

# 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<br>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<br>Cisco Catalyst SD-WAN Manager Release 20.14.1 | This feature adds support for the following features in the UC voice profile:<br>• Analog Interface<br>• Call Routing<br>• Digital Interface<br>• Media Profile<br>• SRST<br>• Server Group<br>• Supervisory Disconnect<br>• Translation Profile<br>• Translation Rule<br>• Trunk Group<br>• Voice Global<br>• Voice Tenant |


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# 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.<br>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.<br>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.<br>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.<br>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.<br>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.<br>Choose the mode for the DID signal type<br>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.<br>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.<br>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.<br>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.<br>For example: 4085550111<br>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.<br>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.<br>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.<br>Range: Integers –6 through 14<br>Default: 0 | **input gain** *decibels* |
| **Attenuation** | Enter the amount of attenuation, in dB, for transmitted voice output.<br>Range: Integers –6 through 14<br>Default: 0 | **output attenuation** *decibels* |
| **Echo Canceller** | Choose **Enable** to apply echo cancellation to voice traffic.<br>This option is disabled by default. | **echo-cancel** *enable* |
| **Voice Activity Detection (VAD)** | Choose **Enable** to apply VAD to voice traffic.<br>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.<br>Options are **U-law** and **A-law**. | **compand-type** {**u-law** \| **a-law**} |
| **Impedance** | Choose the terminating impedance for calls.<br>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**.<br>Click to configure the options for various tuning parameters.<br>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.<br>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.<br>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.<br>Range: Integers 0 through 10<br>Default: 1 | **pre-dial-delay** *seconds* |
| **Supervisory Disconnect** | Applies only to FXO voice interface templates.<br>Choose the type of tone that indicates that a call has been released and that a connection should be disconnected:<br>• **Signal**: A disconnect signal indicates a supervisory disconnect<br>• **Anytone**: Any tone indicates a supervisory disconnect<br>• **Dualtone**: A dual tone indicates a supervisory disconnect<br>Default: signal | • Anytone:<br>**supervisory disconnect anytone**<br>• Signal:<br>**supervisory disconnect**<br>• Dualtone:<br>**supervisory disconnect dualtone** {**mid-call** \| **pre-connect**} |

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

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Battery Reversal** | Applies only to FXO voice interface templates.<br>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.<br>• **Answer**: Configures the port to support answer supervision by detection of battery reversal<br>• **Detection Delay**: Configures the delay time after which the card acknowledges a battery-reversal signal<br>• **Both**: Configures answer and detection delay behavior<br>Detection delay range: Integers 0 through 800<br>Detection delay default: 0 (no delay)<br>**Note**<br>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**]<br>**battery-reversal-detection-delay** *milliseconds* |
| **Timing Hookflash Out** | Applies only to FXO voice interface templates.<br>Enter the duration, in ms, of the hookflash indications that the gateway generates on the FXO interface.<br>Range: Integers 50 through 1550<br>Default: 4000 | **timing hookflash-out** *milliseconds* |
| **Timing Guard Out** | Applies only to FXO voice interface templates.<br>Enter the time, in ms, after a call disconnects before another outgoing call is allowed.<br>Range: Integers 300 through 3000<br>Default: 2000 | **timing guard-out** *milliseconds* |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Timing Hookflash In** | Applies only to FXS voice interface templates.<br>Enter the minimum and maximum duration, in ms, for an on-hook condition to be interpreted as a hookflash by the FXS card.<br>Range for minimum duration: 0 through 400<br>Default minimum range value: 50<br>Range for maximum duration: 50 through 1500<br>Default maximum range value: 1000 | **timing hookflash-in** *maximum-milliseconds minimum-milliseconds* |
| **Pulse Digit Detection** | Applies only to FXS voice interface templates.<br>Enable this option to enable pulse digit detection at the beginning of a call.<br>Default: Enabled | **pulse-digit-detection** |
| **Loop Length** | Applies only to FXS voice interface templates.<br>Choose the length for signaling on FXS ports (**Long** or **Short**).<br>Default: Short | **loop-length** [**long** \| **short**] |
| **Ring Frequency** | Applies only to FXS voice interface templates.<br>Choose the frequency, in Hz, of the alternating current that, when applied, rings a connected device.<br>Default: 23 | **ring frequency** *number* |
| **DC Offset** | Applies only to FXS voice interface templates when **Loop Length** is set to **Long**.<br>Choose the voltage threshold below which a ring does not sound on devices.<br>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.<br>Choose the REN for calls that the port processes. This number specifies the loading effect of a telephone ringer on a line.<br>Range: Integers 1 through 5<br>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**.<br>Click to configure the options for timers for DID calls.<br>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.<br>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.<br>Range: Integers 100 through 6500<br>Default: 550 | **timing wait-wink** *milliseconds* |
| **Wink Duration** | Enter the maximum amount of time, in ms, of the wink signal for the port.<br>Range: Integers 50 through 3000<br>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.<br>Range: Integers 200 through 2000<br>Default: 400 | **timing clear-wait** *milliseconds* |
| **Dial Pulse Min Delay** | Enter the amount of time, in ms, between wink-like pulses for the port.<br>Range: Integers 0, or 140 through 2000<br>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.<br>Range: Integers 110 through 290<br>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.<br>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.<br>**Note**<br>• These caller ID options apply only when the **Signal Type** option in the **Basic** tab is configured as **LoopStart** or **GroundStart**.<br>• 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.<br>• 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. | — |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Port Range** | Enter the port or ports within the voice interface template to which these options apply.<br>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:<br>• **BT**: Frequency-Shift Keying (FSK) with Dual Tone Alerting Signal (DTAS) used by British Telecom<br>• **FSK**: FSK before or during a call<br>• **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.<br>Choose the character that indicates the start of a DTMF string. | **caller-id mode** {**dtmf** {**start** \| **end**} {**#** \| ***** \| **A** \| **B** \| **C** \| **D**}} |
| **DTMF End** | Applies only if you choose **DTMF** for the caller ID mode.<br>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:<br>• **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.<br>• **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.<br>• **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.<br>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** |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Caller ID Block** | Applies only to FXS voice interface templates.<br>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.<br>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).<br>You can add multiple instances of these options so that you can configure different connection PLAR options for different ports. | — |
| **Port Range** | Enter the port or ports within the voice interface template to which these options apply.<br>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.<br>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.<br>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.<br>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.<br>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:<br>• **Incoming**: Applies the corresponding Translation Profile feature to traffic that is incoming to the port<br>• **Outgoing**: Applies the corresponding Translation Profile feature to traffic that is outgoing from the port |
| **Supervisory Disconnect** | Applies only to FXO voice interface templates.<br>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. |
| **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 Peer**s | 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.<br>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**} |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Direction** | Choose the direction of traffic on the dial peer. Options are **incoming** and **outgoing**. | • Incoming:<br>**dial-peer voice** *number* {**pots** \| **voip**}<br>**incoming called-number** *string*<br>• Outgoing:<br>**dial-peer voice** *number* {**pots** \| **voip**}<br>**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.<br>Enter the string as an E.164 format regular expression in the following form:<br>(ipv6:\[([0-9A-Fa-f.:])+\](:[0-9]+)?)) | • Incoming:<br>**dial-peer voice** *number* {**pots** \| **voip**}<br>**incoming called-number** *string*<br>• Outgoing:<br>**dial-peer voice** *number* {**pots** \| **voip**}<br>**destination-pattern** *string* |
| **Forward Digits Type** | Applies only when **Dial peer type** is configured as **pots** and **direction** is configured as **outgoing**.<br>Choose how the dial peer transmits digits in outgoing numbers:<br>• **all**: The dial peer transmits all digits<br>• **none**: The dial peer does not transmit digits that do not match the destination pattern<br>• **some**: The dial peer transmits the specified number of right-most digits<br>Default: none | • All:<br>**dial-peer voice** *number* **pots**<br>**forward-digits all**<br>• None:<br>**dial-peer voice** *number* **pots**<br>**forward-digits 0**<br>• Some:<br>**dial-peer voice** *number* **pots**<br>**forward-digits** *number* |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Forward Digits** | Applies only when you choose **Some** for **Forward Digits Type**.<br>Enter the number of right-most digits in the outgoing number to transmit.<br>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**<br>**forward-digits** *number* |
| **Prefix** | Applies only when **Dial peer type** is configured as **pots** and **direction** is configured as **outgoing**.<br>Enter a string to be pretended to the dial string for outgoing calls.<br>Valid values: Integers 0 through 9 and comma (,) | **dial-peer voice** *number* **pots**<br>**prefix** *string* |
| **Transport Protocol** | Applies only when **Dial peer type** is configured as **sip**. Choose the transport protocol for SIP control signaling.<br>Options are **tcp** and **udp**. | **dial-peer voice** *number* **voip**<br>**session transport** {**tcp** \| **udp**} |
| **Preference** | Enter the preference of the dial peer.<br>If dial peers have the same match criteria, the system uses the one with the highest preference value.<br>Range: Integers 0 through 10<br>Default: 0 | **dial-peer voice** *number* **voip**<br>**preference** *value*<br>**dial-peer voice** *number* **pots**<br>**preference** *value* |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Port** | Applies only when **Dial peer type** is configured as **pots**.<br>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**.<br>For an outgoing dial peer, the router sends the calls that match the dial peer to this port.<br>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**<br>• For an analog port:<br>**port** *slot*/*subslot*/*port*<br>• For a digital port:<br>**port** *slot*/*subslot*/*port***:15**<br>**port** *slot*/*subslot*/*port***:23** |
| **Destination Address** | Applies only when **Dial peer type** is configured as **sip** and **direction** is configured as **outgoing**.<br>Enter the network address of the remote voice gateway to which calls are sent after a local outgoing SIP dial peer is matched.<br>Enter the address in one of these formats:<br>• **dns:***hostname.domain*<br>• **sip-server**<br>• **ipv4:***destination-address*<br>• **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.<br>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.<br>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.<br>For a detailed description of each bundle, see the "Primary Fax Protocol Command Bundles" table in Configure SIP Dial Peers for a Voice Policy.<br>The descriptions of the bundles include the following components:<br>• **nse**: Uses NSEs to switch to T.38 fax relay mode<br>• **force**: Unconditionally uses Cisco Network Services Engines (NSE) to switch to T.38 fax relay<br>• **version**: Specifies a version for configuring fax speed:<br>  • **0**: Configures version 0, which uses T.38 version 0 (1998–G3 faxing)<br>  • **3**: Configures version 3, which uses T.38 version 3 (2004–V.34 or SG3 faxing)<br>• **none**: No fax pass-through or T.38 fax relay is attempted<br>• **Pass-through**: The fax stream uses one of the following high-bandwidth codecs:<br>  • **g711ulaw**: Uses the G.711 ulaw codec<br>  • **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."<br>Choose the fallback mode for fax transmissions. This fallback mode is used if the primary fax protocol cannot be negotiated between device endpoints.<br>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.”<br>Enter the number of redundant T.38 fax packets to be sent for the low-speed V.21-based T.30 fax machine protocol.<br>Range: Integers 0 (no redundancy) to 5<br>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.”<br>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.<br>Range: Integers 0 (no redundancy) to 2<br>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.<br>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:<br>• **None**: Modem pass-through is disabled on the device<br>• **NSE G.711ulaw**: Uses named signaling events (NSEs) to communicate G.711 ulaw codec switchover between gateways<br>• **NSE G.711alaw**: Uses named NSEs to communicate G.711 alaw codec switchover between gateways | • None:<br>**no modem passthrough**<br>• NSE G.711ulaw:<br>**modem passthrough nse codec g711ulaw**<br>NSE G.711alaw:<br>**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.<br>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.<br>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.<br>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:<br>• **Incoming**: Applies the corresponding Translation Profile feature to traffic that is incoming to the port<br>• **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.<br>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.<br>Disable this option if the device does not have any hardware DSP resources and you want to use the software MTP DSP service.<br>This option is enabled by default. | — |
| **Module Location** | If **Services** is enabled, choose the slot and sub-slot location for the hardware DSP.<br>You can configure as many module locations as needed.<br>For a list of supported modules, see Configure UC Voice Services Using the Workflow Library or Configuration Groups. | **voice-card** *slot*/*subslot*<br>**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.<br>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.<br>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**.<br>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.<br>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.<br><br>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.<br><br>The following codecs are supported:<br>• For the **transcode** profile type:<br>  • g711alaw<br>  • g711ulaw<br>  • g729abr8<br>  • g729ar8<br>  • g729br8<br>  • g729r8<br>  • g722-64<br>  • ilbc<br>  • iSAC<br>  • opus<br>  • pass-through<br>• For the **conference** profile type:<br>  • g711alaw<br>  • g711ulaw<br>  • g722r-64<br>  • g729abr8<br>  • g729ar8<br>  • g729br8<br>  • g729r8<br>• For the **mtp** profile type when **MTP Type Hardware** or both **MTP Type Hardware** and **MTP Type Software** are chosen:<br>  • g711ulaw<br>  • g711alaw<br>  • pass-through | |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| | • For the **mtp** profile type when **MTP Type Software** is chosen:<br>• g711ulaw<br>• g711alaw<br>• g722-64<br>• g729abr8<br>• g729ar8<br>• g729br8<br>• g729r8<br>• ilbc<br>• iSAC<br>• 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.<br>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.<br>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.<br>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.<br>You can configure up to 12 Cisco Unified Communications Manager servers.<br>**Note**<br>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.<br>Enter the interface in this format:<br>*interface-type*/*interface-number*/*port*<br>where:<br>• *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.<br>• *interface-number*: Interface number that the services use to register with Cisco Unified Communications Manager.<br>• *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.<br>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.<br>Range: 1 (lowest) through 7 (highest)<br>Default: 5 | **sccp ip precedence** *value* |
| **Add Configure Server List** | Click to display the following options for a Cisco Unified Communications Manager server:<br>• **Server Identifier**: Unique system-generated identifier of the Cisco Unified Communications Manager server<br>• **Server IP**: Enter the IP address of the Cisco Unified Communications Manager server | • Server identifier:<br>*identifier-number*<br>• Server IP:<br>**sccp ccm** {*ipv4-address* \| i*pv6-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.<br>You can add multiple Cisco Unified Communications Manager groups.<br>**Note**<br>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.<br>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.<br><br>The drop-down list displays the server identifiers of the Cisco Unified Communications Manager servers that you configured.<br><br>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.<br><br>The servers in the field appear in descending order of priority, with the highest priority server appearing first.<br><br>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:<br><br>• **guard**: Switchback occurs when active sessions are terminated gracefully or when the guard timer expires, whichever happens first<br><br>• **graceful**: Switchback occurs after all active sessions terminate gracefully<br><br>• **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<br><br>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:<br><br>• **graceful**: Switchover occurs after all active sessions terminate gracefully<br><br>• **immediate**: Switchover occurs immediately, whether or not there is an active connection<br><br>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.<br>Range: Integers 1 to 180<br>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.<br>Range: Integers 1 to 180<br>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.

### Restrictions for Digital Interface

The following Network Interface modules are not supported with Digital Interface Feature:

- NIM-1CE1T1-PRI
- NIM-2CE1T1-PRI
- NIM-8CE1T1-PRI

The provisioning of these modules require **voice-dsp** keyword in the **pri-group timeslots** command.

**Note** As a workaround, you can use the CLI add-on profile to configure the **pri-group timeslots** command with the **voice-dsp** keyword for specific T1/E1 controller ports in the preceding NIMs.

### Digital Interface Configuration

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* |
| **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.<br>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.<br>Default: Unchecked | **no local-bypass** |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Port and Clock Selector** | 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:<br>**1.** 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.<br>**2.** For each port, choose one of the following options to set the clock source:<br>• **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.<br>This option is the default.<br>• **Network**: Sets the backplane clock or the system oscillator clock as the module clock source.<br>• **Primary Clock**: Sets the port to be a primary clock source.<br>• **Secondary Clock**: Sets the port to be a secondary clock source.<br>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.<br>**3.** Click **Save**. | **controller** {**t1** \| **e1**} *slot*/*sub-slot*/*number*<br>**clock source** {**network** \| **line** \| **line primary** \| **line secondary**} |

### Basic

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Add Basic** | Click to configure basic options for the group of digital ports.<br>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.<br>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.<br>Ranges:<br>• For T1 PRI: Time slots 1 through 24. The 24th time slot is the D channel.<br>• For E1 PRI: Time slots 1 through 31. The 16th time slot is the D channel. | **controller e1/t1** *slot*/*sub-slot*/*port*<br>**pri-group timeslots** *timeslots-range* |
| **Line Termination** | Applies to E1 voice interface templates only. Choose the termination type for the interface:<br>• **75-ohm**: 75 ohm unbalanced termination<br>• **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:<br>• **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.<br>This value is the default<br>• **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:<br>• For a **Long** cable length, enter the loss value, decibels (dB).<br>Options are **–7.5**, **–15**, **–22.5**, and **0**.<br>The default value is 0.<br>• 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.<br>For a T1 voice interface template:<br>• **ami**: Use alternate mark inversion as the line code type<br>• **b8zs**: Use binary 8-zero substitution as the line code type (default)<br>For an E1 voice interface template:<br>• **amiami**: Use alternate mark inversion as the line code type<br>• **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.<br>For a T1 voice interface template:<br>• **esf**: Extended super frame (default)<br>• **sf**: Super frame<br>For an E1 voice interface template:<br>• **ccr4**: CRC4 framing type (default)<br>• **no-crc4**: No CRC4 framing type | **controller t1** *slot*/*sub-slot*/*port* **framing** [**esf** \| **sf**]<br>**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.<br>Default: Disabled | **interface serial** *slot*/*sub-slot*/**port:** {**15**\| **23**}<br>**isdn protocol-emulate** [**network** \| **user**] |

| Field | Description | Cisco IOS CLI Equivalent |
|---|---|---|
| **Switch Type** | Choose the ISDN switch type for this interface:<br>• **primary-qsig**: Supports QSIG signaling according to the Q.931 protocol. Network side functionality is assigned with the **isdn protocol-emulate** command.<br>• **primary-4ess**: Lucent (AT&T) 4ESS switch type for the United States.<br>• **primary-5ess**: Lucent (AT&T) 5ESS switch type for the United States.<br>• **primary-dms100**: Nortel DMS-100 switch type for the United States.<br>• **primary-net5**: NET5 ISDN PRI switch types for Asia, Australia, and New Zealand. ETSI-compliant switches for Euro-ISDN E-DSS1 signaling system.<br>• **primary-ni**: National ISDN switch type.<br>• **primary-ntt**: Japanese NTT ISDN PRI switches. | **interface serial** *slot*/*sub-slot*/**port:**{**15** \| **23**}<br>**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.<br>Range: Integers 0 through 200<br>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.<br>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.<br>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**}<br>**isdn timer T200** *value*<br>**isdn timer T203** *value*<br>**isdn timer T301** *value*<br>**isdn timer T303** *value*<br>**isdn timer T306** *value*<br>**isdn timer T309** *value*<br>**isdn timer T310** *value*<br>**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**.<br>• **Port Range**: Displays the ports that you chose<br>• **ISDN Timer**: Displays the ISDN timers that you can provision.<br>• **Value**: Enter the value, in ms, for the corresponding ISDN timer:<br>• For the **T200** ISDN timer:<br>• Range: 400 through 400000<br>• Default for all switch types: 1000<br>• For the **T203** ISDN timer:<br>• Range: Integers 400 through 400000<br>• Default for QSIG, ETSI Net5, and DMS-100 switch types: 10000<br>• Default for 4ESS, 5ESS, NTT, and NI switch types: 30000<br>• For the **T301** ISDN timer:<br>• Range: 180000 through 86400000<br>• Default for NTT and ETSI Net5 switch types: 180000<br>• Default for other switch types: 300000<br>• For the **T303** ISDN timer:<br>• Range: 400 through 86400000<br>• Default for QSIG switch type: 6000<br>• Default for other switch types: 4000<br>• For the **T306** ISDN timer:<br>• Range: 400 through 86400000<br>• Default for all switch types: 30000<br>• For the **T309** ISDN timer:<br>• Range: 0 through 86400000<br>• 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<br>• For the **T310** ISDN timer:<br>• Range: 400 through 400000<br>• Default for NI , 4ESS and 5ESS switch types when network side configuration is false (User): 30000<br>• Default for NI, 4ESS, and 5ESS switch types when network side configuration is true (Network): 10000<br>• Default for ETSI Net5 switch types: 4000<br>• Default for QSIG switch type: 120000<br>• Default for NTT switch type: 3000<br>• Default for DMS-100 switch type when network side configuration is false (User): 1000<br>• Default for DMS-100 switch type when network side configuration is true (Network): 4000<br>• Default for other switch types: 4000<br>• For the **T321** ISDN timer:<br>• Range: 0 through 86400000<br>• Default for ETSI Net5 switch type: 30000<br>• 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.<br>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.<br>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:<br>• **data**: X.121 data numbering plan<br>• **isdn**: E.164 ISDN/Telephony numbering plan<br>• **national**: Number called to reach a subscriber in the same country, but outside the local network<br>• **privacy**: Private numbering plan<br>• **reserved/*extension***: Reserved for the extension | **isdn map address {{** *address* **\|** *reg-exp***} plan** *plan* **type** *type* **\| transparent}** |
| **Type** | Choose an ISDN number type:<br>• **abbreviated**: Abbreviated representation of the complete number as supported by your network<br>• **international**: Number called to reach a subscriber in another country<br>• **national**: Number called to reach a subscriber in the same country, but outside the local network<br>• **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.<br>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.<br>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**<br>**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.<br>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.<br>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.<br>Range: Integers –6 through 14<br>Default: 0 | **input gain** *decibels* |
| **Attenuation** | Enter the amount of attenuation, decibels (dB), for transmitted voice output.<br>Range: Integers –6 through 14<br>Default: 3 | **output attenuation** *decibels* |
| **Echo Canceller** | Choose **Enable** to apply echo cancellation to voice traffic.<br>This option is disabled by default. | **echo-cancel** *enable* |
| **Voice Activity Detection** | Choose **Enable** to apply VAD to voice traffic.<br>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**).<br>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.<br>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.<br>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:<br>To remove an option from the field, click its **X** icon.<br>• **called-number**: Indicates the outgoing call number<br>• **called-subaddr**: Indicates the subaddress of the outgoing call<br>• **caller-number**: Indicates the incoming call number<br>• **caller-subaddr**: Indicates the subaddress of the incoming call<br>• **connected-number**: Indicates the number of the remaining caller if a disconnect occurs during a conference<br>• **connected-subaddr**: Indicates the subaddress of the remaining caller if a disconnect occurs during a conference<br>• **display**: Provides information about the text display<br>• **extended-facility**: Provides information about extended facility requests<br>• **facility**: Provides information about facility requests<br>• **high-layer-compat**: Provides information about higher layer compatibility<br>• **low-layer-compat**: Provides information about lower layer compatibility<br>• **network-facility**: Provides information about network facility requests<br>• **notify-indicator**: Provides information about notifications<br>• **progress-indicator**: Provides information about the call in progress<br>• **redirecting-number**: Indicates the number that is redirecting the call<br>• **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.<br>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.<br>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.<br>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:<br>• **Incoming**: Applies the corresponding Translation Profile feature to traffic that is incoming to the port<br>• **Outgoing**: Applies the corresponding Translation Profile feature to traffic that is outgoing from the port |
| **Supervisory Disconnect** | Applies only to FXO voice interface templates.<br>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.<br>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:<br>• **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.<br>• **sip-notify**: A Cisco proprietary out-of-band DTMF relay mechanism that transports DTMF signals using SIP NOTIFY messages.<br>• **sip-kpml**: Keypad Markup Language (KPML) is used to indicate DTMF tones in SIP messaging. It transmits DTMF tone indications via SIP NOTIFY messages<br>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.<br>The options in the field appear in descending order of priority, with the highest priority option appearing first.<br>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.<br>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.<br>The codecs in the field appear in descending order of priority, with the highest priority option appearing first.<br>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*<br>**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**<br>**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.<br>The maximum values that you can enter depend on the device that you are configuring. | **voice register global**<br>**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.<br>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**<br>**moh** *filename* |
| **System Message** | Enter a message that displays on endpoints when Cisco Unified SRST mode is in effect. | **voice register global**<br>**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<br>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.<br>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*<br>**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.<br>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:<br>• **busy**: Forwards a call to another directory number when a phone is busy<br>• **all**: Forwards all incoming calls to another directory number<br>• **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.<br>Range: Integers 3 to 60000<br>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.<br>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:<br>• **Incoming**: Applies the corresponding Translation Profile feature to traffic that is incoming to the port<br>• **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.<br>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.)<br>Options are:<br>• **none**: No hunt scheme defined.<br>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.)<br>• **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.<br>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**.<br>Choose the order of selection preference of the server for the setting up of outgoing calls.<br>Range: Integers 0 (highest preference) through 5 (lowest preference)<br>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.<br>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.<br>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.<br>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.<br>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.<br>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.<br>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:<br>• **Busy**<br>• **Disconnect**<br>• **Number Unobtainable**<br>• **Out of Service**<br>• **Reorder**<br>• **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.<br>Enter the cadence as an on/off value pair, separated with a space.<br>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.<br>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.<br>You can add multiple instances of these options. | — |
| **Supervisory Number** | Enter a unique number to identify dual-tone detection parameters.<br>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.<br>Range: Integers 0 through 200 (0 through 2000 ms)<br>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.<br>Range: Integers 0 through 100 (0 through 1000 ms)<br>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.<br>Range: Integers 0 through 125<br>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.<br>Range: Integers 0 through 20<br>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.<br>Range: Integers 0 through 35<br>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.<br>Range: Integers 0 through 15<br>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.<br>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.<br>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:<br>• **calling**: Maps a translation rule set for the number that is calling in<br>• **called**: Maps a translation rule set for the number that is being called | • Calling:<br>**translate calling** *translation-rule-number*<br>• Called:<br>**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.<br>Enter the string in regular expression format beginning and ending with a slash (/). For example, /^9/.<br>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:<br>• **reject**: Causes the system to reject the call.<br>• **replace**: Causes the system to replace the string in the **Matching pattern** field with a string that you specify. | **voice translation-rule** *number*<br>• Match and replace rule:<br>**rule** *precedence /match-pattern/ / replace-pattern/*<br>• Reject rule:<br>**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.<br>Enter the number in regular expression format beginning and ending with a slash (/). For example, //, which indicates a replacement of no string.<br>To include the backslash character (\) in a replace string, precede the backslash with a backslash.<br>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.<br>The name can contain up to 32 characters. | **trunk group** *name* |
| **Hunt Scheme** | Choose the hunt scheme in the hunt group for outgoing calls.<br>**Note**<br>Depending on the hunt scheme that you choose, the **Channel** field, **Direction** field, or both appear.<br>• **least-idle**: Searches for an idle channel with the shortest idle time<br>• **least-used**: Searches for a trunk group member that has the highest number of available channels (applies only to PRI ISDN cards)<br>• **longest-idle**: Searches for an idle channel with the longest idle time<br>**round-robin**: Searches trunk group members in turn for an idle channel, starting with the trunk group member that follows the last used<br>• **sequential**: Searches for an idle channel, starting with the trunk group member with the highest preference within the trunk group<br>• **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**]<br>**hunt-scheme least-used** [**even** \| **odd** \| **both** [**up** \| **down**]<br>**hunt-scheme longest-idle** [**even** \| **odd** \| **both**]<br>**hunt-scheme random**<br>**hunt-scheme round-robin** [**even** \| **odd** \| **both** [**up** \| **down**]<br>**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.<br>If the maximum number of incoming calls is reached, the trunk group becomes unavailable for more calls.<br>Range: Integers 0 through 1000 | **trunk group** *name*<br>**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.<br>If the maximum number of outgoing calls is reached, the trunk group becomes unavailable for more calls.<br>Range: Integers 0 through 1000 | **trunk group** *name*<br>**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**.<br>Choose the type of channel that the hunt scheme searches for:<br>• **Both**: Searches both even- and odd-numbered channels.<br>• **Even**: Searches for an idle even-numbered channel. If no idle even-numbered channels are available, an odd-numbered channel is sought.<br>• **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**.<br>Choose the order in which the hunt scheme searches for channels:<br>• **up**: Searches channels in ascending order within a trunk group member.<br>• **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.<br>If you do not enter a value and a call fails, the system does not attempt to make the call again.<br>Range: Integers 1 through 5 | **trunk group** *name*<br>**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.<br>Enter each IPv4 address in CIDR format. For example, 10.1.2.3/32.<br>The router does not communicate with other addresses, which prevents fraudulent calls being placed through the router.<br>A Trusted IPv4 or IPv66 prefix is required for TDM to IP calls. | **voice service voip**<br>**ip address trusted list**<br>**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.<br>This information defines how the return/response to this traffic should be sent. | **voice service voip**<br>**sip**<br>**bind control source-interface** *interface-id*<br>**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.<br>Disable this option to remove the clock synchronization with the backplane clock for the module.<br>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.<br>Designate the priority of up to 6 clock sources for the digital interface.<br>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.<br>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.<br>The ports in the field appear in descending order of priority, with the highest priority port appearing first.<br>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.<br>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.<br>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.<br>Range: Integers 0 through 86400<br>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:<br>• **Both**: Control and media packets<br>• **Control**: Control packets<br>• **Media**: Media packets<br>• **Disabled**: Bind interface is not configured | — |
| **Transport Type** | Choose the transport protocol for SIP control signaling for the tenant.<br>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* |