls-redundancy** *value* **hs-redundancy** *value* **no fax-relay sg3-to-g3**
T.38 Fax Relay Version 0	Primary fax protocol is T.38 fax relay version 0. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 ls-redundancy** *value* **hs-redundancy** *value*

Fax Command Protocol Bundle	Description	Cisco IOS CLI Equivalent
T.38 Fax Relay Version 3 NSE	Primary fax protocol is NSE based T.38 fax relay version 3. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 3 nse ls-redundancy** *value* **hs-redundancy** *value* **no fax-relay sg3-to-g3**
T.38 Fax Relay Version 3 NSE force	Primary fax protocol is NSE force option of T.38 fax relay version 3. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 3 nse force ls-redundancy** *value* **hs-redundancy** *value* **no fax-relay sg3-to-g3**
T.38 Fax Relay Version 0 NSE	Primary fax protocol is NSE option of T.38 fax relay version 0. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse ls-redundancy** *value* **hs-redundancy** *value*
T.38 Fax Relay Version 0 NSE force	Primary fax protocol is NSE force option of T.38 fax relay version 0. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse force ls-redundancy** *value* **hs-redundancy** *value*
T.38 Fax Relay Version 0 No ECM	Primary fax protocol is T.38 fax relay version 0 with ECM disabled. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable**
T.38 Fax Relay Version 0 NSE No ECM	Primary fax protocol is NSE based T.38 fax relay version 0 with ECM disabled. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable**
T.38 Fax Relay Version 0 NSE force No ECM	Primary fax protocol is NSE force option T.38 fax relay version 0 with ECM disabled. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse force ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable**

Fax Command Protocol Bundle	Description	Cisco IOS CLI Equivalent
T.38 Fax Relay Version 0 Rate 14.4 No ECM	Primary fax protocol is T.38 fax relay version 0 with ECM disabled and fax rate of 14,400 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable** **fax rate 14400**
T.38 Fax Relay Version 0 NSE Rate 14.4 No ECM	Primary fax protocol is NSE based T.38 fax relay version 0 with ECM disabled and fax rate of 14,400 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable** **fax rate 14400**
T.38 Fax Relay Version 0 NSE force Rate 14.4 No ECM	Primary fax protocol is NSE force option T.38 fax relay version 0 with ECM disabled and fax rate of 14,400 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse force ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable** **fax rate 14400**
T.38 Fax Relay Version 0 Rate 9.6 No ECM	Primary fax protocol is T.38 fax relay version 0 with ECM disabled and fax rate of 9,600 bps Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable** **fax rate 9600**
T.38 Fax Relay Version 0 NSE Rate 9.6 No ECM	Primary fax protocol is NSE based T.38 fax relay version 0 with ECM disabled and fax rate of 9,600 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable** **fax rate 9600**
T.38 Fax Relay Version 0 NSE force Rate 9.6 No ECM	Primary fax protocol is NSE force option T.38 fax relay version 0 with ECM disabled and fax rate of 9,600 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse force ls-redundancy** *value* **hs-redundancy** *value* **fax-relay ecm disable** **fax rate 9600**

Fax Command Protocol Bundle	Description	Cisco IOS CLI Equivalent
T.38 Fax Relay Version 0 Rate 14.4	Primary fax protocol is T.38 fax relay version 0 with ECM and fax rate of 14,400 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 ls-redundancy** *value* **hs-redundancy** *value* **fax rate 14400**
T.38 Fax Relay Version 0 NSE Rate 14.4	Primary fax protocol is NSE based T.38 fax relay version 0 with ECM and fax rate of 14,400 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse ls-redundancy** *value* **hs-redundancy** *value* **fax rate 14400**
T.38 Fax Relay Version 0 NSE force Rate 14.4	Primary fax protocol is NSE force option T.38 fax relay version 0 with ECM and fax rate of 14,400 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse force ls-redundancy** *value* **hs-redundancy** *value* **fax rate 14400**
T.38 Fax Relay Version 0 Rate 9.6	Primary fax protocol is T.38 fax relay version 0 with ECM and fax rate of 9,600 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 ls-redundancy** *value* **hs-redundancy** *value* **fax rate 9600**
T.38 Fax Relay Version 0 NSE Rate 9.6	Primary fax protocol is NSE based T.38 fax relay version 0 with ECM and fax rate of 9,600 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse ls-redundancy** *value* **hs-redundancy** *value* **fax rate 9600**
T.38 Fax Relay Version 0 NSE force Rate 9.6	Primary fax protocol is NSE force option T.38 fax relay version 0 with ECM and fax rate of 9,600 bps. Options for selecting the low-speed and high-speed redundancy values are available.	**fax protocol t38 version 0 nse force ls-redundancy** *value* **hs-redundancy** *value* **fax rate 9600**
None	Fax protocol is disabled.	**fax protocol none**
Fax Pass-through G711ulaw	Primary fax protocol is fax pass-through with pass-through codec set to g711ulaw.	**fax protocol pass-through g711ulaw**

Fax Command Protocol Bundle	Description	Cisco IOS CLI Equivalent
Fax Pass-through G711ulaw No ECM	Primary fax protocol is fax pass-through with pass-through codec set to g711ulaw and ECM disabled.	**fax protocol pass-through g711ulaw** **fax-relay ecm disable**
Fax Pass-through G711alaw	Primary fax protocol is fax pass-through with pass-through codec set to g711alaw.	**fax protocol pass-through g711alaw**
Fax Pass-through G711alaw No ECM	Primary fax protocol is fax pass-through with pass-through codec set to g711alaw and ECM disabled.	**fax protocol pass-through g711alaw** **fax-relay ecm disable**

The following table describes the selections that are available for the Fallback option when you configure the fax protocol capability for a SIP dial peer endpoint.

Table 25: Fallback protocol options

Fallback Fax Protocol Options	Description	Cisco IOS CLI Equivalent
None	Fallback Fax Protocol is None. All special fax handling is disabled.	**fax protocol t38** [**nse** [**force**]] [**version** {**0** | **3**}] [**ls-redundancy** *value* [**hs-redundancy** *value*]] **fallback none** **fax protocol pass-through** {**g711ulaw** | **g711alaw** } **fallback none**
Fax Pass-through G711ulaw	Fallback Fax Protocol is Fax Pass-through with pass-through codec set to g711ulaw.	**fax protocol t38** [**nse** [**force**]] [**version** {**0** | **3**}] [**ls-redundancy** *value* [**hs-redundancy** *value*]] **fallback pass-through g711ulaw**
Fax Pass-through G711alaw	Fallback Fax Protocol is Fax Pass-through with pass-through codec set to g711alaw.	**fax protocol t38** [**nse** [**force**]] [**version** {**0** | **3**}] [**ls-redundancy** *value* [**hs-redundancy** *value*]] **fallback pass-through g711alaw**

Configure SRST phones for a voice policy

This task defines how the system augments and manipulates calls for the Cisco Unified SRST phone endpoint type within a voice policy. It allows you to configure media profile options for phones operating in SRST mode.

When configuring SRST phones for a voice policy, you define the codecs and DTMF relay options that will be available for phones registered to the local gateway when in Cisco Unified SRST mode. These settings ensure proper call functionality during WAN outages or degradation.

Before you begin

You must be in the process of adding a new voice policy or editing an existing one within Cisco SD-WAN Manager

Follow these steps to configure SRST phones for a voice policy:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Unified Communications**

Step 2 Click **Add Voice Policy**, and choose **SRST Phone**.

Step 3 From the **Add SRST Phone Policy Profile** drop-down list, select **Create New**.

Alternatively, you can select **Copy from Existing** to reuse settings from an existing policy profile.

Step 4 Configure the Media Profile:

a) Click **Media Profile**, and click **Next**.
b) Click **Add New Media Profile**.
c) In the page that displays, configure options as described in the Voice policy SRST phones configuration options, on page 72 topic.

Define the codecs and DTMF relay options for SRST phones.

Step 5 Click **Next**.

Step 6 In **Policy Profile Name**, enter a name for this child policy.

Step 7 In **Policy Profile Description**, enter a description for this child policy.

Step 8 Click **Save**.

The SRST phones for the voice policy are successfully configured.

What to do next

You have now configured all subpolicies for the Add a Voice Policy supertask. Proceed to provision a device template for Unified Communications to apply this voice policy to your devices.

Voice policy SRST phones configuration options

This reference topic describes the options available for configuring SRST (Survivable Remote Site Telephony) phones within a voice policy. These settings define the media profile, including codecs and DTMF relay options, for phones operating in Cisco Unified SRST mode.

Table 26: Voice policy SRST phones configuration options

Option	Description	Cisco IOS CLI Equivalent
Medial Profile Number	Enter a number for this Cisco Unified SRST media profile. Valid range: Integers 1 through 10000.	**voice class codec** *tag-number*

Option	Description	Cisco IOS CLI Equivalent
Codec	Move from the Source list to the Target list the codecs that you want to be available for phones when they are in Cisco Unified SRST mode and communicating with other phones that are in the same site and registered to the same gateway. Codecs in the target list are in descending order of priority, with the highest priority at the top of the list. Drag and drop items in this list to rearrange them.	**voice class codec** *tag-number* **codec preference** *value codec-type*
DTMF field	Move from the source list to the target list the DTMF relay options that you want the system to use when in Cisco Unified SRST mode. Items in the target list are in descending order of priority, with the highest priority at the top of the list. Drag and drop items in this list to rearrange them. If you want to include the **Inband** option in the Target list, it can be the only option in that list. If you want to include other options in the Target list, move the Inband option to the Source list before saving the media profile.	**dtmf-relay** {[[**sip-notify**] [**sip-kpml**] [**rtp-nte**]]}
Save	Click to save the configuration settings that you made.	—

Provision a device template for unified communications

This task allows you to select UC-specific feature templates and set up the voice policy to include with a device template. This process applies the configured Unified Communications settings to your devices.

A device template bundles various feature templates and policies, enabling consistent and scalable deployment of configurations across multiple devices. This task is the final step in preparing your UC voice services for deployment.

Before you begin

Ensure that all necessary UC-specific feature templates (Voice Card, Call Routing, SRST, DSPFarm) and voice policies have been previously created and configured.

Follow these steps to provision a device template for Unified Communications:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Configuration** > **Templates**.

Step 2 Click **Device Templates**, and click **Create Template**.

Note
In Cisco vManage Release 20.7.1 and earlier releases, **Device Templates** is called **Device**.

Step 3 From the **Create Template** drop-down list, select **From Feature Template**.

Step 4 From the **Device Model** drop-down list, select the type of supported device to which you want to attach the UC-specific feature templates and map the voice policy.

Step 5 Click **Unified Communications**.

Step 6 To select UC-specific feature templates to include with the device template, perform these actions:

a) From the **Voice Card** drop-down list, select the voice card feature template.
b) From the Call Routing drop-down list, select the call routing feature template.
c) From the SRST drop-down list, select the SRST feature template.
d) From the DSPFarm drop-down list, select the DSPFarm template.

Step 7 To set up the voice policy to include with the device template, perform these actions:

a) From the **Voice Policy** drop-down list, select the voice policy.
b) Click **Mapping**.
c) From the list of endpoint types in the left pane of the screen that displays, select the type of endpoint that contains the subpolicies that you want to map to specific endpoints.
d) From the list of subpolicies that displays, click **…**, and select **Mapping** for the subpolicy that you want to map to specific endpoints.
e) In the list of endpoints that displays, select each endpoint to which you want to map the subpolicy
f) Click **Map**.
g) Click **Save**.

When you map subpolicies to endpoints, the system generates the CLI commands. See Generated CLI commands for subpolicies to endpoints mapping, on page 75.

Step 8 To create the device template, click **Create**.

A device template for Unified Communications is successfully provisioned, including the selected UC-specific feature templates and voice policy mappings.

What to do next

The device template is now ready to be attached to devices to deploy the Unified Communications configurations.

Generated CLI commands for subpolicies to endpoints mapping

This reference topic describes the CLI commands that are automatically generated when subpolicies are mapped to specific endpoints within a device template. This table provides insight into the underlying configuration applied by the Cisco SD-WAN Manager.

Table 27: Generated CLI commands for subpolicies to endpoints mapping

Endpoint	Subpolicy	Cisco IOS CLI Application Mapping	Remarks
Voice Port FXO Voice Port FXS Voice Port FXS DID Voice Port PRI ISDN POTS Dial Peer SIP Dial Peer	Translation profile	**translation-profile incoming** *profile-name* **translation-profile outgoing** *profile-name*	A translation profile policy is applied to a dial peer or a voice profile.
SRST Phone SIP Dial Peer	Media profile	voice register pool *number* **voice-class codec** *number* **dtmf-relay** {[[**sip-notify**] [**sip-kpml**] [**rtp-nte**]]}	A media profile policy includes voice class codec and DTMF relay configurations. This policy is applied to an incoming SIP dial peer, an outgoing SIP dial peer, or an SRST phone profile.
Voice Port FXO	Supervisory disconnect	voice port *number* **supervisory custom-cptone** *cptone-name* **supervisory dualtone-detect=params** *tag*	A supervisory disconnect policiy such as custom-cptone or dualtone-detect-params is applied to FXO voice interfaces.

Endpoint	Subpolicy	Cisco IOS CLI Application Mapping	Remarks
Voice Port FXO Voice Port FXS Voice Port FXS DID Voice Port PRI ISDN POTS Dial Peer	Trunk group	**trunk-group name** [*preference-num*] **voice-port** *number trunk-group name* [*preference-num*] **interface serial** *slot*/*sub-slot*/**port:**{**15** | **23**} **dial-peer voice** *tag* **pots** **trunkgroup name** *preference-num*	If more than one interface is assigned to the same trunk group, the *preference-num* value determines the order in which the trunk group uses the interfaces. A preference-num value of 1 is the highest preference, so an interface with that value is used first. A value of 64 is the lowest preference so an interface with that value is used last.
SIP Dial Peer	Modem pass-through	None: **no modem passthrough** G.711 ulaw: **modem passthrough nse codec g711ulaw** G.711 alaw: **modem passthrough nse codec g711alaw**	—
SIP Dial Peer	Fax protocol	**fax protocol** {**none** | **pass-through** {**g711ulaw** | **g711alaw**} [**fallback none**] | **t38** [**nse** [**force**]] [**version** {**0** | **3**}] [**ls-redundancy** *value* [**hs-redundancy** *value*]] [**fallback** {**none** | **pass-through** {**g711ulaw** | **g711alaw**}}]}	—

Dial peer CSV file

This reference topic describes the structure and content requirements for a Dial Peer CSV (Comma Separated Values) file. This file includes information for one or more incoming and outgoing SIP and POTS dial peers, allowing for bulk configuration or import into Cisco SD-WAN Manager.

The file must be comma delimited, and each record in the file must include each field that the following table describes, in the order shown.

Table 28: Dial peer CSV file

Field	Description
Dial Peer Tag	Number that is used to reference the dial peer.
Dial Peer Type	Type of dial peer that you are creating (**pots** or **voip**).
Direction	Direction of traffic on the dial peer (**Incoming** or **Outgoing**).
Description	Description of the dial peer.
Forward Digits	How the dial peer transmits digits in outgoing numbers: • **All**—The dial peer transmits all digits in the number. • **None**—The dial peer does not transmit digits in the number that do not match the destination pattern. • *n*—The dial peer transmits the number of right-most digits in the number that the integer *n* represents. For example, if *n* is 7 and the outgoing number is 1112223333, the dial peer transmits 2223333.
Preference	For POTS dial peers, a unique numeric value for the dial peer. If dial peers have the same match criteria, the system uses the one with the highest preference value.
Prefix	Digits to be prepended to outgoing POTS dial peer calls.
Numbering Pattern	String that the router uses to match incoming calls to the dial peer.
Dest. Address	Network address of the remote voice gateway to which calls are sent after a local outgoing SIP dial peer is matched.
Voice Port	Voice port that the router uses to match calls to the dial peer. For an outgoing dial peer, the router sends the calls that match the dial peer to this port. For an incoming dial peer, this port serves as an additional match criterion. The dial peer is matched only if a call comes in on this port.

Field	Description
Transport Protocol	For SIP dial peers, transport protocol (**TCP** or **UDP**) for SIP control signaling.

Example dial peer CSV file

```
Tag,type,Direction,Description,Forward Digits,Preference,Prefix,Pattern,Dest. Address,Voice
 Port,Transport
6545,voip,Outgoing,description To Voice Gateway,,1,,23456,ipv4:166.2.121.17,,udp
6756,voip,Outgoing,description ***Fax Number 6362-6362***,,0,,34567,ipv4:166.2.121.16,,tcp
768,voip,Outgoing, description Fire Alarm Dialer,,8,,5678,ipv4:166.2.121.19,,udp
10,pots,Incoming,,,5,,0115T,,1/0/1,
54,pots,Outgoing,,,6,,.T,,1/0/3,
23,pots,Incoming,,all,0,,76..,,1/0/4,
26,pots,Incoming,,5,1,55,9800.......,,1/0/5,
27,pots,Incoming,,5,1,55,9800.......,,0/1/5:15,
```

Translation rules CSV file

This reference topic describes the structure and content requirements for a Translation Rules CSV (Comma Separated Values) file. This file can be used to import existing translation rule information when configuring translation profiles for voice policies.

The file must be comma delimited, and each record in the file must include each field that the following table describes, in the order shown.

Table 29: Translation rules CSV file

Field	Description
Match	String that you want the translation rule to affect. The string must be in regular expression format beginning and ending with a slash (/). For example, /^9/.
Action	Action that the system performs for calls that match the string in the Match field. Valid values are: • **reject**—Causes the system to reject the call • **replace**—Causes the system to replace the match string with the value in the Replace field
Replace	If the Action field contains **replace**, this field contains the string to which to translate the matched string. Enter the number in regular expression format beginning and ending with a slash (/). For example, //, which indicates a replacement of no string. As an example, if you specify a match string of /^9/ and a replace string of //, the system removes the leading 9 from calls with a number that begins with 9. In this case, the system translates 914085551212 to 14085551212.

Example translation rules CSV file

This section provides an example of a Translation Rules CSV file, illustrating the format and content described above.

```
Match,Action,Replace
/34/,replace,/34/
/23/,reject,
/56/,replace,/100/
/16083652563/,replace,/6083652563/
```

Monitor UC operations

This task allows you to monitor the real-time statuses of lines, calls, interfaces, and related items that a device processes after enabling UC voice services for supported routers.

Monitoring UC operations provides crucial insights into the performance and status of your Unified Communications deployment, helping you verify functionality and troubleshoot issues.

Before you begin

UC voice services must be enabled for the supported routers you wish to monitor.

Follow these steps to monitor UC operations:

Procedure

Step 1 From the Cisco SD-WAN Manager menu, choose **Monitor** > **Devices**.

Note
Cisco vManage Release 20.6.1 and earlier: From the Cisco SD-WAN Manager menu, choose **Monitor** > **Network**.

Step 2 In the table of devices, select the device for which you want to monitor UC operations.

Step 3 From **Security Monitoring**, click **Real Time**.

Step 4 In **Device Options**, select one of these options:

a) **Voice Calls**: Displays information for active voice calls. See Voice calls monitoring information, on page 80 table.
b) **Voice VOIP Calls**: Displays information for active VOIP calls. See Voice VoIP calls monitoring information, on page 80 table.
c) **Voice Phone Info**: Displays information about Cisco Unified SRST registrations. See Voice phone info monitoring information, on page 81 table.
d) **Voice Controller T1 E1 Current 15 mins Stats**: Displays configuration and status information for the T1/E1 voice module that is installed in the device, compiled over the past 15 minutes. See Voice controller T1 E1 current 15 Mins stats monitoring information, on page 82 table.
e) **Voice Controller T1 E1 Total Stats**: Displays configuration and status information for the T1/E1 voice module that is installed in the device, compiled since the module last started. See Voice controller T1 E1 total stats monitoring information, on page 83 table.
f) **Voice ISDN Status**: Displays information about Layer 1 and Layer 2 status for the ISDN controller, and information about active calls. See Voice ISDN status information, on page 84 table.
g) **Voice DSPFarm SCCP CUCM Groups**: Displays detailed information about Cisco Unified Communications Manager groups that are configured for DSP farm services on a device. See Voice DSPFarm SCCP CUCM groups monitoring information, on page 84 table.

h) **Voice DSPFarm Profile**: Displays detailed information about DSP farm service profiles and media resources that are configured on the device. See Voice DSPFarm profile monitoring information, on page 85 table.
i) **Voice DSP Farm SCCP Connections**: Displays detailed information about SCCP connections between the device and Cisco Unified Communications Manager. See Voice DSPFarm SCCP connections monitoring information, on page 86 table.
j) **Voice DSPFarm Active**: Displays operational and status information about DSP farm resources that are active on the device. See Voice DSPFarm active monitoring information, on page 86 table.
k) **Interface Detail**: Displays status and statistical information for interfaces that are configured for the router.
l) **Interface Statistics**: Displays statistical information for interfaces that are configured for the router
m) **Interface T1/E1**: Displays information for the T1/E1 voice module that is installed in the device

Real-time operational data for the selected UC services is displayed, providing insight into their status and performance.

Voice calls monitoring information

This reference topic describes the information displayed when monitoring active voice calls on a device. It provides details such as call ID, voice port, codec used, and packet statistics.

Table 30: Voice calls monitoring information

Field	Description
Call ID	System assigned identifier of a telephony call leg
Voice Port	Voice port used for the call
Codec	Negotiated codec used for the call
VAD	Indicates whether VAD is enabled or disabled for the call
DSP Cannel	DSP channel used for the call
DSP Type	Type of DSP used for the call
Aborted Packets	Number of packets aborted during the call
TX Packets	Number of packets transmitted during the call
RX Packets	Number of packets received during the call
Last Updated	Date and time when the information on this page was last updated

Voice VoIP calls monitoring information

This reference topic describes the information displayed when monitoring active VoIP calls on a device. It provides details such as call ID, codec, destination address, and packet statistics for RTP connections.

Table 31: Voice VoIP calls monitoring information

Field	Description
Call ID	System assigned identifier of an RTP connection for a call leg
Codec	Negotiated codec used for the call
Destination Address	IP address of the destination of the call
Destination Port	RTP port of the destination of the call
TX Packets	Number of packets transmitted during the call
RX Packets	Number of packets received during the call
Duration (ms)	Duration of the call, in milliseconds
Last Updated	Date and time when the information on this page was last updated

Voice phone info monitoring information

This reference topic describes the information displayed when monitoring voice phone information, specifically related to Cisco Unified SRST registrations on a device. It includes details about phone pools, network identifiers, and registration states.

Table 32: Voice phone info monitoring information

Field	Description
Pool Tag	Tag number that is assigned to the Cisco Unified SRST phone pool on the device
ID Network	Identifier of the network subnet that the device uses to register phones that fallback from Cisco Unified Communications Manager to this device
Registration State	Indicates whether phones that are in Cisco Unified SRST mode are registered to this device
Dialpeer Tag	System assigned tag used by the dial peer that is assigned to the directory number of phones that are in Cisco Unified SRST mode and are registered to this device
Address	IP address of the device interface that is used for SIP SRST call control when phones fail over
Directory Number	Directory number of each phone that is in Cisco Unified SRST mode
Last Updated	Date and time when the information on this page was last updated

Voice controller T1 E1 current 15 Mins stats monitoring information

This reference topic describes the configuration and status information displayed for a T1/E1 voice module, compiled over the past 15 minutes. It includes details such as interface status, clock source, and various error statistics.

Table 33: Voice controller T1 E1 current 15 Mins stats monitoring information

Field	Description
Interface-slot-num	Slot number of the controller.
Insterface-subslot-num	Subslot number of the controller.
Interface-port-num	Port number of the controller.
Status	Status of the controller.
Type	Type of the controller.
Clock Source	Clock source used for the controller.
Line Code Violations	Number line code violations that have occurred.
Path Code Violations	Number path code violations that have occurred.
Slip Seconds	Number of slip seconds that have occurred. A slip can occur when there is a difference between the timing of a synchronous receiving terminal and the received signal.
Frame Loss Seconds	Number of seconds in which out of frame (OOF) errors have occurred.
Line Err. seconds	Number of seconds in which Line Errored Seconds (LES) have occurred. A LES is a second in which one or more Line Code Violation errors are detected.
Degraded Minutes	Number of Degraded Minutes that have occurred. A Degraded Minute is one in which the estimated error rate exceeds 1E-6 but does not exceed 1E-3.
Errored Seconds	Number of Errored Seconds that have occurred.
Bursty Errored Seconds	Number of Bursty Error Seconds that have occurred. A Bursty Error Second is a second with less than 320 and more than 1 path coding violation errors, no severely errored frame defects, and no detected incoming AIS defects.
Severely Errored Seconds	Number of Severely Errored Seconds that have occurred.
Unavailable Seconds	Number of Unavailable Seconds that have occurred. This value is calculated by counting the number of seconds that the interface is unavailable.
Last Updated	Date and time when the information on this page was last updated.

Voice controller T1 E1 total stats monitoring information

This reference topic describes the configuration and status information displayed for a T1/E1 voice module, compiled since the device last started. It includes cumulative details such as interface status, clock source, and various error statistics.

Table 34: Voice controller T1 E1 total stats monitoring information

Field	Description
Interface-slot-num	Slot number of the controller.
Insterface-subslot-num	Subslot number of the controller.
Interface-port-num	Port number of the controller.
Status	Status of the controller.
Type	Type of the controller.
Clock Source	Clock source used for the controller.
Line Code Violations	Number line code violations that have occurred.
Path Code Violations	Number path code violations that have occurred.
Slip Seconds	Number of slip seconds that have occurred. A slip can occur when there is a difference between the timing of a synchronous receiving terminal and the received signal.
Frame Loss Seconds	Number of seconds in which out of frame (OOF) errors have occurred
Line Err. seconds	Number of seconds in which Line Errored Seconds (LES) have occurred. A LES is a second in which one or more Line Code Violation errors are detected.
Degraded Minutes	Number of Degraded Minutes that have occurred. A Degraded Minute is one in which the estimated error rate exceeds 1E-6 but does not exceed 1E-3.
Errored Seconds	Number of Errored Seconds that have occurred.
Bursty Errored Seconds	Number of Bursty Error Seconds that have occurred. A Bursty Error Second is a second with less than 320 and more than 1 path coding violation errors, no severely errored frame defects, and no detected incoming AIS defects.
Severely Errored Seconds	Number of Severely Errored Seconds that have occurred.
Unavailable Seconds	Number of Unavailable Seconds that have occurred. This value is calculated by counting the number of seconds that the interface is unavailable.
Last Updated	Date and time when the information on this page was last updated.

Voice ISDN status information

This reference topic describes the information displayed when monitoring voice ISDN status on a device. It provides details about Layer 1 and Layer 2 status for the ISDN controller and information about active calls.

Table 35: Voice ISDN status information

Field	Description
Key ID	Identifier of the table row
Interface	Name of the PRI ISDN digital interface
Switch Type	Switch type used for the PRI ISDN digital interface
Layer 1 Status	Layer 1 status of the PRI ISDN digital interface
Layer 2 Status	Layer 2 status of the PRI ISDN digital interface
Active Calls	Number of active calls on the PRI ISDN digital interface
Last Updated	Date and time when the information on this page was last updated

Voice DSPFarm SCCP CUCM groups monitoring information

This reference topic describes the detailed information displayed when monitoring Cisco Unified Communications Manager groups configured for DSP farm services on a device. It includes group IDs, switchover/switchback methods, and associated profiles.

Table 36: Voice DSPFarm SCCP CUCM groups monitoring information

Field	Description
CUCM Group ID	Identifier of the Cisco Unified Communications Manager group
Description	Description of the Cisco Unified Communications Manager group
Switchover Method	Method that the primary Cisco Unified Communications Manager server in this Cisco Unified Communications Manager group uses for failover
Switchback Method	Method that the secondary Cisco Unified Communications Manager server in this Cisco Unified Communications Manager group uses to switch back after a failover
CUCM ID	Identifier of each Cisco Unified Communications Manager server in the Cisco Unified Communications Manager group
CUCM Priority	Priority in which the Cisco Unified Communications Manager servers in this Cisco Unified Communications Manager group are used

Field	Description
Profile ID	Identifier of the DSP farm profile that is registered to each Cisco Unified Communications Manager server in the Cisco Unified Communications Manager group
Reg. Name	Name of the DSP farm profile that is registered to each Cisco Unified Communications Manager server in the Cisco Unified Communications Manager group
Last Updated	Date and time when the information on this page was last updated

Voice DSPFarm profile monitoring information

This reference topic describes the detailed information displayed when monitoring DSP farm service profiles and media resources configured on a device. It includes profile IDs, service types, operational status, and resource information.

Table 37: Voice DSPFarm profile monitoring information

Field	Description
Profile ID	Identifier of the DSP farm profile.
Service ID	Type of DSP farm service that is configured for this DSP farm profile.
Service Mode	Service mode for this DSP farm profile.
Resource ID	Resource identifier for the DSP resource group in this DSP farm profile.
Admin	Status of this DSP farm profile. If this field displays DOWN, ensure that the Shutdown option is not enabled in the Profile tab of the DSPFarm feature template that defines this DSP farm.
Operation	Status of the registration of the profile with Cisco Unified Communications Manager: • ACTIVE IN PROGRESS—Profile is in the process of registering with Cisco Unified Communications Manager • DOWN—Profile is unable to register with Cisco Unified Communications Manager • ACTIVE— Profile is registered with Cisco Unified Communications Manager
App. Type	Type of application with which the DSP farm services that are provisioned on the device are associated.

Field	Description
App. Status	Status of the association of this profile with Cisco Unified Communications Manager: • app-assoc-done—Profile is associated with Cisco Unified Communications Manager • app-assoc-not-done—Profile is not associated with Cisco Unified Communications Manager
Resource Provider	Information about the mediaresource family that relates to the profile.
Provider Status	Status of the media resources that relate to the profile.
Last Updated	Date and time when the information on this page was last updated.

Voice DSPFarm SCCP connections monitoring information

This reference topic describes the detailed information displayed when monitoring SCCP connections between a device and Cisco Unified Communications Manager. It includes connection IDs, session types, codecs, and remote/local endpoint details.

Table 38: Voice DSPFarm SCCP connections monitoring information

Field	Description
Connection ID	Identifier of an SCCP connection for an active call that uses this DSP farm service
Session ID	Identifier of an SCCP session for an active call that uses this DSP farm service
Session Type	Type of DSP farm service for this SCCP connection
Mode	Mode for direction of traffic for this SCCP connection
Codec	Codec provisioned for this SCCP connection
Remote IP	IP address of the remote endpoint for this SCCP connection
Remote Port	Port number of the remote endpoint for this SCCP connection
Source Port	Port number of the local endpoint for this SCCP connection
Last Updated	Date and time when the information on this page was last updated

Voice DSPFarm active monitoring information

This reference topic describes the operational and status information displayed for active DSP farm resources on a device. It includes details about DSP identifiers, status, and packet counts for active connections.

Table 39: Voice DSPFarm active monitoring information

Field	Description
DSP	Identifier of the DSP for an active call that uses this DSP farm service
Status	Status of the DSP for an active call that uses this DSP farm service
Resource ID	Resource Identifier that is associated with the DSP that this connection uses
Bridge ID	Bridge Identifier that is associated with the DSP that this connection uses
Transmit Packets	Number of packets that this connection has transmitted
Received Packets	Number of packets that this connection has received
Last Updated	Date and time when the information on this page was last updated