

Cisco-Trunk (Private Line) Connections on the Cisco MC3810

Feature Overview

This feature allows you to configure Cisco-trunk (private line) connections for Voice over ATM, and Voice over HDLC on the Cisco MC3810.

Note For more information on how to configure Cisco-trunk (private line) connections for Voice over Frame Relay, see the online feature guide *Voice over Frame Relay Using FRF.11 and FRF.12* for Cisco IOS Release 12.0(4)T. This document describes how to configure Cisco-trunk connections for Voice over ATM and Voice over HDLC only.

Benefits

- Provides true permanent (private line) connections.
- Provides comprehensive busy-out support for trunk connections (for more information, see the online feature guide *Trunk Conditioning on the Cisco MC3810* for Cisco IOS Release 12.0(4)T).
- Provides transparent Channel Associated Signaling (CAS) protocol transport to allow the trunk to carry arbitrary ABCD signaling protocols.
- Provides conversion from North American signaling protocols to CEPT (Conférence Européenne des Postes et des Télécommunications) signaling protocols used for European voice networks.
- Provides remote analog to digital channel-bank operation for converting from Digital Voice Module (DVM) to Analog Voice Module (AVM) configurations.

Supported Platforms

This feature is supported only on the Cisco MC3810.

Supported MIBs and RFCs

None

List of Terms and Acronyms

ABCD signaling—4-bit telephony line signaling coding in which each letter of “ABCD” represents one of the 4 bits. This is often associated with CAS or Robbed-Bit signaling on a T1 or E1 telephony trunk.

CAS—Channel Associated Signaling.

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. It optionally provides a pass-through connection path to pass signaling information between the two telephony interfaces at either end of the connection.

CODEC—Coder-Decoder. (i) 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. (ii) In Voice over IP, Voice over Frame Relay, and Voice over ATM, a DSP software algorithm used to compress/decompress speech or audio signals.

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

DTMF—Dual tone multifrequency. Use of 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 recEive and transMit (or Ear and Mouth). E&M is a trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. Cisco’s 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.

FXO—Foreign Exchange Office. An FXO interface connects to the Public Switched Telephone Network’s (PSTN) central office and is the interface offered on a standard telephone. Cisco’s FXO interface is an RJ-11 connector that allows an analog connection to be directed at the PSTN’s 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. Cisco’s FXS interface is an RJ-11 connector that allows connections to basic telephone service equipment, keysets, and PBXs.

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

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

PBX—Private Branch Exchange. Privately owned central switching office.

Permanent calls—Permanent calls are private line calls that are 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 via 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.

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

Tandem switching—The dynamic switching of voice calls between VoFR, VoATM, or VoHDL 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.

VoATM—Voice over ATM.

VoATM dial peer—Dial peer connected via an ATM network. VoATM peers point to specific VoATM devices.

VoHDL—Voice over High-Level Data Link Control (HDL).

VoHDL dial peer—Dial peer connected via a Frame Relay network. VoHDL peers point to specific VoHDL devices.

Voice over ATM—Voice over ATM enables a router to carry voice traffic (for example, telephone calls and faxes) over an ATM network. When sending voice traffic over ATM, the voice traffic is encapsulated using a special AAL5 encapsulation for multiplexed voice.

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 using FRF.12 encapsulation.

Voice over HDL—Voice over HDL enables a router to carry live voice traffic (for example, telephone calls and faxes) back-to-back to a second router over a serial line.

Functional Description

A Cisco-trunk (private line) call is basically a normal dynamic switched call of indefinite duration that uses a fixed destination telephone number and includes optional transparent end-to-end signaling. The telephone number of the destination endpoint is permanently configured into the router so that it always selects a fixed destination. After the call is established, either at boot-up or when configured, the call stays up until one of the voice ports or network ports is shut down, or until a network disruption occurs.

The **connection trunk** voice-port command is used to establish a Cisco-trunk call; the dial peer is configured using the **session protocol cisco-switched** command, which invokes the Cisco proprietary session protocol.

Cisco-trunk (private line) trunk calls can be configured for Voice over Frame Relay in addition to Voice over ATM and Voice over HDL.

Configuration Tasks

This section describes how to configure Cisco-trunk (private line) calls for Voice over ATM. The following major tasks are covered and are divided into the following sections:

- Configuring Voice over ATM or Voice over HDLC Dial Peers for Cisco-Trunk (Private Line) Calls on page 4
- Configuring Cisco-Trunk Permanent Calls on a Cisco MC3810 on page 6

Configuring Voice over ATM or Voice over HDLC Dial Peers for Cisco-Trunk (Private Line) Calls

If you will be sending Cisco-trunk (private line) calls over the ATM or HDLC network, you must configure the VoATM or VoHDLC dial peers to specifically support Cisco-trunk (private line) calls. Cisco-trunk (private line) calls are permanent calls.

Note You can also create a voice class to configure trunk conditioning values for the idle and out-of-service (OOS) states, and then assign the voice class to the VoATM or VoHDLC dial peer. For more information, see the online feature guide *Trunk Conditioning on the Cisco MC3810* for Cisco IOS Release 12.0(4)T.

To configure a VoATM or VoHDLC dial peer to support Cisco-trunk permanent (private line) trunk calls, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	<code>router(config)# dial-peer voice tag {voatm vohdlc}</code>	Define a VoATM or VoHDLC dial peer and enter dial-peer configuration mode. The <i>tag</i> value identifies the dial peer and must be unique on the router. Do not duplicate a specific tag number. All subsequent commands that you enter in dial-peer voice mode before you exit will apply to this dial peer.
2	<code>router(config-dialpeer)# destination-pattern string</code>	Configure the dial peer's destination pattern.
3	Choose one of the following: <code>router(config-dialpeer)# session target interface pvc {name vpi/vci vci}</code> <code>router(config-dialpeer)# session target interface</code>	For Voice over ATM, configure the ATM session target for the dial peer. Enter ATM0 for the <i>interface</i> value. For Voice over HDLC, configure the HDLC session target for the dial peer.
4	<code>router(config-dialpeer)# session protocol cisco-switched</code>	Configure the session protocol to support Cisco-trunk calls. The cisco-switched option is the default setting, and entering this command is not required.

Configuring Voice over ATM or Voice over HDLC Dial Peers for Cisco-Trunk (Private Line) Calls

Step	Command	Purpose
5	<code>router(config-dialpeer)# codec type [bytes bytes]</code>	<p>Specify the voice coder rate of speech and payload size for the dial peer. The default dial peer CODEC is g729r8. Note that the Cisco MC3810 is limited to a maximum of 12 calls when using g729r8; to support up to 24 calls on the Cisco MC3810, use g729ar8.</p> <p>Specifying the payload size by entering the bytes value is optional. Each CODEC type defaults to a different payload size if you do not specify a value. To obtain a list of the default payload sizes, enter the codec command and the bytes option followed by a question mark (?).</p> <p>Note On the Cisco MC3810, you can also assign codec values to the voice port. When you configure the CODEC type for regular switched voice calls, you must set the CODEC type on the Cisco MC3810 voice port. When you configure the CODEC for permanent calls (cisco-trunk), you must configure the CODEC type on the dial peer. You cannot specify the payload size on the voice port.</p>
6	<code>router(config-dialpeer)# dtmf-relay</code>	<p>(Optional) If the codec type is a low bit-rate CODEC such as g729 or g723, specify support for DTMF relay to improve end-to-end transport of DTMF tones. DTMF tones do not always propagate reliably with low bit-rate CODECs.</p> <p>DTMF relay is disabled by default.</p>
7	<code>router(config-dialpeer)# signal-type {cas cept ext-signal transparent}</code>	<p>Define the flavor of the ABCD signaling packets that are generated by the voice port and sent to the data network. The signal type must be configured to the same setting at both ends of the permanent voice call.</p> <p>Enter cas to support CAS. Enter cept to support the European CEPT standard (related to MEL CAS).</p> <p>Enter ext-signal to indicate that ABCD signaling packets should not be sent, for configurations where the line signaling information is carried externally to the voice port.</p> <p>Enter transparent (for digital T1/E1 interfaces on the Cisco MC3810 only) to read the ABCD signaling bits directly from the T1/E1 interface without interpretation, and to pass them transparently to the data network (this is also known as transparent FRF.11 signaling).</p>
8	<code>router(config-dialpeer)# no vad</code>	<p>(Optional) Disable voice activity detection (VAD) on the dial peer. This command is enabled by default.</p>
9	<code>router(config-dialpeer)# sequence-numbers</code>	<p>(Optional) Enable the voice sequence number if required for your configuration. This command is disabled by default.</p>
10	<code>router(config-dialpeer)# preference value</code>	<p>(Optional) Configure a preference for the VoFR dial peer. The value is a number from 0 to 10 where the lower the number, the higher the preference in hunt groups.</p>
11	<code>router(config-dialpeer)# fax rate {2400 4800 7200 9600 14400 disable voice}</code>	<p>(Optional) Configure the transmission speed (in bps) at which a fax will be sent to the dial peer.</p> <p>The default is voice, which specifies the highest possible transmission speed allowed by the voice rate.</p>
12	To configure another VoATM dial peer, exit dial-peer configuration mode and repeat steps 1 through 11.	

Configuring Cisco-Trunk Permanent Calls on a Cisco MC3810

To configure Cisco-trunk permanent calls on a Cisco MC3810 for Voice over ATM and Voice over HDLC, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<code>router(config)# voice port slot/port</code>	Enter voice port configuration mode on the Cisco MC3810.
2	<code>router(config-voiceport)# connection trunk destination-string [answer-mode]</code>	Configure the trunk connection, specifying the telephone number in the <i>destination-string</i> . When configuring Cisco-trunk permanent calls, one side must be the call initiator (master) and the other side is normally the call answerer (slave). By default, the voice port operates in master mode. Enter the answer-mode keyword to specify that the voice port should operate in slave mode.
3	<code>router(config-voiceport)# shutdown</code>	Shut down the voice port.
4	<code>router(config-voiceport)# no shutdown</code>	Reactivate the voice port to enable the trunk connection to take effect.

Note Every time you enter the **connection trunk** or **no connection trunk** command, you must toggle the voice port (by entering **shutdown**, then **no shutdown**) for the changes to take effect.

Configuration Example

The following example configures a dial peer to support a Cisco-trunk (private line) call for Voice over ATM:

```
router(config)# dial peer voice voatm 20
router(config-dialpeer)# destination pattern 5552000
router(config-dialpeer)# session target atm0 pvc 20
router(config-dialpeer)# session protocol cisco-switched
router(config-dialpeer)# codec g728ar8
router(config-dialpeer)# signal-type ext-signal
router(config-dialpeer)# no vad

router(config)# voice-port 1/1
router(config-voiceport)# connection trunk 5552000 answer-mode
router(config-voiceport)# shutdown
router(config-voiceport)# no shutdown
```

Command Reference

For command reference information, see the online feature guide *Voice over Frame Relay Using FRF.11 and FRF.12* for Cisco IOS Release 12.0(4)T.