Cisco Unified Communications Voice Profile

Table 1: Feature History

Feature Name	Release Information	Description
Support for Cisco Unified Communications DSP Farm Feature	Cisco IOS XE Catalyst SD-WAN Release 17.13.1a Cisco Catalyst SD-WAN Manager Release 20.13.1	This feature introduces the UC voice profile with support for the DSP farm feature.
Support for Additional Unified Communications Features	Cisco IOS XE Catalyst SD-WAN Release 17.14.1a Cisco Catalyst SD-WAN Manager Release 20.14.1	This feature adds support for the following features in the UC voice profile: • Analog Interface • Call Routing • Digital Interface • Media Profile • SRST • Server Group • Supervisory Disconnect • Translation Profile • Translation Rule • Trunk Group • Voice Global • Voice Tenant

Analog Interface

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Analog Interface feature provides options for configuring parameters for a voice card analog interface.

If you are using an NIM-2FX/4FXOP, SM-X-24FXS/4FXO, SM-X-16FXS/2FXO, or SM-X-8FXS/12FXO combo card, configure two instances of this feature, one for FXS and one for FXO. Ensure that you use the same module location for each instance. When you deploy this feature, the configuration preview displays the correct port mapping for the FXS and FXO ports.

Note If you want to remove or replace the analog interface configuration on a device:

1. Delete all configuration instances for this feature (Basic, Station ID, Line Params, Tuning Params, DID Timer, Caller ID, Connection Plar, and Associations).
2. Add one Basic configuration instance with default settings.
3. Deploy this updated interface feature configuration to the device, which resets the analog interface configuration on the device.
4. Delete this feature or configure a new one.

The following tables describe the options for configuring the Analog Interface feature.

Field	Description	Cisco IOS CLI Equivalent
Name	Enter a unique name for the analog interface configuration. The name can contain any characters.	—
Description	Enter a description of the analog interface configuration.	**description** *string*
Voice Interface Templates	Choose a group of voice interface FXO or FXS analog ports to be provisioned.	—
Use DSP	Check this check box if you want to allow local calls between analog ports on the same device to use the built-in DSPs. Default: Unchecked	**no local-bypass**

Field	Description	Cisco IOS CLI Equivalent
Module Location	Choose the slot and sub-slot location for the group of analog ports to be provisioned. For a list of supported modules, see Supported Devices for Cisco Unified Voice Services using the Workflow Library or Configuration Groups.	**voice-card** *slot/subslot*

Basic

Field	Description	Cisco IOS CLI Equivalent
Add Basic	Click to configure the basic options for the group of analog ports. You can add multiple instances of these options so that you can configure different basic options for different ports.	—
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Signal Type	Choose the signal type that indicates an on-hook or off-hook condition for calls that the ports receive. Options are **LoopStart**, **GroundStart**, and **DID**. The **DID** option is available only for FXS voice interface templates.	**signal** {**groundstart** | **loopstart**}
DID Signal Mode	Applies only if you choose **DID** for an FXS voice interface template. Choose the mode for the DID signal type Options are **Delay Dial**, **Immediate**, and **Wink Start**.	**signal did** {**delay-dial** | **immediate** | **wink-start**}
Shutdown	Enable this option to shut down ports that are not being used.	**shutdown**
Description	Enter a description of this basic configuration.	**description** *string*
Action	Click the **Recycle Bin** icon to delete the corresponding Basic options instance.	—

Station ID

Field	Description	Cisco IOS CLI Equivalent
Add Station ID	Click to configure the station name and station number from which caller ID information is sent. You can add multiple instances of these options so that you can configure different station ID options for different ports.	—
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Station Name	Enter the name of the station. The station name can contain up to 50 letters, numbers, spaces, dashes (-), and underscores (_).	**station-id name** *name*
Station Number	Enter the phone number of the station in E.164 format. For example: 4085550111 The station number can contain up to 15 numbers.	**station-id number** *number*
Action	Click the **Recycle Bin** icon to delete the corresponding Station ID options instance.	—

Line Params

Field	Description	Cisco IOS CLI Equivalent
Line Params	Click and configure options for adjusting voice and tone parameters for the port or ports. You can add multiple instances of these options so that you can configure different line parameters options for different ports.	—

Field	Description	Cisco IOS CLI Equivalent
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Gain	Enter the amount of gain, in decibels (dB), for voice input. Range: Integers –6 through 14 Default: 0	**input gain** *decibels*
Attenuation	Enter the amount of attenuation, in dB, for transmitted voice output. Range: Integers –6 through 14 Default: 0	**output attenuation** *decibels*
Echo Canceller	Choose **Enable** to apply echo cancellation to voice traffic. This option is disabled by default.	**echo-cancel** *enable*
Voice Activity Detection (VAD)	Choose **Enable** to apply VAD to voice traffic. This option is disabled by default.	**vad**
Compand Type	Choose the companding standard to be used to convert between analog and digital signals in PCM systems. Options are **U-law** and **A-law**.	**compand-type** {**u-law** | **a-law**}
Impedance	Choose the terminating impedance for calls. Default: 600r	**impedance** {**600c** | **600r 900c** | **900r** | **complex1** | **complex2** | **complex3** | **complex4** | **complex5** | **complex6**}
Call Progress Tone	Choose the locale for the call progress tone.	**cptone** *locale*
Action	Click the **Recycle Bin** icon to delete the corresponding Line Params options instance.	—

Tuning Params

Field	Description	Cisco IOS CLI Equivalent
Tuning Params	Appears only when the **Signal Type** option in the **Basic** tab is configured as **LoopStart** or **GroundStart**. Click to configure the options for various tuning parameters. You can add multiple instances of these options so that you can configure different tuning parameter options for different ports.	—
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Pre Dial Delay	Applies only to FXO voice interface templates. Enter the time, in seconds, of the delay on the FXO interface between the beginning of the off-hook state and the initiation of DTMF signaling. Range: Integers 0 through 10 Default: 1	**pre-dial-delay** *seconds*
Supervisory Disconnect	Applies only to FXO voice interface templates. Choose the type of tone that indicates that a call has been released and that a connection should be disconnected: • **Signal**: A disconnect signal indicates a supervisory disconnect • **Anytone**: Any tone 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**}

Field	Description	Cisco IOS CLI Equivalent
Dial Type	Applies only to FXO voice interface templates. Choose the dialing method for outgoing calls: • **dtmf**: Dual-tone multifrequency dialer • **pulse**: Pulse dialer • **mf**: Multifrequency dialer Default: dtmf	**dial-type** {**dtmf** | **pulse** | **mf**}
Timing Sup-Disconnect	Applies only to FXO voice interface templates. Enter the minimum time, in milliseconds (ms), 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. Range: Integers 50 through 1500 Default: 350	**timing sup-disconnect** *milliseconds*

Field	Description	Cisco IOS CLI Equivalent
Battery Reversal	Applies only to FXO voice interface templates. 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. Choose one of the following options. If you choose **Detection Delay** or **Both**, enter a value, in ms, of the delay time after which the port acknowledges a battery-reversal signal. • **Answer**: Configures the port to support answer supervision by detection of battery reversal • **Detection Delay**: Configures the delay time after which the card acknowledges a battery-reversal signal • **Both**: Configures answer and detection delay behavior Detection delay range: Integers 0 through 800 Detection delay default: 0 (no delay) **Note** If an FXO port or its peer FXS port does not support battery reversal, do not configure this battery reversal option to avoid unpredictable behavior,	**battery-reversal** [**answer**] **battery-reversal-detection-delay** *milliseconds*
Timing Hookflash Out	Applies only to FXO voice interface templates. Enter the duration, in ms, of the hookflash indications that the gateway generates on the FXO interface. Range: Integers 50 through 1550 Default: 4000	**timing hookflash-out** *milliseconds*
Timing Guard Out	Applies only to FXO voice interface templates. Enter the time, in ms, after a call disconnects before another outgoing call is allowed. Range: Integers 300 through 3000 Default: 2000	**timing guard-out** *milliseconds*

Field	Description	Cisco IOS CLI Equivalent
Timing Hookflash In	Applies only to FXS voice interface templates. Enter the minimum and maximum duration, in ms, for an on-hook condition to be interpreted as a hookflash by the FXS card. Range for minimum duration: 0 through 400 Default minimum range value: 50 Range for maximum duration: 50 through 1500 Default maximum range value: 1000	**timing hookflash-in** *maximum-milliseconds minimum-milliseconds*
Pulse Digit Detection	Applies only to FXS voice interface templates. Enable this option to enable pulse digit detection at the beginning of a call. Default: Enabled	**pulse-digit-detection**
Loop Length	Applies only to FXS voice interface templates. Choose the length for signaling on FXS ports (**Long** or **Short**). Default: Short	**loop-length** [**long** | **short**]
Ring Frequency	Applies only to FXS voice interface templates. Choose the frequency, in Hz, of the alternating current that, when applied, rings a connected device. Default: 23	**ring frequency** *number*
DC Offset	Applies only to FXS voice interface templates when **Loop Length** is set to **Long**. Choose the voltage threshold below which a ring does not sound on devices. Options are **10-volts**, **20-volts**, **24-volts**, **30-volts**, and **35-volts**.	**ring dc-offset** *number*

Field	Description	Cisco IOS CLI Equivalent
Ringer Equivalence Number (REN)	Applies only to FXS voice interface templates. Choose the REN for calls that the port processes. This number specifies the loading effect of a telephone ringer on a line. Range: Integers 1 through 5 Default: 1	**ren** *number*
Action	Click the **Recycle Bin** icon to delete the corresponding Tuning options instance.	—

DID Timer

Field	Description	Cisco IOS CLI Equivalent
Add DID Timer	Appears only to FXS voice interface templates when the **Signal Type** option in the **Basic** tab is configured as **DID**. Click to configure the options for timers for DID calls. You can add as multiple instances of these options so that you can configure different DID timer options for different ports.	—
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Wait before Wink	Enter the amount of time, in ms, that the port waits after receiving a call before sending a wink signal to notify the remote side that it can send DNIS information. Range: Integers 100 through 6500 Default: 550	**timing wait-wink** *milliseconds*
Wink Duration	Enter the maximum amount of time, in ms, of the wink signal for the port. Range: Integers 50 through 3000 Default: 200	**timing wait-duration** *milliseconds*

Field	Description	Cisco IOS CLI Equivalent
Clear Wait	Enter the minimum amount of time, in ms, between an inactive seizure signal and the call being cleared for the port. Range: Integers 200 through 2000 Default: 400	**timing clear-wait** *milliseconds*
Dial Pulse Min Delay	Enter the amount of time, in ms, between wink-like pulses for the port. Range: Integers 0, or 140 through 2000 Default: 140	**timing dial-pulse min-delay** *milliseconds*
Answer Winkwidth	Enter the minimum delay time, in ms, between the start of an incoming seizure and the wink signal. Range: Integers 110 through 290 Default: 210	**timing answer-winkwidth** *milliseconds*
Action	Click the **Recycle Bin** icon to delete the corresponding DID Timer options instance.	—

Caller ID

Field	Description	Cisco IOS CLI Equivalent
Caller ID	Click to configure the options for enabling caller ID for the port or ports. Caller ID is an analog service by which a telephone central office switch sends digital information about an incoming call. The Caller ID feature for analog FXS ports is configurable on a per-port basis to phones that are connected to analog FXS voice ports. Caller ID also is available on analog FXO ports. Caller ID-related features are based on the identity of the calling party. **Note** • These caller ID options apply only when the **Signal Type** option in the **Basic** tab is configured as **LoopStart** or **GroundStart**. • If an FXS voice port has caller-id commands configured, remove all the caller-id configurations before changing the signaling type from loop-start or ground-start to DID. • If you remove a voice port from a device after a caller ID command is configured, remove the caller ID configuration from the device. Otherwise, a voice port configuration mismatch occurs between the Cisco IOS configuration and the Cisco Catalyst SD-WAN configuration.	—
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Caller ID Mode	Choose a noncountry, standard caller ID mode for a receiving FXO or a sending FXS voice port: • **BT**: Frequency-Shift Keying (FSK) with Dual Tone Alerting Signal (DTAS) used by British Telecom • **FSK**: FSK before or during a call • **DTMF**: DTMF digits with the start and end digit codes	**caller-id mode** {**BT** | **FSK** | **DTMF**}
DTMF Start	Applies only if you choose **DTMF** for the caller ID mode. Choose the character that indicates the start of a DTMF string.	**caller-id mode** {**dtmf** {**start** | **end**}{**#** | ***** | **A** | **B** | **C** | **D**}}

Field	Description	Cisco IOS CLI Equivalent
DTMF End	Applies only if you choose **DTMF** for the caller ID mode. Choose the character that indicates the end of a DTMF string.	**caller-id mode** {**dtmf** {**start** | **end**} {**#** | ***** | **A** | **B** | **C** | **D**}}
Alerting Options	Choose the alerting method for on-hook caller ID information: • **Line-Reversal**: Sets the line-reversal alerting method for caller ID information for an on-hook (Type 1) caller ID at a sending FXS voice port and for an on-hook caller ID at a receiving FXO voice port. • **Pre-ring**: Sets a 250 ms pre-ring alerting method for caller ID information for an on-hook (Type 1) caller ID at a sending FXS and a receiving FXO voice port. • **Ring 1**, **Ring 2**, **Ring 3**, or **Ring 4**: Sets the ring-cycle method for receiving caller ID information for an on-hook (Type 1) caller ID at a receiving FXO or a sending FXS voice port.	**caller-id alerting** {**line-reversal** | **pre-ring** | **ring** {**1** | **2** | **3** | **4**}}
DSP Pre-Allocate Alerting	Applies only to FXO voice interface templates. Enable this option to statically allocate a DSP voice channel for receiving caller ID information for an on-hook (Type 1) caller ID at a receiving FXO voice port.	**caller-id alerting dsp pre-allocate**
Caller ID Block	Applies only to FXS voice interface templates. Enable this option to request blocking of caller ID information display at the far end of a call that originates from an FXS port.	**caller-id block**
Caller ID Format E911	Applies only to FXS voice interface templates. Enable this option to use the enhanced 911 format for calls that are sent on the FXS port.	**caller-id format e911**
Action	Click the **Recycle Bin** icon to delete the corresponding Caller ID options instance.	—

Connection Plar

Field	Description	Cisco IOS CLI Equivalent
Connection Plar	Click to configure the options for the connection Private Line Automatic Ringdown (PLAR). You can add multiple instances of these options so that you can configure different connection PLAR options for different ports.	—

Field	Description	Cisco IOS CLI Equivalent
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Connection Plar Pattern	Enter the PLAR extension to which the selected ports forward inbound calls.	**connection plar** *digits*
OPX	Applies only to FXO voice interface templates. Check this check box to enable Off-Premises Extension for the PLAR extension.	**connection plar opx** *digits*
Action	Click the **Recycle Bin** icon to delete the corresponding Connection Plar options instance.	—

Association

Field	Description
Association	Click to configure options for associating other configured UC voice features with the port or ports. When you associate a feature in this way, the configuration options in that feature are applied to the designated ports. You can add multiple instances of these options so that you can configure different association options for different ports.
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port 1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** to specify ports 1 through 5.
Trunk Group	Choose a configured Trunk Group feature to associate with the port.
Trunk Group Priority	Enter the priority of the trunk group. The number you enter is the priority of the POTS dial peer in the trunk group for incoming and outgoing calls. Range: Integers 1 through 64
Translation Profile	Choose a configured Translation Profile feature to associate with the port.

Field	Description
Translation Profile Direction	Choose the direction of the traffic to which to apply the selected Translation Profile feature: • **Incoming**: Applies the corresponding Translation Profile feature to traffic that is incoming to the port • **Outgoing**: Applies the corresponding Translation Profile feature to traffic that is outgoing from the port
Supervisory Disconnect	Applies only to FXO voice interface templates. Choose a configured Supervisory Disconnect feature to associate with the port.
Action	Click the **Recycle Bin** icon to delete the corresponding Association options instance.

Call Routing

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Call Routing feature provides options for configuring TDM-SIP trunking, including options for dial peers, fax operations, and modem operations. Dial peers make up a dial plan, which defines how a router routes traffic.

A plain old telephone system (POTS) dial peer defines the characteristics of a traditional telephony network connection. POTS dial peers map a dialed string to a specific voice port on the local router, normally the voice port connecting the router to the local PSTN, PBX, or telephone.

A SIP dial peer defines the characteristics of a packet network connection. SIP dial peers map a dialed string to a remote network device, such as the destination router that is connected to the remote telephony device.

Both POTS and SIP dial peers are needed to establish voice connections over a packet network.

You can configure a standalone call routing feature, or configure multiple call routing features that are mapped to different Analog Interface or Digital Interface features.

The following tables describe the options for configuring the Call Routing feature.

Field	Description
Name	Enter a unique name for the call routing configuration. The name can contain any characters.
Description	Enter a description of the call routing configuration.
Dial Peer Tag Prefix	Enter a unique number to be pretended to a dial peer tag to ensure that the dial peer tag can be uniquely identified across this feature.
Description	Enter a description of the analog or digital interface configuration to which this call routing configuration is to be associated.

Field	Description
Voice Module Location Parcel Name	Choose the Analog or Digital Interface feature to which the POTS dial peer call routing port-related configuration is to be associated.

Dial Peer

Field	Description	Cisco IOS CLI Equivalent
Add Dial Peers	Click to add a dial peer to a dial plan. Configure the following options in the **Add Dial Peer** dialog box, then click **Save**	—
Add Dial Peer Dialog Box Options		
Tag	Enter a number to be used to reference the dial peer. Range: Integers 1 through 214748364	**dial-peer voice** *number* {**pots** | **voip**}
Dial peer type	Choose the type of dial peer that you are creating. Options are **pots** and **sip**.	**dial-peer voice** *number* {**pots** | **voip**}
Direction	Choose the direction of traffic on the dial peer. Options are **incoming** and **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 the dial peer.	**description**
Number pattern	Enter the 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 following form: (ipv6:\[([0-9A-Fa-f.:])+\](:[0-9]+)?))	• Incoming: **dial-peer voice** *number* {**pots** | **voip**} **incoming called-number** *string* • Outgoing: **dial-peer voice** *number* {**pots** | **voip**} **destination-pattern** *string*

Field	Description	Cisco IOS CLI Equivalent
Forward Digits Type	Applies only when **Dial peer type** is configured as **pots** and **direction** is configured as **outgoing**. Choose 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*
Forward Digits	Applies only when you choose **Some** for **Forward Digits Type**. Enter the number of right-most digits in the outgoing number to transmit. For example, if you set this option to 7 and the outgoing number is 1112223333, the dial peer transmits 2223333.	**dial-peer voice** *number* **pots** **forward-digits** *number*
Prefix	Applies only when **Dial peer type** is configured as **pots** and **direction** is configured as **outgoing**. Enter a string to be pretended to the dial string for outgoing calls. Valid values: Integers 0 through 9 and comma (,)	**dial-peer voice** *number* **pots** **prefix** *string*
Transport Protocol	Applies only when **Dial peer type** is configured as **sip**. Choose the transport protocol for SIP control signaling. Options are **tcp** and **udp**.	**dial-peer voice** *number* **voip** **session transport** {**tcp** | **udp**}

Field	Description	Cisco IOS CLI Equivalent
Preference	Enter the preference of the dial peer. If dial peers have the same match criteria, the system uses the one with the highest preference value. Range: Integers 0 through 10 Default: 0	**dial-peer voice** *number* **voip** **preference** *value* **dial-peer voice** *number* **pots** **preference** *value*
Port	Applies only when **Dial peer type** is configured as **pots**. 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 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.	**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**
Destination Address	Applies only when **Dial peer type** is configured as **sip** and **direction** is configured as **outgoing**. 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-address* • **ipv6:***destination-address*	**session target** {**ipv4:***destination-address* | **ipv6:***destination-address* | **sip-server** | **dns:***hostname.domain*}
Dial Peer File Options		

Field	Description	Cisco IOS CLI Equivalent
Download Dial Peer List	To create or edit a dial peer CSV file, click this option to download the Cisco provided file named Dial-Peers.csv. The first time that 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 detailed information about this file, see Dial Peer CSV File.	—
Upload Dial Peer List	To import configuration information from a dial peer CSV file that you have created, click this option, choose the file to upload, then click **Save**.	—
Action	Click **Edit** to edit the corresponding Dial Peer options instance. Click **Delete** to delete the corresponding Dial Peer options.	—

Fax

Field	Description	Cisco IOS CLI Equivalent
Add Fax Protocol	Click to configure the options for the fax protocol capability for a SIP dial peer endpoint.	—
Dial Peer Range	Enter the tag or tags of the SIP dial peers for which to enable fax options. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify SIP dial peer tag 1; **1,2,3** to specify tags 1, 2, and 3; or **1-5** so specify tags 1 through 5.	—

Field	Description	Cisco IOS CLI Equivalent

Primary Protocol	Choose 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 in Configure SIP Dial Peers for a Voice Policy.
	
	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**}}]

Field	Description	Cisco IOS CLI Equivalent
Fallback Protocol	Available when the primary protocol bundle name that you selected in the **Primary Protocol** field begins with "T.38" or "Fax Pass-through." Choose 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 in Configure SIP Dial Peers for a Voice Policy.	**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**}}]}
Low Speed Redundancy	Available when the primary protocol bundle name that you selected in the **Primary Protocol** field begins with "T.38." Enter the number of redundant T.38 fax packets to be sent for the low-speed V.21-based T.30 fax machine protocol. Range: Integers 0 (no redundancy) to 5 Default: 0	**ls-redundancy** *value*
High Speed Redundancy	Available when the primary protocol bundle name that you selected in the **Primary Protocol** field begins with "T.38." Enter 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: Integers 0 (no redundancy) to 2 Default: 0	**hs-redundancy** *value*
Action	Click the **Recycle Bin** icon to delete the corresponding Fax options instance.	—

Modem

Field	Description	Cisco IOS CLI Equivalent
Add Modem Passthrough	Click to configure the modem pass-through feature for a SIP dial peer endpoint.	—
Dial Peer Range	Enter the tag or tags of the SIP dial peers for which to enable modem options. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify SIP dial peer tag 1; **1,2,3** to specify tags 1, 2, and 3; or **1-5** so specify tags 1 through 5.	—
Protocol	Choose 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.711 ulaw codec switchover between gateways • **NSE G.711alaw**: Uses named NSEs to communicate G.711 alaw codec switchover between gateways	• None: **no modem passthrough** • NSE G.711ulaw: **modem passthrough nse codec g711ulaw** NSE G.711alaw: **modem passthrough nse codec g711alaw**
Action	Click the **Recycle Bin** icon to delete the corresponding Modem options instance.	—

Association

Field	Description
Association	Click to configure the following options for associating other configured UC features with the dial plan. When you associate a feature in this way, the configuration options in that feature are applied to the designated POTS or SIP dial peers. You can add multiple instances of these options so that you can configure different association options for different ports.
Dial Peer Range	Enter the dial peer or peers to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify dial peer 1; **1,2,3** to specify dial peers 1, 2, and 3; or **1-5** to specify dial peers 1 through 5.
Media Profile Name	Choose a configured Media Profile feature to associate with the SIP dial peer.
Server Group	Choose a configured Server Group feature to associate with the SIP dial peer.
Trunk Group	Choose a configured Trunk Group feature to associate with the POTS dial peer.

Field	Description
Trunk Group Priority	Enter the priority of the trunk group, which specifies the priority of the POTS dial peer in the trunk group for incoming and outgoing calls. Range: Integers 1 through 64
Translation Profile	Choose a configured Translation Profile feature to associate with the port.
Translation Profile Direction	Choose the direction of the traffic to which to apply the selected Translation Profile feature: • **Incoming**: Applies the corresponding Translation Profile feature to traffic that is incoming to the port • **Outgoing**: Applies the corresponding Translation Profile feature to traffic that is outgoing from the port
Voice Tenant	Choose a configured Voice Tenant feature to associate with the port.
Action	Click the **Recycle Bin** icon to delete the corresponding Association options instance.

DSP Farm

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.13.1a, Cisco Catalyst SD-WAN Manager Release 20.13.1.

The DSP Farm feature provides options for configuring parameters for a Digital Signal Processor (DSP) farm.

A DSP farm is a collection of DSP resources that are available on a voice gateway for conferencing, transcoding, and MTP services. These resources can be configured and managed as out-of-box resources by Cisco Unified Call Manager through the SCCP application, and as inbox transcoder resources by Cisco Unified Border Element (CUBE).

The following tables describe the options for configuring the DSP Farm feature.

Field	Description
Name	Enter a unique name for the DSP farm configuration. The name can contain any characters.
Description	Enter a description of the DSP farm configuration.

Services

Field	Description	Cisco IOS CLI Equivalent
Services	Click to configure options for a DSP farm service. DSP farm services are conferencing, transcoding, and media termination point (MTP).	—

Field	Description	Cisco IOS CLI Equivalent
DSP Services	Enable this option to use hardware DSP resources. Disable this option if the device does not have any hardware DSP resources and you want to use the software MTP DSP service. This option is enabled by default.	—
Module Location	If **Services** is enabled, choose the slot and sub-slot location for the hardware DSP. You can configure as many module locations as needed. For a list of supported modules, see Configure UC Voice Services Using the Workflow Library or Configuration Groups.	**voice-card** *slot/subslot* **dsp service dspfarm**
SCCP	Check this check box to enable the SCCP application for provisioning conference, transcoding, and MTP services. Then configure the **Profile** options as described in the following table.	—
CUBE	Check this check box to enable the CUBE application for provisioning inbox transcoding services. Then configure the **Profile** options as described in the following table.	—
Action	Click the **Recycle Bin** icon to delete the corresponding Services options instance.	—

Profile

Field	Description	Cisco IOS CLI Equivalent
Add Profile	Click and, in the **Profile** dialog box for a DSP farm profile, configure the options that this table describes. Click **Add** in the dialog box to add the profile to the table of profiles. A profile includes options for provisioning a specific DSP farm service type, which can be transcoding, conferencing, or MTP. A profile is associated with either the SCCP application or CUBE, which invokes the resources for a service as needed. You can add multiple instances of these options so that you can configure different profile options for as needed.	—
Profile ID	Displays a unique system-generated identifier for the profile.	**profile-identifier**
Application	Choose the application with which to associate the profile. Options are **sccp** and **cube**.	**associate application** {**sccp** | **cube**}

Field	Description	Cisco IOS CLI Equivalent
Profile Type	For the **sccp** application, choose the service type to provision. Options are **transcode**, **conference**, and **mtp**. For the **cube** application, **transcode** is selected automatically as the service to provision.	**dspfarm profile** *profile-identifier* { **conference** | **mtp** | **transcode**}
Transcode Universal Profile Type	For the **transcode** profile type, check this check box to allow transcoding between codecs of any type. When this check box is unchecked, transcoding is allowed only between the G.711 codec and other codecs.	**dspfarm profile** *profile-identifier* **transcode** [**universal**]
MTP Type Hardware	For the **mtp** profile type, check this check box to have MTP translations and conversions performed by the hardware DSP resources.	**maximum session hardware**
MTP Type Software	For the **mtp** profile type, check this check box to have MTP translations and conversions performed by the router CPU.	**maximum session software**
Profile Name	For the **transcode** or **conference** profile type for the **sccp** application, or for the **cube** application, enter a unique name that you can use to identify the profile.	—

Field	Description	Cisco IOS CLI Equivalent
Codec List		**codec** *codec-name*

Field	Description	Cisco IOS CLI Equivalent

	Choose the codecs to be available for the DSP farm service that this profile defines.
	For the **mtp** profile type, you can choose **pass-through** and one other option. To change a codec, remove the current one before choosing a new one.
	The following codecs are supported:
	• For the **transcode** profile type:
	  • g711alaw
	  • g711ulaw
	  • g729abr8
	  • g729ar8
	  • g729br8
	  • g729r8
	  • g722-64
	  • ilbc
	  • iSAC
	  • opus
	  • pass-through
	• For the **conference** profile type:
	  • g711alaw
	  • g711ulaw
	  • g722r-64
	  • g729abr8
	  • g729ar8
	  • g729br8
	  • g729r8
	• For the **mtp** profile type when **MTP Type Hardware** or both **MTP Type Hardware** and **MTP Type Software** are chosen:
	  • g711ulaw
	  • g711alaw
	  • pass-through	

Field	Description	Cisco IOS CLI Equivalent
	• For the **mtp** profile type when **MTP Type Software** is chosen: • g711ulaw • g711alaw • g722-64 • g729abr8 • g729ar8 • g729br8 • g729r8 • ilbc • iSAC • pass-through	
Feature List	For the **cube** application, choose the features to enable for in-box transcoding.	—
Maximum Sessions	For the **transcode** or **conference** 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 the Cisco DSP Calculator.	**maximum sessions** *number*
MTP Maximum Hardware Sessions	If you checked **MTP Type Hardware**, enter the maximum number of hardware sessions that can be used for MPT translations and conversions. Range: Integers 1 through 4000	**maximum session hardware** *number*
MTP Maximum Software Sessions	If you checked **MTP Type Software**, enter the maximum number of CPU sessions that can be used for MRP translations and conversions. Range: Integers 1 through 6000	**maximum session software** *number*
Shutdown	Enable this option to take this profile out of service.	**shutdown**
Action	Click **Edit** to edit the corresponding Profile options instance. Click **Delete** to delete the corresponding Profile options.	—

CUCM

Field	Description	Cisco IOS CLI Equivalent
CUCM	Click to configure the Cisco Unified Communications Manager servers to which the profiles that you define register. You can configure up to 12 Cisco Unified Communications Manager servers. **Note** These options do not appear if you enable DSP services and check only the **CUBE** option.	—
Configure 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-number*/*port* where: • *interface-type*: Type of interface that the services use to register with Cisco Unified Communications Manager. The type can be a Gigabit Ethernet 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*]
IP Precedence	Enter the IP precedence value to be used by the SCCP application for IP packets. Range: 1 (lowest) through 7 (highest) Default: 5	**sccp ip precedence** *value*
Add Configure Server List	Click to display the following options for a Cisco Unified Communications Manager server: • **Server Identifier**: Unique system-generated identifier of the Cisco Unified Communications Manager server • **Server IP**: Enter the IP address of the Cisco Unified Communications Manager server	• Server identifier: *identifier-number* • Server IP: **sccp ccm** {*ipv4-address* | *ipv6-address* | *dns*} **identifier** *identifier-number* **version 7.0+**

Field	Description	Cisco IOS CLI Equivalent
Action	Click the **Recycle Bin** icon to delete the corresponding CUCM options instance.	—

CUCM Group

Field	Description	Cisco IOS CLI Equivalent
Add CUCM Group	Click and, in the **CUCM Group** dialog box, configure a Cisco Unified Communications Manager group by using the options that this table describes. Each group includes up to 4 Cisco Unified Communications Manager servers that control the DSP farm services that, in turn, are associated with the servers. Click **Add** in the dialog box when you are finished. You can add multiple Cisco Unified Communications Manager groups. **Note** These options do not appear if you enable DSP services and check only the **CUBE** option.	—
CUCM Media Resource Name	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 (_).	**associate ccm** *profile-identifier* **register** *device-name*
Profile Name	Enter the name that you entered for the DSP farm profile that is to be registered to this Cisco Unified Communications Manager group.	—

Field	Description	Cisco IOS CLI Equivalent
Server Groups Priority Order	Designate the priority in which the Cisco Unified Communications Manager servers in this Cisco Unified Communications Manager group are used. The drop-down list displays the server identifiers of the Cisco Unified Communications Manager servers that you configured. Choose the server that you want to be the primary server. This server has the highest priority. Then choose the server that you want to be a redundant server with the next highest priority. Continue in this way to choose other redundant servers. The servers in the field appear in descending order of priority, with the highest priority server appearing first. To remove a server from the field, 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*
CUCM Switchback	Choose the switchback method that the Cisco Unified Communications Manager servers in this Cisco Unified Communications Manager group use to switch back after a failover: • **guard**: Switchback occurs when active sessions are terminated gracefully or when the guard timer expires, whichever happens first • **graceful**: Switchback occurs after all active sessions terminate gracefully • **immediate**: Performs the Cisco Unified Communications Manager switchback to the higher priority Cisco Unified Communications Manager immediately when the timer expires, whether or not there is an active connection Default: graceful	**switchback method**{ **graceful** | **guard** [*timeout-guard-value*] | **immediate**}
Server Switchover	Choose the switchover method that Cisco Unified Communications Manager servers in this Cisco Unified Communications Manager group use when failing over: • **graceful**: Switchover occurs after all active sessions terminate gracefully • **immediate**: Switchover occurs immediately, whether or not there is an active connection Default: graceful	**switchover method** {**graceful** | **immediate**}

Field	Description	Cisco IOS CLI Equivalent
Keep Alive Retries	Enter the number of keepalive retries from the SCCP application to Cisco Unified Communications Manager. Range: Integers 1 to 180 Default: 3	**keepalive retries** *number*
Keep Alive Time Out	Enter the number of seconds between successive keepalive messages from the SCCP application to Cisco Unified Communications Manager. Range: Integers 1 to 180 Default: 20	**keepalive retries** *seconds*
Bind Interface	Enter the interface to bind with the Cisco Unified Communications Manager group.	**bind interface** *interface-name*
Action	Click **Edit** to edit the corresponding CUCM Group options instance. Click **Delete** to delete the corresponding CUCM Group options.	—

Digital Interface

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Digital Interface feature provides options for configuring parameters for a voice card digital interface.

Note If you want to remove or replace the digital interface configuration on a device, delete all configuration instances for this feature (Basic, ISDN Timer, ISDN Map, Shutdown, Line Params, Outgoing IE, and Associations), and add one Basic configuration instance with default settings. Then deploy this updated interface feature configuration to the device, which resets the digital interface configuration on the device. You can then delete this feature or configure a new one.

The following tables describe the options for configuring the Digital Interface feature.

Field	Description	Cisco IOS CLI Equivalent
Name	Enter a unique name for the digital interface configuration. The name can contain any characters.	—
Description	Enter a description of the digital interface configuration.	**description** *string*

Field	Description	Cisco IOS CLI Equivalent
Voice Interface Templates	Choose a group of voice interface T1 or E1 ISDN digital ports to be provisioned for the digital interface.	—
Module Location	Choose the slot and sub-slot location for the group of digital ports to be provisioned. For a list of supported modules, see Configure UC Voice Services Using the Workflow Library or Configuration Groups.	**voice-card** *slot/subslot*
Use DSP	Check this check box if you want to allow local calls between digital ports on the same device to use DSPs. Default: Unchecked	**no local-bypass**

<table>
<tr><th>Field</th><th>Description</th><th>Cisco IOS CLI Equivalent</th></tr>
<tr><td>Port and Clock Selector</td><td>Click Selected and, in the Port and Clock Selector dialog box, follow these steps to configure the clock source for each T1 or E1 port on the voice interface template that you chose:

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 voice interface template that you chose.
For each port, choose one of the following options to set 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.

This option is the default.
Network: Sets the backplane clock or the system oscillator clock as the module clock source.
Primary Clock: Sets the port to be a primary clock source.
Secondary Clock: Sets the port to be a secondary clock source.

You can chose 1 port to be the primary clock source and 1 port to be the secondary clock source. Choosing a primary clock source does not require you to choose a secondary clock source.
Click Save.
</td><td>controller {t1 | e1} slot/sub-slot/number

clock source {network | line | line primary | line secondary}</td></tr>
</table>

Basic

Field	Description	Cisco IOS CLI Equivalent
Add Basic	Click to configure basic options for the group of digital ports. You can add as multiple instances of these options so that you can configure different basic options for different ports.	—
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Time slots	Enter the number of time slots of the interface. 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** *timeslots-range* [**voice-dsp**]
Line Termination	Applies to E1 voice interface templates only. Choose the termination type for the interface: • **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**}

Field	Description	Cisco IOS CLI Equivalent
Cable Length Type	Applies to T1 voice interface templates only. Choose the cable length type for the interface: • **Long**: Applies to cables that are longer than 660 feet (201.2 m). Attenuates the pulse from the transmitter by using pulse equalization and line build-out. This value is the default • **Short**: Applies to cables that are 660 feet (201.2 m) or less in length. Sets transmission attenuation for the cable.	**controller t1** *slot/sub-slot/port* **cablelength** {**short | long**}
Cable Length	Applies to T1 voice interface templates only. Choose the length of the cable for the interface: • For a **Long** cable length, enter the loss value, decibels (dB). Options are **–7.5**, **–15**, **–22.5**, and **0**. The default value is 0. • For a **Short** cable length (up to 660 feet (201.2 m), enter the value that most closely exceeds the length of the cable. For example, if the cable length is 180 feet (55 m) enter **220**.	**controller t1** *slot/subslot/port* **cablelength** {**[short [110ft | 220ft | 330ft | 440ft | 550ft | 660ft]] [long [-15db |-22db |-7.5db | 0db]]**}

Field	Description	Cisco IOS CLI Equivalent
Line Code	Choose the line code type for the interface. For a T1 voice interface template: • **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 voice interface template: • **amiami**: Use alternate mark inversion as the line code type • **hdb3**: Use high-density bipolar 3-zero as the line code type (default)	**linecode** {**ami** | **b8zs** | **hdb3**
Framing	Choose the frame type for the interface. For a T1 voice interface template: • **esf**: Extended super frame (default) • **sf**: Super frame For an E1 voice interface template: • **ccr4**: 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**]
Framing Australia	Applies to E1 voice interface templates only. Enable this option to use the Australia framing type.	**controller e1** *slot*/*sub-slot*/*port* **framing** [**crc4** | **no-crc4**] **australia**
Network Side	Enable this option to have the device to which this configuration is to be associated use the standard PRI network-side interface. Default: Disabled	**interface serial** *slot*/*sub-slot*/**port:**{**15**| **23**} **isdn protocol-emulate** [**network** | **user**]

Field	Description	Cisco IOS CLI Equivalent
Switch Type	Choose the ISDN switch type for this interface: • **primary-qsig**: Supports QSIG signaling according to the Q.931 protocol. Network side functionality is assigned with the **isdn protocol-emulate** command. • **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-net5**: NET5 ISDN PRI switch types for Asia, Australia, and New Zealand. ETSI-compliant switches for Euro-ISDN E-DSS1 signaling system. • **primary-ni**: National ISDN switch type. • **primary-ntt**: Japanese NTT ISDN PRI switches.	**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**]
Delay Connect Timer	Enter the duration, in ms, to delay connect a PRI ISDN hairpin call. Range: Integers 0 through 200 Default: 20	**voice-port** *slot*/*sub-slot*/**port:**{**15** | **23**} **timing delay-connect** *value*
Action	Click the **Recycle Bin** icon to delete the corresponding Basic options instance.	—

ISDN Timer

Field	Description	Cisco IOS CLI Equivalent
Add ISDN Timer	Click to configure options for the ISDN timer for the interface. You can add multiple instances of these options so that you can configure different ISDN timer options for different ports.	—
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—

Field	Description	Cisco IOS CLI Equivalent
ISDN Timer and Value		**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*

Field	Description	Cisco IOS CLI Equivalent

	Click to configure an ISDN timer. Configure the following fields in the **ISDN Timer and Value** dialog box, then click **Save**.
	• **Port Range**: Displays the ports that you chose
	• **ISDN Timer**: Displays the ISDN timers that you can provision.
	• **Value**: Enter the value, in ms, for the corresponding ISDN timer:
	  • For the **T200** ISDN timer:
	    • Range: 400 through 400000
	    • Default for all switch types: 1000
	  • For the **T203** ISDN timer:
	    • Range: Integers 400 through 400000
	    • Default for QSIG, ETSI Net5, and DMS-100 switch types: 10000
	    • Default for 4ESS, 5ESS, NTT, and NI switch types: 30000
	  • For the **T301** ISDN timer:
	    • Range: 180000 through 86400000
	    • Default for NTT and ETSI Net5 switch types: 180000
	    • Default for other switch types: 300000
	  • For the **T303** ISDN timer:
	    • Range: 400 through 86400000
	    • Default for QSIG switch type: 6000
	    • Default for other switch types: 4000
	  • For the **T306** ISDN timer:
	    • Range: 400 through 86400000
	    • Default for all switch types: 30000
	  • For the **T309** ISDN timer:
	    • Range: 0 through 86400000
	    • Default for all switch types when network side configuration is false (User): 90000	

Field	Description	Cisco IOS CLI Equivalent
	• Default for all switch types when network side configuration is true (Network): 5000 • For the **T310** ISDN timer: • Range: 400 through 400000 • Default for NI , 4ESS and 5ESS switch types when network side configuration is false (User): 30000 • Default for NI, 4ESS, and 5ESS switch types when network side configuration is true (Network): 10000 • Default for ETSI Net5 switch types: 4000 • Default for QSIG switch type: 120000 • Default for NTT switch type: 3000 • Default for DMS-100 switch type when network side configuration is false (User): 1000 • Default for DMS-100 switch type when network side configuration is true (Network): 4000 • Default for other switch types: 4000 • For the **T321** ISDN timer: • Range: 0 through 86400000 • Default for ETSI Net5 switch type: 30000 • Default for other switch types: 40000	
Action	Click the **Recycle Bin** icon to delete the corresponding ISDN Timer options instance.	—

ISDN Map

Field	Description	Cisco IOS CLI Equivalent
Add ISDN Map	Click to configure the following options to override with custom values the default ISDN type and plan that the router generates. You can add multiple instances of these options so that you can configure different ISDN mapping options for different ports.	—

Field	Description	Cisco IOS CLI Equivalent
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Digit Range	Enter a digit or range of digits to map to ISDN telephone numbers that are used internally	**isdn map address {{** *address* | *reg-exp*} **plan** *plan* **type** *type* | **transparent}**
Plan	Choose an ISDN numbering plan: • **data**: X.121 data numbering plan • **isdn**: E.164 ISDN/Telephony numbering plan • **national**: Number called to reach a subscriber in the same country, but outside the local network • **privacy**: Private numbering plan • **reserved/*extension***: Reserved for the extension	**isdn map address {{** *address* | *reg-exp*} **plan** *plan* **type** *type* | **transparent}**
Type	Choose an ISDN number type: • **abbreviated**: Abbreviated representation of the complete number as supported by your network • **international**: Number called to reach a subscriber in another country • **national**: Number called to reach a subscriber in the same country, but outside the local network • **reserved/5**: Reserved for the extension	**isdn map address {{** *address* | *reg-exp*} **plan** *plan* **type** *type* **}**
Action	Click the **Recycle Bin** icon to delete the corresponding ISDN Map options instance.	—

Shutdown

Field	Description	Cisco IOS CLI Equivalent
Add Shutdown	Click to configure to disable or enable the controller, serial interface, or voice port that is associated with the interface port. You can add multiple instances of these options so that you can configure different shutdown options for different ports.	—
Port ID	Enter the port or ports to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—
Controller	Enable this option to shut down a controller.	**controller e1/t1** *slot*/*sub-slot*/*port* **shutdown**
Serial	Check this check box to shut down a serial interface.	**interface serial** *slot*/*sub-slot*/**port:**{ **15** | **23**} **shutdown**
Voice Port	Check this check box to shut down a voice port.	**voice-port** *slot*/*sub-slot*/**port:**{ **15** | **23**} **shutdown**
Action	Click the **Recycle Bin** icon to delete the corresponding Shutdown options instance.	—

Line Params

Field	Description	Cisco IOS CLI Equivalent
Add Line Params	Click to configure options for adjusting various line parameters for the port or ports. You can add multiple instances of these options so that you can configure different line parameters for different ports .	—
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—

Field	Description	Cisco IOS CLI Equivalent
Gain	Enter the amount of gain, decibels (dB), for voice input. Range: Integers –6 through 14 Default: 0	**input gain** *decibels*
Attenuation	Enter the amount of attenuation, decibels (dB), for transmitted voice output. Range: Integers –6 through 14 Default: 3	**output attenuation** *decibels*
Echo Canceller	Choose **Enable** to apply echo cancellation to voice traffic. This option is disabled by default.	**echo-cancel** *enable*
Voice Activity Detection	Choose **Enable** to apply VAD to voice traffic. This option is disabled by default.	**vad**
Compand Type	Choose the companding standard to be used to convert between analog and digital signals in PCM systems (**U-law** or **A-law**). The default is **U-law**.	**compand-type** {**u-law** | **a-law**}
Call Progress Tone	Choose the locale for the call progress tone.	**cptone** *locale*
Action	Click the **Recycle Bin** icon to delete the corresponding Line Params options instance.	—

Outgoing IE

Field	Description	Cisco IOS CLI Equivalent
Add Outgoing IE	Click to configure the following options for the outgoing Information Element. You can add multiple instances of these options so that you can configure outgoing Information Element options for different ports.	—
Port Range	Enter the port or ports within the voice interface template to which the following option applies. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** so specify ports 1 through 5.	—

Field	Description	Cisco IOS CLI Equivalent
Type	Choose one or more of the following options to specify the Information Elements to pass in outgoing ISDN messages: To remove an option from the field, click its **X** icon. • **called-number**: Indicates the outgoing call number • **called-subaddr**: Indicates the subaddress of the outgoing call • **caller-number**: Indicates the incoming call number • **caller-subaddr**: Indicates the subaddress of the incoming call • **connected-number**: Indicates the number of the remaining caller if a disconnect occurs during a conference • **connected-subaddr**: Indicates the subaddress of the remaining caller if a disconnect occurs during a conference • **display**: Provides information about the text display • **extended-facility**: Provides information about extended facility requests • **facility**: Provides information about facility requests • **high-layer-compat**: Provides information about higher layer compatibility • **low-layer-compat**: Provides information about lower layer compatibility • **network-facility**: Provides information about network facility requests • **notify-indicator**: Provides information about notifications • **progress-indicator**: Provides information about the call in progress • **redirecting-number**: Indicates the number that is redirecting the call • **user-user**: Provides information about the users at either end of the call	**isdn outgoing ie** *type*
Action	Click the **Recycle Bin** icon to delete the corresponding Outgoing IE options instance.	—

Associations

Field	Description
Association	Click to configure options for associating other configured UC voice features with the port or ports. When you associate a feature in this way, the configuration options in that feature are applied to the designated ports. You can add multiple instances of these options so that you can configure different association options for different ports.

Field	Description
Port Range	Enter the port or ports within the voice interface template to which these options apply. Enter a number, a comma separated string of numbers, or a range of numbers separated with a hyphen. For example, enter **1** to specify port 1; **1,2,3** to specify ports 1, 2, and 3; or **1-5** to specify ports 1 through 5.
Trunk Group	Choose a configured Trunk Group feature to associate with the port.
Trunk Group Priority	Enter the priority of the trunk group. The number you enter is the priority of the POTS dial peer in the trunk group for incoming and outgoing calls. Range: Integers 1 through 64
Translation Profile	Choose a configured Translation Profile feature to associate with the port.
Translation Profile Direction	Choose the direction of the traffic to which to apply the selected Translation Profile feature: • **Incoming**: Applies the corresponding Translation Profile feature to traffic that is incoming to the port • **Outgoing**: Applies the corresponding Translation Profile feature to traffic that is outgoing from the port
Supervisory Disconnect	Applies only to FXO voice interface templates. Choose a configured Supervisory Disconnect feature to associate with the port.
Action	Click the **Recycle Bin** icon to delete the corresponding Associations options instance.

Media Profile

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Media Profile feature provides options for configuring the codecs to be available for the SIP trunk communication with remote dial peers, and DTMF relay options to use for SIP calls. You can configure multiple Media Profile features.

The following table describes the options for configuring the Media Profile feature.

Field	Description	Cisco IOS CLI Equivalent
Name	Enter a unique name for the media profile configuration. The name can contain any characters.	—
Description	Enter a description of the media profile configuration.	—

Field	Description	Cisco IOS CLI Equivalent
Media Profile Number	Enter a number for this SIP media profile. Range: Integers 1 through 10000	**voice class codec** *tag-number*
DTMF Target	Choose the DTMF relay options that you want the system to use for SIP calls: • **rtp-nte**: Real-Time Transport Protocol (RTP) Named Telephone Events (NTE). An in-band DTMF relay method, which uses RTP Named Telephony Event (NTE) packets to carry DTMF information instead of voice. • **sip-notify**: A Cisco proprietary out-of-band DTMF relay mechanism that transports DTMF signals using SIP NOTIFY messages. • **sip-kpml**: Keypad Markup Language (KPML) is used to indicate DTMF tones in SIP messaging. It transmits DTMF tone indications via SIP NOTIFY messages Choose the option that you want to have the highest priority. Then choose the option that you want to have the next highest priority. Continue in this way to choose a third option. The options in the field appear in descending order of priority, with the highest priority option appearing first. To remove an option from the field, click its **X** icon. To change the priority order of options, remove the options and add them back in the desired order.	**dtmf-relay** {[[**sip-notify**] [**sip-kpml**] [**rtp-nte**]]}
Codec List	Choose the codecs that you want to be made available for the SIP trunk to use when communicating with the remote dial peer. Choose the codec that you want to have the highest priority. Then choose the codec that you want to have the next highest priority. Continue in this way to choose other codecs. The codecs in the field appear in descending order of priority, with the highest priority option appearing first. To remove a codec from the field, click its **X** icon. To change the priority order of codecs, remove the codecs and add them back in the desired order.	**voice class codec** *tag-number* **codec preference** *value codec-type*

SRST

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The SRST feature provides options for configuring 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 (device) so that they continue to function and provide PSTN breakout services without requiring the WAN resources that are no longer available.

The following tables describe the options for configuring the SRST feature.

Field	Description
Name	Enter a unique name for the SRST configuration. The name can contain any characters.
Description	Enter a description of the SRST configuration.

Global

Field	Description	Cisco IOS CLI Equivalent
Max Phones	Enter the number of phones that the system can register to the local gateway when the gateway is in Cisco Unified SRST mode.	**voice register global** **max-pool** *max-voice-register-pools*
Max Directory Numbers	Enter the number of directory numbers that the gateway supports when the gateway is in Cisco Unified SRST mode. The maximum values that you can enter depend on the device that you are configuring.	**voice register global** **max-dn** *max-directory-numbers*
Music on Hold	Enable this option to play music on hold on endpoints when a caller is on hold and the gateway is in Cisco Unified SRST mode.	—
Music on Hold File	Enter the path and filename of the audio file for music on hold. The file must be in the system flash and must be in the .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*
System Message	Enter a message that displays on endpoints when Cisco Unified SRST mode is in effect.	**voice register global** **system message** *string*

Phone Profile

Field	Description	Cisco IOS CLI Equivalent
Add New Phone Pool Profile	Click to configure the options for providing registration permission control and certain dial-peer attributes that are applied to the dynamically created VoIP dial peers when SIP phone registrations match the pool You can add multiple instances of these options so that you can configure different options for different pool tags.	—

Field	Description	Cisco IOS CLI Equivalent
Pool Tag	Enter the unique sequence number of the set of SIP phones to be configured. Range: Integers 1 to the number of phones that you configured with the **Max Phones** option.	**voice register pool** *pool-tag*
IPv4/6 Network Access	Enter the IPv4 or IPv6 prefix of the network that contains the set of SIP phones to be configured.	**voice register pool** *pool-tag* **id** [**network** *address* **mask** *mask*]
Action	Click the **Recycle Bin** icon to delete the corresponding Phone Profile options instance.	—

Call Forward

Field	Description	Cisco IOS CLI Equivalent
Add New Call Forward	Click to configure the options for forwarding incoming voice calls to SIP phones. You can add multiple instances of these options so that you can configure different options for different pool tags.	—
Pool Tag	Enter one of the pool tags that you defined for the phone profile to associate with call forwarding actions.	—
Action	Choose the situation that causes a directory number to be forwarded to another directory number when the gateway is in SRST mode: • **busy**: Forwards a call to another directory number when a phone is busy • **all**: Forwards all incoming calls to another directory number • **noan**: Forwards a call to another directory number when no answer is received after a configured timeout	**call-forward b2bua all** {*number* | **busy** *number* | **noan** *number* [**timeout** *seconds*]}
Digit String	Enter the directory number to which forwarded calls are sent.	**call-forward b2bua all** {*number* | **busy** *number* | **noan** *number* [**timeout** *seconds*]}
Timeout	For a call forward **noan** action, enter the number of seconds that a call rings with no answer after which the call is forwarded to the directory number that the**Digit String** option defines. Range: Integers 3 to 60000 Default: 20	**call-forward b2bua noan** {*number* [**timeout** *seconds*]}

Field	Description	Cisco IOS CLI Equivalent
Action	Click the **Recycle Bin** icon to delete the corresponding Call Forward options instance.	—

Association

Field	Description
Association	Click to configure options for associating other configured UC voice features with the port or ports. When you associate a feature in this way, the configuration options in that feature are applied to the designated set of SIP phones. You can add as multiple instances of these options so that you can configure different association options for different phone pools.
Pool Tag	Enter the unique sequence number of the set of SIP phones to be configured.
Media Profile	Choose a configured Media Profile feature to associate with the phone pool profile.
Translation Profile	Choose a configured Translation Profile feature to associate with the port.
Translation Profile Direction	Choose the direction of the traffic to which to apply the selected Translation Profile feature: • **Incoming**: Applies the corresponding Translation Profile feature to traffic that is incoming to the port • **Outgoing**: Applies the corresponding Translation Profile feature to traffic that is outgoing from the port
Action	Click the **Recycle Bin** icon to delete the corresponding Association options instance.

Server Group

Minimum supported releases: .

Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Server Group feature lets you configure a group of up to five destination SIP servers for an outbound dial peer.

When a call matches a dial peer that is configured with a server group, the destination is selected from the list of servers based on the Server Group feature configuration.

When you associate a server group with an outbound dial peer, the session target information in the dial plan must point to the provisioned server group

The following tables describe the options for configuring the Server Group feature

Field	Description
Name	Enter a unique name for the server group configuration. The name can contain any characters.

Field	Description
Description	Enter a description of the server group configuration.

Basic Configuration

Field	Description	Cisco IOS CLI Equivalent
Server Group ID	Enter a unique identification number for this server group. Range: Integers 1 through 10000	**voice class server-group** *server-group-id*
Description	Enter a description of this server group.	**description** *string*
Hunt Scheme	Choose the hunt method for the order of selection of target server IP addresses, which are IP addresses of the servers in the server group, for setting up outgoing calls. (Server addresses are configured as described in the following **Address List** table.) Options are: • **none**: No hunt scheme defined. If a hunt scheme is not defined, an available IP address of the highest **Preference** value is selected. (The preference is configured as described in the following **Address List** table.) • **round-robin**: Searches IP addresses in turn for the next available server, starting with the server that follows the last used member of the server group.	**hunt-scheme round-robin**
Shutdown	Enable this option to put this server group in shutdown mode, which causes the outbound SIP dial peers that use this server group to be out of service.	—

Address List

Field	Description	Cisco IOS CLI Equivalent
Add Address List	Click to configure options for adding a server to the server group. You can add up to 5 instances of these options that you can add up to 5 servers to the server group.	—
IPv4/6 Address	Enter the IPv4 or IPv6 address of the server.	**ipv4|ipv6**} *address*
Port	Enter the number of the server port that is listening for SIP calls.	**port** *port*

Field	Description	Cisco IOS CLI Equivalent
Preference	Applies only if the **Hunt Scheme** Basic Configuration option is set to **none**. Choose the order of selection preference of the server for the setting up of outgoing calls. Range: Integers 0 (highest preference) through 5 (lowest preference) Default: 0	**preference** *preference-order*
Action	Click the **Recycle Bin** icon to delete the corresponding Address List options instance.	—

Hunt Stop Rules

Field	Description	Cisco IOS CLI Equivalent
Add Hunt Stop Rules	Click to configure options for configuring a hunt stop rule. This rule stops hunting for servers in the server group based on configured SIP response codes. You can add up to 10,000 instances of these options so that you can configure different hunt stop rules for different response codes.	—
Rule ID	Enter the identifier of the hunt stop rule. Range: Integers 1 through 1000	**huntstop** *rule-tag* **resp-code** *from_resp_code* to *to_resp_code*
Response Code Start	Enter the first SIP response code in a range of codes for the hunt stop rule. Range: Integers 400 through 599	**huntstop** *rule-tag* **resp-code** *from_resp_code* to *to_resp_code*
Response Code End	Enter the last SIP response code in a range of codes for the hunt stop rule. For example, huntstop 1 resp-code 401. Range: Integers 400 through 599	**huntstop** *rule-tag* **resp-code** *from_resp_code* to *to_resp_code*
Action	Click the **Recycle Bin** icon to delete the corresponding Hunt Stop Rules options instance.	—

Supervisory Disconnect

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Supervisor Disconnect feature provides options for configuring supervisory disconnect events.

The following tables describe the options for configuring the Supervisory Disconnect feature.

Field	Description
Name	Enter a unique name for the supervisory disconnect configuration. The name can contain any characters.
Description	Enter a description of the supervisory disconnect configuration.

Custom CPTone

Field	Description	Cisco IOS CLI Equivalent
Add Custom CPTone	Click to configure options for custom call progress tones for a supervisory disconnect event. You can add as multiple instances of these options so that you can configure different dual-tone options for a supervisory name.	—
Supervisory Name	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	Choose the type of dual-tone that causes a supervisory disconnect event: • **Busy** • **Disconnect** • **Number Unobtainable** • **Out of Service** • **Reorder** • **Ringback**	**dualtone {ringback | busy | reorder | out-of-service | number-unobtainable | disconnect}**
Cadence	Enter the cadence interval, in ms, of the dual-tones that cause a supervisory disconnect event. 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 spaces.	**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*]]]
Dualtone Frequency	Enter the frequency, in Hz, for each tone in the dual tone. Range for each tone: Integers 300 through 3600	**frequency** *frequency-1 [frequency-2]*
Action	Click the **Recycle Bin** icon to delete the corresponding Custom CPTone options instance.	—

Dual Tone Detection Params

Field	Description	Cisco IOS CLI Equivalent
Add Dual Tone Detection Params	Click to configure the following options for dual-tone detection parameters for a supervisory disconnect event. You can add multiple instances of these options.	—
Supervisory Number	Enter a unique number to identify dual-tone detection parameters. Range: Integers 1 through 10000	**voice class dualtone-detect-params** *tag-number*
Cadence-Variation	Enter the maximum time, in ms, by which the tone onset can vary from the specified onset time and still be detected. The system multiplies the value that you enter by 10. Range: Integers 0 through 200 (0 through 2000 ms) Default: 10 (100 ms)	**cadence-variation** *time*
Frequency Max Delay	Enter the maximum delay, in milliseconds, before a supervisory disconnect occurs after the dual-tone is detected. The system multiplies the value that you enter by 10. Range: Integers 0 through 100 (0 through 1000 ms) Default: 10 (100 ms)	**freq-max-delay** *time*
Frequency Max Deviation	Enter the maximum deviation, in Hz, by which each tone can deviate from configured frequencies and be detected. Range: Integers 0 through 125 Default: 10	**freq-max-deviation** *hertz*
Frequency Max Power	Enter the power of the dual-tone, in dBm0, above which a supervisory disconnect is not detected. Range: Integers 0 through 20 Default: 10	**freq-max-power** *dBm0*
Frequency Min Power	Enter the power of the dual-tone, in dBm0, below which a supervisory disconnect is not detected. Range: Integers 0 through 35 Default: 3	**freq-min-power** *dBm0*
Frequency Power Twist	Enter the difference, in dBm0, between the minimum power and the maximum power of the dual-tone above which a supervisory disconnect is not detected. Range: Integers 0 through 15 Default: 6	**freq-power-twist** *dBm0*

Field	Description	Cisco IOS CLI Equivalent
Action	Click the **Recycle Bin** icon to delete the corresponding Dual Tone Detection Params options instance.	—

Translation Profile

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Translation Profile feature provides options for configuring translation profiles.

The following table describes the options for configuring the Translation Profile feature.

Note You must configure the Translation Rule feature before you can configure the Translation Profile feature.

Field	Description
Name	Enter a unique name for the translation profile configuration. The name can contain any characters.
Description	Enter a description of the translation profile configuration.

Basic Configuration

Field	Description	Cisco IOS CLI Equivalent
Name	Enter a unique name for the translation profile. If you do not enter a name, “Translation Profile” is used as the name.	—
Add Translation Profile Configuration	Click to configure options for mapping rules that are defined by the Translation Rule feature for calling and called numbers. You can add up to 2 instances of these options, one instance for the **calling** call type and one for the **called** call type.	—
Select Call Type	Choose the type of call to which to map a translation rule set: • **calling**: Maps a translation rule set for the number that is calling in • **called**: Maps a translation rule set for the number that is being called	• Calling: **translate calling** *translation-rule-number* • Called: **translate called** *translation-rule-number*

Field	Description	Cisco IOS CLI Equivalent
Select Translation Rule	Choose a provisioned Translation Rule feature to associate with to the call type that you chose.	—
View Rule	Click to view the translation rule that you chose.	—

Translation Rule

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Translation Rule feature provides options for creating translation rules for calling and called numbers. You can create up to 100 translation rules for a card.

The Translation Rule feature is used to match called party or calling party numbers for configured digit manipulation. Because the Translation Rule feature can contain a set of rules, it can be used to match one or more patterns of numbers and have each pattern manipulated in a different way.

The following table describes the options for configuring the Translation Rule feature.

Field	Description
Name	Enter a unique name for the translation rule configuration. The name can contain any characters.
Description	Enter a description of the translation rule configuration.

Basic Settings

Field	Description	Cisco IOS CLI Equivalent
Translation rule set number	Enter a unique number to assign to a translation rule set that you are creating.	**voice translation rule** *number*
Import	Click to copy translation rules from a CSV file to Cisco Catalyst SD-WAN Manager.	—
Export	Click to save existing translation rules that your created in a CSV file.	—
Add Rule	Click to configure the options for the Translation Rule feature.	—
Rule number	Displays a number that designates the precedence for this rule.	—
Matching pattern	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.	—

Field	Description	Cisco IOS CLI Equivalent
Action	Choose one of the following options to designate the action that the system performs for calls that match the string in the **Matching pattern** field: • **reject**: Causes the system to reject the call. • **replace**: Causes the system to replace the string in the **Matching pattern** field with a string that you specify.	**voice translation-rule** *number* • Match and replace rule: **rule** *precedence /match-pattern/ / replace-pattern/* • Reject rule: **rule** *precedence* **reject** */match-pattern/*
Replacement pattern	If you choose the **replace** action for the rule, 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. To include the backslash character (\) in a replace string, precede the backslash with a backslash. For example, if you specify a matching pattern of /^9/ and a replacement pattern 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.	—
Action	Click the **Recycle Bin** icon to delete the corresponding Rule options instance.	—

Trunk Group

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Trunk Group feature provides options for configuring voice ports as members of a trunk group. You can configure one trunk group for a voice card.

The following tables describe the options for configuring the Trunk Group feature.

Field	Description
Name	Enter a unique name for the trunk group configuration. The name can contain any characters.
Description	Enter a description of the trunk group configuration.

Basic Settings

Field	Description	Cisco IOS CLI Equivalent
Name	Enter the name of the trunk group. The name can contain up to 32 characters.	**trunk group** *name*
Hunt Scheme	Choose the hunt scheme in the hunt group for outgoing calls. **Note** Depending on the hunt scheme that you choose, the **Channel** field, **Direction** field, or both appear. • **least-idle**: Searches for an idle channel with the shortest idle time • **least-used**: Searches for a trunk group member that has the highest number of available channels (applies only to PRI ISDN cards) • **longest-idle**: Searches for an idle channel with the longest idle time **round-robin**: Searches trunk group members in turn for an idle channel, starting with the trunk group member that follows the last used • **sequential**: Searches for an idle 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	**hunt-scheme least-idle** [**even** | **odd** | **both**] **hunt-scheme least-used** [**even** | **odd** | **both** [**up** | **down**] **hunt-scheme longest-idle** [**even** | **odd** | **both**] **hunt-scheme random** **hunt-scheme round-robin** [**even** | **odd** | **both** [**up** | **down**] **hunt-scheme sequential** [**even** | **odd** | **both** [**up** | **down**]
Max Calls In	Enter the maximum number of incoming calls that are allowed for the trunk group. If you do not enter a value, there is no limit on the number of incoming calls. If the maximum number of incoming calls is reached, the trunk group becomes unavailable for more calls. Range: Integers 0 through 1000	**trunk group** *name* **max-calls voice** *number-of-calls* **direction in**
Max Calls Out	Enter the maximum number of outgoing calls that are allowed for the trunk group. If you do not enter a value, there is no limit on the number of outgoing calls. If the maximum number of outgoing calls is reached, the trunk group becomes unavailable for more calls. Range: Integers 0 through 1000	**trunk group** *name* **max-calls voice** *number-of-calls* **direction out**

Field	Description	Cisco IOS CLI Equivalent
Channel	This option does not appear when the **Hunt Scheme** option is set to **random**. Choose the type of channel that the hunt scheme searches for: • **Both**: Searches both even- and odd-numbered channels. • **Even**: Searches for an idle even-numbered channel. If no idle even-numbered channels are available, an odd-numbered channel is sought. • **Odd**: Searches for an idle odd-numbered channel. If no idle odd-numbered channels are available, an even-numbered channel is sought.	—
Direction	This option appears when the **Hunt Scheme** option is set to **round-robin** or **sequential**. Choose the order in which the hunt scheme searches for channels: • **up**: Searches channels in ascending order within a trunk group member. • **down**: Searches channels in descending order within a trunk group member.	—
Max Retry	Enter 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. Range: Integers 1 through 5	**trunk group** *name* **max-retry** *attempts*

Voice Global

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Voice Global feature provides options for configuring system-wide call routing and network clock parameters.

The following tables describe the options for configuring the Voice Global feature

Field	Description
Name	Enter a unique name for the voice global configuration. The name can contain any characters.
Description	Enter a description of the voice global configuration.

Call Routing

Field	Description	Cisco IOS CLI Equivalent
Trusted IPv4/6 Prefix List	Enter a comma separated list of IPv4 or IPv6 addresses with which the router can communicate through SIP. Enter each IPv4 address in CIDR format. For example, 10.1.2.3/32. The router does not communicate with other addresses, which prevents fraudulent calls being placed through the router. A Trusted IPv4 or IPv66 prefix is required for TDM to IP calls.	**voice service voip** **ip address trusted list** **ipv4** *ipv4-address*/*ipv4-network-mask*
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*

Network Clock

Field	Description	Cisco IOS CLI Equivalent
Participation	Enable this option to configure all T1 or E1 digital interfaces to participate in the backplane clock. Disable this option to remove the clock synchronization with the backplane clock for the module. Default: Enabled	**network-clock synchronization participate** *slot sub-slot*

Field	Description	Cisco IOS CLI Equivalent
Clock Priority Sorting	Appears only if you have configured a digital interface and selected either a primary or secondary clock source for the interface. Designate the priority of up to 6 clock sources for the digital interface. 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. Choose the port that you want to have the highest priority. Then choose the port that you want to have the next highest priority. Continue in this way to choose other ports. The ports in the field appear in descending order of priority, with the highest priority port appearing first. To remove a port from the field, click its **X** icon. To change the priority order of ports, remove the ports and add them back in the desired 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	Choose **true** to enable network synchronization between all modules and the router. Choose **false** to disable network synchronization between all modules and the router. Default: False	**network-clock synchronization automatic**
Wait to restore clock	Enter the amount of time, in ms, that the router waits before including a primary clock source in the clock selection process. Range: Integers 0 through 86400 Default: 300	**network-clock wait-to-restore** *milliseconds*

Voice Tenant

Minimum supported releases: Cisco IOS XE Catalyst SD-WAN Release 17.14.1a, Cisco Catalyst SD-WAN Manager Release 20.14.1.

The Voice Tenant feature provides options for configuring SIP-specific attributes for a tenant. The voice tenant configuration can be then applied to individual dial peers.

The following tables describe the options for configuring the Voice Tenant feature.

Field	Description
Name	Enter a unique name for the voice tenant configuration. The name can contain any characters.
Description	Enter a description of the voice tenant configuration.

Basic Configuration

Field	Description	Cisco IOS CLI Equivalent
Tag	Enter a unique name for this voice tenant configuration.	**voice class tenant** *tag*
Bind Interface	Choose the type of packets that are bound to network interfaces for advertising the source IP address of the tenant: • **Both**: Control and media packets • **Control**: Control packets • **Media**: Media packets • **Disabled**: Bind interface is not configured	—
Transport Type	Choose the transport protocol for SIP control signaling for the tenant. Options are **TCP**, **UDP**, and **TCP TLS**.	**session transport** {**udp** | **tcp** [**tls**]}
Bind Control Interface Name	Enter a network interface name for binding control packets.	**bind control source-interface** *interface-id*
Bind Media Interface Name	Enter a network interface name for binding media packets.	**bind media source-interface** *interface-id*