Voice over Frame Relay Using FRF.11 and FRF.12 Configuration Updates

Voice over Frame Relay functionality has been updated in this release, so that configuration on all supported platforms is nearly identical. In Cisco IOS Release 12.0(4)T, when support for Voice over Frame Relay Using FRF.11 and FRF.12 was introduced, configuration procedures were different depending on the router platform used.

Some commands introduced in earlier Cisco IOS releases have been removed or modified. This document describes the configuration procedures effective in this release.

In addition, this release provides support for digital voice calls for Voice over Frame Relay on the Cisco 2600 and 3600 series routers. In previous releases, the Cisco 2600 and 3600 series only supported analog voice calls for Voice over Frame Relay.

Benefits

Configuration Consistency Across all Voice over Frame Relay Routers

This feature provides consistency for configuration requirements across the hardware models that support Voice over Frame Relay: the Cisco 2600 and 3600 series routers, the Cisco 7200 series routers, and the Cisco MC3810 multiservice access concentrator. In previous releases, configuration procedures on the Cisco MC3810 were different from those on other routers.

Digital Voice Support on the Cisco 2600 and 3600 Series Routers

Beginning in this release, the Cisco 3600 series routers support digital voice calls for Voice over Frame Relay. This feature requires that a Digital T1 Packet Voice Trunk network module is installed on the router.

Restrictions

- The Cisco 2600 series and 3600 series routers cannot terminate calls initiated by a Cisco MC3810 using VoFR implementations before Cisco IOS Release 12.0(3)XG or 12.0(4)T.
- Cisco MC3810 concentrators running Cisco IOS versions before release 12.0(3)XG or 12.0(4)T cannot tandem VoFR calls from Cisco 2600 series, 3600 series, and 7200 series routers.
- It is currently not possible to translate from the VoIP transport protocol to other protocols, such as VoFR. As a result, a call coming in on a VoIP connection is not (tandem) switched to a VoFR connection.
- Hookflash for dial tone recall from the router is not supported. However, the router can pass-through hookflash on FXO-FXS permanent connections and E&M-E&M connections by using the **connection trunk** voice-port configuration command.

- Voice over ATM Switched Virtual Circuits (SVCs) are not supported in this release.

Caution Voice over ATM SVCs were first supported on the Cisco MC3810 in Cisco IOS Release 12.0(5)XK and 12.0(7)T. If upgrading a Cisco MC3810 from IOS release 12.0(5)XK or 12.0(7)T to this release to obtain Voice over Frame Relay improvements, you will lose support for your Voice over ATM SVCs.

Related Documents

- For complete information about Voice over Frame Relay configuration, see the Cisco IOS 12.0(4)T online document *Voice over Frame Relay Using FRF.11 and FRF.12*. Unless otherwise stated in this document, the procedures in the *Voice over Frame Relay Using FRF.11 and FRF.12* document are still valid.
- For information about voice port changes in Cisco IOS Release 12.0(7)XK, see the following Cisco IOS 12.0(7)XK online documents:
  - *Voice Port Enhancements in Cisco 2600 and 3600 Series Routers and MC3810 Series Concentrators*
  - *Voice Port Testing Enhancements in Cisco 2600 and 3600 Series Routers and MC3810 Series Concentrators*
  - *Voice Busyout Enhancements on the Cisco 2600 and 3600 Series Routers and MC3810 Series Concentrators*
- For information about software configuration requirements for the Digital T1 Packet Voice trunk network modules on the Cisco 2600 and Cisco 3600, see the Cisco IOS 12.0(7)T online document *Configuring Digital T1 Packet Voice Trunk Network Modules on Cisco 2600 and 3600 Series Routers.*
- For more information about voice technologies, see the Cisco IOS 12.0 *Voice, Video, and Home Applications Configuration Guide*, and the *Voice, Video, and Home Applications Command Reference*.

Supported Platforms

- Cisco 2600
- Cisco 3600
- Cisco MC3810
- Cisco 7200

Note In Cisco IOS releases before release 12.0(5)XE, the Cisco 7200 could only act as a tandem router in Voice over Frame Relay topologies, and could not terminate VoFR calls. Beginning in Cisco IOS Release 12.0(5)XE2 and 12.1(1)T, the Cisco 7200 can terminate VoFR calls.

Prerequisites

Before you can configure a Cisco router to use Voice over Frame Relay, you must do the following:

- Complete your company's dial plan.
- Establish a working Frame Relay network. For more information about configuring Frame Relay, refer to Cisco IOS Release 12.0 *Wide-Area Networking Configuration Guide*.
- Establish a working telephony network based on your company's dial plan:
  - Integrate your dial plan and telephony network into your existing Frame Relay network topology. Make routing and/or dialing transparent to the user—for example, avoid secondary dial tones from secondary switches where possible.
  - Contact your PBX vendor for instructions about how to reconfigure the appropriate PBX interfaces.

After you have analyzed your dial plan and decided how to integrate it into your existing Frame Relay network, you are ready to configure your network devices to support Voice over Frame Relay.

Supported MIBs and RFCs

None

Configuration Tasks

This section describes the following new and modified configuration procedures for Voice over Frame Relay in this release:

- Configuring Dial Peer Digit Manipulation on page 4
- Configuring Dial Peer Hunting on page 4
- Configuring Voice over Frame Relay Connections on page 5

For all remaining Voice over Frame Relay procedures, see the Cisco IOS 12.0(4)T release online document *Voice over Frame Relay Using FRF.11 and FRF.12*.

Configuring Dial Peer Digit Manipulation

In Cisco IOS Release 12.0(7)XK, the forward-digits command used for POTS dial-peer digit manipulation has been changed. To configure dial-peer digit manipulation to forward digits, perform the following steps beginning in global configuration mode.

Step	Command	Purpose
1	`Router(config)#` **`dial-peer voice`** *`tag`* **`pots`**	Enter dial-peer configuration mode for a POTS dial peer.
2	`Router(config-dial-peer)#` **`forward-digits`** `{`*`num-digit`* `|` **`all`** `|` **`extra`**`}` or `Router(config-dial-peer)#` **`default forward-digits`** or `Router(config-dial-peer)#` **`no forward-digits`**	If using the forward-digits feature, configure the digit-forwarding method. The range for the number of digits forwarded (*num-digit*) is 0 to 32. See the “Command Reference” section on page 32 for an explanation of the command options. In the default condition, dialed digits not matching the destination pattern are forwarded. **Note** The **no** state is not the **default** state.

Configuring Dial Peer Hunting

After you have configured dial peers, you can configure how the router performs dial peer hunting functions. To configure the dial peer hunting behavior on the router, perform the following steps beginning in configuration mode:

Step	Command	Purpose
1	`Router(config)#` **`dial-peer hunt`** *`hunt-order-number`*	Specify the hunt selection order for dial peers.
2	`Router(config)#` **`dial-peer terminator`** *`character`*	(Optional) Designate a special character to be used as a terminator for variable length dialed numbers.

Disabling Dial-Peer Hunting on a Specific Dial Peer

If using dial peer hunting, there may be situations when you want to disable dial-peer hunting on a specific dial peer. To disable dial-peer hunting on a dial peer, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	`Router(config)#` **`dial-peer voice`** *`tag`* `{`**`pots`** `|` **`vofr`**`}`	Enter dial-peer configuration mode for the specified dial peer.
2	`Router(config-dial-peer)#` **`huntstop`**	Disable dial-peer hunting on the dial peer. Once you enter this command, no further hunting is allowed if a call fails on the specified dial peer.

To reenable dial-peer hunting on a dial peer, enter the **no huntstop** command.

Configuring Voice over Frame Relay Connections

After you have configured the Frame Relay DLCI settings and you have configured your dial plan, you are ready to configure specific VoFR connections.

There are many different scenarios for VoFR connections. For information on the different connection types, see the next section, "Overview of Voice over Frame Relay Connection Types."

For procedures on how to configure the different connection types, see the following sections:

- Configuring Switched Calls (User Dialed or Auto-Ringdown) on page 7
- Configuring Cisco-Trunk Permanent (Private Line) Calls on page 9
- Configuring FRF.11 Trunk (Private Line) Calls on page 13

In addition, special consideration is required for configuring calls for tandem nodes. For more information, see "Configuring Connections for Tandem Nodes" section on page 14.

Note Use of Cisco trunks for permanent calls (private-line) is recommended over FRF.11 trunk calls unless FRF.11 compliant standards-based interworking is required with non-Cisco devices. The Cisco trunk protocol is a superset of the FRF.11 protocol and contains Cisco proprietary extensions designed to support switched call routing and other advanced features.

Overview of Voice over Frame Relay Connection Types

When you configure VoFR connections, there are many different connection types possible depending on the hardware platform, whether the call is to be a regular switched (user dialed or auto-ringdown) call, or whether the call is a permanent call (Cisco-trunk or FRF.11-trunk). You configure these specific connection types by using combinations of several commands.

Table 1 lists the different connection types for VoFR connections supported on the Cisco 2600 and 3600 series, and the combinations of commands to enter for each call type.

Note Calls to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK require different commands. Specific procedures for this configuration are described in separate sections.

Table 1 Voice over Frame Relay Connection Types that are Supported

Type of Call	Frame Relay DLCI interface Command to Enter[1]	Data Fragmentation Supported by vofr Command	Session Protocol Command to enter in Dial-Peer mode	Voice Port Connection Command to Enter
Switched call (user dialed or auto-ringdown) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** [*cid*]][2]	FRF.11 Annex C	**session protocol cisco-switched**[3]	For user dialed calls: none For auto-ringdown calls: **connection plar** *destination-string*
Switched call (user dialed or auto-ringdown) to a Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK	**vofr cisco**[4]	Cisco proprietary[5]	**session protocol cisco-switched**	For user dialed calls: none For auto-ringdown calls: **connection plar** *destination-string*
Cisco-trunk permanent call (private-line) to other routers supporting VoFR	**vofr data** *cid* **call-control** *cid*	FRF.11 Annex C	**session protocol cisco-switched**	**connection trunk** *destination-string* [**answer mode**]
Cisco-trunk permanent call (private-line) to a Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK	**vofr cisco**	Cisco proprietary	**session protocol cisco-switched**	**connection trunk** *destination-string* [**answer mode**]
FRF.11 trunk call (private-line) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** *cid*][6]	FRF.11 Annex C	**session protocol frf11-trunk**	**connection trunk** *destination-string* [**answer mode**]

1 The **voice-encap** option of the **frame-relay interface-dlci** command on the Cisco MC3810 is no longer supported as of Cisco IOS Release 12.0(7)XK.

2 The recommended use of this command is **vofr data 4 call-control 5**.

3 The **session protocol cisco-switched** option is the default setting. If you do not enter this command, the setting still applies.

4 This command consumes data CID 4 and call-control CID 5.

5 Cisco proprietary fragmentation is based on an early draft of FRF.12 and is compatible with Cisco MC3810 concentrators running software versions before Cisco IOS Release 12.0(3)XG or 12.0(4)T.

6 For FRF.111 trunk calls, the call-control option is not required. It is only required if you mix FRF.11 trunk calls with other types of voice calls on the same PVC.

Configuring Switched Calls (User Dialed or Auto-Ringdown)

This section describes how to configure switched calls (user dialed or auto-ringdown) on the different router platforms. This section is divided into the following procedures:

- Configuring Switched Calls to Other Voice over Frame Relay Routers on page 7
- Configuring Switched Calls to a Cisco MC3810 Running Cisco IOS Releases Before 12.0(7)XK on page 8

Configuring Switched Calls to Other Voice over Frame Relay Routers

You can configure switched calls on Cisco 2600, 3600, and 7200 series routers, and on Cisco MC3810 concentrators.

Note If configuring switched calls to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK , see "Configuring Switched Calls to a Cisco MC3810 Running Cisco IOS Releases Before 12.0(7)XK" section on page 8.

To configure switched calls on routers that support VoFR, use the following commands from interface configuration mode:

Step	Command	Purpose
1	`Router(config-if)#` **`frame-relay interface-dlci`** *`dlci`*	Configure the Frame Relay DLCI and enter DLCI configuration mode.
2	`Router(config-if)#` **`vofr`** `[`**`data`** *`cid`*`] [`**`call-control`** `[`*`cid`*`]]`	Configure the Frame Relay DLCI to support VoFR and set the data and call-control CIDs. The recommended setting for this command is **vofr data 4 call-control 5**. **Note** When the **vofr** command is used, all subchannels on the DLCI are configured for FRF.11 encapsulation. If you enter the **vofr** command without any keywords or arguments, the data subchannel is CID 4 and there is no call-control subchannel. If configuring user-dialed calls, this procedure is completed. If configuring auto-ringdown calls, proceed to the next step.
3	For Cisco 2600 and 3600 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `Router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `Router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
4	`Router(config-voiceport)#` **`connection plar`** *`string`*	(Optional) For auto-ringdown calls, configure the PLAR connection, specifying the telephone number in the *destination-string*.

This configuration uses standard FRF.11 Annex C fragmentation.

Configuring Switched Calls to a Cisco MC3810 Running Cisco IOS Releases Before 12.0(7)XK

You can configure switched calls to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK. However, the configuration is different from standard switched calls because earlier Cisco MC3810 versions used the Cisco proprietary version of FRF.12.

Note The Cisco 2600/3600 series routers cannot terminate or initiate calls with a Cisco MC3810 running software versions before Cisco IOS Releases 12.0(3)XG and 12.0(4)T.

To configure switched calls to a Cisco MC3810 running Cisco IOS releases before 12.0(7)XK , use the following commands beginning in interface configuration mode:

Step	Command	Purpose
1	`Router(config-if)#` **`frame-relay interface-dlci`** *`dlci`*	Configure the Frame Relay DLCI and enter DLCI configuration mode. **Note** The **voice-encap** option of the **frame-relay interface-dlci** command on the Cisco MC3810 is no longer supported beginning in this release.
2	`Router(config-if)#` **`vofr cisco`**	Configure the Frame Relay DLCI to support VoFR and the Cisco proprietary fragmentation implementation. When this command is entered, data CID 4 and call-control CID 5 are automatically assigned. If you are configuring user-dialed calls, this procedure is complete. If configuring auto-ringdown calls, proceed to the next step.
3	For Cisco 2600 and 3600 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `Router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `Router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.
4	`Router(config-voiceport)#` **`connection plar`** *`destination-string`*	(Optional) For auto-ringdown calls, configure the PLAR connection, specifying the telephone number in the *destination-string*.

This configuration uses Cisco proprietary data fragmentation.

Configuring Cisco-Trunk Permanent (Private Line) Calls

This section describes how to configure Cisco-trunk permanent (private line) calls on the different router platforms. This section is divided into the following procedures:

- "Configuring Voice over Frame Relay Dial Peers for Cisco-Trunk (Private Line) Calls" section on page 9
- Configuring Cisco Trunk Permanent Calls on page 11
- Configuring Cisco Trunk Permanent Calls to a Cisco MC3810 Running Cisco IOS Releases Before 12.0(7)XK on page 12

Configuring Voice over Frame Relay Dial Peers for Cisco-Trunk (Private Line) Calls

If you are sending Cisco-trunk (private line) calls over the Frame Relay network, you must configure the Voice over Frame Relay dial peers to specifically support Cisco-trunk (private line) calls. Cisco-trunk (private line) calls are permanent calls.

One key task when you configure Cisco-trunk (private line) connections is to configure the signal type for the dial peer. The **signal-type** dial-peer command supports the following options:

- **cas**—Use the **cas** option to support North American CAS/robbed-bit signaling. This is the default signaling type.
- **cept**—Use the **cept** option to provide a basic E1 ABCD protocol, primarily for CEPT E&M signaling. This option is primarily used for European voice networks. If this option is used with FXS or FXO voice ports, the signaling used is equivalent to MEL CAS.
- **ext-signal**—Use the **ext-signal** option in cases where some external signaling channel is being used (for example, common channel signaling), or where no signaling information is being sent at all over a permanent "dumb" voice pipe. Applications where no signaling is required include using a simple voice pipe to carry audio for a public address system.
- **transparent—**Use the **transparent** option when the ABCD signaling bits are copied through from the T1/E1 interface "transparently" without modification or interpretation (also known as transparent FRF.11 signaling). This allows the router to handle or transport unknown signaling protocols. On a router with analog voice ports, the **transparent** option does not apply. If the **transparent** option is sent over analog voice ports, the signaling is equivalent to the **cept** option.

Configure the signal type so that the signal type that is selected in the dial peers on the routers at both ends of the permanent voice call are the same. When you configure a permanent connection between a T1/E1 Cisco MC3810 and an analog voice port on a Cisco 2600 or Cisco 3600, set the signal type to **cas**, which is the default.

Cisco 2600 and Cisco 3600 analog voice ports do not support the **cept** or **transparent** signal types. The T1/E1 Cisco MC3810 can also be set to **transparent**, which simply passes the signaling through from the Cisco 2600/3600 without interpretation. However, when **transparent** is used, the Cisco MC3810 makes no assumptions regarding the on-hook and off-hook state of the call. By default, when configured by using **transparent**, the Cisco MC3810 operates the voice path in the permanently open state, so that voice packets are sent (and network bandwidth consumed) regardless of the state of the call.

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

Step	Command	Purpose
1	`Router(config)#` **`dial-peer voice`** *`number`* **`vofr`**	Define a VoFR dial peer and enter dial-peer configuration mode. All subsequent commands that you enter in dial-peer voice mode before you exit will apply to this dial peer. The *number* tag value identifies the dial peer and must be unique on the router. Do not duplicate a specific tag number.
2	`Router(config-dial-peer)#` **`destination-pattern`** *`string`*	Configure the dial peer's destination pattern. The same restrictions for the string listed in the POTS dial-peer configuration also apply to the VoFR destination pattern.
3	`Router(config-dial-peer)#` **`session target`** *`interface dlci`* `[`*`cid`*`]`	Configure the Frame Relay session target for the dial peer.
4	`Router(config-dial-peer)#` **`session protocol cisco-switched`**	Configure the session protocol to support switched calls. This is the default setting, and entering this command is not required.
5	`Router(config-dial-peer)#` **`codec`** *`type`* `[`**`bytes`** *`bytes`*`]`	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**. 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 (?). **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** and **frf11-trunk**), you must configure the codec type on the dial peer. You cannot specify the payload size on the voice port.
6	`Router(config-dial-peer)#` **`dtmf-relay`**	(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. DTMF relay is disabled by default.
7	`Router(config-dial-peer)#` **`signal-type`** `{`**`cas`** `|` **`cept`** `|` **`ext-signal`** `|` **`transparent`**`}`	Define the flavor of the ABCD signaling packets that are generated by the voice port and sent to the data network. Enter **cas** to support CAS. Enter **cept** to support the European CEPT standard (related to MEL CAS). 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. Enter **transparent** (for digital T1/E1 interfaces) 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).
8	`Router(config-dial-peer)#` **`no vad`**	(Optional) Disable voice activity detection (VAD) on the dial peer. This command is enabled by default.

Step	Command	Purpose
9	`Router(config-dial-peer)#` **`sequence-numbers`**	(Optional) Enable the voice sequence number if required for your configuration. This command is disabled by default.
10	`Router(config-dial-peer)#` **`preference`** *`value`*	(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.
11	`Router(config-dial-peer)#` **`fax rate`** `{`**`2400`** `|` **`4800`** `|` **`7200`** `|` **`9600`** `|` **`14400`** `|` **`disable`** `|` **`voice`**`}`	(Optional) Configure the transmission speed (in bps) at which a fax will be sent to the dial peer. The default is **voice**, which specifies the highest possible transmission speed allowed by the voice rate.
12	To configure another VoFR dial peer, exit dial-peer configuration mode and repeat steps 1 through 11.	

Configuring Cisco Trunk Permanent Calls

You can configure Cisco trunk permanent calls on Cisco 2600, 3600, and 7200 series routers, and on Cisco MC3810 concentrators.

Note If configuring Cisco trunk permanent calls to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK , see "Configuring Cisco Trunk Permanent Calls to a Cisco MC3810 Running Cisco IOS Releases Before 12.0(7)XK" section on page 12.

To configure Cisco trunk permanent calls on these routers, use the following commands from interface configuration mode:

Step	Command	Purpose
1	`Router(config-if)#` **`frame-relay interface-dlci`** *`dlci`*	Configure the Frame Relay DLCI and enter DLCI configuration mode. **Note** The **voice-encap** option of the **frame-relay interface-dlci** command on the Cisco MC3810 is no longer supported beginning in this release.
2	`Router(config-if)#` **`vofr`** `[`**`data`** *`cid`*`] [`**`call-control`** `[`*`cid`*`]]`	Configure the Frame Relay DLCI to support VoFR. **Note** When you enter the **vofr** command is used, all subchannels on the DLCI are configured for FRF.11 encapsulation. If you enter the **vofr** command is entered without any keywords or arguments, the data subchannel is CID 4 and there are no call-control subchannel. If configuring tandem calls, this step ends your configuration.
3	For Cisco 2600 and 3600 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `Router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `Router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.

Step	Command	Purpose
4	Router(config-voiceport)# **connection trunk** *destination-string* [**answer-mode**]	For private line calls, configure the trunk connection by specifying the telephone number in the *destination-string*. 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 operates in master mode. Enter the **answer-mode** keyword to specify that the voice port operates in slave mode.
5	Router(config-voiceport)# **shutdown**	Shut down the voice port.
6	Router(config-voiceport)# **no shutdown**	Reactivate the voice port to enable the trunk connection to take effect.

This configuration uses standard FRF.11 Annex C fragmentation.

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.

Configuring Cisco Trunk Permanent Calls to a Cisco MC3810 Running Cisco IOS Releases Before 12.0(7)XK

To configure Cisco trunk permanent calls to a Cisco MC3810 running Cisco IOS releases before 12.0(7)XK , use the following commands from interface configuration mode:

Step	Command	Purpose
1	Router(config-if)# **frame-relay interface-dlci** *dlci*	Configure the Frame Relay DLCI and enter DLCI configuration mode.
2	Router(config-if)# **vofr cisco**	Configure the Frame Relay DLCI to support VoFR and the Cisco proprietary data implementation. When this command is entered, data CID 4 and call-control CID 5 are automatically assigned.
3	For Cisco 2600 and 3600 series analog voice ports: Router(config)# **voice-port** *slot/subunit/port* For Cisco 2600 and 3600 series digital voice ports: Router(config)# **voice-port** *slot/port:ds0-group* For Cisco MC3810 series analog voice ports: Router(config)# **voice-port** *slot/port* For Cisco MC3810 series digital voice ports: Router(config)# **voice-port** *slot:ds0-group*	Identify the voice port you want to configure and enter voice-port configuration mode.
4	Router(config-voiceport)# **connection trunk** *destination-string* [**answer-mode**]	For private line calls, configure the trunk connection, specifying the telephone number in the *destination-string*. 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 operates in master mode. Enter the **answer-mode** keyword to specify that the voice port should operate in slave mode.
5	Router(config-voiceport)# **shutdown**	Shut down the voice port.

Step	Command	Purpose
6	`Router(config-voiceport)#` **`no shutdown`**	Reactivate the voice port to enable the trunk connection to take effect.

This configuration uses Cisco proprietary data fragmentation.

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.

Configuring FRF.11 Trunk (Private Line) Calls

On the Cisco MC3810 and on Cisco 2600 and 3600 series routers, you can configure FRF.11 trunk calls to a second router.

You cannot configure FRF.11 trunk calls for tandem VoFR configurations.

Note This configuration requires that you set the **session protocol** dial-peer configuration command to **frf11-trunk**.

To configure FRF.11 trunk (private line) calls, use the following commands from interface configuration mode:

Step	Command	Purpose
1	`Router(config-if)#` **`frame-relay interface-dlci`** *`dlci`*	Configure the Frame Relay DLCI and enter DLCI configuration mode.
2	Choose one of the following:	
	`Router(config-if)#` **`vofr`** `[`**`data`** *`cid`*`] [`**`call-control`** *`cid`*`]`	(Cisco 2600 and Cisco 3600) Configure the Frame Relay DLCI to support VoFR and to optionally enter the data and call-control CIDs.
	`Router(config-if)#` **`vofr`** `[`**`data`** *`cid`*`]`	(Cisco MC3810) Configure the Frame Relay DLCI to support VoFR and to optionally enter the data CID. The **call-control** option is not supported on the Cisco MC3810. **Note** When you enter the **vofr** command is used, all subchannels on the DLCI are configured for FRF.11 encapsulation except the data subchannel. If you enter the **vofr** command without any keywords or arguments, the data subchannel is CID 4 and there is no call-control subchannel.
3	For Cisco 2600 and 3600 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/subunit/port`* For Cisco 2600 and 3600 series digital voice ports: `Router(config)#` **`voice-port`** *`slot/port:ds0-group`* For Cisco MC3810 series analog voice ports: `Router(config)#` **`voice-port`** *`slot/port`* For Cisco MC3810 series digital voice ports: `Router(config)#` **`voice-port`** *`slot:ds0-group`*	Identify the voice port you want to configure and enter voice-port configuration mode.

Step	Command	Purpose
4	`Router(config-voiceport)#` **`connection trunk`** *`destination-string`* `[`**`answer-mode`**`]`	For private line calls, configure the trunk connection by specifying the telephone number in the *destination-string*. When configuring FRF.11 trunk calls, one side must be the call initiator (master) and the other side is normally the call answerer (slave). By default, the voice port is the master. Enter the **answer-mode** keyword to specify that the voice port is the slave.
5	`Router(config-voiceport)#` **`shutdown`**	Shut down the voice port.
6	`Router(config-voiceport)#` **`no shutdown`**	Reactivate the voice port to enable the FRF.11 trunk connection to take effect.

This configuration uses FRF.11 Annex C data fragmentation.

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.

Configuring Connections for Tandem Nodes

Tandeming is switching incoming VoFR calls on a Frame Relay DLCI to an outgoing VoFR enabled DLCI. Tandeming works for switched calls and Cisco-trunk permanent calls only. You cannot tandem FRF.11 trunk calls over a multihop network.

Tandeming is supported on the Cisco MC3810 and on Cisco 2600, 3600, and 7200 series routers.

Note When creating voice networks with a mixture of router types, the Cisco MC3810 must be running Cisco IOS Release 12.0(3)XG, 12.0(4)T, or later releases to act as a tandem node.

Depending on which router is used as the end node and which router is used as the tandem node, you must use the correct Frame Relay PVC type when configuring your connections. Table 2 shows the different combinations of routers that can serve as end nodes and tandem nodes, and the Frame Relay PVC type required.

Table 2 VoFR End Node and Tandem Node Combinations That Are Supported

End Node(s)	Tandem Node	vofr Command to Enter for the Frame Relay DLCI
Cisco 2600/3600, Cisco MC3810 or Cisco 7200	Cisco 2600, Cisco 3600, Cisco MC3810 or Cisco 7200	**vofr call-control**
Cisco 2600/3600, Cisco MC3810	Cisco MC3810 running Cisco IOS releases before 12.0(7)XK	**vofr cisco**
Cisco MC3810 running Cisco IOS releases before 12.0(7)XK	Cisco 2600, Cisco 3600, or Cisco 7200	**vofr cisco**

When you configure a tandem node, you must configure two VoFR dial peers, one for each tandem connection.

Verifying Your Voice Connections

Verify that the voice connection for switched calls is working by following these steps:

1. Pick up the handset on a telephone connected to the configuration and verify that you can get a dial tone.
2. Make a call from the local telephone to a configured dial peer and verify that the call attempt is successful.

Verify that the voice connection for FXO-FXS trunk calls from a telephone to a remote PBX is working by doing the following:

1. Pick up the telephone and listen to hear the dial tone from the remote PBX.
2. Dial digits so that the remote PBX routes the call.

You can check the validity of your dial-peer and voice-port configuration by performing the following tasks:

- If you have relatively few dial peers configured, enter the **show dial-peer voice** command to verify that the data configured is correct. On the Cisco MC3810, enter the **show dial-peer voice summary** command.
- To show the status of the voice ports, enter the **show voice port** command.
- To show the call status for all voice ports, enter the **show call active voice [brief]**. On the Cisco MC3810, use the **show voice call** command.
- To show the current status of all DSP voice channels on the Cisco MC3810, enter the **show voice dsp** command.

You can check the validity of your VoFR configuration on the DLCI by performing the following task:

- To show the VoFR configuration, enter the **show frame-relay vofr** [*interface* [*dlci* [*cid*]]] command. This command is not supported on the Cisco MC3810 when the **vofr cisco** command is configured.
- To show the status of Cisco-trunk permanent calls (private-line) on the Cisco MC3810, enter the **show voice permanent** command.

Troubleshooting Tips

If you are having trouble connecting a call, you can try to resolve the problem by performing the following tasks:

- If no calls are going through, make sure the **frame-relay voice bandwidth** command is configured.
- If you have Voice over Frame Relay configured on a PVC and are experiencing problems with data connectivity on that PVC, make sure the **frame-relay fragment** command has been configured.
- If data is not being transmitted but fragmentation is configured, make sure that Frame Relay traffic shaping is turned on.
- If you suspect the problem is with the dial plan or the dial peers, use the **show dial-plan number** *dial string* command to display which dial peers are used when a specific number is called.
- If you have problems connecting an FRF.11 trunk call, make sure the **session protocol** dial-peer command is set to **frf11-trunk**.

- If configuring FRF.11 trunk calls on the Cisco 2600/3600, verify that the **called-number vofr** dial-peer command is configured, and that its number matches the destination-pattern of the corresponding POTS dial peer.
- Be sure the voice port is set to **no shutdown**.
- Be sure the serial port or the T1/E1 controller is set to **no shutdown**.
- Be sure to toggle the voice port (by first entering **shutdown**, then n**o shutdown**) every time you enter the **connection trunk** or **no connection trunk** commands.

Configuration Examples

This section provides specific configuration examples for different VoFR connections and call type scenarios. This section includes the following examples:

- Two Routers Using Frame Relay Fragmentation on page 18
- Two Routers Using a VoFR PVC on page 19
- Router Using a VoFR PVC to a Cisco MC3810 Running Cisco IOS Versions Before 12.0(7)XK on page 20
- Cisco-Trunk (Private Line) Calls between Two Routers on page 21
- FRF.11 Trunk Calls between Two Routers on page 22
- Tandem Configuration with Three Routers for Switched Calls on page 24
- Tandem Configuration with a Cisco MC3810 Tandem Node for Switched Calls on page 25
- Tandem Configuration with a Cisco MC3810 Endpoint Node for Cisco-Trunk (Private Line) Calls on page 26
- Tandem Configuration with All Cisco MC3810 Concentrators for Switched Calls on page 28
- Cisco Trunk Call with Hunt Groups on page 30

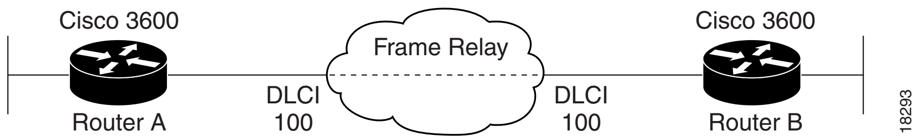
Note In the examples, some commands are shown with a lowercase letter in boldface. These letters indicate command settings that must match on the different routers. For example, the **frame-relay cir** ***s*** value indicates that the committed information rate “s” must match on the routers as shown.

The examples do not provide complete configurations, but show the required commands to configure Voice over Frame Relay.

Two Routers Using Frame Relay Fragmentation

Figure 1 shows an example of Frame Relay fragmentation between two routers.

Figure 1 Two Routers Using Frame Relay Fragmentation



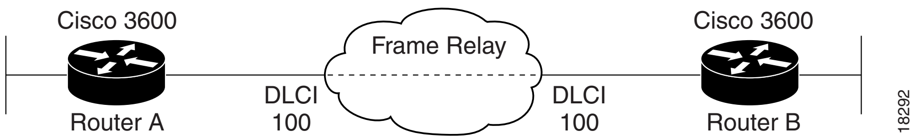
Router A (Cisco 3600)	Router B (Cisco 3600)
`interface serial 0/0`	`interface serial 0/0`
`ip address xxx.xxx.xxx 255.255.255.0`	`ip address xxx.xxx.xxx 255.255.255.0`
`frame-relay traffic shaping`	`frame-relay traffic shaping`
	`frame-relay class toto`
`frame-relay interface-dlci 100`	`frame-relay interface-dlci 100`
`class toto`	
	`map-class frame-relay toto`
`map-class frame-relay toto`	`encapsulation frame-relay`
`encapsulation frame-relay`	`frame-relay cir` **s**
`frame-relay cir` **s**	`frame-relay bc` **u**
`frame-relay bc` **u**	`frame-relay fragment` **y**
`frame-relay fragment` **y**	

This configuration uses FRF.12 fragmentation.

Two Routers Using a VoFR PVC

Figure 2 shows an example of two routers with connections using a VoFR PVC.

Figure 2 **Two Routers Using a VoFR PVC**



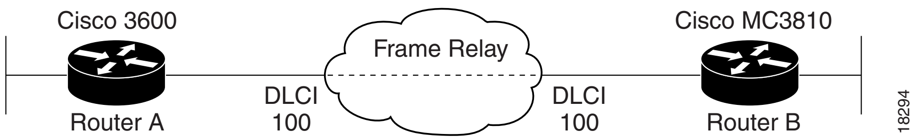
Router A (Cisco 3600)	Router B (Cisco 3600)
`interface serial 0/0`	`interface serial 0/0`
`  frame-relay traffic shaping`	`  frame-relay traffic shaping`
	`  frame-relay class toto`
`frame-relay interface-dlci 100`	
`  vofr data` **`z`**	`frame-relay interface-dlci 100`
`  class toto`	`  vofr data` **`z`**
`map-class frame-relay toto`	`map-class frame-relay toto`
`  frame-relay voice-bandwidth` **`t`**	`  frame-relay voice-bandwidth` **`t`**
`  frame-relay min-cir` **`x`**	`  frame-relay min-cir` **`x`**
`  frame-relay cir` **`s`**	`  frame-relay cir` **`s`**
`  frame-relay bc` **`u`**	`  frame-relay bc` **`u`**
`  frame-relay fragment` **`y`**	`  frame-relay fragment` **`y`**

This configuration uses FRF.11 Annex C fragmentation.

Router Using a VoFR PVC to a Cisco MC3810 Running Cisco IOS Versions Before 12.0(7)XK

Figure 3 shows an example of a Cisco 3600 series router with connections to a Cisco MC3810 running a Cisco IOS version before 12.0(7)XK. In this example, the Voice over Frame Relay interface on both the Cisco 3600 and the Cisco MC3810 is configured by using the **vofr cisco** command.

Figure 3 Router Using a VoFR PVC to a Cisco MC3810 Running Cisco IOS Versions Before 12.0(7)XK



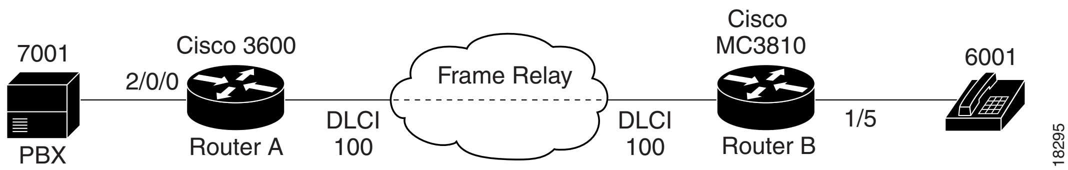
Router A (Cisco 3600)	Router B (Cisco MC3810)
`interface serial 0/0`	`interface serial 0`
`ip address xxx.xxx.xxx 255.255.255.0`	`ip address xxx.xxx.xxx 255.255.255.0`
`frame-relay traffic shaping`	`frame-relay traffic shaping`
	`frame-relay class toto`
`frame-relay interface-dlci 100`	
`vofr cisco`	`frame-relay interface-dlci 100`
`class toto`	`vofr cisco`
`map-class frame-relay toto`	`map-class frame-relay toto`
`frame-relay voice-bandwidth` **`t`**	`frame-relay voice-bandwidth` **`t`**
`frame-relay min-cir` **`x`**	`frame-relay min-cir` **`x`**
`frame-relay cir` **`s`**	`frame-relay cir` **`s`**
`frame-relay bc` **`u`**	`frame-relay bc` **`u`**
`frame-relay fragment` **`y`**	`frame-relay fragment` **`y`**

This configuration uses FRF.11 Annex C fragmentation.

Cisco-Trunk (Private Line) Calls between Two Routers

Figure 4 shows an example of VoFR Cisco-trunk (private line) calls between two routers.

Figure 4 Cisco-Trunk (Private Line) Calls Between Two Routers

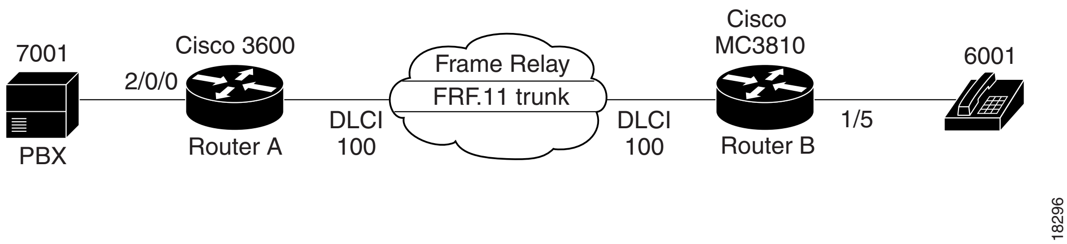


Router A (Cisco 3600)	Router B (Cisco MC3810)
`interface serial 0/0`	`interface serial 0`
`ip address xxx.xxx.xxx 255.255.255.0`	`ip address xxx.xxx.xxx 255.255.255.0`
`encapsulation frame-relay`	`encapsulation frame-relay`
`frame-relay traffic shaping`	`frame-relay traffic shaping`
`frame-relay interface-dlci 100`	`frame-relay interface-dlci 100`
`class voice`	`class voice`
`vofr data 4 call-control 5`	`vofr data 4 call-control 5`
`map-class frame-relay voice`	`map-class frame-relay voice`
`frame relay cir` **s**	`frame relay cir` **s**
`frame relay bc` **u**	`frame relay bc` **u**
`frame-relay voice bandwidth` **v**	`frame-relay voice bandwidth` **v**
`frame-relay min-cir` **x**	`frame-relay min-cir` **x**
`frame-relay fragment y`	`frame-relay fragment` **y**
`voice-port 2/0/0`	`voice-port 1/5`
`connection trunk 6001 answer-mode`	`connection trunk 7001`
`dial-peer voice 1 pots`	`dial-peer voice 2 pots`
`destination-pattern 7001`	`destination-pattern 6001`
`port 2/0/0`	`port 1/5`
`dial-peer voice 2 vofr`	`dial-peer voice 4 vofr`
`codec` **x** `bytes` **y**	`codec` **x** `bytes` **y**
`destination-pattern 6001`	`destination-pattern 7001`
`session protocol cisco-switched`	`session protocol cisco-switched`
`session target` **Sn** `100`	`session target` **Sn** `100`

FRF.11 Trunk Calls between Two Routers

Figure 5 shows an example of FRF.11 trunk calls configured between two routers.

Figure 5 FRF.11 Trunk Calls between Two Routers



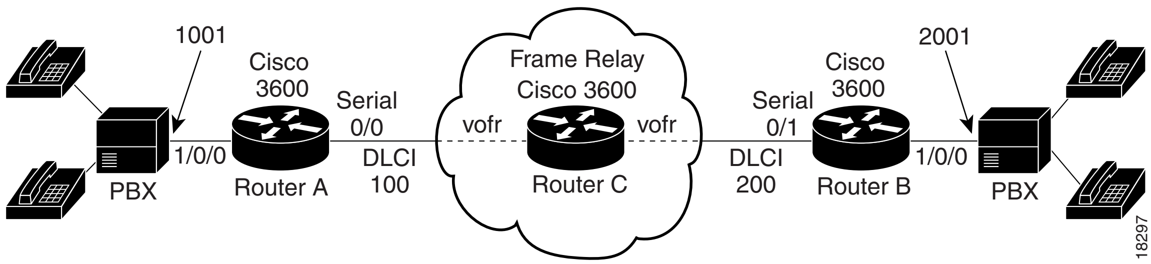
Router A (Cisco 3600)	Router B (Cisco MC3810)
`interface serial 0/0`	`interface serial 0`
`ip address xxx.xxx.xxx 255.255.255.0`	`ip address xxx.xxx.xxx 255.255.255.0`
`encapsulation frame-relay`	`encapsulation frame-relay`
`frame-relay traffic shaping`	`frame-relay traffic shaping`
`frame-relay interface-dlci 100`	`frame-relay interface-dlci 100`
`class voice`	`class voice`
`vofr data 4`	`vofr data 4`
`map-class frame-relay voice`	`map-class frame-relay voice`
`frame relay cir` **s**	`frame relay cir` **s**
`frame-relay min-cir in` **x**	`frame-relay min-cir in` **x**
`frame relay bc` **u**	`frame relay bc` **u**
`frame-relay voice bandwidth` **v**	`frame-relay voice bandwidth` **v**
`frame-relay fragment` **y**	`frame-relay fragment` **y**
`voice-port 2/0/0`	`voice-port 1/5`
`connection trunk 6001`	`connection trunk 7001`
`dial-peer voice 1 pots`	`dial-peer voice 2 pots`
`destination-pattern 7001`	`destination-pattern 6001`
`port 2/0/0`	`port 1/5`
`dial-peer voice 2 vofr`	`dial-peer voice 4 vofr`
`codec` **x** `bytes` **y** `bytes`	`codec` **x** `bytes` **y**
`destination-pattern 6001`	`destination-pattern 7001`
`session protocol frf11-trunk`	`session protocol frf11-trunk`
`session target` **Sn** `100` **d**	`session target` **Sn** `100` **d**
`called-number 7001`	`dtmf-relay`

Router A (Cisco 3600)	Router B (Cisco MC3810)
dtmf-relay	vad
vad	

Tandem Configuration with Three Routers for Switched Calls

Figure 6 shows an example of a tandem configuration with two Cisco 3600 routers as endpoints and a third Cisco 3600 as a tandem node.

Figure 6 Tandem Configuration with Three Routers for Switched Calls

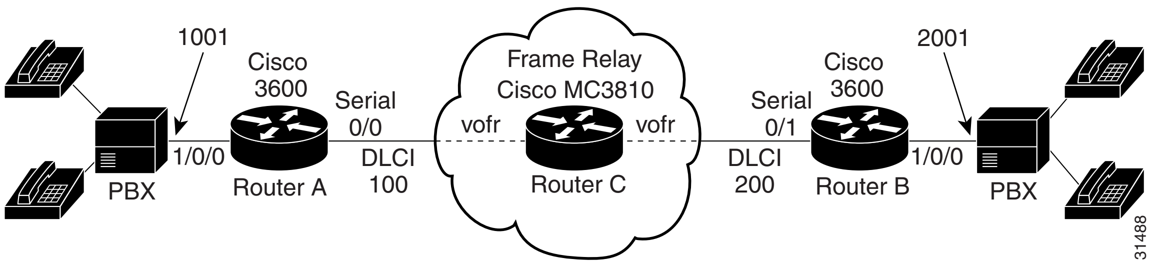


Router A (Cisco 3600) Endpoint	Router C (Cisco 3600) Tandem Node	Router B (Cisco 3600) Endpoint
interface serial 0/0	interface serial 0/0	interface serial 0/0
encapsulation frame-relay	encapsulation frame-relay	encapsulation frame-relay
frame-relay traffic-shaping	frame-relay traffic-shaping	frame-relay traffic-shaping
frame-relay interface-dlci 100	frame-relay interface-dlci 100	frame-relay interface-dlci 100
class voice	class voice	class voice
vofr data 4 call-control 5	vofr data 4 call-control 5	vofr data 4 call-control 5
map-class frame-relay voice	interface serial 0/1	map-class frame-relay voice
frame-relay cir **a**	encapsulation frame-relay	frame-relay cir **a**
frame-relay min-cir **t**	frame-relay traffic-shaping	frame-relay min-cir **t**
frame-relay bc **b**	frame-relay interface-dlci 200	frame-relay bc **b**
frame-relay voice bandwidth **c**	class voice	frame-relay voice bandwidth **c**
frame-relay fragment **d**	vofr	frame-relay fragment **d**
dial-peer voice 1 pots	map-class frame-relay voice	dial-peer voice 1 pots
destination-pattern 1001	frame-relay cir **a**	destination-pattern 2001
port 1/0/0	frame-relay min-cir **t**	port 1/0/0
	frame-relay bc **b**	
	frame-relay voice bandwidth **c**	
dial-peer voice 2 vofr	frame-relay fragment **d**	dial-peer voice 2 vofr
destination-pattern 2...		destination-pattern 1...
session target serial 0/0 100	dial-peer voice 1 vofr	session target serial 0/0 200
	destination-pattern 1...	
voice-port 1/0/0	session target serial 0/0 100	voice-port 1/0/0
	dial-peer voice 2 vofr	
	destination-pattern 2...	
	session target serial 0/1 200	

Tandem Configuration with a Cisco MC3810 Tandem Node for Switched Calls

Figure 7 shows an example of a tandem configuration with a Cisco MC3810 acting as a tandem node.

Figure 7 Tandem Configuration with a Cisco MC3810 Tandem Node for Switched Calls

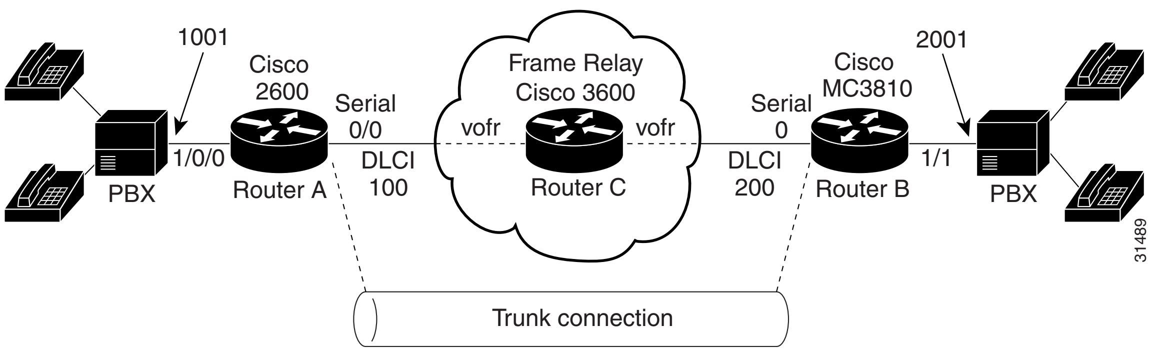


Router A (Cisco 3600) Endpoint	Router C (Cisco MC3810) Tandem Node	Router B (Cisco 3600) Endpoint
`interface serial 0/0`	`interface serial 0`	`interface serial 0/0`
`encapsulation frame-relay`	`encapsulation frame-relay`	`encapsulation frame-relay`
`frame-relay traffic-shaping`	`frame-relay traffic-shaping`	`frame-relay traffic-shaping`
`frame-relay interface-dlci 100`	`frame-relay interface-dlci 100`	`frame-relay interface-dlci 100`
`class voice`	`class voice`	`class voice`
`vofr data 4 call-control 5`	`vofr data 4 call-control 5`	`vofr data 4 call-control 5`
`map-class frame-relay voice`	`interface serial 1`	`map-class frame-relay voice`
`frame-relay cir` **a**	`encapsulation frame-relay`	`frame-relay cir` **a**
`frame-relay min-cir` **t**	`frame-relay traffic-shaping`	`frame-relay min-cir` **t**
`frame-relay bc` **b**	`frame-relay interface-dlci 200`	`frame-relay bc` **b**
`frame-relay voice bandwidth` **c**	`class voice`	`frame-relay voice bandwidth` **c**
`frame-relay fragment` **d**	`vofr data 4 call-control 5`	`frame-relay fragment` **d**
`dial-peer voice 1 pots`	`map-class frame-relay voice`	`dial-peer voice 1 pots`
`destination-pattern 1001`	`frame-relay cir` **a**	`destination-pattern 2001`
`port 1/0/0`	`frame-relay min-cir` **t**	`port 1/0/0`
	`frame-relay bc` **b**	
	`frame-relay voice bandwidth` **c**	
`dial-peer voice 2 vofr`	`frame-relay fragment` **d**	`dial-peer voice 2 vofr`
`destination-pattern 2...`		`destination-pattern 1...`
`session target serial 0/0 100`	`dial-peer voice 1 vofr`	`session target serial 0/0 200`
	`destination-pattern 1...`	
`voice-port 1/0/0`	`session target serial 0/0 100`	`voice-port 1/0/0`
	`dial-peer voice 2 vofr`	
	`destination-pattern 2...`	
	`session target serial 0/1 200`	

Tandem Configuration with a Cisco MC3810 Endpoint Node for Cisco-Trunk (Private Line) Calls

Figure 8 shows an example of a tandem configuration with a Cisco MC3810 acting as an endpoint node for Cisco-trunk (private line) calls.

Figure 8 Tandem Configuration with a Cisco MC3810 Endpoint Node for Permanent Switched Calls



Router A (Cisco 2600) Endpoint	Router C (Cisco 3600) Tandem Node	Router B (Cisco MC3810) Endpoint
`interface serial 0/0`	`interface serial 0/0`	`interface serial 0`
`encapsulation frame-relay`	`encapsulation frame-relay`	`encapsulation frame-relay`
`frame-relay traffic-shaping`	`frame-relay traffic-shaping`	`frame-relay traffic-shaping`
`frame-relay interface-dlci 100`	`frame-relay interface-dlci 100`	`frame-relay interface-dlci 200`
`class voice`	`class voice`	`class voice`
`vofr data 4 call-control 5`	`vofr data 4 call-control 5`	`vofr data 4 call-control 5`
`map-class frame-relay voice`	`interface serial 0/1`	`map-class frame-relay voice`
`frame-relay cir` **a**	`encapsulation frame-relay`	`frame-relay cir` **a**
`frame-relay min-cir` **t**	`frame-relay traffic-shaping`	`frame-relay min-cir` **t**
`frame-relay bc` **b**	`frame-relay interface-dlci 200`	`frame-relay bc` **b**
`frame-relay voice bandwidth` **c**	`class voice`	`frame-relay voice bandwidth` **c**
`frame-relay fragment` **d**	`vofr data 4 call-control 5`	`frame-relay fragment` **d**
`dial-peer voice 1 pots`	`map-class frame-relay voice`	`dial-peer voice 1 pots`
`destination-pattern 1001A`	`frame-relay cir` **a**	`destination-pattern 2001A`
`port 1/0/0`	`frame-relay min-cir` **t**	`port 1/1`
	`frame-relay bc` **b**	
`dial-peer voice 2 vofr`	`frame-relay voice bandwidth` **c**	`dial-peer voice 2 vofr`
`destination-pattern 2...`	`frame-relay fragment` **d**	`destination-pattern 1...`
`session target serial 0/0 100`		`session target serial 0 200`
	`dial-peer voice 1 vofr`	
`voice-port 1/0/0`	`destination-pattern 1...`	`voice-port 1/1`

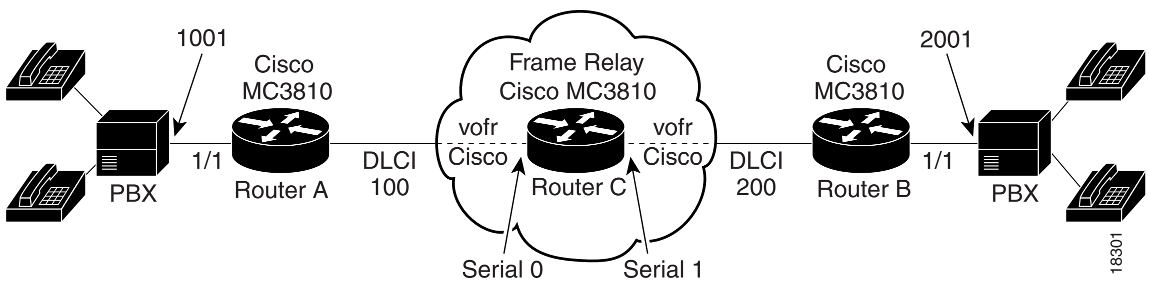
Router A (Cisco 2600) Endpoint	Router C (Cisco 3600) Tandem Node	Router B (Cisco MC3810) Endpoint
`connection trunk 2001A answer-mode`	`session target serial 0/0 100`	`connection trunk 1001A`
	`dial-peer voice 2 vofr`	
	`destination-pattern 2...`	
	`session target serial 0/1 200`	

Tandem Configuration with All Cisco MC3810 Concentrators for Switched Calls

Figure 9 shows an example of a tandem configuration with Cisco MC3810 concentrators as both endpoint and tandem nodes.

Note When a Cisco MC3810 running Cisco IOS versions before 12.0(7)XK is on a VoFR network, the configuration for connections to and from that Cisco MC3810 is slightly different than for other routers that support VoFR. The **vofr cisco** command is required for these connections on the Cisco MC3810.

Figure 9 Configuration with All Cisco MC3810 Concentrators as Endpoint and Tandem Nodes



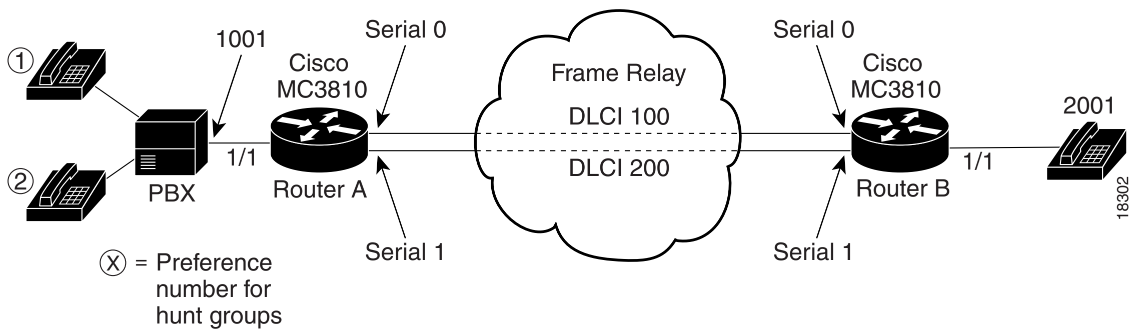
Router A (Cisco MC3810) Endpoint	Router C (Cisco MC3810) Tandem Node Running Cisco IOS Version Before 12.0(7)XK	Router B (Cisco MC3810) Endpoint
interface serial 0	interface serial 0	interface serial 0
encapsulation frame-relay	encapsulation frame-relay	encapsulation frame-relay
frame-relay traffic-shaping	frame-relay traffic-shaping	frame-relay traffic-shaping
frame-relay interface-dlci 100	frame-relay interface-dlci 100	frame-relay interface-dlci 200
class voice	class voice	class voice
vofr cisco	vofr cisco	vofr cisco
map-class frame-relay voice	interface serial 1	map-class frame-relay voice
frame-relay cir **a**	encapsulation frame-relay	frame-relay cir **a**
frame-relay bc **b**	frame-relay traffic-shaping	frame-relay bc **b**
frame-relay voice bandwidth **c**	frame-relay interface-dlci 200	frame-relay voice bandwidth **c**
frame-relay min-cir **t**	class voice	frame-relay fragment **d**
	vofr cisco	frame-relay min-cir **t**
dial-peer voice 1 pots	map-class frame-relay voice	dial-peer voice 1 pots
destination-pattern 1001	frame-relay cir **a**	destination-pattern 2001
port 1/1	frame-relay min-cir **t**	port 1/1
	frame-relay bc **b**	
dial-peer voice 2 vofr	frame-relay voice bandwidth **c**	dial-peer voice 2 vofr
destination-pattern 2...	frame-relay fragment **d**	destination-pattern 1...

Router A (Cisco MC3810) Endpoint	Router C (Cisco MC3810) Tandem Node Running Cisco IOS Version Before 12.0(7)XK	Router B (Cisco MC3810) Endpoint
`session target serial 0 100`		`session target serial 0 200`
	`dial-peer voice 1 vofr`	
`voice-port 1/1`	`destination-pattern 1...`	`voice-port 1/1`
	`session target serial 0 100`	
	`dial-peer voice 2 vofr`	
	`destination-pattern 2...`	
	`session target serial 1 200`	

Cisco Trunk Call with Hunt Groups

Figure 10 shows an example of a Cisco trunk (private line) call that is configured with hunt groups. In this example, the two routers are in master-slave mode with a backup path. Router B is configured as a slave and Router A is configured as the master. The master makes periodic attempts to establish the trunk until the trunk is established. Two dial peers match the destination string configured in the voice port, but because one dial peer has a higher preference than the other dial peer, the call setup is attempted through that dial peer. If the call setup fails, the master can continue attempting call setups by using the next available dial peer. After all dial peers are exhausted, the master can continue following the list cyclically by starting again from the dial peer with the highest preference.

Figure 10 Cisco Trunk (Private Line) Call with Hunt Groups



Router A (Cisco MC3810)	Router B (Cisco MC3810)
`interface serial 0`	`interface serial 0`
`  encapsulation frame-relay`	`  encapsulation frame-relay`
`  frame-relay traffic-shaping`	`  frame-relay traffic-shaping`
`  frame-relay interface-dlci 100`	`  frame-relay interface-dlci 100`
`  class voice`	`  class voice`
`  vofr data 4 call-control 5`	`  vofr data 4 call-control 5`
`interface serial 1`	`interface serial 1`
`  encapsulation frame-relay`	`  encapsulation frame-relay`
`  frame-relay traffic-shaping`	`  frame-relay traffic-shaping`
`  frame-relay interface-dlci 200`	`  frame-relay interface-dlci 200`
`  class voice`	`  class voice`
`  vofr data 4 call-control 5`	`  vofr data 4 call-control 5`
`map-class frame-relay voice`	`map-class frame-relay voice`
`  frame-relay cir` **a**	`  frame-relay cir` **a**
`  frame-relay bc` **b**	`  frame-relay bc` **b**
`  frame-relay voice bandwidth` **c**	`  frame-relay voice bandwidth` **c**
`  frame-relay min-cir` **t**	`  frame-relay min-cir` **t**
`dial-peer voice 1 pots`	`dial-peer voice 1 pots`

Router A (Cisco MC3810)	Router B (Cisco MC3810)
`  destination-pattern 1001A`	`  destination-pattern 2001A`
`  port 1/1`	`  port 1/1`
`dial-peer voice 100 vofr`	`dial-peer voice 100 vofr`
`  destination-pattern 2...`	`  destination-pattern 1...`
`  session target serial0 100`	`  session target serial0 100`
`  preference 1`	`  preference 1`
`dial-peer voice 200 vofr`	`dial-peer voice 200 vofr`
`  destination-pattern 2...`	`  destination-pattern 1...`
`  session target serial1 200`	`  session target serial1 200`
`  preference 2`	`  preference 2`
`voice-port 1/1`	`voice-port 1/1`
`  connection trunk 2005A`	`  description FXS port`
`  description FXO port`	`  connection trunk 1001A answer-mode`

Command Reference

This section documents new or modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.0 command reference publications, or in the Cisco IOS 12.0(4)T online document *Voice over Frame Relay Using FRF.11 and FRF.12*.

See new and modified debug commands in the "Debug Commands" section on page 52.

The following new and modified commands are described in this section (modified commands are marked by an asterisk):

- **dial-peer hunt**
- **dial-peer terminator***
- **dial-peer voice***
- **forward-digits**
- **frag-pre-queuing***
- **frame-relay interface-dlci***
- **huntstop***
- **signal-type***
- **vofr***

In Cisco IOS Release 12.0(1)T or later releases, you can search and filter the output for **show** and **more** commands. This functionality is useful when you need to sort through large amounts of output, or if you want to exclude output that you do not need to see.

To use this functionality, enter a **show** or **more** command followed by the "pipe" character (|), one of the keywords **begin**, **include**, or **exclude**, and an expression that you want to search or filter on:

command **|** {**begin** | **include** | **exclude**} *regular-expression*

See the following example of the **show atm vc** command where you want the command output to begin with the first line where the expression "PeakRate" appears:

show atm vc | begin PeakRate

For more information on the search and filter functionality, refer to the Cisco IOS Release 12.0(1)T feature module titled *CLI String Search*.

dial-peer hunt

To specify a hunt selection order for dial-peers, use the **dial-peer hunt** dial-peer configuration command. Use the **no** form of this command to restore the default selection order.

dial-peer hunt *hunt-order-number*
no dial-peer hunt

Syntax Description

hunt-order-number	A number from 0 to 7 that selects a predefined hunting selection order: 0—Longest match in phone number, explicit preference, random selection. This is the default hunt order number. 1—Longest match in phone number, explicit preference, least recent use. 2—Explicit preference, longest match in phone number, random selection. 3—Explicit preference, longest match in phone number, least recent use. 4—Least recent use, longest match in phone number, explicit preference. 5—Least recent use, explicit preference, longest match in phone number. 6—Random selection. 7—Least recent use.

Default

The default is the longest match in phone number, explicit preference, and random selection (hunt order number 0).

Command Mode

Global configuration

Command History

Release	Modification
12.0(7)XK	This command was introduced, and was first supported on the Cisco 2600 and 3600 series routers and on the Cisco MC3810 multiservice access concentrator.

Usage Guidelines

Use the **dial-peer hunt** dial-peer configuration command if you have configured hunt groups. “Longest match in phone number” refers to the destination pattern that matches the greatest number of the dialed digits. “Explicit preference” refers to the **preference** setting in the dial-peer configuration. “Least recent use” refers to the destination pattern that has waited the longest since being selected. “Random selection” weighs all of the destination patterns equally in a random selection mode.

Example

The following example configures the dial peers to hunt in the following order: (1) longest match in phone number, (2) explicit preference, (3) random selection.

```
Router# configure terminal
Router(config)# dial-peer hunt 0
```

Related Commands

Command	Description
preference	Specifies the preferred selection order of a dial peer within a hunt group.
destination-pattern	Specifies the prefix or the complete telephone number for a dial peer.
show dial-peer voice	Displays configuration information for dial peers.

dial-peer terminator

To change the character used as a terminator for variable length dialed numbers, use the **dial-peer terminator** global configuration command. Use the **no** form of this command to restore the default terminating character.

dial-peer terminator *character*
no dial-peer terminator

Syntax Description

character	Designates the terminating character for a variable-length dialed number. Valid numbers and characters are #, *, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, a, b, c, and d. The default is #.

Default

The default terminating character is #.

Command Mode

Global configuration

Command History

Release	Modification
12.0	This command was introduced.
12.0(7)XK	Usage was restricted to variable-length dialed numbers.

Usage Guidelines

There are certain areas in the world, 0for example, in certain European countries, where telephone numbers can vary in length. When a dialed-number string is identified as a variable length dialed-number, the system does not place a call until the configured value for the **timeouts interdigits** command has expired, or until the caller dials the terminating character. Use the **dial-peer terminator** global configuration command to change the terminating character.

Example

The following example specifies “9” as the terminating character for variable-length dialed numbers:

```
Router# configure terminal
Router(config)# dial-peer terminator 9#
```

Related Commands

Command	Description
answer-address	Specifies the preferred selection order of a dial peer within a hunt group.
destination-pattern	Specifies the prefix or the complete telephone number for a dial peer.
timeouts interdigit	Specifies the interdigit timeout value for a voice port in seconds.
show dial-peer voice	Displays configuration information for dial peers.

dial-peer voice

To enter dial-peer configuration mode and specify the method of voice encapsulation, use the **dial-peer voice** global configuration command.

For the Cisco 2600 series:

dial-peer voice *tag* {**pots** | **voip** | **vofr**}
no dial-peer voice *tag*

For the Cisco 3600 series and the Cisco MC3810:

dial-peer voice *tag* {**pots** | **voip** | **vofr** | **voatm**}
no dial-peer voice *tag*

Syntax Description

tag	A number identifying a particular dial peer. Valid entries are 1 to 2147483647.
pots	POTS dial peer using basic telephone service.
voip	VoIP dial peer using voice encapsulation on the POTS network.
vofr	Voice over Frame Relay dial peer using encapsulation on the Frame Relay backbone network.
voatm	(Cisco 3600 and MC3810 only) Voice over ATM dial peer using real-time AAL5 voice encapsulation on the ATM backbone network.

Default

No default behavior or values.

Command Mode

Global configuration

Command History

Release	Modification
11.3(1)T	This command was introduced.
11.3(1)MA	This command was first supported on the Cisco MC3810 with support for POTS, VoFR and VoATM.
12.0(3)XG	This command added VoFR to the Cisco 2600 and 3600 series routers.
12.0(4)T	This command added VoFR to the Cisco 7200 series platform.
12.0(7)XK	This command added VoIP to the Cisco MC3810 and VoATM to the Cisco 3600 series routers. Support for VoHDLC on the Cisco MC3810 was removed in this release.

Usage Guidelines

Use the **dial-peer voice** global configuration command to switch to the dial-peer configuration mode from the global configuration mode. Use the **exit** command to exit the dial-peer configuration mode and return to the global configuration mode.

Example

The following example accesses dial-peer configuration mode and configures a POTS peer identified as dial peer 10:

```
Router# configure terminal
Router(config)# dial-peer voice 10 pots
```

Related Commands

Command	Description
voice-port	Enters voice-port configuration mode.

forward-digits

To specify which digits to forward for voice calls, use the **forward-digits** dial-peer configuration command. If the **no** form of this command is entered, any digits not matching the destination-pattern are not forwarded. Use the **default** form of this command to restore the default state.

forward-digits {*num-digit* | **all** | **extra**}
no forward-digits
default forward-digits

Syntax Description

num-digit	The number of digits to be forwarded. If the number of digits is greater than the length of a destination phone number, the length of the destination number is used. The valid range is 0 to 32. Setting the value to 0 is equivalent to entering **no forward-digits**.
all	Forward all digits. If **all** is entered, the full length of the destination pattern is used.
extra	If the length of the dialed digit string is greater than the length of the dial-peer destination pattern, the extra right-justified digits are forwarded. However, if the dial-peer destination pattern is variable length (ending with character "T", for example: T, 123T, 123...T), extra digits are not forwarded.

Default

Dialed digits not matching the destination-pattern are forwarded.

Command Mode

Dial-peer configuration

Command History

Release	Modification
11.3(1) MA	This command was introduced on the Cisco MC3810.
12.0(2) T	The **implicit** option was added.
12.0(4) T	This command was modified to support ISDN PRI QSIG signaling calls.
12.0(7)XK	This command was first supported on the Cisco 2600 and 3600 platforms, the **implicit** keyword was removed, and the **extra** keyword was added.

Usage Guidelines

This command applies only to POTS dial peers.

Forwarded digits are always right-justified, so that extra leading digits are stripped.

The destination pattern includes both explicit digits and wildcards if present.

Use the **default** form of this command if a non-default digit-forwarding scheme was entered previously and you wish to restore the default.

For QSIG ISDN connections, entering **forward-digits all** implies that all the digits of the called party number are sent to the ISDN connection. When you enter **forward-digits** *num-digit* and enter a number from 1 to 32, the number of digits specified (right justified) of the called part number are sent to the ISDN connection.

Examples

The following example forwards all of the digits in the destination pattern of a POTS dial peer:

```
Router(config)# dial-peer voice 1 pots
Router(config-dial-peer)# destination-pattern 8...
Router(config-dial-peer)# forward-digits all
```

The following example forwards 4 of the digits in the destination pattern of a POTS dial peer:

```
Router(config)# dial-peer voice 1 pots
Router(config-dial-peer)# destination-pattern 555....
Router(config-dial-peer)# forward-digits 4
```

The following example forwards the extra right-justified digits that exceed the length of the destination pattern of a POTS dial peer:

```
Router(config)# dial-peer voice 1 pots
Router(config-dial-peer)# destination-pattern 555....
Router(config-dial-peer)# forward-digits extra
```

Related Commands

Command	Description
destination-pattern	Defines the prefix or the full E.164 telephone number to be used for a dial peer.
show dial-peer voice	Displays configuration information for dial peers.

frag-pre-queuing

This command was added in Cisco IOS Release 12.0(2)T on the Cisco MC3810. Beginning with Cisco IOS Release 12.0(7)XK, this command is no longer supported.

frame-relay interface-dlci

To assign a data link connection identifier (DLCI) to a specified Frame Relay subinterface on the router or access server, use the **frame-relay interface-dlci** interface configuration command. Use the **no** form of this command to remove this assignment.

frame-relay interface-dlci *dlci* [**ietf** | **cisco**] [**voice-cir** *cir*]
no frame-relay interface-dlci *dlci* [**ietf** | **cisco**] [**voice-cir** *cir*]

Syntax Description

dlci	DLCI number to be used on the specified subinterface.
ietf | **cisco**	(Optional) Encapsulation type: Internet Engineering Task Force (IETF) Frame Relay encapsulation or Cisco Frame Relay encapsulation.
voice-cir *cir*	(Optional; supported on the Cisco MC3810 only.) Specifies the upper limit on the voice bandwidth that may be reserved for this DLCI. The default is the CIR configured for the Frame Relay map class. For more information, see the "Usage Guidelines" section.

Default

No DLCI is assigned.

Command Mode

Interface configuration

Command History

Release	Modification
10.0	This command was introduced.
11.3(1) MA	The **voice-encap** option was added for the Cisco MC3810.
12.0(2) T	The **voice-cir** option was added for the Cisco MC3810.
12.0(3)XG and 12.0(4)T	Additional usage guidelines added.
12.0(7)XK	The **voice-encap** option on the Cisco MC3810 was removed. This option is no longer supported.

huntstop

To disable all further dial-peer hunting if a call fails when using hunt groups, enter the **huntstop** dial-peer configuration command. To reenable dial-peer call hunting, enter the **no** form of this command.

huntstop
no huntstop

Syntax Description

This command has no arguments or keywords.

Default

Disabled

Command Mode

Dial-peer configuration

Command History

Release	Modification
12.0(5)T	This command was introduced on the Cisco MC3810.
12.0(7)XK	Support for this command was extended to the Cisco 2600 and 3600 series routers.

Usage Guidelines

Once you enter this command, no further hunting is allowed if a call fails on the specified dial peer.

This command can be used with all types of dial peers.

Examples

The following example shows how to disable dial-peer hunting on a specific dial peer:

```
Router(config)# dial peer voice 100 vofr
Router(config-dial-peer)# huntstop
```

The following example shows how to reenable dial-peer hunting on a specific dial peer:

```
Router(config)# dial peer voice 100 vofr
Router(config-dial-peer)# no huntstop
```

Related Commands

Command	Description
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.

signal-type

To set the signaling type to be used when connecting to a dial peer, use the **signal-type** command from dial-peer configuration mode. To return to the default signal-type, use the **no** form of this command.

signal-type {**cas** | **cept** | **ext-signal** | **transparent**}
no signal-type

Syntax Description

cas	North American EIA-464 Channel-Associated Signaling (robbed bit signaling). If the Digital T1 Packet Voice Trunk Network Module is installed, this option may not be available.
cept	Provides a basic E1 ABCD signaling protocol. Used primarily for E&M interfaces. When used with FXS/FXO interfaces, this protocol is equivalent to MELCAS.
ext-signal	External signaling. The DSP does not generate any signaling frames. Use this option when there is an external signaling channel, for example, CCS, or when you need to have a permanent “dumb” voice pipe.
transparent	On the Cisco MC3810, selecting this option produces different results depending on whether you are using a digital voice module (DVM) or an analog voice module (AVM). For a DVM: The ABCD signaling bits are copied from or transported through the T1/E1 interface “transparently” without modification or interpretation. This enables the MC3810 to handle arbitrary or unknown signaling protocols. For an AVM: It is not possible to provide “transparent” behavior because the Cisco MC3810 must interpret the signaling information in order to read and write the correct state to the analog hardware. This option is mapped to be equal to “cas.”

Default

cas

Command Mode

Dial-peer configuration

Command History

Release	Modification
12.0(3)XG	This command was introduced.
12.0(4)T	Support was added for the Cisco 7200 series routers.
12.0(7)XK	In previous releases, the **cept** and **transparent** options were only supported on the Cisco MC3810. Beginning in this release, these options are supported on the Cisco 2600, Cisco 3600 and Cisco 7200 routers.

Usage Guidelines

This command applies to VoFR and VoATM dial peers. It is used with permanent connections only (Cisco trunks and FRF.11 trunks), not with switched calls.

This command is used to inform the local telephony interface of the type of signaling it should expect to receive from the far-end dial peer. To turn signaling off at this dial peer, select the **ext-signal** option. If signaling is turned off and there are no external signaling channels, a “hot” line exists, enabling this dial peer to connect to anything at the far end.

When you connect an FXS to another FXS, or if you have anything other than an FXS/FXO or E&M/E&M pair, the appropriate signaling type on Cisco 2600 series and 3600 series routers is **ext-signal** (disabled).

If you have a digital E1 connection at the remote end that is running cept/MELCAS signaling and you then trunk that across to an analog port, you should make sure that you configure both ends for the **cept** signal-type.

If you have a T1 or E1 connection at both ends and the T1/E1 is running a signaling protocol that is neither EIA-464 or cept/MELCAS, you may want to configure the signal-type for the transparent option in order to pass through the signaling.

Examples

The following example shows how to disable signaling on a Cisco 2600 series or 3600 series router or on an MC3810 concentrator for VoFR dial peer 200, starting from global configuration mode:

```
Router(config)# dial-peer voice 200 vofr
Router(config-dial-peer)# signal-type ext-signal
Router(config-dial-peer)#
```

Related Commands

Command	Description
codec (dial-peer)	Specifies the voice coder rate of speech for a dial peer.
connection	Specifies the connection mode for a voice port.
destination-pattern	Specifies the telephone number associated with a dial peer.
dtmf-relay	Enables the DSP to generate FRF.11 Annex A frames for a dial peer.
preference	Enables the preferred dial peer to be selected when multiple dial peers within a hunt group are matched for a dial string.
session protocol	Establishes the VoFR protocol for calls between local and remote routers.
session target	Specifies a network-specific address for a dial peer.
sequence-numbers	Enables the generation of sequence numbers in each frame generated by the DSP.

vofr

To enable Voice over Frame Relay (VoFR) on a specific DLCI and to configure specific subchannels on that DLCI, use the **vofr** command from Frame Relay DLCI configuration mode. Use the **no** form of the command to disable VoFR on a specific DLCI.

For switched calls:

vofr [data *cid*] **[call-control [***cid***]]**
no vofr [data *cid*] **[call-control [***cid***]]**

For switched calls to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK:

vofr [**cisco**]
no vofr [**cisco**]

For Cisco-trunk permanent calls:

vofr data *cid* **call-control** *cid*
no vofr data *cid* **call-control** *cid*

For Cisco-trunk permanent calls to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK:

vofr cisco
no vofr cisco

For FRF-11 trunk calls:

vofr [data *cid*] **[call-control** *cid***]**
no vofr [data *cid*] **[call-control** *cid***]**

Syntax Description

data	(Required for Cisco-trunk permanent calls. Optional for switched calls.) Used to select a subchannel (CID) for data other than the default subchannel, which is 4.
cid	(Optional) Specifies the subchannel to be used for data. Valid values are from 4 to 255; the default is 4. If **data** is specified, enter a valid CID.
call-control	(Optional) Used to specify that a subchannel will be reserved for call-control signaling. This option is not supported on the Cisco MC3810.
cid	(Optional) Specifies the subchannel to be used for call-control signaling. Valid values are from 4 to 255; the default is 5. If you specify **call-control** and you do not enter a CID, the default CID is used.
cisco	(Optional) Cisco proprietary voice encapsulation for VoFR with data is carried on CID 4 and call-control on CID 5. This option is required when configuring switched calls or Cisco trunks to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK. If configuring switched calls or Cisco trunks to Cisco MC3810 concentrators running Cisco IOS release 12.0(7)XK and later releases, do not use this option.

Default

Disabled

Command Mode

Frame Relay DLCI

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced.
12.0(7)XK	The use of the **cisco** option was modified. Beginning in this release, use the **cisco** option only when configuring connections to Cisco MC3810 concentrators running Cisco IOS releases before 12.0(7)XK.

Usage Guidelines

Table 3 lists the different options of the **vofr** command and which combination of options is used beginning in Cisco IOS Release 12.0(7)XK.

Table 3 Combinations of the vofr Command

Type of Call	vofr Command Combination to Use
Switched call (user dialed or auto-ringdown) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** [*cid*]][1]
Switched call (user dialed or auto-ringdown) to a Cisco MC3810 running Cisco IOS releases before 12.0(7)XK	**vofr cisco**[2]
Cisco-trunk **permanent call** (private-line) to other routers supporting VoFR	**vofr data** *cid* **call-control** *cid*
Cisco-trunk **permanent call** (private-line) to a Cisco MC3810 running Cisco IOS Releases before 12.0(7)XK	**vofr cisco**
FRF.11 trunk call (private-line) to other routers supporting VoFR	**vofr** [**data** *cid*] [**call-control** *cid*][3]

1 The recommended use of this command is **vofr data 4 call-control 5**.
2 This command consumes data CID 4 and call-control CID 5.
3 For FRF.111 trunk calls, the call-control option is not required. It is only required if you mix FRF.11 trunk calls with other types of voice calls on the same PVC.

Usage Restrictions for Cisco IOS Releases Prior to 12.0(7)XK

This section describes restrictions for using the **vofr** command in releases prior to Cisco IOS Release 12.0(7)XK. Beginning in Cisco IOS Release 12.0(7)XK, these restrictions no longer apply.

When you use the **vofr** command without the **cisco** option, all subchannels on the DLCI are configured for FRF.11 encapsulation. If you enter the **vofr** command is entered without any keywords or arguments, the data subchannel is CID 4 and there is no call-control subchannel.

Table 4 describes special conditions and restrictions for the use of the **vofr** command on the Cisco MC3810.

Table 4 Using the vofr Command with the Cisco MC3810

Type of Call	Conditions and Restrictions
FRF.11 trunks	1. Do NOT use **cisco** option or **call-control** option. 2. Use **vofr** or **vofr data** *cid*.
Cisco trunks	1. Must use **vofr cisco**.
switched-vofr	1. Must use **vofr cisco**.

If you select the “data” option, enter a numeric value to complete the command. If you select the “call-control” option, you do not enter a numeric value if you wish to accept the default call-control subchannel. See the following examples for clarification.

When you use the **vofr** command on a Cisco MC3810 without the “cisco” option, switched calls are not permitted. You can only make permanent FRF.11-trunk calls.

Note It is not possible to configure the **call-control** option on a Cisco MC3810. If you configure this option, the setting is ignored.

Examples

The following example shows how to enable VoFR on Serial 1/1, DLCI 100 on a Cisco 2600 series, 3600 series, or 7200 series router or on an MC3810 concentrator, starting from global configuration mode:

```
Router(config)# interface serial 1/1
Router(config-if)# frame-relay interface-dlci 100
Router(config-fr-dlci)# vofr
Router(config-fr-dlci)#
```

The above example configures CID 4 for data; no call-control CID is defined.

To configure CID 4 for data and CID 5 for call-control (both defaults), enter the following command:

```
Router(config-fr-dlci)# vofr call-control
Router(config-fr-dlci)#
```

To configure CID10 for data and CID 15 for call-control, enter the following command:

```
Router(config-fr-dlci)# vofr data 10 call-control 15
Router(config-fr-dlci)#
```

To configure CID 4 for data and CID 15 for call-control, enter the following command:

```
Router(config-fr-dlci)# vofr call-control 15
Router(config-fr-dlci)#
```

To configure CID 10 for data and CID 5 for call-control, enter the following command:

```
Router(config-fr-dlci)# vofr data 10 call-control
Router(config-fr-dlci)#
```

To configure CID 10 for data with no call-control, enter the following command:

```
Router(config-fr-dlci)# vofr data 10
Router(config-fr-dlci)#
```

To configure a Cisco router or MC3810 for a VoFR application with an older release of the MC3810 (before Release 12.0(3)XG), enter the following command:

```
Router(config-fr-dlci)# vofr cisco
Router(config-fr-dlci)#
```

Related Commands

Command	Description
frame-relay interface-dlci	Assigns a data link connection identifier (DLCI) to a specified Frame Relay subinterface.
class	Assigns a VC class to a PVC.

Debug Commands

This section provides information on new and modified VoFR debug commands for the Cisco 2600 series, 3600 series, and the Cisco MC3810 multiservice access concentrator.

All other debug commands used with Voice over Frame Relay are documented in the Cisco IOS Release 12.0 command references.

The following new and modified commands are described in this section:

- **debug ccfrf11 session**
- **debug ccswvoice vofr-debug**
- **debug ccswvoice vofr-session**
- **debug vtsp vofr subframe**

debug ccfrf11 session

To display the ccfrf11 function calls during call setup and teardown, use the **debug ccfrf11 session** command from privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug ccfrf11 session
no debug ccfrf11 session

Syntax Description

This command has no keywords or arguments.

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810.

Usage Guidelines

Use this command to display debug information about the various FRF.11 VoFR service provider interface (SPI) functions. Note that this debug command does not display any information regarding the proprietary Cisco switched-VoFR SPI.

This debug is only useful when the session protocol is “frf11-trunk.”

Examples

The following example shows sample output from the **debug ccfr11 session** command:

```
Router# debug ccfrf11 session
INCOMING CALL SETUP (port setup for answer-mode):
*Mar  6 18:04:07.693:ccfrf11_process_timers:scb (0x60EB6040) timer (0x60EB6098) expired
*Mar  6 18:04:07.693:Setting accept_incoming to TRUE
*Mar  6 18:04:11.213:ccfrf11_incoming_request:peer tag 800:callingNumber=+2602100,
        calledNumber=+3622110
*Mar  6 18:04:11.213:ccfrf11_initialize_ccb:preffered_codec set(-1)(0)
*Mar  6 18:04:11.213:ccfrf11_evhandle_incoming_call_setup_request:calling +2602100,
        called +3622110 Incoming Tag 800
*Mar  6 18:04:11.217:ccfrf11_caps_ind:PeerTag = 800
*Mar  6 18:04:11.217:     codec(preferred) = 4, fax_rate = 2, vad = 2
*Mar  6 18:04:11.217:     cid = 30, config_bitmask = 0, codec_bytes = 20, signal_type=2
*Mar  6 18:04:11.217:     required_bandwidth 8192
*Mar  6 18:04:11.217:ccfrf11_caps_ind:Bandwidth reservation of 8192 bytes succeeded.
*Mar  6 18:04:11.221:ccfrf11_evhandle_call_connect:Entered
```

```
CALL SETUP (MASTER):
5d22h:ccfrf11_call_setup_request:Entered
5d22h:ccfrf11_evhandle_call_setup_request:Entered
5d22h:ccfrf11_initialize_ccb:preffered_codec set(-1)(0)
5d22h:ccfrf11_evhandle_call_setup_request:preffered_codec set(9)(24)
5d22h:ccfrf11_call_setup_trunk:subchannel linking successful
5d22h:ccfrf11_caps_ind:PeerTag = 810
5d22h:     codec(preferred) = 512, fax_rate = 2, vad = 2
5d22h:     cid = 30, config_bitmask = 1, codec_bytes = 24, signal_type=2
5d22h:     required_bandwidth 6500
5d22h:ccfrf11_caps_ind:Bandwidth reservation of 6500 bytes succeeded.

CALL TEARDOWN:
*Mar  6 18:09:14.805:ccfrf11_call_disconnect:peer tag 0
*Mar  6 18:09:14.805:ccfrf11_evhandle_call_disconnect:Entered
*Mar  6 18:09:14.805:ccfrf11_call_cleanup:freeccb 1, call_disconnected 1
*Mar  6 18:09:14.805:ccfrf11_call_cleanup:Setting accept_incoming to FALSE and starting
        incoming timer
*Mar  6 18:09:14.809:timer 2:(0x60EB6098)starts - delay (70000)
*Mar  6 18:09:14.809:ccfrf11_call_cleanup:Alive timer stopped
*Mar  6 18:09:14.809:timer 1:(0x60F64104) stops
*Mar  6 18:09:14.809:ccfrf11_call_cleanup:Generating Call record
*Mar  6 18:09:14.809:cause=10 tcause=10    cause_text="normal call clearing."
*Mar  6 18:09:14.809:ccfrf11_call_cleanup:Releasing 8192 bytes of reserved bandwidth
*Mar  6 18:09:14.809:ccfrf11_call_cleanup:ccb 0x60F6404C, vdbPtr 0x610DB7A4
        freeccb_flag=1, call_disconnected_flag=1
```

Related Commands

Command	Description
debug ccswvoice vofr-debug	Displays the ccswvoice function calls during call setup and teardown.
debug ccswvoice vofr-session	Displays the ccswvoice function calls during call setup and teardown.
debug vtsp vofr subframe	Displays the first 10 bytes (including header) of selected VoFR subframes for the interface.

debug ccswvoice vofr-debug

To display the ccswvoice function calls during call setup and teardown, use the **debug ccswvoice vofr-debug** command from privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug ccswvoice vofr-debug
no debug ccswvoice vofr-debug

Syntax Description

This command has no arguments or keywords.

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810.

Usage Guidelines

Use this command when troubleshooting a VoFR call that uses the "cisco-switched" session protocol. This command provides the same information as the **debug ccswvoice vofr-session** command, but includes additional debugging information relating to the calls.

Examples

The following example shows sample output from the **debug ccswvoice vofr-debug** command:

```
Router# debug ccswvoice vofr-debug
CALL TEARDOWN:
3640_vofr(config-voiceport)#
*Mar  1 03:02:08.719:ccswvofr_bridge_drop:dropping bridge calls src 17 dst 16 dlci 100
      cid 9 state ACTIVE
*Mar  1 03:02:08.727:ccswvofr:callID 17 dlci 100 cid 9 state ACTIVE event O/G REL
*Mar  1 03:02:08.735:ccswvofr:callID 17 dlci 100 cid 9 state RELEASE event I/C RELCOMP
*Mar  1 03:02:08.735:ccswvofr_store_call_history_entry:cause=22 tcause=22
      cause_text=no circuit.
3640_vofr(config-voiceport)#

CALL SETUP (outgoing):
*Mar  1 03:03:22.651:ccswvofr:callID 23 dlci -1 cid -1 state NULL event O/G SETUP
*Mar  1 03:03:22.651:ccswvofr_out_callinit_setup:callID 23 using dlci 100 cid 10
*Mar  1 03:03:22.659:ccswvofr:callID 23 dlci 100 cid 10 state O/G INIT event I/C PROC
*Mar  1 03:03:22.667:ccswvofr:callID 23 dlci 100 cid 10 state O/G PROC event I/C CONN
ccfrf11_caps_ind:codec(preferred) = 0
```

Related Commands

Command	Description
debug ccfrf11 session	Displays the ccfrf11 function calls during call setup and teardown.
debug ccswvoice vofr-session	Displays the ccswvoice function calls during call setup and teardown.
debug vtsp vofr subframe	Displays the first 10 bytes (including header) of selected VoFR subframes for the interface.

debug ccswvoice vofr-session

To display the ccswvoice function calls during call setup and teardown, use the **debug ccswvoice vofr-session** command from privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug ccswvoice vofr-session
no debug ccswvoice vofr-session

Syntax Description

This command has no arguments or keywords.

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(3)XG and 12.0(4)T	This command was introduced on the Cisco 2600 and Cisco 3600 series routers.
12.0(7)XK	This command was first supported on the Cisco MC3810.

Usage Guidelines

Use this command to show the state transitions of the cisco-switched-vofr state machine as a call is processed, and when attempting to troubleshoot a VoFR call that uses the "cisco-switched" session protocol.

Examples

The following example shows sample output from the **debug ccswvoice vofr-session** command:

```
Router# debug ccswvoice vofr-session
CALL TEARDOWN:
3640_vofr(config-voiceport)#
*Mar  1 02:58:13.203:ccswvofr:callID 14 dlci 100 cid 8 state ACTIVE event O/G REL
*Mar  1 02:58:13.215:ccswvofr:callID 14 dlci 100 cid 8 state RELEASE event I/C RELCOMP
3640_vofr(config-voiceport)#

CALL SETUP (outgoing):
*Mar  1 02:59:46.551:ccswvofr:callID 17 dlci -1 cid -1 state NULL event O/G SETUP
*Mar  1 02:59:46.559:ccswvofr:callID 17 dlci 100 cid 9 state O/G INIT event I/C PROC
*Mar  1 02:59:46.567:ccswvofr:callID 17 dlci 100 cid 9 state O/G PROC event I/C CONN
3640_vofr(config-voiceport)#
```

Related Commands

Command	Description
debug ccfrf11 session	Displays the ccfrf11 function calls during call setup and teardown.
debug ccswvoice vofr-debug	Displays the ccswvoice function calls during call setup and teardown.
debug vtsp vofr subframe	Displays the first 10 bytes (including header) of selected VoFR subframes for the interface.

debug vtsp vofr subframe

To display the first 10 bytes (including header) of selected VoFR subframes for the interface, use the **debug vtsp vofr subframe** command in privileged EXEC mode. Use the **no** form of this command to turn off the debug function.

debug vtsp vofr subframe *payload* **[from-dsp] [to-dsp]**
no debug vtsp vofr subframe

Syntax Description

payload	Number used to selectively display subframes of a specific payload. The payload types are: 0: Primary Payload - WARNING! This option may cause network instability. 1: Annex-A 2: Annex-B 3: Annex-D 4: All other payloads 5: All payloads - WARNING! This option may cause network instability.
from-dsp	(Optional) Displays only the subframes received from the DSP.
to-dsp	(Optional) Displays only the subframes going to the DSP.

Command Mode

Privileged EXEC

Command History

Release	Modification
12.0(3)XG	This command was introduced on the Cisco 2600 and 3600.
12.0(7)XK	Support for this command was extended to the Cisco MC3810.

Usage Guidelines

Each debug output displays the first 10 bytes of the FRF.11 subframe, including header bytes. Use the **from-dsp** and **to-dsp** options to limit the debugs to a single direction. If not specified, debugs are displayed for subframes when they are received from the DSP and before they are sent to the DSP.

Use extreme caution in selecting payload options 0 and 5. These options may cause network instability.

Examples

The following example shows sample output from the **debug vtsp vofr subframe** command:

```
Router# debug vtsp vofr subframe 2
vtsp VoFR subframe debugging is enabled for payload 2 to and from DSP 3620_vofr#
*Mar  6 18:21:17.413:VoFR frame received from Network (24 bytes):9E 02 19 AA AA AA AA
      AA AA AA
*Mar  6 18:21:17.449:VoFR frame received from DSP (18 bytes):9E 02 19 AA AA AA AA AA AA
      AA
*Mar  6 18:21:23.969:VoFR frame received from Network (24 bytes):9E 02 19 AA AA AA AA
      AA AA AA
*Mar  6 18:21:24.005:VoFR frame received from DSP (18 bytes):9E 02 19 AA AA AA AA AA AA
      AA
```

Related Commands

Command	Description
debug vtsp all	Enables debugging of all VTSP areas.

Glossary

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.

CID—Channel ID. Designates the Frame Relay subchannel ID for Voice over Frame Relay.

CIR—Committed Information Rate. The average rate of information transfer a subscriber (for example, a network administrator) has stipulated for a Frame Relay PVC.

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.

DLCI—Data-link connection identifier.

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 four kinds of dial peers: POTS, VoIP, VoFR, and VoATM.

DS0—A 64-kbps B channel on an E1 or T1 WAN interface.

DTMF—Dual tone multifrequency. 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 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.

FIFO—First-in, first-out. In data communication, FIFO refers to a buffering scheme where the first byte of data entering the buffer is the first byte retrieved by the CPU. In telephony, FIFO refers to a queuing scheme where the first calls received are the first calls processed.

FRF—Frame Relay Forum. An association of corporate members consisting of vendors, carriers, users, and consultants committed to the implementation of Frame Relay in accordance with national and international standards. Go to 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. Go to 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 smaller pieces and interleaved with real-time frames. In this way, real-time voice and non real-time data frames can be carried together on lower speed links without causing excessive delay to the real-time traffic. Go to http://www.frforum.com.

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.

hookflash—A short on-hook period usually generated by a telephone-like device during a call to
MEL CAS—Mercury Exchange Limited (MEL) Channel Associated Signaling. A voice signaling protocol used primarily in the United Kingdom.

PBX—Private Branch Exchange. 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).

PLAR—Private Line, Automatic Ringdown. A leased voice circuit that connects two single endpoints together. When either telephone handset is taken off-hook, the remote telephone automatically rings.

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.

PVC—Permanent virtual circuit.

SVC—Switched virtual circuit.

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 E&M to E&M fixed point-to-point connections.

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