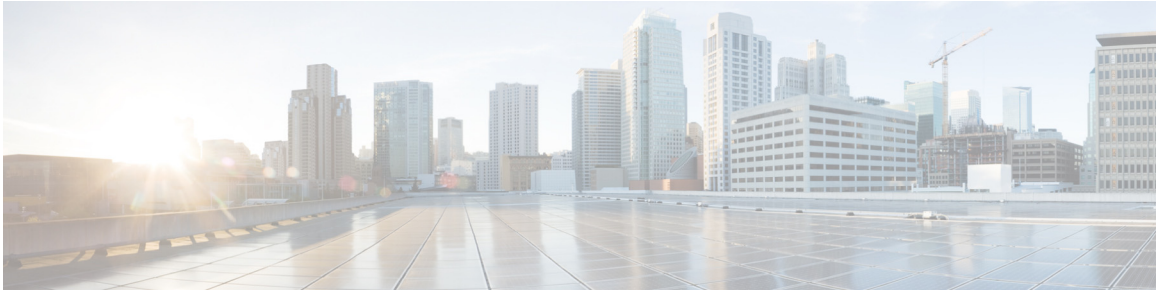

Configuring FXS Ports for Basic Calls

First Published: October 2, 2008
Last updated: September 4, 2015

This module describes how to configure analog Foreign Exchange Station (FXS) ports on a Cisco Integrated Services Router (ISR) or Cisco VG224 Analog Phone Gateway for basic calls.

Finding Feature Information in This Module

Your Cisco IOS software release may not support all of the features documented in this module. To reach links to specific feature documentation and to see a list of the releases in which each feature is supported, see the "Feature Information for Configuring FXS Ports for Basic Calls" section on page 45.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS and Catalyst OS software image support. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Contents

Prerequisites for Configuring FXS Ports for Basic Calls

Cisco IOS Gateway

- The Cisco voice gateway must be set up and configured for operation. For a list of supported Cisco voice gateways, see the "Overview of Supplementary Services Features for FXS Ports on Cisco Voice Gateways" section on page 9. For configuration information, see the appropriate Cisco configuration documentation.
- The analog FXS voice ports are set up and configured for operation. For information, see the *Cisco IOS Voice Port Configuration Guide*.

Analog Endpoints in Cisco Unified Communications Manager

- Cisco Unified Communications Manager 4.2 or a later version.
- Cisco voice gateway analog Foreign Exchange Station (FXS) ports are added in Cisco Unified Communications Manager. Each analog FXS port on which SCCP is enabled counts as a single IP phone for licensing purposes. For example, to register all 24 ports on a Cisco VG224 Analog Phone Gateway, the 24 ports count toward the 2500 limit if you purchase a Cisco Unified Communications Manager license for 2500 devices.

Analog Endpoints in Cisco Unified CME

- Basic feature license and phone user licenses are purchased. For licensing purposes, each analog FXS port on which SCCP is enabled counts as a single phone.

Information About FXS Ports for Basic Calls

To configure FXS ports for basic calls, you should understand the following concepts:

- Hookflash Duration, page 18
- PLAR with DTMF Out-Pulse Digits, page 19
- Dial Tone Generation after Remote Onhook, page 19
- Ground Start FXS Ports as SCCP Analog Endpoints, page 20
- Supervisory Disconnect, page 20

Hookflash Duration

Analog phones use hookflash to access a second dial tone to initiate certain SCCP phone features such as transfer and conference. Hookflash is an on-hook condition of short duration that is usually generated when a phone user presses the Flash button on a phone. The duration of an on-hook condition generated by a Flash button varies for different phone makes and models. Cisco voice gateways measure the duration of detected on-hook conditions to determine whether they should be interpreted as hookflash or not. The duration of a detected on-hook condition is interpreted by Cisco IOS software as follows:

- An on-hook condition that lasts for a time period that falls inside the hookflash duration range is considered a hookflash.
- An on-hook condition that lasts for a shorter period than the lower limit of the range is ignored.
- An on-hook condition that lasts for a longer period than the higher limit of the range is considered a disconnect.

The hookflash duration range for FXS ports is defined as follows:

- The lower limit of the range is set in software at 150 ms, although there is also a hardware-imposed lower limit that is typically about 20 ms, depending on platform type. An on-hook condition that lasts for a shorter time than this hardware-imposed lower limit is not reported to the Cisco IOS software.
- The upper limit of the range is set in software at 1000 ms by default, although this value can be changed on the voice gateway. The upper limit can be set to any value from 50 to 1550 ms.

- If the upper limit of the hookflash duration range is X, a value greater than 150, then any on-hook duration between 150 and X is interpreted as a hookflash. For example, if X is 1550, the hookflash duration range is 150 to 1550 ms. An on-hook signal that lasts for 1250 ms is interpreted as a hookflash, and an on-hook signal of 55 ms is ignored.
- If the upper limit of the hookflash duration range is X, a value less than 150, then any on-hook duration between Y, the hardware lower limit, and X is interpreted as a hookflash. For example, if X is 65, the hookflash duration range is Y to 65 ms (assume Y is 20 ms). An on-hook signal that lasts for 1250 ms is interpreted as a disconnect, and an on-hook signal of 55 ms is interpreted as a hookflash. An on-hook signal of less than Y is ignored.

For information about modifying the upper limit of the hookflash duration range, see the "Modifying Hookflash" section on page 27.

PLAR with DTMF Out-Pulse Digits

A private line automatic ring-down (PLAR) connection allows an analog phone user to make a call without dialing any digits. When the user goes off-hook on the phone, the Cisco voice gateway automatically rings a predefined extension or PSTN number. The PLAR number is configured on the analog FXS port to which the corresponding analog phone is connected.

The PLAR with DTMF out-pulse digits feature in Cisco IOS Release 12.4(9)T is an enhancement that enables the voice gateway to out-pulse additional DTMF digits after the PLAR connection is up. These DTMF digits are configurable and can include 0 to 9, A to D, a comma (,) for a one-second pause, an asterisk (*), and number sign (#). If an analog phone user presses a string of digits (0-9, *, #) after taking a PLAR phone off-hook, the voice gateway buffers the digit string until the DTMF digits are done being out-pulsed. After the voice gateway sends all the DTMF digits, it sends the buffered digits to the destination port.

Although users do not hear a dial tone when taking a PLAR phone off-hook, PLAR phones support the same features as other analog phones. PLAR phones can receive incoming calls and support hookflash for basic supplementary features such as call transfer, call waiting, and conference. Feature access codes (FACs) and speed-dial codes are not valid immediately after taking a PLAR phone off-hook, but after connecting to the destination port, a user can press hookflash to get a dial tone and then dial an access code for features such as speed dial, redial, and call transfer.

For configuration information, see the "Configuring PLAR with DTMF Out-Pulse Digits" section on page 28.

Dial Tone Generation after Remote Onhook

The Dial Tone Generation after Remote Onhook feature provides PBX interoperability by enabling configurable automatic dial tone capability after remote call disconnect. Dial tone is automatically generated to the remaining party in a basic A-B call scenario once one party disconnects, in the same way that a PBX user gets immediate dial tone after remote party disconnect. This allows the user to make a new call without hookflash or going onhook, then off hook. If automatic dial tone generation is disabled, the user is required to go onhook then off-hook, or perform a hookflash, in order to make a new call.

After remote onhook there are two ways for an SCCP analog phone to redial, either by the user pressing a redial button or entering a feature access code (FAC). Some phone model redial buttons do not function with the dial tone after remote onhook feature enabled, resulting in redial digits not being sent. For this reason, the dial tone generation after remote onhook feature supports redial only when activated by FAC.

Dial tone generation immediately after the remote party goes onhook is configurable on a per port basis and is enabled by default. Automatic dial tone is supported only on STC application-controlled loop start FXS ports. For devices such as interactive voice response (IVR) systems that require power denial to disconnect properly, power denial is triggered prior to dial tone generation after one party disconnects. For a PLAR port, dial tone is played instead of triggering another PLAR after remote party disconnect. You cannot configure the dial tone generation after remote onhook feature using the Cisco Unified Communications Manager auto configuration capability.

For configuration information, see the "Configuring SCCP Gateway Dial Tone Generation After Remote Onhook" section on page 32.

Ground Start FXS Ports as SCCP Analog Endpoints

SCCP enhanced supplementary features provide support on the SCCP analog gateway for ground start FXS ports, used for PBX and key system connections, enabling disconnect supervision and Cisco Unified Communications Manager registration. The ground start FXS port feature is supported for basic calls only and supports PBX interoperability by providing supervisory disconnect to FXS ports and analog endpoints. Prior to ground start FXS support, there was no disconnect supervision to signal the end of a call. The ground start FXS ports feature provides power denial-based supervisory disconnect to indicate remote party disconnect using the loop current feed open (LCFO) mechanism.

For configuration information, see the "Configuring SCCP Gateway Ground Start FXS Ports" section on page 34.

Supervisory Disconnect

The Supervisory Disconnect feature provides a disconnect indication to the remote party in a two-party call after one side disconnects. This enables external applications connected to the Cisco voice gateway to promptly clear a call after receiving the disconnect indication. This feature triggers a power denial on FXS ports with loop-start signaling when a voice call disconnects. The power denial is only generated in a two-party call scenario when one party disconnects. Power denial is not generated if a call is on hold and either the active party or the on-hold party hangs up. Power denial is also not generated for a three-way conference call when one party hangs up. This feature is enabled and disabled on a voice port basis. The remote party receives the power denial signal for the duration that is set on the analog FXS port.

Because the Cisco voice gateway cannot distinguish the type of device connected to the analog FXS port, a power denial signal is sent to all FXS ports that have the power denial feature enabled. This can result in analog phones also receiving a power denial signal after one party disconnects in a two-party call. The remaining party hears a brief click sound. To prevent this behavior on analog phones, you can disable the power denial feature on the analog FSX voice port. For configuration information, see the "Configuring Supervisory Disconnect" section on page 36.

How to Configure FXS Ports for Basic Calls

Note This document does not contain details about configuring Cisco Unified Communications Manager or Cisco Unified CME. See the documentation for these products for installation and configuration instructions.

This section contains the following tasks for setting up SCCP analog phone support:

- Enabling SCCP on the Voice Gateway, page 21 (required)
- Enabling the STC Application for Analog FXS Ports, page 24 (required)
- Modifying Hookflash, page 27 (optional)
- Configuring PLAR with DTMF Out-Pulse Digits, page 28 (optional)
- Configuring SCCP Gateway Dial Tone Generation After Remote Onhook, page 32 (optional)
- Configuring SCCP Gateway Ground Start FXS Ports, page 34 (optional)
- Configuring Supervisory Disconnect, page 36 (optional)
- Verifying and Troubleshooting the Configuration, page 38 (optional)

Enabling SCCP on the Voice Gateway

To enable SCCP on the local interface that communicates with your Cisco call-control system and identify priority levels to Cisco Unified Communications Manager servers or Cisco Unified CME routers, perform the following steps.

Note If more than 72 end points are configured with SCCP in a single voice gateway, we recommend you to increase the hold-queue size on the interface of the gateway to 300.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sccp local** *interface-type interface-number* [**port** *port-number*]
4. **sccp ccm** {*ip-address* | *dns*} **identifier** *identifier-number* [**port** *port-number*] [**version** *version-number*]
5. **sccp**
6. **sccp ccm group** *group-number*
7. **associate ccm** *identifier-number* **priority** *priority-number*
8. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**sccp local** *interface-type interface-number* [**port** *port-number*] **Example:** `Router(config)# sccp local FastEthernet0/0`	Selects the local interface that SCCP applications (transcoding and conference) use to register with Cisco Unified Communications Manager and Cisco Unified CME. • *port-number*—(Optional) TCP or UDP port number used by the selected interface. Range: 1025 to 65535. Default: 2000.
Step 4	**sccp ccm** {*ip-address* | *dns*} **identifier** *identifier-number* [**port** *port-number*] [**version** *version-number*] **Example:** `Router(config)# sccp ccm 10.8.1.2 identifier 10 version 4.1`	Adds a Cisco Unified Communications Manager server or Cisco Unified CME router to the list of available call-control systems. • Repeat this step to add a backup system.
Step 5	**sccp** **Example:** `Router(config)# sccp`	Enables SCCP and its related applications (transcoding and conferencing).
Step 6	**sccp ccm group** *group-number* **Example:** `Router(config)# sccp ccm group 1`	Creates a group of Cisco Unified Communications Manager or Cisco Unified CME systems and enters SCCP ccm configuration mode. • *group-number*—Number that identifies the group. Range: 1 to 50. • Repeat this step to add a backup system.

	Command or Action	Purpose
Step 7	**associate ccm** *identifier-number* **priority** *priority-number* **Example:** `Router(config-sccp-ccm)# associate ccm 1 priority 1`	Adds a Cisco Unified Communications Manager server or Cisco Unified CME router to the group and establishes its priority within the group. • *identifier-number*—Identifier that was defined in the **sccp ccm** command in Step 4. • **priority** *priority-number*—Number that indicates the priority of this Cisco Unified Communications Manager server or Cisco Unified CME router. Range: 1 to 4, where 1 is the highest priority. **Note** A second Cisco Unified Communications Manager or Cisco Unified CME with a lower priority number becomes a backup system. **Note** The priority must match the order of the call manager group associated with this device within call manager.
Step 8	**registration timeout** *timeout-value* **Example:** `Router(config-sccp-ccm)# registration timeout 3`	(Optional) Sets the length of time between registration messages sent from SCCP to the Cisco Unified Communications Manager. • Time, in seconds, between registration messages. Range is 1 to 180. Default is 3.
Step 9	**keepalive timeout** *seconds* **Example:** `Router(config-sccp-ccm)# keepalive timeout 3`	(Optional) Sets the length of time between keepalive messages from Skinny Client Control Protocol (SCCP) to Cisco Unified Communications Manager. • Time between keepalive messages. Range is 1 to 180. Default is 30.
Step 10	**connect interval** *seconds* **Example:** `Router(config-sccp-ccm)# connect interval 3`	(Optional) Specifies the amount of time that a given digital signal processor (DSP) farm profile waits before attempting to connect to a Cisco Unified Communications Manager when the current Cisco Unified Communications Manager fails to connect. • Timer value, in seconds. Range is 1 to 3600. Default is 60.
Step 11	**switchback method graceful** **Example:** `Router(config-sccp-ccm)# switchback method graceful`	(Optional) Sets the Cisco Unified Communications Manager switchback method. • **graceful**—The Cisco Unified Communications Manager switchback happens only after all the active sessions are terminated gracefully.
Step 12	**end** **Example:** `Router(config-sccp-ccm)# end`	Exits SCCP ccm configuration mode and returns to privileged EXEC mode.

Examples

The following example shows the configuration for SCCP communication on Cisco VG224 Fast Ethernet interface 0/0 to two Cisco Unified Communications Manager servers.

```
Router# show running-config
.
.
.
sccp local FastEthernet0/0
sccp ccm 10.4.13.20 identifier 10
sccp ccm 10.4.13.70 identifier 12
sccp
!
sccp ccm group 1
 associate ccm 10 priority 1
 associate ccm 12 priority 2
!
```

Enabling the STC Application for Analog FXS Ports

To enable the SCCP telephony control (STC) application and configure analog voice ports on the voice gateway for control by the STC application, perform the following steps.

Prerequisites

- SCCP is enabled on the Cisco voice gateway. For configuration information, see the "Enabling SCCP on the Voice Gateway" section on page 21.
- To enable the STC application for analog ports on a voice gateway on which the **station-id number** command is configured, remove the configuration for the **station-id number** command before performing this task.

Note Only for FXS voice ports on a Cisco VG224 that is used with Cisco Unified CME or for FXS voice ports that are on a *different* router from Cisco Unified CME: To retain the station-id number configuration on your voice gateway, configure the **answer-address** command and do not remove the configuration for the **station-id number** command before performing this task.

Restrictions

- If Cisco Unified CME and the FXS voice ports to be controlled by the STC application are on the *same* voice gateway and the **station-id number** and **destination-pattern** commands are already configured on that gateway, the dial-peer matches the wrong entry and cannot access the STC application. To enable the STC application for FXS ports on a voice gateway on which Cisco Unified CME is configured, *remove* the configuration for the **station-id number** command before performing this task.
- If Cisco Unified CME and the FXS voice ports to be controlled by the STC application are on the *same* voice gateway and the **station-id number** command is configured for a voice-port which is controlled by the STC application with FACs, any feature code or speed dial code with ** will drop the call immediately. To use FACs that include **, *remove* the configuration for the **station-id number** command before performing this task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **stcapp ccm-group** *group-number*
4. **stcapp**
5. **dial-peer voice** *tag* **pots**
6. **service stcapp**
7. **port** *slot-number/port-number*
8. **exit**
9. **voice-port** *slot-number/port-number*
10. **caller-id enable**
11. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**stcapp ccm-group** *group-id* **Example:** `Router(config)# stcapp ccm-group 1`	Associates the STC application with a specific Cisco Unified Communications Manager group that controls calls and features. • *group-id*—Number that identifies the group. Use the number that was assigned with the **sccp ccm group** command in the "Enabling SCCP on the Voice Gateway" section on page 21.
Step 4	**stcapp** **Example:** `Router(config)# stcapp`	Enables the STC application.
Step 5	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 102 pots`	Defines a specific dial peer and enters dial-peer configuration mode. • *tag*—Number that identifies the dial peer. Range: 1 to 2147483647.
Step 6	**service stcapp** **Example:** `Router(config-dial-peer)# service stcapp`	Enables the STC application on the dial peer.

Step 7	**port** *slot-number/port-number*
	or
	port *slot-number/subunit-number/port*
	
	Example:
	`Router(config-dial-peer)# port 2/2`
	or
	`Router(config-dial-peer)# port 0/1/0`
		Assigns an analog voice port to the dial peer.
		• Format and values for analog FXS voice port number is platform-dependent. Type **?** for values.

Step 8	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 9	**voice-port** *slot-number/port-number* **Example:** `Router(config)# voice-port 2/0`	Enters voice-port configuration mode. • *slot-number/port-number*—Analog FXS voice port number. Range: 2/0 to 2/23.

	Command or Action	Purpose
Step 10	**`caller-id enable`** **Example:** `Router(config-voiceport)# caller-id enable`	(Optional) Enables caller ID for this voice port. **Note** Other parameters, such as caller-ID name and number, must be configured on the Cisco call-control system.
Step 11	**`end`** **Example:** `Router(config-voiceport)# end`	Exits voice-port configuration mode and returns to privileged EXEC mode.

Examples

The following example enables the STC application for Cisco Unified Communications Manager group 1 and associates the STC application with dial peer 102, to which the Cisco VG224 analog FXS port 2/2 has been assigned. This configuration also enables caller ID on voice port 2/2.

```
Router# show running-config
.
.
.
stcapp ccm-group 1
stcapp
!
dial-peer voice 102 pots
 service stcapp
 port 2/2
!
voice-port 2/2
 caller-id enable
.
.
.
```

Modifying Hookflash

To change the upper limit of the hookflash duration range for an analog FXS port, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port-number*
4. **timing hookflash-input** *milliseconds*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *slot-number/port-number* or **voice-port** *slot-number/subunit-number/port* **Example:** `Router(config)# voice-port 2/1`	Enters voice-port configuration mode. • Format and values for analog FXS voice port number is platform-dependent. Type **?** for values.
Step 4	**timing hookflash-input** *milliseconds* **Example:** `Router(config-voiceport)# timing hookflash-input 175`	Specifies the maximum duration of an on-hook condition that will be interpreted as a hookflash. • *milliseconds*—Range: 50 to 1550. Default: 1000.
Step 5	**end** **Example:** `Router(config-voiceport)# end`	Exits voice-port configuration mode and returns to privileged EXEC mode.

Configuring PLAR with DTMF Out-Pulse Digits

To configure an analog foreign exchange station (FXS) port to support PLAR, perform the following steps.

Prerequisites

- Cisco IOS Release 12.4(6)T or a later release for Cisco VG224 Analog Phone Gateways.
- Cisco IOS Release 12.4(9)T or a later release for Cisco ISRs.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**
4. **service stcapp**
5. **port** *slot-number/port*
6. **exit**

7. **sccp plar**
8. **voiceport** *port-number* **dial** *dial-string* [**digit** *dtmf-digits* [**wait-connect** *wait-msecs*] [**interval** *inter-digit-msecs*]]
9. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 102 pots`	Defines a specific dial peer and enters dial-peer configuration mode. • *tag*—Number that identifies the dial peer. Range: 1 to 2147483647.
Step 4	**service stcapp** **Example:** `Router(config-dial-peer)# service stcapp`	Enables the STC application for the dial peer.
Step 5	**port** *slot-number*/*port* **Example:** `Router(config-dial-peer)# port 2/2`	Assigns a voice port to the dial peer.
Step 6	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.
Step 7	**sccp plar** **Example:** `Router(config)# sccp plar`	Enters SCCP PLAR configuration mode.

	Command or Action	Purpose
Step 8	**voiceport** *port-number* **dial** *dial-string* [**digit** *dtmf-digits* [**wait-connect** *wait-msecs*] [**interval** *inter-digit-msecs*]] **Example:** `Router(config-sccp-plar)# voiceport 2/1 dial 3660 digit 1234 wait-connect 100 interval 100`	Enables PLAR on analog FXS ports that use SCCP for call control. • *port-number*—Analog FXS voice port number. Range is 2/0 to 2/23. • **dial** *dial-string*—Up to 16 characters that can be dialed on a telephone keypad (0 to 9, A to D, *, #). The voice gateway sends this string to the call-control system when the analog phone goes off hook. • **digit** *dtmf-digits*—Up to 16 characters (0 to 9, A to D, *, #, and a comma (,) for a one-second pause). The voice gateway sends this string to the call-control system after the *wait-msecs* expires. • **wait-connect** *wait-msecs*—Number of milliseconds that the voice gateway waits after voice cut-through before out-pulsing the DTMF digits. Range: 0 to 30000, in multiples of 50. Default: 50. If 0, DTMF digits are sent automatically by the voice gateway after the call is connected. • **interval** *inter-digit-msecs*—Number of milliseconds between the DTMF digits. Range: 50 to 500, in multiples of 50. Default: 50.
Step 9	**end** **Example:** `Router(config-sccp-plar)# end`	Exits SCCP PLAR configuration mode and returns to privileged EXEC mode.

Examples

The following example shows PLAR enabled on voice port 2/0 and 2/1.

```
Router# show running-config
.
.
.
sccp plar
  voiceport 2/0 dial 3660 digit 1234 wait-connect 100 interval 100
  voiceport 2/1 dial 3660 digit 6789 interval 100
!
!
!
dial-peer voice 500 pots
 service stcapp
 port 2/0
!
dial-peer voice 501 pots
 service stcapp
 port 2/1
!
```

Configuring SCCP Gateway Dial Tone Generation After Remote Onhook

This task configures dial tone generation after remote onhook. This feature enables the SCCP gateway to generate dial tone to the remaining party in basic call mode once the remote party disconnects. Perform this task to allow PBX interoperability by enabling configurable automatic dial tone capability after remote call disconnect.

Prerequisites

- Cisco IOS Release 12.(6)XE or a later release.

Restrictions

- The SCCP Gateway Dial Tone Generation After Remote Onhook feature is supported only on SCCP loop-start FXS ports that are registered with Cisco Unified Communications Manager and Cisco Unified CME.
- You cannot configure the SCCP Gateway Dial Tone Generation After Remote Onhook feature using Cisco Unified Communications Manager automatic download capability.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **pots**

4. **tone dialtone remote-onhook**
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **pots** **Example:** `Router(config)# dial-peer voice 101 pots`	Defines a particular dial peer and enters dial peer voice configuration mode. • *tag*—Number that identifies the dial peer range: 1 to 2147483647.
Step 4	**tone dialtone remote-onhook** **Example:** `Router(config-dial-peer)# tone dialtone remote-onhook`	Enables dial tone generation after remote onhook.
Step 5	**end** **Example:** `Router(config-dial-peer)# end`	Exits dial-peer configuration mode and returns to privileged EXEC mode.

Examples

The following examples show the dial tone generation after the remote onhook feature is enabled. Because the dial tone generation after remote onhook feature is enabled by default, it does not display in the **show running-config** output.

```
Router# show running-config

service stcapp
 dial-peer voice 3001 pots
 port 1/1/1

Router# show dial-peer voice 3001

VoiceEncapPeer3001
 peer type = voice, system default peer = FALSE, information type = voice,
!
!
!
 in bound application associated: 'stcapp'
 dial tone generation after remote-onhook = enabled

Router# show stcapp device voice-port 1/1/1
```

```
Port Identifier:  1/1/1
!
Dialtone after remote-onhook feature: activated
```

The following examples show the dial tone generation after the remote onhook feature disabled.

```
Router# show running-config

no tone dialtone remote-onhook
dial-peer voice 3002 pots
 service stcapp
 port 1/1/0

Router# show dial-peer voice 3002

VoiceEncapPeer3002
!
dial tone generation after remote-onhook = disabled

Router# show stcapp device voice-port 1/1/0

Port Identifier:  1/1/0
!
Dialtone after remote-onhook feature: not activated
```

Troubleshooting Tips

The following commands can help troubleshoot the dial tone after remote onhook feature:

- **debug voip application stcapp all**
- **debug voip application stcapp port** *port-number*

Configuring SCCP Gateway Ground Start FXS Ports

To configure ground-start FXS ports, perform the following steps.

Prerequisites

- Cisco IOS Release 12.4(6)XE or a later release.

Restrictions

- Supports for basic calls only.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** [*slot-number*/*subunit-number*/*port*]
4. **signal ground-start**

5. **shutdown**
6. **no shutdown**
7. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** [*slot-number*/*subunit-number*/*port*] **Example:** `Router(config)# voice-port 1/1/1`	Enters voice-port configuration mode.
Step 4	**signal ground-start** **Example:** `Router(config-voiceport)# signal ground-start`	Configures ground-start signaling as the signaling type for the voice port.
Step 5	**shutdown** **Example:** `Router(config-voiceport)# shutdown`	Takes the specified voice port offline and triggers deregistration of the device with Cisco Unified Communications Manager.
Step 6	**no shutdown** **Example:** `Router(config-voiceport)# no shutdown`	Puts the specified voice port back online and triggers reregistration of the device with Cisco Unified Communications Manager.
Step 7	**end** **Example:** `Router(config-voiceport)# end`	Exits voice-port configuration mode and returns to privileged EXEC mode.

Examples

The following example shows the ground start FXS port feature enabled and verifies the port type:

```
Router# show voice port 1/1/1

Foreign Exchange Station 1/1/1 Slot is 1, Sub-unit is 1, Port is 1
 Type of VoicePort is FXS  VIC2-2FXS
 Operation State is DORMANT
 Administrative State is UP
!
Voice card specific Info Follows:
 Signal Type is groundStart
```

Troubleshooting Tips Ground Start FXS Ports

The following command can help troubleshoot ground start FXS ports:

- **debug vpm signal**—Use this command to collect debugging information for signaling events.

Configuring Supervisory Disconnect

To configure the Supervisory Disconnect feature on an analog FXS voice port, perform the following steps.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice-port** *port-number*
4. **supervisory disconnect lcfo**
5. **timeouts power-denial** *ms*
6. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice-port** *port-number* **Example:** `Router(config)# voiceport 2/1`	Enters voice port configuration mode. • *port-number*—Format and values for analog FXS voice port number is platform-dependent. Type ? for values.
Step 4	**supervisory disconnect lcfo** **Example:** `Router(config-voiceport)# supervisory disconnect lcfo`	Sends a loop current feed open (LCFO) signal to indicate a disconnect on an FXS loop-start port. **Note** This command is enabled by default.
Step 5	**timeouts power-denial** *ms* **Example:** `Router(config-voiceport)# timeouts power-denial 500`	Sets the duration of the power denial timeout on the specified FXS voice port. • *ms*—Number of milliseconds that the power-denial is invoked. Range: 0 to 2500. Default: 750.
Step 6	**end** **Example:** `Router(config-voiceport)# end`	Exits voice-port configuration mode and returns to privileged EXEC mode.

Examples

The following example shows that the duration of the power denial is 500 ms on port 2/0 (which has supervisory disconnect enabled by default) and supervisory disconnect on port 2/1 is disabled.

```
Router# show running-config
.
.
.
voice-port 2/0
 timeouts power-denial 500
!
voice-port 2/1
 no supervisory disconnect
.
.
.
```

Verifying and Troubleshooting the Configuration

Use the following commands on the voice gateway to verify the configuration and status of the STC application and SCCP:

- **show call application voice summary**—Displays whether the STC application is running.
- **show call application voice stcapp**—Displays detailed application state and statistics.
- **show call active voice**—Displays the number of calls that are currently active. Call legs associated with this feature are included in the "Call agent controlled call-legs" listing.
- **show sccp** [**all** | **connections** | **statistics**]—Displays SCCP information such as administrative and operational status.
- **show stcapp device summary**—Displays a summary of endpoints associated with the STC application, and their states, types, and directory numbers.
- **show stcapp device** [**name** *device-name* | **voice-port** *port*]—Displays information for a single endpoint associated with the STC application. If an active call is in progress, the output will display additional call-related information.
- **show stcapp statistics** [**all** | **voice-port** *port*]—Displays call statistics for endpoints associated with the STC application.
- **show running-config**—Displays running configuration nondefault values.

Use the following commands on the voice gateway to troubleshoot the STC application and SCCP:

- **debug** [**voip** | **voice**] **application stcapp all**—Displays detailed debugging for all ports.
- **debug** [**voip** | **voice**] **application stcapp error**—Displays error debugging for all ports.
- **debug** [**voip** | **voice**] **application stcapp events**—Displays call flow event debugging for all ports.
- **debug** [**voip** | **voice**] **application stcapp functions**—Displays function debugging for all ports.
- **debug** [**voip** | **voice**] **application stcapp port** *port*—Displays detailed debugging only for the specified port.
- **debug sccp all**—Displays detailed debugging for all SCCP debug trace information.
- **debug sccp config**—Displays SCCP auto-configuration/download debugging.
- **debug sccp errors**—Displays SCCP error debugging.
- **debug sccp events**—Displays SCCP events debugging.
- **debug sccp packets**—Displays SCCP packets debugging.
- **debug sccp parser**—Displays SCCP parser and builder debugging.

Use the following commands on the voice gateway to capture and view a log of STCAPP events:

- **debug voip application stcapp buffer-history**—Enables event logging for STCAPP ports.
- **show stcapp buffer-history**—Displays call flow and device events saved to the event log.

Use the following command on the voice gateway to filter output for debug commands based on the individual voice port:

- **debug condition voice-port** *port*

Configuration Examples for Configuring FXS Ports for Basic Calls

This section contains the following examples:

- Example: Cisco IOS Gateway SCCP Analog Ports Configuration, page 39
- Example: PLAR with DTMF Out-Pulse Digits, page 41

Example: Cisco IOS Gateway SCCP Analog Ports Configuration

The following example shows a configuration for a Cisco VG224 Analog Phone Gateway in Cisco IOS Release 12.4(2)T:

```
Router# show running-config

Building configuration...

Current configuration : 3442 bytes
!
version 12.4
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
boot-start-marker
boot-end-marker
!
!
no aaa new-model
!
resource manager
!
ip subnet-zero
no ip dhcp use vrf connected
!
!
no ftp-server write-enable
!
stcapp ccm-group 1
stcapp
!
stcapp feature access-code
 prefix *#
 call forward all 2
 call forward cancel 9
!
stcapp feature speed-dial
 prefix ##
 redial 9
 voicemail 8
 speed dial from 3 to 7
!
!
template address
!
voice-card 0
```

```
!
!
!
!
!
interface FastEthernet0/0
 ip address 10.4.138.5 255.255.0.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip default-gateway 10.4.0.1
ip classless
ip route 223.255.254.0 255.255.255.0 10.4.0.1
!
ip http server
!
!
!
control-plane
!
!
voice-port 2/0
 caller-id enable
!
voice-port 2/1
 caller-id enable
!
voice-port 2/2
 caller-id enable
!
voice-port 2/3
 caller-id enable
!
voice-port 2/4
!
.
.
.
!
voice-port 2/23
!
!
!
sccp local FastEthernet0/0
sccp ccm 10.4.131.200 identifier 7815
sccp ccm 10.4.138.77 identifier 7825
sccp
!
sccp ccm group 1
 associate ccm 7815 priority 1
 associate ccm 7825 priority 2
!
!
dial-peer voice 500 pots
 service stcapp
 port 2/0
!
dial-peer voice 501 pots
```

```
 service stcapp
 port 2/1
!
dial-peer voice 502 pots
 service stcapp
 port 2/2
!
dial-peer voice 503 pots
 service stcapp
 port 2/3
!
dial-peer voice 504 pots
 service stcapp
 port 2/4
!
.
.
.
!
dial-peer voice 523 pots
 service stcapp
 port 2/23
!
!
line con 0
 exec-timeout 0 0
 transport preferred all
 transport output all
line aux 0
 transport preferred all
 transport output all
line vty 0 4
 login
 transport preferred all
 transport input all
 transport output all
!
end
```

Example: PLAR with DTMF Out-Pulse Digits

The following example shows PLAR with DTMF Out-Pulse Digits configured on a Cisco VG224 voice gateway:

```
Router# show running-config

Building configuration...

Current configuration : 3442 bytes
.
.
.
!
stcapp ccm-group 1
stcapp
!
stcapp feature access-code
 prefix *#
 call forward all 2
 call forward cancel 9
!
stcapp feature speed-dial
```

```
 digit 2
 voicemail 55
 speed dial from 11 to 17
!
!
template address
!
voice-card 0
!
!
!
!
!
!
!
interface FastEthernet0/0
 ip address 10.4.138.5 255.255.0.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
ip default-gateway 10.4.0.1
ip classless
ip route 223.255.254.0 255.255.255.0 10.4.0.1
!
ip http server
!
!
!
control-plane
!
!
voice-port 2/0
 caller-id enable
!
voice-port 2/1
 caller-id enable
!
voice-port 2/2
 caller-id enable
!
voice-port 2/3
 caller-id enable
!
voice-port 2/4
!
.
.
.
!
voice-port 2/23
!
!
!
sccp local FastEthernet0/0
sccp ccm 10.4.131.200 identifier 7815
sccp ccm 10.4.138.77 identifier 7825
sccp ccm 172.16.18.4 identifier 37454
sccp
```

```
!
sccp ccm group 1
 associate ccm 7815 priority 1
 associate ccm 7825 priority 2
 associate ccm 37454 priority 3
 registration timeout 3
 keepalive retries 1
 keepalive timeout 3
 switchback method graceful
!
sccp plar
  voiceport 2/0 dial 3660 digit 1234 wait-connect 500 interval 200
  voiceport 2/1 dial 3264 digit 678,,,9*0,,#123 interval 100
  voiceport 2/3 dial 3478 digit 34567 wait-connect 500
!
!
dial-peer voice 500 pots
 service stcapp
 port 2/0
!
dial-peer voice 501 pots
 service stcapp
 port 2/1
!
dial-peer voice 502 pots
 service stcapp
 port 2/2
!
dial-peer voice 503 pots
 service stcapp
 port 2/3
!
dial-peer voice 504 pots
 service stcapp
 port 2/4
!
.
.
.
!
dial-peer voice 523 pots
 service stcapp
 port 2/23
!
!
line con 0
 exec-timeout 0 0
 transport preferred all
 transport output all
line aux 0
 transport preferred all
 transport output all
line vty 0 4
 login
 transport preferred all
 transport input all
 transport output all
!
end
```

Additional References

The following sections provide references related to SCCP analog phone support for FXS ports on the Cisco voice gateway.

Related Documents

Related Topic	Document Title
Cisco Unified Communications Manager	*Cisco Unified Communications Manager documentation*
Cisco Unified Communications Manager Express	*Cisco Unified Communications Manager Express documentation*
Cisco IOS debugging	*Cisco IOS Debug Command Reference*
Cisco IOS voice commands	*Cisco IOS Voice Command Reference*
Cisco IOS voice configuration	Cisco IOS Voice Configuration Library
Cisco voice gateway	• *Cisco VG200 Series documentation* • *Cisco 1800 Series Integrated Services Routers documentation* • *Cisco 2800 Integrated Services Routers documentation* • *Cisco 3800 Series Integrated services Routers documentation* • *Cisco Unified 500 Series documentation*
Conferencing and transcoding resources	• "Configuring Enhanced Conferencing and Transcoding for Voice Gateway Routers" chapter in the *Cisco Unified CallManager and Cisco IOS Interoperability Guide*. • *Cisco CallManager and IOS Gateway DSP Farm Configuration Example*

Technical Assistance

Description	Link
The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies. To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds. Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.	http://www.cisco.com/techsupport

Feature Information for Configuring FXS Ports for Basic Calls

Table 1 lists the features in this module and provides links to specific configuration information. Only features that were introduced or modified in Cisco IOS Release 12.4(6)XE or a later release appear in the table.

For information on a feature in this technology that is not documented here, see the "Supplementary Services Features Roadmap" section on page 1.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note Table 1 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 1 Feature Information for Configuring FXS Ports for Basic Calls

Feature Name	Releases	Feature Information
Hookflash Duration	12.4(6)XE 12.4(11)T	Enables modification of the upper limit of the hookflash duration range for an analog FXS port. The following sections provide information about this feature: • Hookflash Duration, page 18. • Modifying Hookflash, page 27. No new commands were introduced by this feature.

Table 1 Feature Information for Configuring FXS Ports for Basic Calls (continued)

Feature Name	Releases	Feature Information
SCCP Dial Tone Generation After Remote On-Hook	12.4(6)XE 12.4(11)T	Enables the SCCP gateway to generate dial tone to the remaining party in basic call mode once the remote party disconnects. The following sections provide information about this feature: • Dial Tone Generation after Remote Onhook, page 19. • Configuring SCCP Gateway Dial Tone Generation After Remote Onhook, page 32. The following commands were introduced or modified by this feature: **tone dialtone remote-onhook**
SCCP Gateway Ground Start FXS Ports		Supports ground-start FXS ports for disconnect supervision and Cisco Communications Manager registration of PBX and key system connections. The following sections provide information about this feature: • Ground Start FXS Ports as SCCP Analog Endpoints, page 20 • Configuring SCCP Gateway Ground Start FXS Ports, page 34 No new commands were introduced by this feature.
SCCP PLAR with DTMF Out-Pulse Digits for FXS Analog Phones	12.4(6)T	Adds private line automatic ring-down (PLAR) support for SCCP analog ports on a Cisco VG224 Analog phone gateway. The following sections provide information about this feature: • PLAR with DTMF Out-Pulse Digits, page 19 • Configuring PLAR with DTMF Out-Pulse Digits, page 28 The following commands were introduced or modified by this feature: **sccp plar**, **voiceport**.
	12.4(9)	Adds supports for FXS ports on Cisco ISRs.

Table 1 Feature Information for Configuring FXS Ports for Basic Calls (continued)

Feature Name	Releases	Feature Information
Supervisory Disconnect	12.4(6)T	Provides a disconnect indication to the remote party in a two-party call after one side disconnects and enables external applications connected to the Cisco voice gateway to promptly clear a call after receiving the disconnect indication. The following sections provide information about this feature: • Supervisory Disconnect, page 20 • Configuring Supervisory Disconnect, page 36. The following commands were introduced or modified by this feature: **supervisory disconnect**, **timeouts power-denial**.
	12.4(9)T	Adds supports for FXS ports on Cisco ISRs.