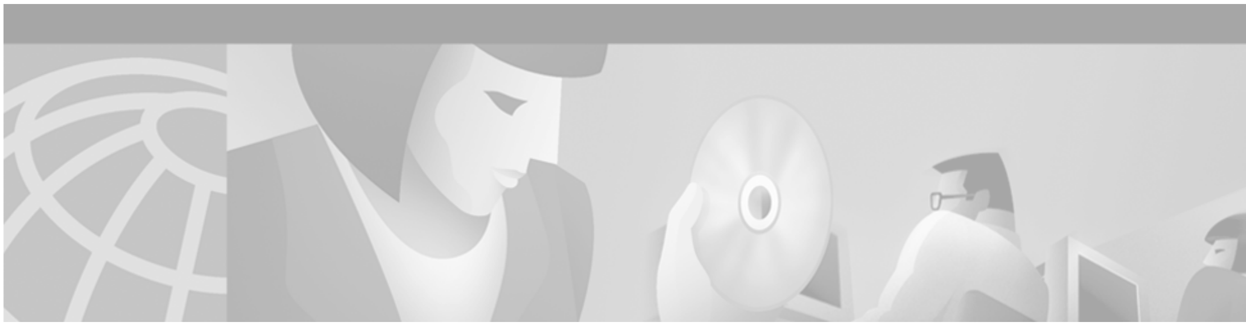

FXO Answer and Disconnect Supervision

Document Update Alert

This document was originally produced for Cisco IOS Release 12.1(3)T and released as the FXO Supervisory Disconnect Tone feature. This feature has been updated in subsequent releases, and more recent documentation is available.

If you are using Cisco IOS Release 12.2(2)T or higher, refer to the following sections in the *Voice Port Configuration Guide* of the Cisco IOS Voice Configuration Library, Release 12.3:

- Configuring Disconnect Supervision
- FXO Supervisory Disconnect Tone Commands

If you are using Cisco IOS Release 12.1(3)T or higher, refer to the following sections in the Configuring Voice Ports chapter of the *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2:

- Disconnect Supervision Commands
- FXO Supervisory Disconnect Tone Commands

This feature module describes the FXO Supervisory Disconnect Tone feature for analog FXO ports with loop start signaling, introduced in Cisco IOS Release 12.1(3)T on the Cisco 2600 and 3600 series routers and Cisco MC3810 series concentrators.

This document includes the following sections:

Feature Overview

If the FXO Supervisory Disconnect Tone feature is configured and a detectable tone from the PSTN or PBX is detected by the digital signal processor (DSP), the analog FXO port goes on-hook. This feature prevents an analog FXO port from remaining in an off-hook state after an incoming call is ended.

You can configure a voice port to detect either of the following tone types:

- Disconnect tones from the PBX or PSTN

  You can configure the FXO Supervisory Disconnect Tone feature to function in either of the following ways:

  - Continuously throughout the call duration
  - Before a call is answered

  As part of the tone detection process by the DSP, a DSP event is reported to the host software.

- Any tone received from the PBX or PSTN

  Detection of any tone is effective only during call set-up (before a call is answered), and echo cancellation must be enabled to prevent disconnection due to detection of the router's own ringback tone.

Note The cadence configuration commands are visible in Cisco IOS Release 12.1(3)T; however, they are not implemented in the Cisco IOS Release 12.1(3)T software. Support for cadence detection is available in the FXO Answer and Disconnect Supervision feature in Cisco IOS Release 12.2(2)T. This allows detection of tones in custom cadence patterns with up to four on/off time cycles.

Benefits

The FXO Supervisory Disconnect Tone feature allows interoperability with PSTN and PBX systems whether or not they transmit supervisory tones.

Restrictions

The FXO Supervisory Disconnect Tone feature is applicable only to analog FXO ports with loop-start signaling. Cadence detection is not supported.

The Cisco MC3810 series concentrators must be equipped with high-performance compression modules (HCMs) to support tone detection. Standard voice compression modules (VCMs) do not support the FXO Supervisory Disconnect Tone feature.

Related Documents

- *Cisco IOS Multiservice Applications Configuration Guide*, Cisco IOS Release 12.1
- *Cisco IOS Multiservice Applications Command Reference*, Cisco IOS Release 12.1
- *Software Configuration Guide For Cisco 3600 Series and Cisco 2600 Series Routers*
- *Cisco MC3810 Multiservice Concentrator Configuration Guide*

Supported Platforms

- Cisco 2600 series
- Cisco 3600 series
- Cisco MC3810 series

Supported Standards, MIBs, and RFCs

None

Prerequisites

The FXO Supervisory Disconnect Tone feature described in this document requires Cisco IOS Release 12.1(3)T or later.

Configuration Tasks

You can configure a voice port to disconnect when it detects a specific tone or tone pattern from a PBX or PSTN, or you can configure it to disconnect if it detects any tone before a call is answered.

For supervisory disconnect based on detection of a specific tone from a PBX or PSTN, complete the following tasks:

- Configuring Voice Ports to Detect Supervisory Disconnect Tones, page 3
- Verifying Configuration of Disconnect Tone Detection Parameters, page 6

For supervisory disconnect based on detection of any tone from a PBX or PSTN before a call is answered, complete the following tasks:

- Configuring an FXO Voice Port to Disconnect with any Detected Tone, page 6
- Verifying Configuration of Disconnect Tone Detection Parameters, page 6

Configuring Voice Ports to Detect Supervisory Disconnect Tones

To enable detection of supervisory disconnect tones, first create a voice class that defines the tone detection parameters, and then apply the voice class to the applicable analog FXO voice ports.

Configuring a Voice Class for FXO Supervisory Disconnect Tone

To configure a voice port to detect incoming tones, you need to know the parameters of the tones expected from the PBX or PSTN. This procedure configures the voice port to go on hook when it detects the specified tones. The parameters of the tones need to be precisely specified to prevent unwanted disconnects due to detection of non-supervisory tones or noise.

A supervisory disconnect tone is normally a dual tone consisting of two frequencies; however, tones of only one frequency can also be detected.

Caution If a voice port is configured to detect non-dual tones, unwanted disconnects can result from detection of random tone frequencies—the phenomenon of "talkoff."

To create a voice class that defines the specific tone or tones to be detected, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`voice class dualtone`** *`tag`*	Create a voice class for defining one tone detection pattern. The range for the *tag* number is 1 to 10000. The *tag* number must be unique on the router.
Step 2	`Router(config-voice-class)#` **`freq-pair`** *`tone-id frequency-1 frequency-2`*	Specify the two frequencies in Hz for a tone to be detected (or one frequency if a non-dual tone is to be detected). If the tone to be detected contains only one frequency, enter 0 for *`frequency-2`*. Repeat this command for each additional tone to be specified. The *`tone-id`* range is 1 to 16. There is no default. The range for *`frequency-1`* and *`frequency-2`* is 300 to 3600, or you can enter 0 for *`frequency-2`*. There is no default.
Step 3	`Router(config-voice-class)#` **`exit`**	Exit from the voice-class configuration mode.

Assigning an FXO Supervisory Disconnect Tone Voice Class to an FXO Voice Port

To assign an FXO supervisory disconnect tone voice class to an analog FXO voice port, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	For Cisco 2600 and 3600 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco MC3810 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/port`*	Identify the voice port you want to configure and enter voice-port configuration mode.
Step 2	`Router(config-voiceport)#` **`supervisory disconnect dualtone {mid-call | pre-connect} voice-class`** *`tag`*	Assign an FXO supervisory disconnect tone voice class to the voice port. The voice class is one that you created in the "Configuring a Voice Class for FXO Supervisory Disconnect Tone" section. Specify **mid-call** for tone detection during the entire call. Specify **pre-connect** for tone detection only during call set-up.
Step 3	`Router(config-voiceport)#` **`exit`**	Exit from voice-port configuration mode.

Configuring an FXO Voice Port to Disconnect with any Detected Tone

To configure an analog FXO voice port to go on-hook upon receipt of any tone received from a PBX or PSTN before the call is answered, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	For Cisco 2600 and 3600 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco MC3810 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/port`*	Identify the voice port you want to configure and enter voice-port configuration mode.
Step 2	`Router(config-voiceport)#` **`supervisory disconnect anytone`**	Configure the voice port to disconnect on receipt of any tone.
Step 3	`Router(config-voiceport)#` **`exit`**	Exit from voice-port configuration mode.

Verifying Configuration of Disconnect Tone Detection Parameters

Use either or both of the following methods to verify that the tone detection parameters have been properly configured on a voice port:

Use the "show running-config" Command

Enter the **show running-config** command to review the tone detection parameters you have configured.

The following example shows part of the output from the **show running-config** command on a Cisco MC3810, in which voice class 70 defines the tone parameters:

:

```
router# show running-config
Building configuration...
.
.
.
voice class dualtone 70
 freq-pair 1 350 440
 freq-pair 2 500 800
.
.
.
voice-port 1/6
 supervisory disconnect dualtone mid-call voice-class 70
.
.
.
```

Additional display lines are not shown.

Send a Supervisory Disconnect Tone from a PBX, the PSTN, or a SAGE Call Generator

Step 1 Invoke a call from a PBX, the PSTN, or a SAGE call generator to the router, and wait a few seconds to be sure that the call is connected.

Step 2 Use the PBX, PSTN, or SAGE to send a supervisory disconnect tone.

Note The PBX or PSTN must be configured to send a supervisory disconnect tone when the call is terminated. The tone must match the parameters you have configured in the router for supervisory disconnect.

If you use a SAGE, the tone generated by the SAGE must match the parameters you have configured in the router for supervisory disconnect.

Step 3 On the router enter the **show voice port summary** command, and verify that the FXO port changes back to the on-hook state.

The following is a sample display from the **show voice port summary** command on a Cisco MC3810, in which analog FXO voice port 1/2 is on hook:

```
router# show voice port summary
                                IN       OUT
PORT   CH SIG-TYPE   ADMIN OPER STATUS   STATUS   EC
====== == ========== ===== ==== ======== ======== ==
1/2    -- fxo-ls     up    dorm idle     on-hook  y
```

Configuration Examples

Applying a Voice Class to a Voice Port

This example applies voice class 70 to FXO voice port 1/2 on a Cisco MC3810, and specifies tone detection for the entire duration of the call.

```
Router(config)# voice port 1/1
Router(config-voiceport)# no echo-cancel enable
Router(config-voiceport)# connection plar 12
Router(config-voiceport)# supervisory disconnect dualtone mid-call voice-class 70
Router(config-voiceport)# exit
```

This example applies voice class 80 to FXO voice port 0/1/0 on a Cisco 3600 series router, and specifies tone detection for the call set-up time only.

```
Router(config)# voice port 0/1/0
Router(config-voiceport)# no echo-cancel enable
Router(config-voiceport)# connection plar 12
Router(config-voiceport)# supervisory disconnect dualtone pre-connect voice-class 80
Router(config-voiceport)# exit
```

Configuring a Voice Port to Disconnect with Any Detected Tone

This example configures voice port 1/1 to go on-hook upon receipt of any tone from a PBX or PSTN if the caller goes on-hook before the call is answered:

```
Router(config)# voice port 1/1
Router(config-voiceport)# echo-cancel enable
Router(config-voiceport)# connection plar 12
Router(config-voiceport)# supervisory disconnect anytone
Router(config-voiceport)# exit
```

Command Reference

This section documents new commands. All other commands used on these platforms are documented in the Cisco IOS Release 12.1 command reference publications.

- **freq-pair**
- **supervisory disconnect anytone**
- **supervisory disconnect dualtone voice-class**
- **voice class dualtone**

freq-pair

To specify the frequency components of a tone to be detected, use the **freq-pair** voice-class command. To cancel detection of a tone, use the **no** form of this command.

freq-pair *tone-id frequency-1 frequency-2*

no freq-pair *tone-id*

Syntax Description

tone-id	A tag identifier for a tone to be detected. The range is 1 to 16. There is no default.
frequency-1	One frequency component of the tone to be detected, in Hz. The range is 300 to 3600. There is no default.
frequency-2	A second frequency component of the tone to be detected, in Hz. The range is 300 to 3600 or you can specify 0. There is no default.

Defaults

No tone is specified for detection.

Command Modes

Voice-class configuration.

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600, 3600, and MC3810 series.

Usage Guidelines

To detect a tone with two frequency components (a dualtone), configure frequencies for *frequency-1* and *frequency-2*.

To detect a tone with only one frequency component, configure a frequency for *frequency-1* and enter 0 for *frequency-2*.

You can configure a router to detect up to 16 tones.

Examples

The following example configures tone number 1 (tone-id 1) with frequency components of 480 and 2400 Hz:

```
Router(config)# voice class dualtone 100
Router(config-voice-class)# freq-pair 1 480 2400
Router(config-voice-class)# exit
```

The following example configures tone number 1 (tone-id 1) with frequency components of 480 Hz and 2400 Hz, and tone number 2 (tone-id 2) with frequency components of 560 Hz and 880 Hz:

```
Router(config)# voice class dualtone 50
Router(config-voice-class)# freq-pair 1 480 2400
Router(config-voice-class)# freq-pair 2 560 880
Router(config-voice-class)# exit
```

Related Commands

Command	Description
voice class dualtone	Creates a voice class for FXO tone detection parameters.

supervisory disconnect anytone

To configure an FXO voice port to go on-hook if the router detects any tone from a PBX or PSTN before the call is answered, use the **supervisory disconnect anytone** voice-port command. To restore the default, use the **no** form of this command.

supervisory disconnect anytone

no supervisory disconnect anytone

Syntax Description

This command has no arguments or keywords.

Defaults

The supervisory disconnect function is not enabled on voice ports.

Command Modes

Voice-port configuration.

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600, 3600, and MC3810 series.

Usage Guidelines

The **supervisory disconnect anytone** voice-port command can be used to provide the disconnect function in cases where the PBX or PSTN does not provide a supervisory tone.

This function is enabled only during call setup (before the call is answered); examples of tones that trigger a disconnect include busy tone, fast busy tone, and dial tone.

You must enable echo cancellation; otherwise, the router's own ringback tone can trigger a disconnect.

This command replaces the **no supervisory disconnect signal** command.

If you enter the **no supervisory disconnect signal** command, the supervisory disconnect anytone feature will be enabled, and `supervisory disconnect anytone` will be displayed when **show** commands are entered.

Examples

The following example configures voice ports 1/4 and 1/5 to go on hook if any tone from the PBX or PSTN is detected before the call is answered:

```
Router(config)# voice-port 1/4
Router(config-voice-class)# supervisory disconnect anytone
Router(config-voice-class)# exit
Router(config)# voice-port 1/5
Router(config-voice-class)# supervisory disconnect anytone
Router(config-voice-class)# exit
```

The following example disables the disconnect function on voice port 1/5:

```
Router(config)# voice-port 1/5
Router(config-voice-class)# no supervisory disconnect anytone
Router(config-voice-class)# exit
```

Related Commands

Command	Description
voice class dualtone	Creates a voice class for FXO tone detection parameters.

supervisory disconnect dualtone voice-class

To assign a previously-configured voice class for FXO supervisory disconnect tone to a voice port, use the **supervisory disconnect dualtone voice-class** voice port configuration command. To remove a voice class from a voice port, use the **no** form of this command.

supervisory disconnect dualtone {**mid-call** | **pre-connect**} **voice-class** *tag*

no supervisory disconnect dualtone voice-class *tag*

Syntax Description

mid-call	Configures tone detection to operate throughout the duration of the call.
pre-connect	Configures tone detection to operate during call set-up, and to stop when the called telephone goes off-hook.
tag	A unique identification number assigned to one voice class. The tag number maps to the tag number assigned using the **voice class dualtone** global configuration command. The range is 1 to 10000.

Defaults

No voice class is assigned to a voice port.

Command Modes

Voice-port configuration.

Command History

Release	Modification
12.1(3)T	This command was introduced on the Cisco 2600, 3600, and MC3810 series.
12.2(2)T	This command was replaced by the **supervisory dualtone-detect-params** and the **supervisory custom-cptone** commands.

Usage Guidelines

You can apply an FXO supervisory disconnect tone voice class to multiple voice ports.

You can assign only one FXO supervisory disconnect tone voice class to a voice port. If a second voice class is assigned to a voice port, the second voice class replaces the one previously assigned.

You cannot assign separate FXO supervisory disconnect tone commands directly to the voice port.

This feature is applicable to analog FXO voice ports with loop-start signaling.

Examples

The following example assigns voice class 70 to FXO voice port 1/5 of a Cisco MC3810 series concentrator, and specifies tone detection during the entire call duration:

```
Router(config)# voice-port 1/5
Router(config)# no echo-cancel enable
Router(config-voiceport)# supervisory disconnect dualtone mid-call voice-class 70
Router(config-voiceport)# exit
```

The following example assigns voice class 80 to FXO voice port 0/1/1 of a Cisco 3600 series router, and specifies tone detection only during call set-up:

```
Router(config)# voice-port 0/1/1
Router(config)# no echo-cancel enable
Router(config-voiceport)# supervisory disconnect dualtone pre-connect voice-class 80
Router(config-voiceport)# exit
```

Related Commands

Command	Description
voice class dualtone	Creates a voice class for FXO tone detection parameters.

voice class dualtone

To create a voice class for FXO supervisory disconnect tone detection parameters, use the **voice class dualtone** global configuration command. To delete the voice class, use the **no** form of this command.

voice class dualtone *tag*

no voice class dualtone *tag*

Syntax Description

tag	A unique identification number assigned to one voice class. The range is 1 to 10000.

Defaults

No voice class is configured for tone detection parameters.

Command Modes

Global configuration.

Command History

Release	Modification
12.1(3)T	This command was introduced.
12.2(2)T	This command was replaced by the **voice-class dualtone-detect-params** and the **voice-class custom-cptone** commands.

Usage Guidelines

Use this command first to create the voice class. Then use the **supervisory disconnect dualtone voice-class** command to assign the voice class to a voice port.

A voice class can define any number of tones to be detected. You need to define a matching tone for each supervisory disconnect tone expected from a PBX or from the PSTN.

Examples

The following example configures voice class dualtone 70, which defines one tone with two frequency components, and does not configure a cadence list:

```
Router(config)# voice class dualtone 70
Router(config-voice-class)# freq-pair 1 350 440
Router(config-voice-class)# freq-max-deviation 10
Router(config-voice-class)# freq-max-power 6
Router(config-voice-class)# freq-min-power 25
Router(config-voice-class)# freq-power-twist 15
Router(config-voice-class)# freq-max-delay 16
Router(config-voice-class)# exit
```

The following example configures voice class dualtone 100, which defines one tone with two frequency components, and configures a cadence list:

```
Router(config)# voice class dualtone 70
Router(config-voice-class)# freq-pair 1 350 440
Router(config-voice-class)# freq-pair 2 480 850
Router(config-voice-class)# freq-max-deviation 10
Router(config-voice-class)# freq-max-power 6
Router(config-voice-class)# freq-min-power 25
Router(config-voice-class)# freq-power-twist 15
Router(config-voice-class)# freq-max-delay 16
Router(config-voice-class)# exit
```

The following example configures voice class dualtone 90, which defines three tones, each with two frequency components, and configures two cadence lists:

```
Router(config)# voice class dualtone 90
Router(config-voice-class)# freq-pair 1 350 440
Router(config-voice-class)# freq-pair 2 480 850
Router(config-voice-class)# freq-pair 3 1000 1250
Router(config-voice-class)# freq-max-deviation 10
Router(config-voice-class)# freq-max-power 6
Router(config-voice-class)# freq-min-power 25
Router(config-voice-class)# freq-power-twist 15
Router(config-voice-class)# freq-max-delay 16
Router(config-voice-class)# exit
```

Related Commands

Command	Description
supervisory disconnect dualtone voice-class	Assigns a previously-configured voice class for FXO supervisory disconnect tone to a voice port.

Glossary

ABCD signaling—Four-bit telephony line signaling coding in which each letter of “ABCD” represents one of the four bits. This is often associated with CAS or Robbed-Bit signaling on a T1 or E1 telephony trunk.

AIS—Alarm Indication Signal.

AVBO—Advanced Voice Busy Out.

Cisco trunk (private line) call—A Cisco trunk (private line) call is established by the forced connection of a dynamic switched call. A Cisco trunk call is established during configuration of the trunk and stays up for the duration of the configuration. Optionally, it provides a pass-through connection path to pass signaling information between the two telephony interfaces at either end of the connection.

CLI—Command line interface.

codec—coder-decoder. An integrated circuit device that typically uses pulse code modulation to transform analog signals into a digital bit stream, and digital signals back into analog signals. In Voice over IP, Voice over Frame Relay, and Voice over ATM, a DSP software algorithm used to compress/decompress speech or audio signals.

DLCI—Data-link connection identifier.

Dial peer—An addressable call endpoint that contains configuration information including voice protocol, a CODEC type, and a telephone number associated with the call endpoint. There are five kinds of dial peers: POTS, VoIP, VoFR, VoATM, and VoHDLC.

DSP—Digital Signaling Processor.

DTMF—Dual tone multi frequency. Uses two simultaneous voice-band tones for dial such as touch tone.

DTMF relay—Enables the generation of FRF.11 Annex A frames for a VoFR dial peer. The DSP generates Annex A frames instead of passing a DTMF tone through the network as a voice sample.

Dynamic switched call—A telephone call dynamically established across a packet data network based on a dialed telephone number. In the case of VoFR, a Cisco proprietary session protocol similar to Q.931 is used to achieve call switching and negotiation between calling endpoints. The proprietary session protocol runs over FRF.11-compliant subchannels.

E&M—Stands for 2-wire or 4-wire interfaces with separate signaling paths (from “Ear and Mouth,” also “recEive and transMit”). E&M is a trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. The Cisco analog E&M interface is an RJ-48 connector that allows connections to PBX trunk lines (tie lines). E&M is also available on E1 and T1 digital interfaces.

E1—European equivalent of T1. 32-64kbps channels include 1-channel for framing and 1-channel for D-channel information. The clock rate is 2.048 Mhz.

FRF—Frame Relay Forum. An association of corporate members consisting of vendors, carriers, users, and consultants committed to implementing Frame Relay in accordance with national and international standards. See http://www.frforum.com.

FRF.11—Frame Relay Forum implementation agreement for Voice over Frame Relay (v1.0 May 1997). This specification defines multiplexed data, voice, fax, DTMF digit-relay, and CAS/Robbed-bit signaling frame formats, but does not include call setup, routing, or administration facilities. See http://www.frforum.com.

FRF.11 Annex C—See FRF.12.

FRF11-trunk—A point-to-point permanent voice connection (private line) conforming to the FRF.11 specification.

FRF.12—The FRF.12 Implementation Agreement (also known as FRF.11 Annex C) was developed to allow long data frames to be fragmented into small pieces and interleaved with real-time frames. In this way, real-time voice and non real-time data frames can be carried together on low speed links without causing excessive delay to the real-time traffic.

FXO—Foreign Exchange Office. An FXO interface connects to the Public Switched Telephone Network (PSTN) central office and is the interface offered on a standard telephone. The Cisco FXO interface is an RJ-11 connector that allows an analog connection to be directed to the PSTN central office or to a station interface on a PBX.

FXS—Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. The Cisco FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment, keysets, and PBXs.

ICPIF—Calculated Planning Impairment Factor.

LVBO—Local Voice Busy Out.

MEL CAS—Mercury Exchange Limited (MEL) Channel Associated Signaling (CAS). A voice signaling protocol used primarily in the United Kingdom.

OOS—Out of Service state of the call or trunk.

PBX—Private Branch Exchange. A privately owned central switching office.

Permanent calls—Permanent calls are private line calls used for fixed point-to-point calls, connections between PBXs (E&M to E&M), or for remote telephone extensions (FXO to FXS).

POTS—Plain old telephone service. Basic telephone service supplying standard single line telephones, telephone lines, and access to the PSTN.

POTS dial peer—Dial peer connected by a traditional telephony network. POTS peers point to a particular voice port on a voice network device.

PSTN—Public Switched Telephone Network. PSTN refers to the local telephone company.

RTR—Response Time Reporter.

Switched calls—Switched calls are normal telephone calls when a user picks up a phone, hears a dial tone, and enters the destination phone number to reach the other phone. Switched calls can also be private line auto-ringdown (PLAR) calls, or tie-line calls for fixed point-to-point connections.

T1—Digital WAN carrier facility. T1 transmits DS-1-formatted data at 1.544 Mbps through the telephone-switching network by using AMI or B8ZS coding.

Tandem switching—The dynamic switching of voice calls between VoFR, VoATM, or VoHDLC PVCs and subchannels; also called tandeming. Tandem switching is often encountered in multi-hop VoFR call connection paths.

Trunk—Service that allows quasi-transparent connections between two PBXs, a PBX and a local extension, or some other combination of telephony interfaces with signaling passed transparently through the packet data network.

VoFR—Voice over Frame Relay.

VoFR dial peer—Dial peer connected by a Frame Relay network. VoFR peers point to specific VoFR devices.

Voice over Frame Relay—Voice over Frame Relay enables a router to carry voice traffic, for example, telephone calls and faxes, over a Frame Relay network. When sending voice traffic over Frame Relay, the voice traffic is segmented and encapsulated for transit across the Frame Relay network by using FRF.12 encapsulation.

Voice over IP—Voice over IP enables a router to carry voice traffic, for example, telephone calls and faxes, over an IP network. In Voice over IP, the DSP segments the voice signal into frames, which are then coupled in groups of two and stored in voice packets that are transported by using IP in compliance with ITU-T specification H.323.

VoIP—Voice over IP through Ethernet.