CHAPTER 31

PSTN Connection Settings

Impacted GUIs: Cisco Unified Communications Manager Business Edition 3000 First Time Setup Wizard and Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.

The PSTN Connection Settings page allows you to configure additional PSTN connections, edit or delete existing connections for the Cisco Unified Communications Manager Business Edition 3000 server. Table 31-1 describes the settings on the PSTN Connections page (**Connections > PSTN Connections**). Click **PSTN Connections** to view the existing PSTN connections for the Cisco Unified Communications Manager Business Edition 3000.

Caution For DNS, make sure that you map the IPv4 address of the Cisco Unified Communications Manager Business Edition 3000 server to the hostname on the DNS server. Cisco recommends that you update the DNS server before you change the hostname or IP address on the Network page.

If you update the IP address for the server, you need to reissue the CLI commands that are listed on the Gateway page (**System Settings > Gateway**).

PSTN Connection Settings

Table 31-1 Settings on the PSTN Connections Page

Setting	Description
Name	Specifies the name of the PSTN connection.
Description	Specifies the description for the PSTN connection.
Connection Type	Specifies the connection type for the PSTN connection.
Device Name	Specifies the device name for the PSTN connection.
Add PSTN Connections Information	
Edit	Click the **Edit** link corresponding to the PSTN connection to edit the configuration settings. Click **Show Advanced Settings** or Hide Advanced Settings to view or edit the advanced settings for the PSTN connection. Click **Save** to save the changes. Click **Reset** to revert to the previous saved configuration.

Table 31-1 Settings on the PSTN Connections Page (continued)

Setting	Description
Delete	Click the **Delete** link corresponding to the PSTN connection to delete the existing connection. A warning message appears indicating that the connection will be deleted and calls through that connection will be dropped. You can choose either to delete the PSTN connection or to cancel the operation. **Note** Ensure that you retain minimum of one T1/E1 PSTN connection configured with the internal gateway. The media transcoding and conferencing will not work properly if all the internal gateway connections are deleted.
Add PSTN Connection	Click **Add PSTN Connection** to configure a new PSTN connection.

Connection Type

Table 31-2 Settings on the Connection Type page

Setting	Description
Connection Type	Select the required connection type from the drop-down list. Select one of the following: • E1 PRI • T1 PRI • T1 CAS • FXO • SIP Trunk Click **Next** to go to the next screen.

Device

Table 31-3 Settings on the Device page

Setting	Description
Device Type	Select the required device type: • **BE3000**—Select this option for the MCS7890-C1 internal gateway. The device name is auto-filled with internal gateway. • **ISR2901**—Select this option to configure an ISR2901 gateway.
Device	If you have selected device type as CUCMBE3000, the device name field will be auto-filled. Select the device for the ISR2901 gateway if available. Select **Add Device** to add a new device for ISR2901 gateway.
Add ISR2901 Gateway Information	
Hostname	Enter a hostname for the device.
IP Settings	Determine how you want the gateway to obtain its IP address. • If you want the gateway to obtain an IP address through DHCP, click **Obtain an IP Address Automatically**. • To assign a static IP address to the gateway, click **Use the following IP address** and enter the IPv4 address, the subnet mask, and the default gateway. The IP address, subnet mask, and default gateway must be in the format ddd.ddd.ddd.ddd where ddd is a value between 0 and 255 (except 0.0.0.0).
Enable DNS resolution	• To enable DNS for the gateway, check the **Enable DNS resolution** check box and enter the primary and secondary DNS servers. The IP addresses of primary and secondary DNS servers must be in the format ddd.ddd.ddd.ddd where ddd is a value between 0 and 255 (except 0.0.0.0). **Note** For DNS, make sure that you map the IPv4 address of the Cisco Unified Communications Manager Business Edition 3000 server to the hostname on the DNS server. Cisco recommends that you update the DNS server before you change the hostname or IP address on the Network page.
OK	Click **OK** to add a device for the gateway. The newly configured device is selected.
Cancel	Click **Cancel** to cancel the configuration and return to the previous page. Click **Next** to go to Provider section.

Adding/Editing a Device

Table 31-4 Settings on the Add Device page

Setting	Description
Hostname	Enter a hostname for the device.
IP Settings	Determine how you want the gateway to obtain its IP address. • If you want the gateway to obtain an IP address through DHCP, click **Obtain an IP Address Automatically**. • To assign a static IP address to the gateway, click **Use the following IP address** and enter the IPv4 address, the subnet mask, and the default gateway. The IP address, subnet mask, and default gateway must be in the format ddd.ddd.ddd.ddd where ddd is a value between 0 and 255 (except 0.0.0.0).
Enable DNS resolution	To enable DNS for the gateway, check the **Enable DNS resolution** check box and enter the primary and secondary DNS servers. The IP addresses of primary and secondary DNS servers must be in the format ddd.ddd.ddd.ddd where ddd is a value between 0 and 255 (except 0.0.0.0). **Note** For DNS, make sure that you map the IPv4 address of the Cisco Unified Communications Manager Business Edition 3000 server to the hostname on the DNS server. Cisco recommends that you update the DNS server before you change the hostname or IP address on the Network page.
OK	Click **OK** to add a device for the gateway. The newly configured device is selected.
Cancel	Click **Cancel** to cancel the configuration and return to the previous page.

Provider

Table 31-5 Settings on the Service Provider page

Setting	Description
Provider	Specifies the service provider who adequately supports Cisco Unified Communications Manager Business Edition 3000 server. The service provider provides a list of attributes for specific connection settings and the default layout for the respective Connection Settings page. The following are the service providers for Cisco Unified Communications Manager Business Edition 3000: • Cisco Digital Access E1 PRI (if the Connection Type is EI PRI) • Cisco Digital Access T1 PRI (if the Connection Type is TI PRI) • Cisco Digital Access T1 CAS (if the Connection Type is T1 CAS) • Cisco Unified Border Element (CUBE) (default, if the Connection Type is SIP Trunk) **Note** If you have installed a Connection Pack, the associated service provider name will appear in the list. • SPA8800 (if the Connection Type is FXO) Click **Next** to go to the Connection Settings page.

Connection Settings

Table 31-6 describes the settings on the connection settings page for General section.

Table 31-6 Settings on the Connection Settings Page for General Section

Setting	Description
Connection Name	Displays the name of the PSTN connection.
Description	Displays the description provided for the PSTN connection. You can enter a new description if required.
Connection Type	Displays the type of PSTN connection.
Device Type	Displays the type of the gateway device.
Device Name	Displays the name of the gateway.
Device Port	Select the required port for the gateway device from the drop down list

The following are the description of Advanced Settings for Connection Types for Cisco Unified Communications Manager Business Edition 3000:

- Connection Type: E1 PRI, page 31-6
- Connection Type: T1 PRI, page 31-14
- Connection Type: T1 CAS, page 31-22
- Connection Type: SIP Trunk, page 31-24
- Connection Type: FXO, page 31-32

Connection Type: E1 PRI

Table 31-7 describes the settings on the **Add PSTN Connection > Connection Settings** page when your chosen Connection Type is **E1 PRI**.

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Connection Settings	
Protocol Type	Select the communications protocol type for the PSTN connection. EI PRI provides two options: • PRI EURO • PRI AUSTRALIAN
Show Advanced Settings/Hide Advanced Settings	
Interface Settings	
Protocol Side	This setting specifies whether the gateway connects to a Network device or to a User device. Make sure that the two ends of the PRI connection use opposite settings. For example, if you connect to a PBX and the PBX uses User as its protocol side, select Network for this device. Typically, use User for this option for central office connections.
Clock	Select Internal or External for the clock source.
PCM Type	Specify the digital encoding format. Select one of the following formats: • a-law—Use for Europe and other countries, except North America, Hong Kong, Taiwan, and Japan • mu-law—Use for North America, Hong Kong, Taiwan, and Japan
Line Coding	Select the line coding from one of the following: • High Density Bi-polar 3 (HDB3) • Alternate mark inversion (AMI)

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Framing	Select the multiframe format of the span from one of the following: • Cyclic Redundancy Check 4 (CRC4) • Non Cyclic Redundancy Check 4 (NonCRC4)
Echo Cancellation	Select whether to enable or disable echo cancellation.
Coverage (ms)	If an issue occurs with echo cancellation, select a value to address the issue. Choose one of the following values: • 24 • 32 • 48 • 64 • 128 (available with MCS7890C1 internal gateway only) **Note** This option is available only if echo cancellation is enabled.
Channel Selection Order	Select the order in which channels or ports are enabled from first (lowest number port) to last (highest number port), or from last to first. Valid entries include TOP DOWN (first to last) or BOTTOM UP (last to first). If you are not sure which port order to use, select TOP DOWN.
Channel IE Type	Select one of the following values to specify whether channel selection is presented as a channel map or a slot map: • Timeslot Number—B-channel usage always indicates actual time slot map format (such as 1-15 and 17-31 for E1). • Slotmap—B-channel usage always indicates a slot map format. • Use Number When 1B—Channel usage indicates a channel map for one B-channel but indicates a slot map if more than one B-channel exists. • Continuous Number—Configures a continuous range of slot numbers (1-30) as the E1 logical channel number instead of the noncontinuous actual time slot number (1-15 and 17-31).
Delay for first restart (ticks)	Enter the rate at which the spans are brought in service. The delay occurs when many PRI spans are enabled on a system and the Inhibit Restarts at PRI Initialization check box is unchecked. For example, set the first five cards to 0 and set the next five cards to 16. (Wait 2 seconds before bringing them in service.)

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Delay between restarts (ticks)	Enter the time between restarts. The delay occurs when a PRI RESTART is sent if the Inhibit Restarts check box is unchecked.
Inhibit Restarts at PRI Initialization	A RESTART or SERVICE message confirms the status of the ports on a PRI span. If RESTART or SERVICE messages are not sent, Cisco Unified Communications Manager Business Edition 3000 assumes the ports are in service. When the D-Channel successfully connects with another PRI D-Channel, it sends a RESTART or SERVICE message when this check box is unchecked.
Enable G. Clear	Check this check box to enable G. Clear Codec support. Checking this check box causes echo cancellation and zero suppression for outbound calls to be disabled.
Trasmit UTF-8 for Calling Party Name	If you check the Transmit UTF-8 for Calling Party Name check box, the gateway sends unicode for the calling party name.
PRI-Specific Settings	
Display IE Delivery	Check the check box to enable delivery of the display information element (IE) in SETUP and NOTIFY messages (for DMS protocol) for the calling and connected party name delivery service.
Redirecting Number IE Delivery–Inbound	Check this check box to indicate the first redirecting number and the redirecting reason of the call when the call is forwarded. (The UUIE part of the outgoing SETUP message from the Cisco Unified Communications Manager Business Edition 3000 includes the Redirecting Number IE.) Uncheck the check box to exclude the first redirecting number and the redirecting reason. You use Redirecting Number IE for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number IE, check the check box.
Redirecting Number IE Delivery–Outbound	Check this check box to accept the Redirecting Number IE in the incoming SETUP message to the Cisco Unified Communications Manager Business Edition 3000. (The UUIE part of the SETUP message includes the Redirecting Number IE.) Uncheck the check box to exclude the Redirecting Number IE. You use Redirecting Number IE for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number IE, you should check the check box.

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Setup non-ISDN Progress Indicator IE Enable	Check this check box only if users are not receiving ringback tones on outbound calls. When this check box is checked, the Cisco Unified Communications Manager Business Edition 3000 sends Q.931 Setup messages out digital (that is, non-H.323) gateways with the Progress Indicator field set to non-ISDN. This message notifies the destination device that the gateway is non-ISDN and that the destination device should play in-band ringback. This problem usually associates with a Cisco Unified Communications Manager Business Edition 3000 that connect to PBXs through digital gateways.
Outbound Call Routing	
Calling Party Presentation	Select whether you want the Cisco Unified Communications Manager Business Edition 3000 to allow or restrict the display of the calling party phone number. • Default—If you do not want to change the calling line ID presentation • Allowed—To indicate that the "Calling Line ID is Allowed" on outbound calls • Restricted—To indicate that "Calling Line ID is Restricted" on outbound calls
Calling Party Selection	Any outbound call on a gateway can send directory number information. Select which directory number is sent. Select one of the following options: • Originator—Send the directory number of the calling device. • First Redirect Number—Send the directory number of the redirecting device. • Last Redirect Number—Send the directory number of the last device to redirect the call. • First Redirect Number (External)—Send the directory number of the first redirecting device with the external phone mask applied. • Last Redirect Number (External)—Send the directory number of the last redirecting device with the external phone mask applied.

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Called Party IE Type Unknown	Select the format for the number type in called party directory numbers. Cisco Unified Communications Manager Business Edition 3000 sets the called directory number (DN) type. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified Communications Manager Business Edition 3000 does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the called directory number to be encoded to a non-national type numbering plan. Select one of the following options: • Cisco Unified Communications Manager—Use when the Cisco Unified Communications Manager Business Edition 3000 sets the directory number type. • Unknown—Use when the dialing plan is unknown. • National—Use when you are dialing within the dialing plan for your country. • International—Use when you are dialing outside the dialing plan for your country. • Subscriber—Use when you are dialing a subscriber by using a shortened subscriber number.

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Calling Party IE Type Unknown	Select the format for the number type in calling party directory numbers. Cisco Unified Communications Manager Business Edition 3000 sets the calling directory number (DN) type. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified Communications Manager Business Edition 3000 does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the calling directory number to be encoded to a non-national type numbering plan. Select one of the following options: • Cisco Unified Communications Manager—Use when the Cisco Unified Communications Manager Business Edition 3000 sets the directory number type. • Unknown—Use when the dialing plan is unknown. • National—Use when you are dialing within the dialing plan for your country. • International—Use when you are dialing outside the dialing plan for your country. • Subscriber—Use when you are dialing a subscriber by using a shortened subscriber number.

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Called Numbering Plan	Select the format for the numbering plan in called party directory numbers. Cisco Unified Communications Manager Business Edition 3000 sets the called DN numbering plan. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified Communications Manager Business Edition 3000 does not recognize European national dialing patterns. You can also change this setting when you are connecting to PBXs by using routing as a non-national type number. Select one of the following options: • Cisco Unified Communications Manager—Use when the Cisco Unified Communications Manager Business Edition 3000 sets the Numbering Plan in the directory number. • ISDN—Use when you are dialing outside the dialing plan for your country. • National Standard—Use when you are dialing within the dialing plan for your country. • Private—Use when you are dialing within a private network. • Unknown—Use when the dialing plan is unknown.

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Calling Numbering Plan	Select the format for the numbering plan in calling party directory numbers. Cisco Unified Communications Manager Business Edition 3000 sets the calling DN numbering plan. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified Communications Manager Business Edition 3000 does not recognize European national dialing patterns. You can also change this setting when you are connecting to PBXs by using routing as a non-national type number. Select one of the following options: • Cisco Unified Communications Manager—Use when the Cisco Unified Communications Manager Business Edition 3000 sets the Numbering Plan in the directory number. • ISDN—Use when you are dialing outside the dialing plan for your country. • National Standard—Use when you are dialing within the dialing plan for your country. • Private—Use when you are dialing within a private network. • Unknown—Use when the dialing plan is unknown.
Sound Settings	
Input Gain (dB)	This allows you to change the amplitude of the voice signal coming into the gateway created by adjusting the decibel (dB) level of the signal. You can increase the volume of a signal coming into the gateway by either increasing input gain or decreasing attenuation. You can decrease the volume of the incoming signal by either decreasing the input gain value or increasing the output attenuation. The minimum value is –6, and the maximum value is 14.
Output Attenuation (dB)	Attenuation is measured in decibels, and the lower the number, the better the voice quality. Adjust the attenuation and input gain values to achieve maximum voice quality. The minimum value is –6, and the maximum value is 14.
Output Level (dB)	This option sets the attenuation of the signal before it enters the line. To reduce the line loss set the output level to 0, –7.5, or –15 dB.
PRI Channels	

Table 31-7 Settings On the Connection Settings Page for E1 PRI connection type

Setting	Description
Enable All Channels	If you leased the entire PRI span from your service provider, check this check box. For E1 PRI connection type, the channel range available is from 1 to 32 lines.
Enable Channel Ranges	If you did not lease the entire PRI span from your service provider, enter the partial channel range that is available for use. Enter commas to separate the ranges; for example, 6-8, 10-12. **Note** You can use up to five partial PRI channel ranges, beyond which you will have to lease the entire channel range.

Connection Type: T1 PRI

Table 31-8 describes the settings on the **Add PSTN Connection > Connection Settings** page when your chosen Connection Type is **T1 PRI**.

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Connection Settings	
Protocol Type	T1 PRI spans provide several options, depending on the carrier or switch. Determine the switch to which you are connecting and the preferred protocol.
Show Advanced Settings/Hide Advanced Settings	
Interface Settings	
Protocol Side	This setting specifies whether the gateway connects to a Network device or to a User device. Make sure that the two ends of the PRI connection use opposite settings. For example, if you connect to a PBX and the PBX uses User as its protocol side, select Network for this device. Typically, use User for this option for central office connections.
Clock	Select Internal or External for the clock source.
PCM Type	Specify the digital encoding format. Select one of the following formats: • a-law—Use for Europe and other countries, except North America, Hong Kong, Taiwan, and Japan • mu-law—Use for North America, Hong Kong, Taiwan, and Japan
Line Coding	Select the line coding from one of the following: • Binary 8-zero substitution (B8ZS) • Alternate mark inversion (AMI)

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Framing	Select the multiframe format of the span from one of the following: • Extended Superframe Format (ESF) • Superframe Format (SF)
Echo Cancellation	Select whether to enable or disable echo cancellation.
Coverage (ms)	If an issue occurs with echo cancellation, select a value to address the issue. Choose one of the following values: • 24 • 32 • 48 • 64 • 128 (available with MCS7890C1 internal gateway only) **Note** This option is available only if echo cancellation is enabled.
Channel Selection Order	Select the order in which channels or ports are enabled from first (lowest number port) to last (highest number port), or from last to first. Valid entries include TOP DOWN (first to last) or BOTTOM UP (last to first). If you are not sure which port order to use, select TOP DOWN.
Channel IE Type	Select one of the following values to specify whether channel selection is presented as a channel map or a slot map: • Timeslot Number—B-channel usage always indicates actual time slot map format (such as 1-15 and 17-31 for E1). • Slotmap—B-channel usage always indicates a slot map format. • Use Number When 1B—Channel usage indicates a channel map for one B-channel but indicates a slot map if more than one B-channel exists. • Continuous Number—Configures a continuous range of slot numbers (1-30) as the E1 logical channel number instead of the noncontinuous actual time slot number (1-15 and 17-31).
Delay for first restart (ticks)	Enter the rate at which the spans are brought in service. The delay occurs when many PRI spans are enabled on a system and the Inhibit Restarts at PRI Initialization check box is unchecked. For example, set the first five cards to 0 and set the next five cards to 16. (Wait 2 seconds before bringing them in service.)

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Delay between restarts (ticks)	Enter the time between restarts. The delay occurs when a PRI RESTART is sent if the Inhibit Restarts check box is unchecked.
Inhibit Restarts at PRI Initialization	A RESTART or SERVICE message confirms the status of the ports on a PRI span. If RESTART or SERVICE messages are not sent, Cisco Unified Communications Manager Business Edition 3000 assumes the ports are in service. When the D-Channel successfully connects with another PRI D-Channel, it sends a RESTART or SERVICE message when this check box is unchecked.
Enable G. Clear	Check this check box to enable G. Clear Codec support. Checking this check box causes echo cancellation and zero suppression for outbound calls to be disabled.
Trasmit UTF-8 for Calling Party Name	If you check the Transmit UTF-8 for Calling Party Name check box, the gateway sends unicode for the calling party name.
PRI-Specific Settings	
Display IE Delivery	Check the check box to enable delivery of the display information element (IE) in SETUP and NOTIFY messages (for DMS protocol) for the calling and connected party name delivery service.
Redirecting Number IE Delivery–Inbound	Check this check box to indicate the first redirecting number and the redirecting reason of the call when the call is forwarded. (The UUIE part of the outgoing SETUP message from the Cisco Unified Communications Manager Business Edition 3000 includes the Redirecting Number IE.) Uncheck the check box to exclude the first redirecting number and the redirecting reason. You use Redirecting Number IE for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number IE, check the check box.
Redirecting Number IE Delivery–Outbound	Check this check box to accept the Redirecting Number IE in the incoming SETUP message to the Cisco Unified Communications Manager Business Edition 3000. (The UUIE part of the SETUP message includes the Redirecting Number IE.) Uncheck the check box to exclude the Redirecting Number IE. You use Redirecting Number IE for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number IE, you should check the check box.

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Setup non-ISDN Progress Indicator IE Enable	Check this check box only if users are not receiving ringback tones on outbound calls. When this check box is checked, the Cisco Unified Communications Manager Business Edition 3000 sends Q.931 Setup messages out digital (that is, non-H.323) gateways with the Progress Indicator field set to non-ISDN. This message notifies the destination device that the gateway is non-ISDN and that the destination device should play in-band ringback. This problem usually associates with a Cisco Unified Communications Manager Business Edition 3000 that connect to PBXs through digital gateways.
Send Calling Name In Facility IE	Check the check box to send the calling name in the Facility IE field. By default, the Cisco Unified Communications Manager leaves the check box unchecked. Set this feature for a private network that has a PRI interface that is enabled for ISDN calling name delivery. When this check box is checked, the calling party name gets sent in the Facility IE of the SETUP or FACILITY message, so the name can display on the called party device. Set this feature for PRI trunks in a private network only. Do not set this feature for PRI trunks that are connected to the PSTN. **Note** This field applies to the NI2 protocol only.
Outbound Call Routing	
Calling Party Presentation	Select whether you want the Cisco Unified Communications Manager Business Edition 3000 to allow or restrict the display of the calling party phone number. • Default—If you do not want to change the calling line ID presentation • Allowed—To indicate that the “Calling Line ID is Allowed” on outbound calls • Restricted—To indicate that “Calling Line ID is Restricted” on outbound calls

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Calling Party Selection	Any outbound call on a gateway can send directory number information. Select which directory number is sent. Select one of the following options: • Originator—Send the directory number of the calling device. • First Redirect Number—Send the directory number of the redirecting device. • Last Redirect Number—Send the directory number of the last device to redirect the call. • First Redirect Number (External)—Send the directory number of the first redirecting device with the external phone mask applied. • Last Redirect Number (External)—Send the directory number of the last redirecting device with the external phone mask applied.
Called Party IE Type Unknown	Select the format for the number type in called party directory numbers. Cisco Unified Communications Manager Business Edition 3000 sets the called directory number (DN) type. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified Communications Manager Business Edition 3000 does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the called directory number to be encoded to a non-national type numbering plan. Select one of the following options: • Cisco Unified Communications Manager—Use when the Cisco Unified Communications Manager Business Edition 3000 sets the directory number type. • Unknown—Use when the dialing plan is unknown. • National—Use when you are dialing within the dialing plan for your country. • International—Use when you are dialing outside the dialing plan for your country. • Subscriber—Use when you are dialing a subscriber by using a shortened subscriber number.

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Calling Party IE Type Unknown	Select the format for the number type in calling party directory numbers. Cisco Unified Communications Manager Business Edition 3000 sets the calling directory number (DN) type. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified Communications Manager Business Edition 3000 does not recognize European national dialing patterns. You can also change this setting when you are connecting to a PBX that expects the calling directory number to be encoded to a non-national type numbering plan. Select one of the following options: • Cisco Unified Communications Manager—Use when the Cisco Unified Communications Manager Business Edition 3000 sets the directory number type. • Unknown—Use when the dialing plan is unknown. • National—Use when you are dialing within the dialing plan for your country. • International—Use when you are dialing outside the dialing plan for your country. • Subscriber—Use when you are dialing a subscriber by using a shortened subscriber number.

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Called Numbering Plan	Select the format for the numbering plan in called party directory numbers. Cisco Unified Communications Manager Business Edition 3000 sets the called DN numbering plan. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified Communications Manager Business Edition 3000 does not recognize European national dialing patterns. You can also change this setting when you are connecting to PBXs by using routing as a non-national type number. Select one of the following options: • Cisco Unified Communications Manager—Use when the Cisco Unified Communications Manager Business Edition 3000 sets the Numbering Plan in the directory number. • ISDN—Use when you are dialing outside the dialing plan for your country. • National Standard—Use when you are dialing within the dialing plan for your country. • Private—Use when you are dialing within a private network. • Unknown—Use when the dialing plan is unknown.

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Calling Numbering Plan	Select the format for the numbering plan in calling party directory numbers. Cisco Unified Communications Manager Business Edition 3000 sets the calling DN numbering plan. Cisco recommends that you do not change the default value unless you have advanced experience with dialing plans such as NANP or the European dialing plan. You may need to change the default in Europe because Cisco Unified Communications Manager Business Edition 3000 does not recognize European national dialing patterns. You can also change this setting when you are connecting to PBXs by using routing as a non-national type number. Select one of the following options: • Cisco Unified Communications Manager—Use when the Cisco Unified Communications Manager Business Edition 3000 sets the Numbering Plan in the directory number. • ISDN—Use when you are dialing outside the dialing plan for your country. • National Standard—Use when you are dialing within the dialing plan for your country. • Private—Use when you are dialing within a private network. • Unknown—Use when the dialing plan is unknown.
Sound Settings	
Input Gain (dB)	This allows you to change the amplitude of the voice signal coming into the gateway created by adjusting the decibel (dB) level of the signal. You can increase the volume of a signal coming into the gateway by either increasing input gain or decreasing attenuation. You can decrease the volume of the incoming signal by either decreasing the input gain value or increasing the output attenuation. The minimum value is –6, and the maximum value is 14.
Output Attenuation (dB)	Attenuation is measured in decibels, and the lower the number, the better the voice quality. Adjust the attenuation and input gain values to achieve maximum voice quality. The minimum value is –6, and the maximum value is 14.
PRI Channels	

Table 31-8 Settings On the Connection Settings Page for T1 PRI connection type

Setting	Description
Enable All Channels	If you leased the entire PRI span from your service provider, check this check box. For E1 PRI connection type, the channel range available is from 1 to 32 lines.
Enable Channel Ranges	If you did not lease the entire PRI span from your service provider, enter the partial channel range that is available for use. Enter commas to separate the ranges; for example, 6-8, 10-12. **Note** You can use up to five partial PRI channel ranges, beyond which you will have to lease the entire channel range.

Connection Type: T1 CAS

Table 31-9 describes the settings on the **Add PSTN Connection > Connection Settings** page when your chosen Connection Type is **T1 CAS**.

Table 31-9 Settings On the Connection Settings Page for T1 CAS connection type

Setting	Description
Connection Settings	
Show Advanced Settings/Hide Advanced Settings	
Interface Settings	
Clock	Select Internal or External for the clock source.
Channel Selection Order	Select the order in which channels or ports are enabled from first (lowest number port) to last (highest number port), or from last to first. Valid entries include TOP DOWN (first to last) or BOTTOM UP (last to first). If you are not sure which port order to use, select TOP DOWN.
Product Specific Configuration Layout	
Line Coding	Select the line coding from one of the following: • Binary 8-zero substitution (B8ZS) • Alternate mark inversion (AMI)
Framing	Select the multiframe format of the span from one of the following: • Extended Superframe Format (ESF) • Superframe Format (SF)
Echo Cancellation	Select whether to enable or disable echo cancellation.

Table 31-9 Settings On the Connection Settings Page for T1 CAS connection type

Setting	Description
Coverage (ms)	If an issue occurs with echo cancellation, select a value to address the issue. Choose one of the following values: • 24 • 32 • 48 • 64 • 128 (available with MCS7890C1 internal gateway only) **Note** This option is available only if echo cancellation is enabled.
Input Gain (dB)	This allows you to change the amplitude of the voice signal coming into the gateway created by adjusting the decibel (dB) level of the signal. You can increase the volume of a signal coming into the gateway by either increasing input gain or decreasing attenuation. You can decrease the volume of the incoming signal by either decreasing the input gain value or increasing the output attenuation. The minimum value is –6, and the maximum value is 14.
Output Attenuation (dB)	Attenuation is measured in decibels, and the lower the number, the better the voice quality. Adjust the attenuation and input gain values to achieve maximum voice quality. The minimum value is –6, and the maximum value is 14.
Channel Configuration	
Channel Range	Specifies the channel range available for TI CAS connection type. Enter channels with dashes to specify ranges and commas to separate values, for example, 1-5,7,12. A maximum of 24 channels can be configured.
Expected Digits	Specifies the number of digits the user dials to place a call through the PSTN connection. The Cisco Unified Communications Manager Business Edition 3000 communicates this number to the Cisco Unified Communications Manager. This field works in tandem with the Extension Length field on the Dial Plan settings, which tells the Cisco Unified Communications Manager how many of the expected digits are significant.
Signalling type	Select one of the following: • Wink Start • Delay Dial

Connection Type: SIP Trunk

Table 31-10 describes the settings on the **Add PSTN Connection > Connection Settings** page when your chosen Connection Type is **SIP trunk**.

Table 31-10 Connection Settings for SIP Trunk Connection Type

Setting	Description
General Information	
Connection Name	Displays the name of the PSTN connection.
Description	Displays the description provided for the PSTN connection.
Connection Type	Displays the type of PSTN connection.
Device Type	Displays the type of gateway device.
Device Name	Displays the name of the gateway.
Connection Settings	
Provider IP Address	Specify the IP address of the service provider.
Provider Port	Specify the port of the service provider
BE3000 Port	Specify the port of the Cisco Unified Communications Manager Business Edition 3000 to which the service provider is connected.
Show/Hide Advanced Settings **Note** In this section, the parameters that are exposed by the service provider for editing appear on the Cisco Unified Communications Manager Business Edition 3000 Administrative Interface.	
Transport Type Information	
Outgoing Transport	From the drop-down list box, choose the outgoing transport mode. Select one of the following: • TCP • UDP
Media Information	
SDP Session-level Bandwidth Modifier for Early Offer and Re-INVITEs	The Session Level Bandwidth Modifier specifies the maximum amount of bandwidth needed when all the media streams are used. There are three Session Level Bandwidth Modifiers: Transport Independent Application Specific (TIAS), Application Specific (AS), and Conference Total (CT). Select one of the following options to specify which Session Level Bandwidth Modifier to include in the SDP portion of SIP Early Offer or Reinvite requests. • TIAS and AS • TIAS only • AS only • CT only

Table 31-10 Connection Settings for SIP Trunk Connection Type (continued)

Setting	Description
Include Inactive SDP attribute in mid-call media changes	Check this check box to allow the Cisco Unified Communications Manager Business Edition 3000 to send an INVITE a=inactive SDP message during call hold or media break during supplementary services.
Include Send-Receive SDP attribute in mid-call INVITEs	When you enable send-receive SDP in mid-call INVITE for an early offer SIP trunk in tandem mode, Cisco Unified Communications Manager inserts Media Termination Point (MTP) to provide send/receive SDP when a SIP device sends offer SDP with *a=inactive* or *sendonly* or *recvonly* in audio media line. In tandem mode, Cisco Unified Communications Manager depends on the SIP devices to initiate reestablishment of media path by sending either a delayed INVITE or mid-call INVITE with send-recv SDP.
Disable Early Media when 180 Ringing message received	By default, Cisco Unified Communications Manager Business Edition 3000 will signal the calling phone to play local ringback if SDP is not received in the 180 or 183 response. If SDP is included in the 180 or 183 response, instead of playing ringback locally, Cisco Unified Communications Manager Business Edition 3000 will connect media, and the calling phone will play whatever the called device is sending (such as ringback or busy signal). If you do not receive ringback, the device to which you are connecting may be including SDP in the 180 response, but it is not sending any media before the 200 OK response. In this case, check this check box to play local ringback on the calling phone and connect the media upon receipt of the 200 OK response **Note** Even though the phone that is receiving ringback is the calling phone, you need the configuration on the called device profile because it determines the behavior.
Include audio mline in outgoing T.38 INVITE	The parameter allows the system to accept a signal from Microsoft Exchange that causes it to switch the call from audio to T.38 fax. To use this feature, you must also configure a SIP trunk with this SIP profile. For more information, see trunk Configuration. **Note** The parameter applies to SIP Trunks only, not phones that are running SIP or other endpoints.
Enable Early Offer support for voice and video calls	Check this check box if you want to create a trunk that supports early offer. Early Offer configurations on SIP profile apply to SIP trunk calls.

Table 31-10 Connection Settings for SIP Trunk Connection Type (continued)

Setting	Description
Early Offer for G.Clear Calls	The Early Offer for G.Clear Calls feature supports both standards-based G.Clear (CLEARMODE) and proprietary Cisco Session Description Protocols (SDP). To enable or disable Early Offer for G.Clear Calls, choose one of the following options: • Disabled • CLEARMODE • CCD • G.nX64 • X-CCD
DTMF Signaling Method	Choose from the following options: • No Preference (default)—Cisco Unified Communications Manager will pick the DTMF method to negotiate DTMF, so the call does not require an MTP. If Cisco Unified Communications Manager has no option but to allocate an MTP (if the Media Termination Point Required check box is checked), SIP trunk will negotiate DTMF to RFC 2833. • RFC 2833—Choose this configuration if the preferred DTMF method to be used across the trunk is RFC 2833. Cisco Unified Communications Manager makes every effort to negotiate RFC 2833, regardless of a media termination point (MTP) usage. Out-of-band provides the fallback method if the peer endpoint supports it. • OOB and RFC 2833—Choose this configuration if both out of band and RFC 2833 should be used for DTMF. **Note** If the peer endpoint supports both out of band and RFC 2833, Cisco Unified Communications Manager will negotiate both out-of-band and RFC 2833 DTMF methods. As a result, two DTMF events would get sent for the same DTMF keypress (one out of band and the other RFC 2833).
SIP Rel1xx Options	This field configures SIP Rel1xx, which determines whether all SIP provisional responses (other than 100 Trying messages) get sent reliably to the remote SIP endpoint. Valid values follow: • Disabled—Disables SIP Rel1xx. • Send PRACK if 1xx contains SDP—Acknowledges a 1xx message with PRACK, only if the 1xx message contains SDP. • Send PRACK for all 1xx messages—Acknowledges all1xx messages with PRACK.
Presentation and Identification Information	

Table 31-10 Connection Settings for SIP Trunk Connection Type (continued)

Setting	Description
Transmit UTF-8 for Calling Party Name	This device uses the user locale setting of the SIP Trunks to determine whether to send unicode and whether to translate received Unicode information. For the sending device, if you check this check box and the user locale setting in the device pool at the device matches the terminating phone user locale, the device sends unicode. If the user locale settings do not match, the device sends ASCII. The receiving device translates incoming unicode characters based on the user locale setting of the sending device pool of the device. If the user locale setting matches the terminating phone user locale, the phone displays the characters. **Note** The phone may display garbled characters if the two ends of the trunk configure user locales that do not belong to the same language group.
Deliver Conference Bridge Identifier	Check this check box for the SIP trunk to pass the b-number that identifies the conference bridge across the trunk instead of changing the b-number to the null value. The terminating side does not require that this option be enabled. Checking this check box is not required for the Open Recording Architecture (ORA) SIP header enhancements to the Recording feature to work. Enabling this option allows the recorder to coordinate recording sessions for conference calls.
Include Remote-Party-ID	Cisco Unified Communications Manager Business Edition 3000 uses the following as supplementary services: • Calling Line ID Presentation (CLIP/CLIR)—To allow or restrict the originating called phone number on a call-by-call basis • Calling Name Presentation (CNIP/CNIR)—To allow or restrict the originating caller name on a call-by-call basis Check this check box to enable this option on the Cisco Unified Communications Manager Business Edition 3000.
Calling Line ID Presentation–Outgoing	Choose whether you want the Cisco Unified Communications Manager to allow or restrict the display of the calling party phone number on the called party phone display for this SIP trunk. • Default—If you do not want to change calling line ID presentation • Allowed—If you want Cisco Unified Communications Manager to allow the display of the calling number • Restricted—If you want Cisco Unified Communications Manager to block the display of the calling number

Table 31-10 Connection Settings for SIP Trunk Connection Type (continued)

Setting	Description
Calling Name Presentation–Outgoing	Choose whether you want the Cisco Unified Communications Manager to allow or restrict the display of the calling party name on the called party phone display for this SIP route pattern. Choose one of the following: • Default—if you do not want to change calling name presentation. • Allowed—if you want Cisco Unified Communications Manager Business Edition 3000 to display the calling name information. • Restricted—if you want Cisco Unified Communications Manager Business Edition 3000 to block the display of the calling name information.
Connected Line ID Presentation–Incoming	Choose whether you want the Cisco Unified Communications Manager to allow or restrict the display of the calling party phone number on the called party phone display for this SIP route pattern. • Default—if you do not want to change calling line ID presentation. • Allowed—if you want Cisco Unified Communications Manager to allow the display of the calling number. • Restricted—if you want Cisco Unified Communications Manager to block the display of the calling number.
Connected Name Presentation–Incoming	Choose whether you want the Cisco Unified Communications Manager to allow or restrict the display of the calling party name on the caller party phone display for this SIP route pattern. Choose one of the following: • Default—if you do not want to change calling name presentation. • Allowed—if you want Cisco Unified Communications Manager Business Edition 3000 to display the calling name information. • Restricted—if you want Cisco Unified Communications Manager Business Edition 3000 to block the display of the calling name information.
Include Asserted-Identity	Check this check box to define the asserted type and SIP privacy for SIP trunk messages.

Table 31-10 Connection Settings for SIP Trunk Connection Type (continued)

Setting	Description
Asserted-Type	From the drop-down list, choose one of the following values to specify the type of Asserted Identity header that SIP trunk messages should include: • Default—This option represents the default value; Screening indication information that the SIP trunk receives from Cisco Unified Communications Manager Call Control determines the type of header that the SIP trunk sends. • PAI—The Privacy-Asserted Identity (PAI) header gets sent in outgoing SIP trunk messages; this value overrides the Screening indication value that comes from Cisco Unified Communications Manager. • PPI—The Privacy Preferred Identity (PPI) header gets sent in outgoing SIP trunk messages; this value overrides the Screening indication value that comes from Cisco Unified Communications Manager. **Note** These headers get sent only if the Include Asserted-Identity check box is checked.

Table 31-10 Connection Settings for SIP Trunk Connection Type (continued)

Setting	Description
SIP Privacy	From the drop-down list, choose one of the following values to specify the type of SIP privacy header for SIP trunk messages to include: • Default—This option represents the default value; Name/Number Presentation values that the SIP trunk receives from the Cisco Unified Communications Manager Call Control compose the SIP Privacy header. For example, if Name/Number presentation specifies Restricted, the SIP trunk sends the SIP Privacy header; however, if Name/Number presentation specifies Allowed, the SIP trunk does not send the Privacy header. • None—The SIP trunk includes the Privacy:none header and implies Presentation allowed; this value overrides the Presentation information that comes from Cisco Unified Communications Manager. • ID—The SIP trunk includes the Privacy:id header and implies Presentation restricted for both name and number; this value overrides the Presentation information that comes from Cisco Unified Communications Manager. • ID Critical—The SIP trunk includes the Privacy:id; critical header and implies Presentation restricted for both name and number. The label "critical" implies that privacy services that are requested for this message are critical, and, if the network cannot provide these privacy services, this request must be rejected. This value overrides the Presentation information that comes from Cisco Unified Communications Manager. **Note** These headers get sent only if the Include Asserted-Identity check box is checked.
Redirecting Information	
Send Redirecting Diversion Header	Check this check box to accept the Redirecting Number in the incoming INVITE message to the Cisco Unified Communications Manager. **Note** You use Redirecting Number for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number, you should check the check box.
Accept incoming Redirecting Diversion Header	Uncheck the check box to exclude the Redirecting Number in the incoming INVITE message to the Cisco Unified Communications Manager. **Note** You use Redirecting Number for voice-messaging integration only. If your configured voice-messaging system supports Redirecting Number, you should check the check box.

Table 31-10 Connection Settings for SIP Trunk Connection Type (continued)

Setting	Description
Redirect by application	Checking this check box and configuring this SIP Profile on the SIP trunk allows the Cisco Unified Communications Manager administrator to • Apply digit analysis to the redirected contacts to make sure that the call get routed correctly. • Prevent DOS attack by limiting the number of redirection (recursive redirection) that a service parameter can set. • Allow other features to be invoked while the redirection is taking place. Getting redirected to a restricted phone number (such as an international number) means that handling redirection at the stack level will cause the call to be routed instead of being blocked. This represents the behavior that you will get if the Redirect by Application check box is unchecked.
SIP OPTIONS Ping Information	
Enable OPTIONS ping	Check this check box if you want to enable the SIP OPTIONS feature. SIP OPTIONS are requests to the configured destination address on the SIP trunk. If the remote SIP device fails to respond or sends back a SIP error response such as 503 Service Unavailable or 408 Timeout, Cisco Unified Communications Manager tries to reroute the calls by using other Trunks or by using a different address. If this check box is not checked, the SIP trunk does not track the status of SIP trunk destinations. When this check box is checked, you can configure two request timers.
In-Service Trunk Ping Interval (sec)	This field configures the time duration between SIP OPTIONS requests when the remote peer is responding and the trunk is marked as In Service. If at least one IP address is available, the trunk is In Service; if all IP addresses are unavailable, the trunk is Out of Service. The default value specifies 60 seconds. Valid values range from 5 to 600 seconds.
Out-of-service Trunk Ping Interval (sec)	This field configures the time duration between SIP OPTIONS requests when the remote peer is not responding and the trunk is marked as Out of Service. The remote peer may be marked as Out of Service if it fails to respond to OPTIONS, if it sends 503 or 408 responses, or if the Transport Control Protocol (TCP) connection cannot be established. If at least one IP address is available, the trunk is In Service; if all IP addresses are unavailable, the trunk is Out of Service. The default value specifies 120 seconds. Valid values range from 5 to 600 seconds.

Table 31-10 Connection Settings for SIP Trunk Connection Type (continued)

Setting	Description
Retry Timer (ms)	This field specifies the maximum waiting time before retransmitting the OPTIONS request. Valid values range from 100 to 1000 milliseconds. The default value specifies 500 milliseconds.
Retry Count	This field specifies the number of times that Cisco Unified Communications Manager resends the OPTIONS request to the remote peer. After the configured retry attempts are used, the destination is considered to have failed. To obtain faster failure detection, keep the retry count low. Valid values range from 1 to 10. The default value specifies 6.

Connection Type: FXO

Table 31-11 Connection Settings for SPA8800 Connection Type

Settings	Description
Basic Parameters	
Port	Also referred to as Line Number, this is the corresponding FXO analog port on the SPA8800. Ports assigned to other PSTN connections are not available for selection.
Description	This field describes the FXO connection.
Direct Inward Dial (DID) Number	This field specifies the number associated to the PSTN connection. By default, this number will route to the auto-attendant unless additional configurations to make the number routable to a callable endpoint are performed.

<table>
<tr><th>Settings</th><th>Description</th></tr>
<tr><td>Line Usage</td><td>There are two main Line Usage options:
All Call Types
Emergency Calls Only All Call Types: The PSTN connection may be used for any incoming and outgoing calls to the PSTN. If you are using the All Call Types, you should have a directory number, translation, or hunt pilot set up to receive inbound calls on that DID. Emergency Calls Only: The outbound PSTN connection is only used for calls to designated emergency service numbers that have been identified by the country chosen during First Time Setup or to alternate emergency service numbers as defined on the site page. If this PSTN connection is selected as Emergency Calls Only, the DID Number for the PSTN connection will be configured as an Emergency Locator ID Number (ELIN) if not previously configured. If you remove ELIN from a site, the setting reverts back to the All Call Types option. Conversely, if an Emergency Calls Only trunk is converted to All Call Types, ELIN is removed from that site. Inbound calls from the PSTN to PSTN connections designated as Emergency Calls Only should only be call-backs on emergency calls. The DID cannot be used for non-emergency calls. To do so could prevent emergency calls from going out, given that the analog line only allows one call at a time. </td></tr>
<tr><td colspan="2">Advanced Parameters</td></tr>
<tr><td colspan="2">PSTN Disconnect Detection</td></tr>
<tr><td>Detect CPC</td><td>CPC, or Calling Party Control, is a brief removal of tip-and-ring voltage to indicate the disconnection of a party from the call. If enabled, the SPA disconnects both call legs when this signal is detected.</td></tr>
<tr><td>Detect Polarity Reversal</td><td>If enabled, the SPA8800 disconnects both call legs when a polarity reversal occurs. On calls originating from the PSTN, the first polarity reversal is ignored and the second one triggers the disconnect. For all other calls across the SPA, the first polarity reversal triggers the disconnect.</td></tr>
<tr><td>Detect PSTN Long Silence</td><td>If enabled, the SPA8800 disconnects both call legs when the PSTN side has no voice activity for a duration longer than the length specified in the PSTN Long Silence Duration parameter during a call across the connection.</td></tr>
<tr><td>Detect VoIP Long Silence</td><td>If enabled, the SPA8800 disconnects both call legs when the VoIP side (non-PSTN) has no voice activity for a duration longer than the length specified in the VoIP Long Silence Duration parameter during a call across the connection.</td></tr>
</table>

Settings	Description
PSTN Long Silence Duration	This field defines the minimum length of PSTN silence (or inactivity) in seconds to trigger a disconnect of a call across the SPA8800 device if Detect PSTN Long Silence is enabled.
VoIP Long Silence Duration	This field defines the minimum length of VoIP silence (or inactivity) in seconds to trigger a disconnect of a call across the SPA8800 device if Detect VoIP Long Silence is enabled.
PSTN Silence Threshold	This parameter adjusts the sensitivity of PSTN silence detection. Choose from {very low, low, medium, high, very high}. The higher the setting, the easier it is to detect silence and trigger a disconnection.
Min CPC Duration	This field defines the minimum duration, in seconds, of a low tip-and-ring voltage for the SPA8800 device to recognize the drop in voltage as a CPC signal or PSTN line removal.
Detect Disconnect Tone	If enabled, the SPA8800 device disconnects both call legs when it detects the disconnect tone from the PSTN side during a call across the device. The disconnect tone varies by country and can be affected by the country of deployment chosen during First Time Setup (FTS).
Device Settings	
Fax Mode	Choose between G.711 Passthrough or T.38 Fax
FXS Port Input Gain	This field defines the input gain for the FXS port in dB, up to three decimal places. The range is 6.000 to -12.000.
FXS Port Output Gain	This field defines the FXS output gain in dB, up to three decimal places. The range is 6.000 to -12.000. The Call Progress Tones and DTMF playback level are not affected by the FXS Port Output Gain parameter.
DTMF Playback Level	This field defines the local DTMF playback level in dBm, up to one decimal place.
DTMF Playback Length	This field defines the local DTMF playback duration in milliseconds.
FXS Port Impedance	This field sets the electrical impedance of the FXS port. Choices are: 600, 900, 600+2.16uF, 900+2.16uF, 270+750||150nF, 220+850||120nF, 220+820||115nF, or 200+600||100nF.
FXO Port Impedance	This field defines the desired impedance of the FXO Port. The impedance values for various countries are: • US—600 • EU (UK, Germany, Netherlands, Sweden, Norway, Italy, Spain, Portugal, Poland, and Denmark)—270+750||150nF • France—270+750||150nF • Australia—220+820||120nF • New Zealand—370+620||310nF
International Control	

Settings	Description
Ring Frequency Min	The value entered in this field should match or be slightly less than the lower limit of the ring frequency (Hz) used to detect the ring signal. This value differs based upon the country of deployment and should only be adjusted should a call across the SPA8800 device to known working number not generate a ring tone.
SPA To PSTN Gain	This field defines the amount of digital gain/attenuation, in decibels (dB), to be applied to the signal sent from the SPA8800 device to the PSTN. Increase this value if parties on the PSTN side have trouble hearing parties on your system or lower it if they are coming across too loud. Range is -15 to 12 dB.
Ring Frequency Max	This value entered in this field should match or be slightly above the higher limit of the ring frequency (Hz) used to detect the ring signal. This value differs based upon the country of deployment and should only be adjusted should a call across the SPA8800 device to known working number not generate a ring tone.
PSTN To SPA Gain	This field defines the amount of digital gain/attenuation, in decibels (dB) applied to the signal sent from the PSTN to the SPA8800 device. Increase this value if parties on the PSTN side are difficult to hear by parties on your system or lower it if they are coming across too loud. Range is -15 to 12 dB.
Ring Validation Time	This field defines the smallest amount of time required by the SPA8800 device to recognize a signal sent across the PSTN connection as a ring signal. If a call from the PSTN is not signaling a ring by a phone on your system, try increasing the validation time until a phone on your system rings.
Tip Ring Voltage Adjust	
Ring Indication Delay	This field defines the amount of time to wait prior to the ring indication occurring in a signal coming across the SPA8800 device. Some amount of time, dependent on country, is required for caller ID to be recognized for the call. Increase this time if a call that is supposed to have caller ID is not showing anything on the display. Decrease it if a ring appears to be occurring late or appears to be cut off.
Operational Loop Current Min	This field defines the minimum loop current that maintains the device in an off-hook state. If the current is lower than this value, the device will enter an on-hook state.
Ring Timeout	This field defines the maximum time allowed for the ring to cross between the higher and lower thresholds. It is expressed as the maximum allowed 1/f in which f is the ring frequency.
On Hook Speed	This field defines the speed with which a call goes from off-hook to on-hook when the phone is hung up.
Ring Threshold	This field defines the minimum voltage that can be detected as a ring.
Current Limiting Enable	This field controls the current alarm threshold.

Settings	Description
Ringer Impedance	This field sets the ringer impedance. Certain countries, like South Africa, require a low ringer impedance.
Line In Use Voltage	This field sets the voltage that must be maintained across a line to designate that the line is in use. This parameter should match the value used by the PSTN.

Click **Finish** to complete adding a PSTN connection to Cisco Unified Communications Manager Business Edition 3000 server. You can add upto 300 connections.