
Configuration Tasks

This section describes how to configure VoFR, including the Cisco implementations for FRF.11 and FRF.12. The following major tasks are covered and are divided into the following sections:

- Preliminary Frame Relay Configuration for Voice on page 12
 - Configuring a Map Class to Support Voice over Frame Relay on page 12
 - Verifying Your Frame Relay Configuration on page 15
- Preparing to Configure Voice Dial Peers on page 16
 - Organize Voice Network Information on page 16
 - Create a Peer Configuration Table on page 16
- Configuring Dial Peers on page 17
 - Configuring POTS Dial Peers on page 20
 - Configuring Voice over Frame Relay Dial Peers on page 22
- Configuring Voice over Frame Relay Connections on page 29
 - Overview of Voice over Frame Relay Connection Types on page 30
 - Configuring Switched Calls (User Dialed or Auto-Ringdown) on page 33
 - Configuring Cisco-Trunk Permanent (Private Line) Calls on page 37
 - Configuring Connections for Tandem Nodes on page 42

This section specifically describes the commands to configure VoFR applications. It is assumed you have already configured your Frame Relay backbone network, including the map class and the LMI. For more information about Frame Relay configuration, see the *Wide-Area Networking Configuration Guide*.

Note The Cisco MC3810 first supported VoFR in Cisco IOS Release 11.3(1)MA. The other platforms that now support VoFR with this release (Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series) can act as tandem nodes with a Cisco MC3810 running Cisco IOS Release 11.3(1)MA or later, but cannot terminate calls initiated by a Cisco MC3810 using VoFR implementations prior to Cisco IOS Release 12.0(3)XG or 12.0(4)T. In addition, Cisco MC3810 concentrators running Cisco IOS versions prior to release 12.0(3)XG or 12.0(4)T cannot tandem calls from Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series routers.

Preliminary Frame Relay Configuration for Voice

This section describes preliminary Frame Relay configuration tasks that are necessary to support VoFR:

- Configuring a Map Class to Support Voice over Frame Relay on page 12
- Verifying Your Frame Relay Configuration on page 15
- Verifying Your Frame Relay Configuration on page 15

Configuring a Map Class to Support Voice over Frame Relay

Before configuring a Frame Relay DLCI for voice traffic, you must create a Frame Relay map class and configure it to support voice traffic. Configuring a Frame Relay map class is required because the voice bandwidth, fragmentation size, and traffic shaping attributes are configured on the map class. These attributes are required for sending voice traffic on the PVC.

This section is divided into the following procedures:

- Configure a Frame Relay Map Class to Support Voice Traffic on page 13
- Configure a Frame Relay Map Class for Traffic Shaping Parameters on page 14
- Configure a Frame Relay Map Class to Support FRF.12 Fragmentation on page 14

A map class applies to a single DLCI or to a group of DLCIs, depending on how the class has been applied to the virtual circuit. If you have a large number of PVCs to configure, you can assign the PVCs the same traffic shaping properties without statically defining the values for each PVC. You can create multiple map classes with different variables for each map class.

Note If configuring the **frame-relay interface-dlci voice-encap** command on a Cisco MC3810, then configuring a Frame Relay map class is not required. For procedures for assigning this command to a DLCI in conjunction with configuring connection types, see the “Configuring Switched Calls on a Cisco MC3810” section on page 34 and the “Configuring Cisco Trunk Permanent Calls on a Cisco MC3810” section on page 39.

Configure a Frame Relay Map Class to Support Voice Traffic

To configure a Frame Relay map class to support voice traffic on a single DLCI or a group of DLCIs, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	<code>router(config)# map-class frame-relay map-class-name</code>	Create a map-class name you will assign to a group of PVCs. The map-class name must be unique.
2	<code>router(config-map-class)# frame-relay voice bandwidth bps reserved</code>	<p>Enter the bandwidth in bits per second, which will determine the number of voice calls allowed on the DLCIs where this map class is associated to. Cisco recommends that this value be set to no higher than the minimum CIR if you do not want to impact voice quality when burst is being transmitted. The valid range is from 8000 to 45,000,000 bps.</p> <p>This command must be configured for voice calls to take place. The default for this command is 0, which disables all voice calls.</p> <p>For more information on determining the amount of voice bandwidth to set, see the section “Configuring Voice Bandwidth” after this procedure.</p> <p>Note This command does not apply if configuring the frame-relay interface-dlci voice-encap command on the Cisco MC3810.</p>

To configure the map class to support FRF.12 fragmentation, see the “Configure a Frame Relay Map Class to Support FRF.12 Fragmentation” section on page 14. To configure the map class to support traffic shaping if you want to send both voice traffic and data traffic on the same PVC, see the “Configure a Frame Relay Map Class for Traffic Shaping Parameters” section on page 14.

Calculating Voice Bandwidth

The **frame-relay voice-bandwidth** map-class command is used to configure how much bandwidth is reserved for voice traffic. If there is not enough reserved voice bandwidth remaining on the PVC, then any new call attempted will be rejected.

When considering the amount of voice bandwidth to allocate to voice, the overall bandwidth calculation must include the voice packetization overhead and not just the raw compressed speech CODEC bandwidth. For VoFR voice packets, there are a total of 6 or 7 bytes total overhead per packet (including standard Frame Relay headers and flags). For subchannels (CIDs) less than number 64, the overhead is 6 bytes. For subchannels greater than or equal to number 64, the overhead is 7 bytes. Add one additional byte if voice sequence numbers are enabled in the voice packets.

To determine the required voice bandwidth, use the following calculation:

$$\text{required_bandwidth} = \text{codec_bandwidth} * (1 + \text{overhead/payload_size})$$

This calculation addresses the amount of bandwidth consumed on the physical network interface. This does not necessarily represent the amount of connection bandwidth used within the Frame Relay network itself, which may be higher due to the overhead of switching small packets.

When using 30-millisecond duration voice packets, an approximate rule-of-thumb is to add 2000 bps overhead to the raw voice compressed speech CODEC rate. With the 32 kbps G.726 ADPCM speech coder, a 30-millisecond speech frame uses 120 bytes voice payload plus 6-7 bytes overhead, and the overall bandwidth requirement is around 34 kbps for each call.

The **codec** command is configured as part of the dial peer configuration procedures in the “Configuring Dial Peers” section on page 17.

Configure a Frame Relay Map Class to Support FRF.12 Fragmentation

To configure the map class to support FRF.12 fragmentation, use the following commands in map-class configuration mode:

Step	Command	Purpose
1	<pre>router(config-map-class)# frame-relay fragment fragment_size</pre>	<p>Configure Frame Relay fragmentation for the map class. The <i>fragment_size</i> defines the payload size of a fragment, and excludes the Frame Relay headers and any Frame Relay fragmentation header. The valid range is from 16 to 1600 bytes, and the default is 53.</p> <p>The <i>fragment_size</i> should be less than or equal to the MTU size.</p> <p>Set the fragmentation size such that the largest data packet is not larger than the voice packets.</p>
2	<pre>router(config-map-class)# frame-relay fair-queue [Congestive_Discard_Threshold] [Number_Dynamic_Conversation_Queues] [Number_Reservable_Conversation_Queues]</pre>	<p>Enable weighted fair queuing for the map class.</p> <p>This command applies to both fragmented and nonfragmented data. This command is equivalent to the fair-queue interface command but applies to a Frame Relay PVC.</p> <p>The default <i>Congestive_Discard_Threshold</i> is 64. The default <i>Number_Dynamic_Conversation_Queues</i> is 16. The default <i>Number_Reservable_Conversation_Queues</i> is 2. The default for <i>Max_Buffer_Size_for_Fair_Queues</i> is 600.</p>

Note When Frame Relay fragmentation is configured, weighted fair queueing is mandatory. If a map class is configured for Frame Relay fragmentation and the queueing type on that map class is not fair-queue, the configured queueing type is automatically overridden by WFQ with the default values.

To configure the map class to support traffic shaping if you want to send both voice traffic and data traffic on the same PVC, see the next section, “Configure a Frame Relay Map Class for Traffic Shaping Parameters.”

Configure a Frame Relay Map Class for Traffic Shaping Parameters

When you configure a Frame Relay PVC to support voice traffic, you must ensure that the carrier can accommodate the traffic rate or profile transmitted on the PVC. If too much traffic is sent at once, the carrier might discard frames, which causes disruptions to real-time voice traffic. The carrier might also deal with traffic bursts by queuing up the bursts and delivering them at a metered rate. Excessive queuing also causes disruption to real-time voice traffic.

To compensate for this condition, traffic shaping is required if sending both voice traffic and data traffic over the same PVC.

Note When you configure the outgoing Excess Burst size, the Committed Burst size, and the committed information rate (CIR) values, obtain the appropriate values from your carrier. The values configured on the router must match those of the carrier. Traffic shaping is necessary to prevent your carrier from discarding Discard Eligible (DE) bits on ingress or to prevent excessive burst data from affecting voice quality.

To configure a Frame Relay map class to determine the traffic shaping characteristics for one or more DLCIs, use the following commands in map-class configuration mode:

Step	Command	Purpose
1	<code>router(config-map-class)# frame-relay bc out bits</code>	Configure the outgoing Committed Burst size for this group of PVCs. Configure the <i>bits</i> value to a minimum of 1000 for voice traffic. Configure the Committed Burst size to match your carrier to prevent the carrier from discarding DE bits on ingress. This command is required.
2	<code>router(config-map-class)# frame-relay be out bits</code>	Configure the outgoing Excess Burst size for this group of PVCs. Configure the outgoing Excess Burst size to match your carrier to prevent the carrier from discarding discard eligible (DE) bits on ingress. This command is required.
3	<code>router(config-map-class)# frame-relay min-cir {in out} bps</code>	Configure the minimum acceptable incoming or outgoing committed information rate (CIR) for this group of PVCs.
4	<code>router(config-map-class)# frame-relay cir out bits</code>	Configure the outgoing excess CIR for this group of PVCs. Configure the CIR size to match your carrier to prevent the carrier from discarding DE bits on ingress. This command is required.
5	<code>router(config-map-class)# frame-relay cir in bits</code>	(Optional) Configure the incoming CIR size for this group of PVCs.
6	<code>router(config-map-class)# frame-relay adaptive shaping becn</code>	(Optional) Configure the adaptive traffic rate adjustment to support backward explicit congestion notification (BECN) on this group of PVCs.

If your Frame Relay map-class configuration is complete, see the “Verifying Your Frame Relay Configuration” section on page 15. To begin your dial peer configuration, see either the “Preparing to Configure Voice Dial Peers” section on page 16, or the “Configuring Dial Peers” section on page 17.

Verifying Your Frame Relay Configuration

You can check the validity of your Frame Relay configuration by performing the following tasks:

- To show the status of your PVCs, use the **show frame-relay pvc** command.
- To show statistics and information on the open sub-channels, use the **show frame-relay vofr** `[[interface] [dlci[cid]]` command. (This command does not display if the **vofr cisco** command is entered on the Cisco MC3810.)
- To show the Frame Relay fragmentation configuration, use the **show frame-relay fragment** `[interface number [dlci]]` command.
- If you are using Frame Relay traffic shaping, use the **show traffic-shape queue** command to display the traffic shaping information. Use the `queue` option to display the queuing statistics.

Preparing to Configure Voice Dial Peers

After you have analyzed your dial plan and decided how to integrate it into your existing network, you are ready to configure your network devices to support VoFR. The actual configuration procedure depends entirely upon the topology of your voice network, but, in general, you need to perform the following tasks:

- Organize Voice Network Information
- Create a Peer Configuration Table



Timesaver If possible, you might want to configure the Frame Relay dial peers in a back-to-back configuration before separating them across the Frame Relay network. You can use a back-to-back configuration to test your VoFR and dial-peer configuration to determine if you can make a voice connection. Then, when you place both peers on the network, failure to make a voice connection will isolate the cause as a network problem.

Organize Voice Network Information

After you have configured your Frame Relay network, you should collect all of the data directly related to each dial peer by creating a peer configuration table to prepare for configuring VoFR.

Create a Peer Configuration Table

Specific information relative to each dial peer needs to be identified before you can configure VoFR. One way to identify this information is to create a peer configuration table.

Figure 1 shows a diagram of a small voice network in which router No. 1 connects a small sales branch office to the main office through router No. 2. Only two devices in the sales branch office that need to be established as dial peers: a telephone and a fax machine. Router No. 2 is the primary gateway to the main office; as such, it needs to be connected to the company's PBX. Two telephones and one fax machine connected to the PBX need to be established as dial peers in the main office.

Table 2 shows the peer configuration table for the example illustrated in Figure 1.

Figure 1 Sample VoFR Network

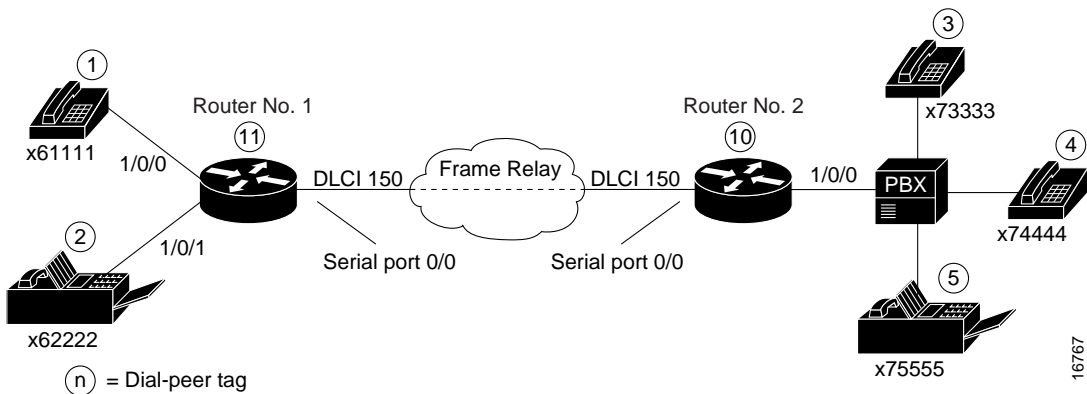


Table 2 Peer Configuration Table for Sample VoFR Network

Dial Peer	Extension	Prefix	Destination Pattern	Type	Voice Port	Session Target
Router No. 1						
1	61111		13107661111	POTS	1/0/0	
2	62222		13107662222	POTS	1/0/1	
10			1310767....	VOFR		S0/0 150
Router No. 2						
11			1310766....	VOFR		S0/0 150
3	73333	7	1310767....	POTS	1/0/0	
4	74444	7	1310767....	POTS	1/0/0	
5	75555	7	1310767....	POTS	1/1	

The dial plan shown in Table 2 lists a simple dial-peer configuration table. No configuration for forwarding digits to a PBX is shown. Additional configuration for this is required.

Configuring Dial Peers

Dial peers describe the entities to and from which a call is established. Dial-peer configuration tasks define the address or set of addresses serviced by that dial peer and the call parameters required to establish a call to and from that dial peer.

Two different kinds of dial peers pertain to VoFR configurations:

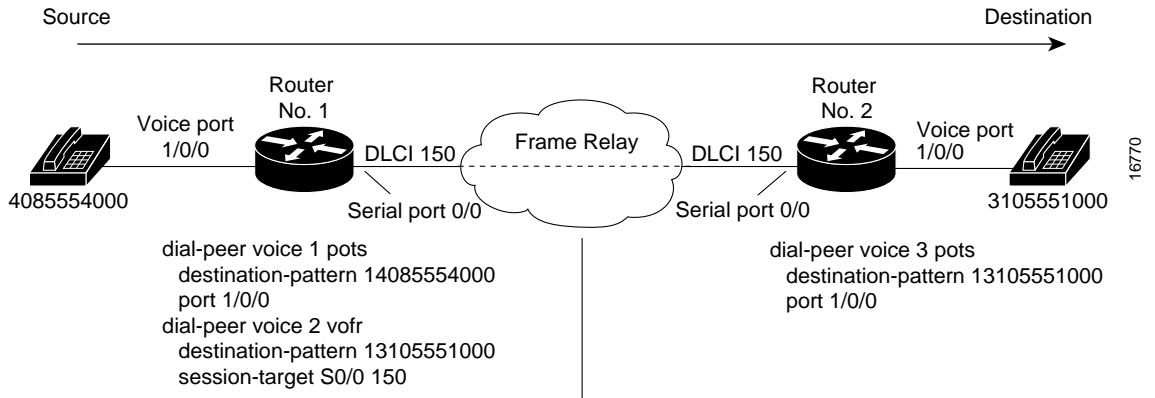
- POTS—Dial peer connected via a traditional telephony network. Voice-telephony peers point to a particular voice port on a voice-network device.
- VoFR—Dial peer connected via a Frame Relay WAN backbone. VoFR dial peers point to specific voice-network devices. For Voice over Frame Relay, you can configure VoFR dial peers for the following call types:
 - Switched calls
 - Cisco-trunk permanent calls
 - FRF.11 trunk calls

POTS peers associate a telephone number with a particular voice port so that incoming calls for that port can be received. VoFR peers point to specific voice-network devices (by associating destination telephone numbers with a specific Frame Relay DLCI) so that outgoing calls can be placed. Both POTS and VoFR dial peers are needed to establish VoFR connections if you want both to send and receive calls.

For tandem voice nodes, POTS dial peers are not configured.

Establishing two-way communication using VoFR requires establishing a specific voice connection between two defined endpoints. As shown in Figure 2, for outgoing calls (from the perspective of the voice-telephony dial peer 1), the voice-telephony dial peer establishes the source (the originating telephone number and voice port) of the call. The voice-network dial peer establishes the destination by associating the destination telephone number with a specific Frame Relay DLCI.

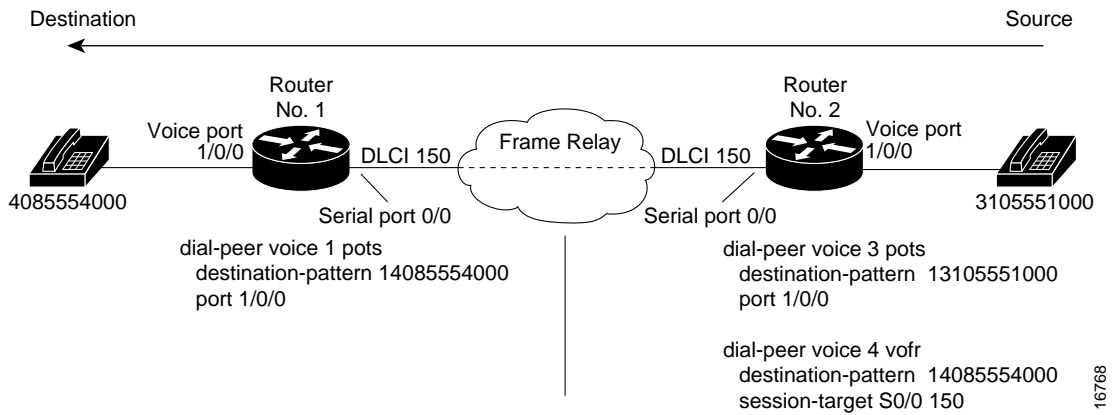
Figure 2 Calls from the Perspective of Router No.1



In the example, the destination pattern 14085554000 string maps to a U.S. telephone number 555-4000, with the digit 1 plus the area code (408) preceding the number. When you configure the destination pattern, set the dial string to match the local dial conventions.

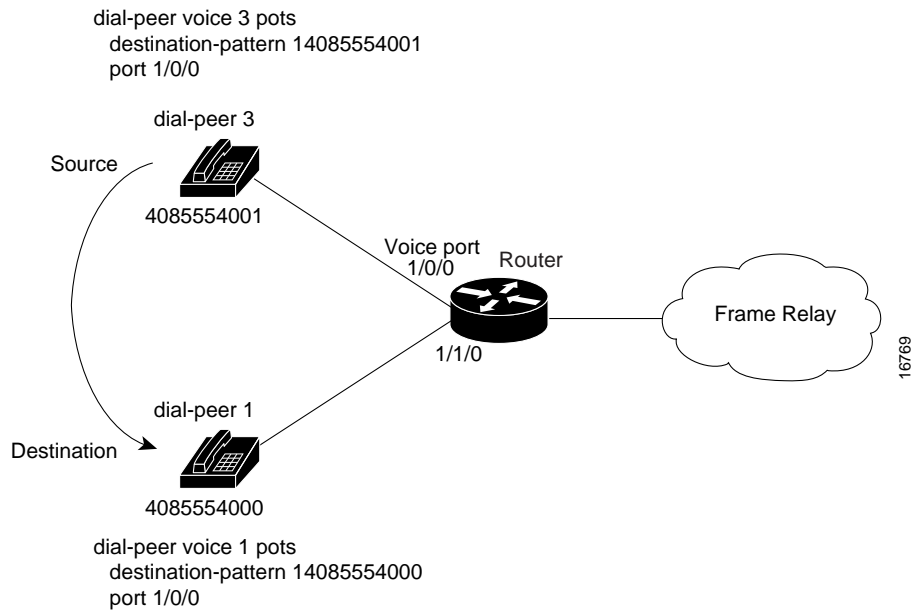
To complete the two-way communications loop, you need to configure VoFR dial peer 4 as shown in Figure 3.

Figure 3 Calls from the Perspective of Router No. 2



The only exception to this example is when both POTS dial peers are connected to the same router, as shown in Figure 4. In this circumstance, you would not need to configure a VoFR dial peer. Figure 4 shows an example for switched calls only.

Figure 4 Communication between Dial Peers Sharing the Same Router



When you configure dial peers, ensure that you understand the relationship between the destination pattern and the session target. The destination pattern is the telephone number of the voice device attached to the voice port. The session target represents the route to a serial port on the peer router at the other end of the Frame Relay connection. Figure 5 and Figure 6 show the relationship between the destination pattern and the session target, as seen from the perspective of both routers in a VoFR configuration. These examples show a topology for switched calls only.

Figure 5 Relationship between Destination Pattern and Session Target from the Perspective of Router No. 1

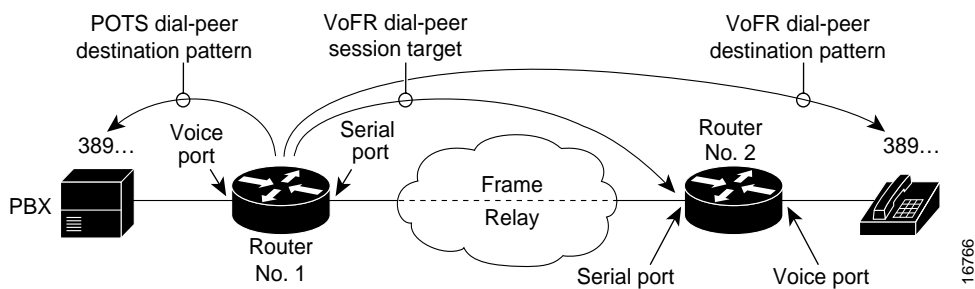
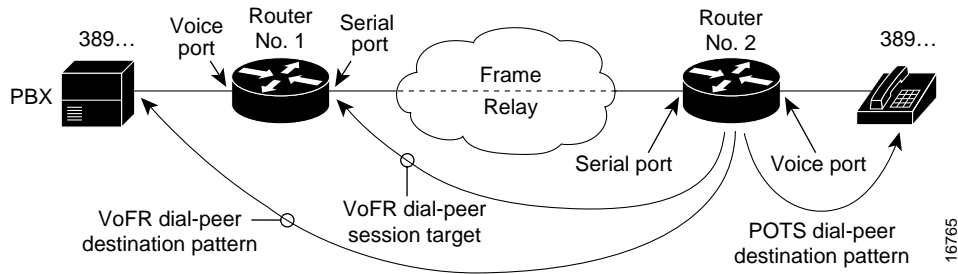


Figure 6 Relationship between Destination Pattern and Session Target from the Perspective of Router No. 2



The following sections describe how to configure POTS peers and VoFR peers.

Configuring POTS Dial Peers

To configure a POTS dial peer, you need to uniquely identify the peer (by assigning it a unique tag number), define its telephone number, and associate it with a voice port through which calls will be established. Under most circumstances, the default values for the remaining dial-peer configuration commands are sufficient to establish connections.

Depending on your dial plan configuration, you might need to consider how to configure voice networks with variable-length dial plans, number expansion, excess digit ployout, forward digits, and default voice routes, or use hunt groups with dial-peer preferences.

Note The Cisco 7200 series routers only perform VoFR tandeming. As a result, you cannot configure POTS dial peers on the 7200 series routers.

To configure POTS dial peers, use the following commands from global configuration mode:

Step	Command	Purpose
1	<code>router(config)# dial-peer voice <i>number</i> pots</code>	Define a POTS 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 <i>number</i> value tag identifies the dial peer and must be unique on the router. Do not duplicate a specific tag number.

Step	Command	Purpose
2	<code>router(config-dialpeer)# destination-pattern [+]string[t]</code>	<p>Configure the dial peer's destination pattern.</p> <p>The <i>string</i> is a series of digits that specify the E.164 or private dialing plan telephone number. Valid entries are the digits 0–9 and the letters A–D. The following special characters can be entered in the string:</p> <ul style="list-style-type: none"> • The plus symbol (+) can be used to indicate an E.164 standard number. On the Cisco MC3810, the plus symbol (+) is not a valid character in the string. • The star character (*) and the pound sign (#) that appear on standard touch-tone dial pads can be used in any dial string. However, these characters cannot be used as leading characters in a string (for example, *650). • The period (.) can be used as a trailing character, and is used as a wildcard character. Multiple periods as trailing characters indicate multiple wildcard digits, such as for the 789... wildcard. • The comma (,) can be used only in prefixes, and is used to insert a one-second pause or a delay. • The timer (T) character can be used to configure variable length dial plans.
3	<p>Choose one of the following:</p> <p><code>router(config-dialpeer)# port slot/subunit/port</code></p> <p><code>router(config-dialpeer)# port slot/port</code></p>	<p>(Cisco 2600 series and 3600 series) Associate this voice-telephony dial peer with a specific voice port.</p> <p>(Cisco MC3810) Associate this voice-telephony dial peer with a specific voice port.</p>

Configuring Dial Plan Options for the POTS Dial Peer

When you configure the dial plan, you have different options for how the dial plan is designed.

To configure dial plan options, use the following commands in dial-peer configuration mode:

Step	Command	Purpose
1	<code>router(config-dialpeer)# preference value</code>	(Optional) Configure a preference for the POTS dial peer. The value is a number from 0–10 where the lower the number, the higher the preference. If POTS and voice-network (VoFR) peers are mixed in the same hunt group, POTS dial peers will be searched first, even if a voice-network peer has a higher preference number.
2	<code>router(config-dialpeer)# prefix string</code>	<p>(Optional) Assign the prefix of the dialed digits for the dial peer. Valid numbers for the <i>string</i> are 0 through 9, and a comma (.). Use a comma to include a pause in the prefix</p> <p>When an outgoing call is initiated to this dial peer, the prefix string value is sent to the telephony interface first, before the telephone number associated with the dial peer.</p>
3	<code>router(config-dialpeer)# direct-inward-dial</code>	(Optional for the Cisco 2600/3600 series only) Enable the Direct Inward Dial (DID) call treatment for the incoming called number.

Step	Command	Purpose
4	<code>router(config-dialpeer)# incoming called-number string</code>	(Optional for the Cisco 2600/3600 series only) Identify the service type for a call on a router handling both voice and modem calls.
5	<code>router(config-dialpeer)# maximum-connections number</code>	(Optional for the Cisco 2600/3600 series only) Specify the maximum number of allowed connections to and from the POTS dial peer. The valid range is 1–2147483647.
6	<code>router(config-dialpeer)# forward-digits {num-digit all implicit}</code>	(Optional for the Cisco MC3810 only) If you are using the forward-digits feature, configure the digit-forwarding method that will be used on the dial peer. The valid range for the number of digits forwarded (<i>num-digit</i>) is 0–32. The default value is implicit , in which the exactly matched digits are not forwarded. Only digits matched by the wildcard pattern are forwarded.

To configure the next POTS dial peer, exit dial-peer configuration mode by entering **exit**, and repeat the previous steps. To configure VoFR dial peers, see the next section “Configuring Voice over Frame Relay Dial Peers.”

Configuring Voice over Frame Relay Dial Peers

To configure a VoFR dial peer, you need to uniquely identify the peer (by assigning it a unique tag number) and define the outgoing serial port number and the virtual circuit number.

Depending on your dial plan configuration, you might need to consider how to configure voice networks with variable-length dial plans, number expansion, excess digit playout, forward digits, and default voice routes, or use hunt groups with dial peer preferences.

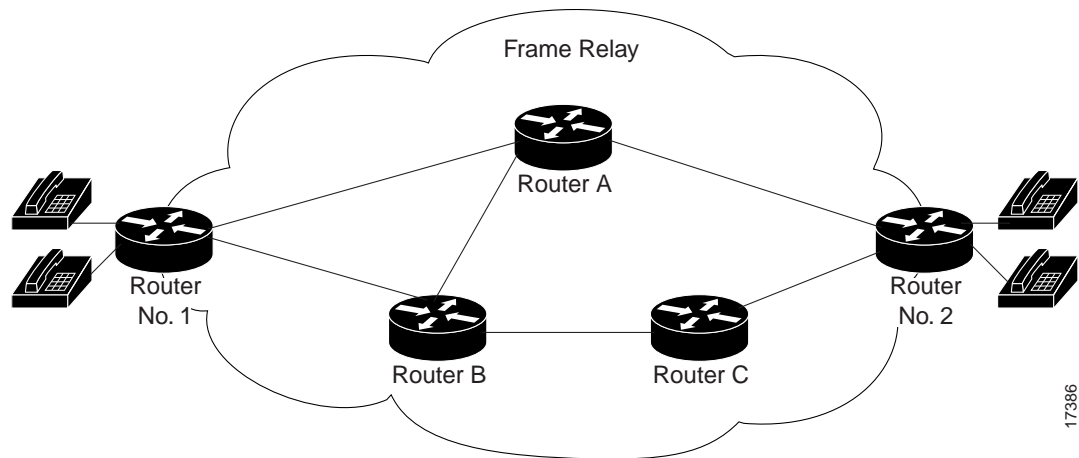
For VoFR dial peer configuration procedures, refer to the following sections:

- Configure Voice over Frame Relay Dial Peers for Switched Calls on page 23
- Configuring Voice over Frame Relay Dial Peers for Cisco-Trunk (Private Line) Calls on page 25
- Configure Voice over Frame Relay Dial Peers for FRF.11 Trunk Calls on page 27
- Configure Voice over Frame Relay Dial Peers for Tandem Nodes on page 29

Note On the Cisco MC3810, you can also configure a voice class to assign idle state and out-of-service (OOS) signaling attributes to a VoFR dial peer. For more information, see the online feature guide *Trunk Conditioning on the Cisco MC3810* for Cisco IOS Release 12.0(4)T.

Figure 7 shows an example of a Frame Relay network with switched calls. In the example, there are two routers (Routers No. 1 and No. 2) with telephone devices at the endpoints. In between the two endpoint routers are tandem routers (Routers A, B, and C) with switched connections in between. Standard VoFR dial peers are configured on the intermediate nodes in the Frame Relay network. Support for switched calls are configured on the dial peers on both Router No. 1 and Router No. 2.

Figure 7 Endpoint Nodes and Tandem Nodes in VoFR Network for Switched Calls



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In this example, different types of VoFR dial peers have to be configured on the different routers.

VoFR dial peers for switched connections must be configured between Router No. 1 and Router No. 2 (the endpoints for the voice connection)

On the tandem routers (Routers A, B, and C) VoFR dial peers for switched calls must be configured for the following:

- Between Router No. 1 and Router A
- Between Router No. 1 and Router B
- Between Router No. 2 and Router A
- Between Router No. 2 and Router C
- Between Router A and Router B
- Between Router B and Router C

Configure Voice over Frame Relay Dial Peers for Switched Calls

If you will be sending switched calls over the Frame Relay network, you must configure the Voice over Frame Relay dial peers to specifically support switched calls.

To configure a VoFR dial peer to support switched calls, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	<code>router(config)# dial-peer voice tag-number vofr</code>	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 <i>tag-number</i> value identifies the dial peer and must be unique on the router. Do not duplicate a specific tag number.
2	<code>router(config-dialpeer)# destination-pattern string</code>	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	<code>router(config-dialpeer)# session target interface dlci [cid]</code>	Configure the Frame Relay session target for the dial peer.

Step	Command	Purpose
4	<code>router(config-dialpeer)# session protocol cisco-switched</code>	Configure the session protocol to support switched calls. Note This is the default setting, and entering this command is not required.
5	<code>router(config-dialpeer)# codec type [bytes bytes]</code>	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 If configuring regular switched voice calls on the Cisco MC3810, you must configure the CODEC type on the voice port.
6	<code>router(config-dialpeer)# dtmf-relay</code>	(Optional) If the codec type configured 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	<code>router(config-dialpeer)# no vad</code>	(Optional) Disable voice activity detection (VAD) on the dial peer. This command is enabled by default.
8	<code>router(config-dialpeer)# sequence-numbers</code>	(Optional) Enable the voice sequence number if required for your configuration. This command is disabled by default.
9	<code>router(config-dialpeer)# preference value</code>	(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.
10	<code>router(config-dialpeer)# fax rate {2400 4800 7200 9600 14400 disable voice}</code>	(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.
11	To configure another VoFR dial peer, exit dial-peer configuration mode and repeat steps 1 through 10.	

To configure VoFR dial peers for tandem nodes, see the “Configure Voice over Frame Relay Dial Peers for Tandem Nodes” section on page 29.

For procedures on how to configure switched calls, go to the “Configuring Switched Calls (User Dialed or Auto-Ringdown)” section on page 33. For information on configuring all call types, go to the “Overview of Voice over Frame Relay Connection Types” section on page 30.

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

If you will be 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. Depending on the router you are using, you might have several options. 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**—On the Cisco MC3810, 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. This option is not supported on the Cisco 2600/3600 series.
- **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**—On the Cisco MC3810 with digital voice ports, 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 Cisco MC3810 to handle or transport unknown signaling protocols. On the Cisco MC3810 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. This option is not supported on the Cisco 2600 series and 3600 series.

The signal type normally must be configured such that the signal type 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, the signal type normally should be set 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 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. For more information on controlling the flow of voice packets with transparent signaling, see the “Verifying Your Frame Relay Configuration” section on page 15.

Configuring Dial Peers

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	<code>router(config)# dial-peer voice number vofr</code>	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 <i>number</i> tag value identifies the dial peer and must be unique on the router. Do not duplicate a specific tag number.
2	<code>router(config-dialpeer)# destination-pattern string</code>	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	<code>router(config-dialpeer)# session target interface dldci [cid]</code>	Configure the Frame Relay session target for the dial peer.
4	<code>router(config-dialpeer)# session protocol cisco-switched</code>	Configure the session protocol to support switched calls. This is the default setting, and entering this command is not required.
5	<code>router(config-dialpeer)# codec type [bytes bytes]</code>	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	<code>router(config-dialpeer)# dtmf-relay</code>	(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	<code>router(config-dialpeer)# signal-type {cas cept ext-signal transparent}</code>	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 on the Cisco MC3810 only) to read the ABCD signaling bits directly from the T1/E1 interface without interpretation, and to pass them transparently to the data network (this is also known as transparent FRF.11 signaling).

Step	Command	Purpose
8	<code>router(config-dialpeer)# no vad</code>	(Optional) Disable voice activity detection (VAD) on the dial peer. This command is enabled by default.
9	<code>router(config-dialpeer)# sequence-numbers</code>	(Optional) Enable the voice sequence number if required for your configuration. This command is disabled by default.
10	<code>router(config-dialpeer)# preference value</code>	(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	<code>router(config-dialpeer)# fax rate {2400 4800 7200 9600 14400 disable voice}</code>	(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.	

To configure VoFR dial peers for tandem nodes, see the “Configure Voice over Frame Relay Dial Peers for Tandem Nodes” section on page 29.

For procedures on how to configure Cisco-trunk permanent (private line) calls, see the “Configuring Cisco-Trunk Permanent (Private Line) Calls” section on page 37. For information on configuring all call types, see the “Overview of Voice over Frame Relay Connection Types” section on page 30.

Configure Voice over Frame Relay Dial Peers for FRF.11 Trunk Calls

If you will be sending FRF.11 trunk calls over the Frame Relay network, you must configure the Voice over Frame Relay dial peers to specifically support FRF.11 trunk calls. For FRF.11 trunk calls, you must statically configure the VoFR dial peers on both sides of the FRF.11 trunk.

Note FRF.11 trunk calls cannot be used in conjunction with dial plans.

To configure a VoFR dial peer to support FRF.11 trunk calls, use the following commands beginning in global configuration mode:

Step	Command	Purpose
1	<code>router(config)# dial-peer voice number vofr</code>	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 <i>number</i> tag value identifies the dial peer and must be unique on the router. Do not duplicate a specific tag number.
2	<code>router(config-dialpeer)# destination-pattern string</code>	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	<code>router(config-dialpeer)# session protocol frf11-trunk</code>	Configure the session protocol to support FRF.11 trunk calls. Note You cannot send FRF.11 trunk calls through tandem nodes.
4	<code>router(config-dialpeer)# session target interface dlci cid</code>	Configure the Frame Relay session target for the dial peer. Note The <i>cid</i> value is required for FRF.11 trunk calls.

Step	Command	Purpose
5	<code>router(config-dialpeer)# codec type bytes bytes</code>	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 . For FRF.11 trunk calls, the codec values must be set the same on both sides of the connection. 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 (?).
6	<code>router(config-dialpeer)# dtmf-relay</code>	(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	<code>router(config-dialpeer)# called-number termination-string</code>	(Required for the Cisco 2600/3600 series only) Configure the termination string for the dial peer for FRF.11 trunk calls. This command is required to enable the router to establish an incoming trunk connection. This command only applies when the session protocol command is set to frf11-trunk . Note Although this command is visible on the Cisco MC3810, the command is ignored if entered.
8	<code>router(config-dialpeer)# signal type {cas external}</code>	(Optional on the Cisco 2600/3600 series only) Configure the signal type when creating an FXS-FXS trunk. The default on the Cisco 2600/3600 series is cas .
9	<code>router(config-dialpeer)# no vad</code>	(Optional) Disable voice activity detection (VAD) on the dial peer. This command is enabled by default.
10	<code>router(config-dialpeer)# sequence-numbers</code>	(Optional) Enable the voice sequence number if required for your configuration. This command is disabled by default.
11	<code>router(config-dialpeer)# preference value</code>	(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.
12	<code>router(config-dialpeer)# fax rate {2400 4800 7200 9600 14400 disable voice}</code>	(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.
13	To configure another VoFR dial peer, exit dial-peer configuration mode and repeat steps 1 through 12.	

Repeat this procedure on the destination router on the other side of the FRF.11 trunk.

To configure VoFR dial peers for tandem nodes, see the “Configure Voice over Frame Relay Dial Peers for Tandem Nodes” section on page 29.

For procedures on how to configure FRF.11 trunk calls, go to the “Configuring FRF.11 Trunk (Private Line) Calls” section on page 41. For information on configuring all call types, go to the “Overview of Voice over Frame Relay Connection Types” section on page 30.

Configure Voice over Frame Relay Dial Peers for Tandem Nodes

You configure standard VoFR dial peers for switched calls on the tandem routers in the network topology. Tandeming is not allowed when the call type is an FRF.11 trunk call.

You can configure VoFR dial peers for tandem routers on the Cisco MC3810 and on the Cisco 2600 series, Cisco 3600 series, and Cisco 7200 series routers.

Note The Cisco 7200 Series routers can only serve as tandem nodes in the VoFR network. This is the only dial peer procedure supported on the Cisco 7200 series.

To configure VoFR dial peers on tandem routers, use the following commands from configuration mode:

Step	Command	Purpose
1	<code>router(config)# dial-peer voice <i>number</i> vofr</code>	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 <i>number</i> value tag identifies the dial peer and must be unique on the router. Do not duplicate a specific number tag.
2	<code>router(config-dialpeer)# destination-pattern [+]<i>string</i>[t]</code>	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	<code>router(config-dialpeer)# session target interface <i>dlci</i></code>	Configure the Frame Relay session target for the dial peer.
4	<code>router(config-dialpeer)# preference <i>value</i></code>	(Optional) Configure a preference for the VoFR dial peer. The <i>value</i> is a number from 0 to 10 where the lower the number, the higher the preference in hunt groups.
5	To configure the next VoFR dial peer, exit dial-peer configuration mode by entering exit , and repeat the previous steps. On tandem nodes, at least two VoFR dial peers are required, one for each call leg in the router.	

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 33
- Configuring Cisco-Trunk Permanent (Private Line) Calls on page 37
- Configuring FRF.11 Trunk (Private Line) Calls on page 41

In addition, special consideration is required for configuring calls for tandem nodes. For more information, see "Configuring Connections for Tandem Nodes" section on page 42.

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 using combinations of several commands.

Table 3 lists the different connection types for VoFR connections supported on the Cisco 2600/3600 series, and the combinations of commands to enter for each call type.

Table 4 lists the different connection types for VoFR connections supported on the Cisco MC3810 series, and the combinations of commands to enter for each call type.

Table 3 Voice over Frame Relay Connection Types Supported on the Cisco 2600/3600 Series

Type of Call	Frame Relay DLCI interface Command to Enter	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 Cisco 2600/3600 routers	vofr [<i>data cid</i>] [call-control [<i>cid</i>]] ¹	FRF.11 Annex C	session protocol cisco-switched ²	For user dialed calls: none For auto-ringdown calls: connection plar <i>destination-string</i>
Switched call (user dialed or auto-ringdown) to a Cisco MC3810	vofr cisco ³	Cisco proprietary ⁴	session protocol cisco-switched	For user dialed calls: none For auto-ringdown calls: connection plar <i>destination-string</i>
Cisco-trunk permanent call (private-line) to other Cisco 2600/3600 routers	vofr <i>data cid</i> call-control <i>cid</i>	FRF.11 Annex C	session protocol cisco-switched	connection trunk <i>destination-string</i> [answer mode]
Cisco-trunk permanent call (private-line) to a Cisco MC3810	vofr cisco	Cisco proprietary	session protocol cisco-switched	connection trunk <i>destination-string</i> [answer mode]
FRF.11 trunk call (private-line)	vofr [<i>data cid</i>] [call-control <i>cid</i>]] ⁵	FRF.11 Annex C	session protocol frf11-trunk	connection trunk <i>destination-string</i> [answer mode]

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

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

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

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

5 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.

Table 4 Voice over Frame Relay Call Types Supported on the Cisco MC3810

Type of Call	Frame Relay DLCI interface Command to Enter	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)	vofr cisco ¹ or frame-relay interface-dlci dlci voice-encap size [voice-cir cir] ²	Cisco proprietary ³	session protocol cisco-switched	For user dialed calls: none For auto-ringdown calls: connection plar destination-string For tie-line connections: connection tie-line destination-string
Cisco-trunk permanent call (private line)	vofr cisco or frame-relay interface-dlci dlci voice-encap size [voice-cir cir]	Cisco proprietary	session protocol cisco-switched	connection trunk destination-string [answer mode]
FRF.11 trunk call (private-line)	vofr [data cid] ⁴	FRF.11 Annex C	session protocol frf11-trunk	connection trunk destination-string [answer mode]

1 This command consumes data CID 4 and call-control CID 5.
 2 The **voice-encap** value is required, but the **voice-cir** value is optional.
 3 Cisco proprietary fragmentation is based on an early draft of FRF.12, and is compatible with Cisco MC3810 concentrators running software versions prior to Cisco IOS Release 12.0(3)XG or 12.0(4)T.
 4 If the **vofr** command is entered without the **cisco** option, only FRF.11 trunks are supported. You cannot mix FRF.11 trunk calls with other call types on the same PVC on the Cisco MC3810.

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 on Cisco 2600, 3600, and 7200 Series Routers on page 33
- Configuring Switched Calls on Cisco 2600, 3600, or 7200 series Routers to a Cisco MC3810 on page 33
- Configuring Switched Calls on a Cisco MC3810 on page 34

Configuring Switched Calls on Cisco 2600, 3600, and 7200 Series Routers

You can configure switched calls on Cisco 2600, 3600, and 7200 series routers. To configure switched calls on these routers, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<code>router(config-if)# frame-relay interface-dlci dlci</code>	Configure the Frame Relay DLCI and enter DLCI configuration mode.
2	<code>router(config-if)# vofr [data cid] [call-control [cid]]</code>	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 the vofr command is entered without any keywords or arguments, the data subchannel will be CID 4 and there will be no call-control subchannel. If configuring user-dialed calls, this procedure is completed. If configuring auto-ringdown calls, proceed to the next step.
3	<code>router(config)# voice-port slot/subunit/port</code>	Enter voice port configuration mode on the Cisco 2600/3600 routers.
4	<code>router(config-voiceport)# connection plar string</code>	(Optional) For auto-ringdown calls, configure the PLAR connection, specifying the telephone number in the <i>destination-string</i> .

This configuration uses standard FRF.11 Annex C fragmentation.

Configuring Switched Calls on Cisco 2600, 3600, or 7200 series Routers to a Cisco MC3810

On the Cisco 2600, 3600, and 7200 series routers, you can configure switched calls to a Cisco MC3810. However, the configuration is different from standard switched calls because the 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 prior to Cisco IOS Releases 12.0(3)XG or 12.0(4)T.

Configuring Voice over Frame Relay Connections

To configure switched calls to a Cisco MC3810, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<code>router(config-if)# frame-relay interface-dlci dlci</code>	Configure the Frame Relay DLCI and enter DLCI configuration mode.
2	<code>router(config-if)# vofr cisco</code>	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 completed. If configuring auto-ringdown calls, proceed to the next step.
3	<code>router(config)# voice-port slot/subunit/port</code>	Enter voice-port configuration mode on the Cisco 2600/3600 series routers.
4	<code>router(config-voiceport)# connection plar destination-string</code>	(Optional) For auto-ringdown calls, configure the PLAR connection, specifying the telephone number in the <i>destination-string</i> .

This configuration uses Cisco proprietary data fragmentation.

Configuring Switched Calls on a Cisco MC3810

On the Cisco MC3810 you can configure switched calls to another Cisco MC3810 or to Cisco 2600, Cisco 3600, and Cisco 7200 series systems. However, the configuration is different from switched calls on other routers because the 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 prior to Cisco IOS Release 12.0(3)XG or 12.0(4)T.

When configuring switched calls on a Cisco MC3810, you must enter one of the following two commands:

- **vofr cisco**

This command was added in Cisco IOS Release 12.0(3)XG, and uses weighted fair queuing at the PVC level. Using the **vofr cisco** command, you cannot disable support for weighted fair queuing.

- **frame-relay interface-dlci dlci voice-encap size [voice-cir cir]**

The **voice-encap** option for the **frame-relay interface-dlci** command was introduced in Cisco IOS Release 11.3(1)MA and does not support weighted fair queuing. However, you also have the option to configure fancy queuing with the **voice-encap** option by entering the **no frag-pre-queuing** command and configuring the appropriate fancy queuing commands.

The procedures for using each command are different. Refer to the appropriate procedure for the command you plan to use. These configurations both use Cisco proprietary data fragmentation.

Note The **vofr** command and the **frame-relay interface-dlci voice-encap** command are mutually exclusive on the same interface, so you must choose which command to use.

To configure switched calls on a Cisco MC3810 using the **vofr cisco** command, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<code>router(config-if)# frame-relay interface-dlci dlci</code>	Configure the Frame Relay DLCI and enter DLCI configuration mode. Note When configuring switched calls on the Cisco MC3810 using the vofr cisco command, do not enter the voice-encap or voice-cir options.
2	<code>router(config-if)# vofr cisco</code>	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 configuring user dialed calls, this procedure is completed. If configuring auto-ringdown calls or tie-line connections on the Cisco MC3810, proceed to the next step.
3	<code>router(config)# voice-port slot/port</code>	Enter voice-port configuration mode on the Cisco MC3810.
4	Choose one of the following: <code>router(config-voiceport)# connection plar destination-string</code> <code>router(config-voiceport)# connection tie-line destination-string</code>	For auto-ringdown calls, configure the PLAR connection, specifying the telephone number in the <i>destination-string</i> . For tie-line calls, configure the tie-line connection, specifying the telephone number in the <i>destination-string</i> .

To configure switched calls on a Cisco MC3810 using the **frame-relay interface-dlci voice-encap** command, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<code>router(config-if)# frame-relay interface-dlci dlci voice-encap size [voice-cir cir]</code>	Configure the Frame Relay DLCI, enter DLCI configuration mode and specify that the DLCI will support voice traffic. For recommended fragmentation sizes to use with the voice-encap option, see Table 5. In this configuration, the voice-encap value is required, but the voice-cir value is optional. The default voice-cir is the CIR configured for the Frame Relay map class. Do not configure the voice-cir option to be higher than the physical link speed. If Frame Relay traffic shaping is enabled for a PVC sharing voice and data, do not configure the voice-cir option to be higher than the value set with the frame-relay mincir command.

Step	Command	Purpose
2	<code>router(config-if)# no frag pre-queuing</code>	<p>(Optional) Enter this command to disable first-come-first-served (FCFS) queuing on the interface.</p> <p>By default, the frag pre-queuing command is enabled. This command supports only FCFS queuing, and performs the queuing before the data fragmentation takes place.</p> <p>If you want to configure fancy queuing (weighted fair queuing, priority queuing, or custom queuing) on the interface, enter the no frag pre-queuing command. Note that after you enter the no frag pre-queuing command, you still must configure the commands to support the fancy queuing method desired.</p> <p>Note The frag pre-queuing command is only supported on the interface if the voice-encap option for the frame-relay interface-dlci command is configured. It is not supported if the vofr cisco command is configured.</p>
3	<code>router(config)# voice-port slot/port</code>	Enter voice-port configuration mode on the Cisco MC3810.
4	<p>Choose one of the following:</p> <pre>router(config-voiceport)# connection plar destination-string</pre> <pre>router(config-voiceport)# connection tie-line destination-string</pre>	<p>For auto-ringdown calls, configure the PLAR connection, specifying the telephone number in the <i>destination-string</i>.</p> <p>For tie-line calls, configure the tie-line connection, specifying the telephone number in the <i>destination-string</i>.</p>

Table 5 lists recommended data fragmentation sizes to use when configuring the **voice-encap** option for the **frame-relay interface-dlci** command.

Table 5 Recommended Data Fragmentation Sizes

Access Rate	Recommended Data Fragmentation Size ¹
64 kbps	80 bytes
128 kbps	160 bytes
256 kbps	320 bytes
512 kbps	640 bytes
1536 kbps (full T1)	1600 bytes
2048 kbps (full E1)	1600 bytes

¹ The data fragmentation size is based on back-to-back Frame Relay. If you are sending traffic through an IGX node with standard Frame Relay, subtract 6 bytes from the recommended data fragmentation size.

Note When you configure the data fragmentation size, use the slower access rate of either the local or remote device to calculate which data fragmentation size to use. If you configure a data fragmentation size too high for either the local or remote device, the access rate will become throttled because the slower device cannot handle the larger data fragmentation size. For example, if the access rate at the local device is 512 kbps and the access rate of the remote device is 256 kbps, configure the data segmentation size based on the slower 256-kbps access rate.

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 Cisco Trunk Permanent Calls on Cisco 2600, 3600, and 7200 Series Routers on page 37
- Configuring Cisco Trunk Permanent Calls on Cisco 2600 and 3600 Routers to a Cisco MC3810 on page 38
- Configuring Cisco Trunk Permanent Calls on a Cisco MC3810 on page 39
- Configuring FRF.11 Trunk (Private Line) Calls on page 41

Configuring Cisco Trunk Permanent Calls on Cisco 2600, 3600, and 7200 Series Routers

You can configure Cisco trunk permanent calls on Cisco 2600, 3600, and 7200 series routers. To configure Cisco trunk permanent calls on these routers, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<code>router(config-if)# frame-relay interface-dlci dlci</code>	Configure the Frame Relay DLCI and enter DLCI configuration mode.
2	<code>router(config-if)# vofr [data cid] [call-control cid]</code>	Configure the Frame Relay DLCI to support VoFR. Note When the <code>vofr</code> command is used, all subchannels on the DLCI are configured for FRF.11 encapsulation. If the <code>vofr</code> command is entered without any keywords or arguments, the data subchannel will be CID 4 and there will be no call-control subchannel. If configuring tandem calls, this step ends your configuration.
3	<code>router(config)# voice-port slot/subunit/port</code>	Enter voice port configuration mode on the Cisco 2600/3600 series routers.
4	<code>router(config-voiceport)# connection trunk destination-string [answer-mode]</code>	For private line calls, configure the trunk connection, specifying the telephone number in the <i>destination-string</i> . When configuring Cisco trunk permanent calls, one side must be the call initiator (master) and the other side is normally the call answerer (slave). By default, the voice operates in master mode. Enter the answer-mode keyword to specify that the voice port should operate in slave mode.
5	<code>router(config-voiceport)# shutdown</code>	Shut down the voice port.
6	<code>router(config-voiceport)# no shutdown</code>	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 on Cisco 2600 and 3600 Routers to a Cisco MC3810

To configure Cisco trunk permanent calls to a Cisco MC3810 from a 2600 series or 3600 series router, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<code>router(config-if)# frame-relay interface-dlci dlci</code>	Configure the Frame Relay DLCI and enter DLCI configuration mode.
2	<code>router(config-if)# vofr cisco</code>	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	<code>router(config)# voice port slot/subunit/port</code>	Enter voice port configuration mode on the Cisco 2600/3600 series routers.
4	<code>router(config-voiceport)# connection trunk destination-string [answer-mode]</code>	For private line calls, configure the trunk connection, specifying the telephone number in the <i>destination-string</i> . When configuring Cisco trunk permanent calls, one side must be the call initiator (master) and the other side is normally the call answerer (slave). By default, the voice operates in master mode. Enter the answer-mode keyword to specify that the voice port should operate in slave mode.
5	<code>router(config-voiceport)# shutdown</code>	Shut down the voice port.
6	<code>router(config-voiceport)# no shutdown</code>	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 Cisco Trunk Permanent Calls on a Cisco MC3810

When configuring Cisco trunk permanent calls on a Cisco MC3810 interface, you must enter one of the following two commands:

- **vofr cisco**

This command was added in Cisco IOS Release 12.0(3)XG, and uses weighted fair queuing at the PVC level. Using the **vofr cisco** command, you cannot disable support for weighted fair queuing.

- **frame-relay interface-dlci** *dlci* **voice-encap** *size* [**voice-cir** *cir*]

The **voice-encap** option for the **frame-relay interface-dlci** command was introduced in Cisco IOS Release 11.3(1)MA and does not support weighted fair queuing. However, you also have the option to configure fancy queuing with the **voice-encap** option by entering the **no frag-pre-queuing** command and configuring the appropriate fancy queuing commands.

The procedures for using each command are different. Refer to the appropriate procedure for the command you plan to use. These configurations both use Cisco proprietary data fragmentation.

Note The **vofr** command and the **frame-relay interface-dlci voice-encap** command are mutually exclusive on the same interface, so you must choose which command to use.

To configure Cisco trunk permanent calls on a Cisco MC3810 using the **vofr cisco** command, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<code>router(config-if)# frame-relay interface-dlci <i>dlci</i></code>	Configure the Frame Relay DLCI and enter DLCI configuration mode. Note When configuring Cisco trunk calls on the Cisco MC3810 using the vofr cisco command, do not enter the voice-encap or voice-cir options.
2	<code>router(config-if)# vofr cisco</code>	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	<code>router(config)# voice port <i>slot/port</i></code>	Enter voice port configuration mode on the Cisco MC3810.
4	<code>router(config-voiceport)# connection trunk <i>destination-string</i> [answer-mode]</code>	For private line calls, configure the trunk connection, specifying the telephone number in the <i>destination-string</i> . When configuring Cisco trunk permanent calls, one side must be the call initiator (master) and the other side is normally the call answerer (slave). By default, the voice operates in master mode. Enter the answer-mode keyword to specify that the voice port should operate in slave mode.
5	<code>router(config-voiceport)# shutdown</code>	Shut down the voice port.
6	<code>router(config-voiceport)# no shutdown</code>	Reactivate the voice port to enable the trunk connection to take effect.

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

Configuring Voice over Frame Relay Connections

To configure Cisco trunk permanent calls on a Cisco MC3810 using the **frame-relay interface-dlci voice-encap** command, use the following commands from interface configuration mode:

Step	Command	Purpose
1	<pre>router(config-if)# frame-relay interface-dlci dlci voice-encap size [voice-cir cir]</pre>	<p>Configure the Frame Relay DLCI, enter DLCI configuration mode and specify that the DLCI will support voice traffic.</p> <p>For recommended fragmentation sizes to use with the voice-encap option, see Table 5.</p> <p>In this configuration, the voice-encap value is required, but the voice-cir value is optional. The default voice-cir is the CIR configured for the Frame Relay map class</p> <p>Note Do not configure the voice-cir option to be higher than the physical link speed. If Frame Relay traffic shaping is enabled for a PVC sharing voice and data, do not configure the voice-cir option to be higher than the value set with the frame-relay mincir command</p>
2	<pre>router(config-if)# no frag pre-queuing</pre>	<p>(Optional) Enter this command if you want to disable first-come-first-served (FCFS) queuing on the interface.</p> <p>By default, the frag pre-queuing command is enabled. This command supports only FCFS queuing, and performs the queuing before the data fragmentation takes place.</p> <p>If you want to configure fancy queuing (weighted fair queuing, priority queuing, or custom queuing) on the interface, enter the no frag pre-queuing command. Note that after you enter the no frag pre-queuing command, you still must configure the commands to support the fancy queuing method desired.</p> <p>Note The frag pre-queuing command is only supported on the interface if the voice-encap option for the frame-relay interface-dlci command is configured. It is not supported if the vofr cisco command is configured.</p>
3	<pre>router(config)# voice port slot/port</pre>	Enter voice port configuration mode on the Cisco MC3810.
4	<pre>router(config-voiceport)# connection trunk destination-string [answer-mode]</pre>	<p>For private line calls, configure the trunk connection, specifying the telephone number in the <i>destination-string</i>.</p> <p>When configuring Cisco trunk permanent calls, one side must be the call initiator (master) and the other side is normally the call answerer (slave). By default, the voice port is the master. Enter the answer-mode keyword to specify that the voice port will be the slave.</p>
5	<pre>router(config-voiceport)# shutdown</pre>	Shut down the voice port.
6	<pre>router(config-voiceport)# no shutdown</pre>	Reactivate the voice port to enable the trunk connection to take effect.

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

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 the **session protocol** dial-peer configuration command be set to **frf11-trunk**. See the “Configuring Voice over Frame Relay Dial Peers” section on page 22.

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

Step	Command	Purpose
1	<code>router(config-if)# frame-relay interface-dlci <i>dlci</i></code>	Configure the Frame Relay DLCI and enter DLCI configuration mode.
2	Choose one of the following: <code>router(config-if)# vofr [<i>data cid</i>] [call-control <i>cid</i>]</code> <code>router(config-if)# vofr [<i>data cid</i>]</code>	(Cisco 2600 and Cisco 3600) Configure the Frame Relay DLCI to support VoFR and to optionally enter the data and call-control CIDs. (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 the vofr command is used, all subchannels on the DLCI are configured for FRF.11 encapsulation except the data subchannel. If the vofr command is entered without any keywords or arguments, the data subchannel will be CID 4 and there will be no call-control subchannel.
3	Choose one of the following: <code>router(config)# voice port <i>slot/subunit/port</i></code> <code>router(config)# voice port <i>slot/port</i></code>	Enter voice port configuration mode on the Cisco 2600 or 3600 series routers. Enter voice port configuration mode on the Cisco MC3810.
4	<code>router(config-voiceport)# connection trunk <i>destination-string</i> [answer-mode]</code>	For private line calls, configure the trunk connection, specifying the telephone number in the <i>destination-string</i> . 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 will be the slave.
5	<code>router(config-voiceport)# shutdown</code>	Shut down the voice port.
6	<code>router(config-voiceport)# no shutdown</code>	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 the switching of an incoming VoFR call 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. The Cisco 7200 series can only act as a tandem router in a VoFR network.

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 higher 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 6 shows the different combinations of routers that can serve as end nodes and tandem nodes, and the Frame Relay PVC type required.

Table 6 VoFR End Node and Tandem Node Combinations Supported

End Node(s)	Tandem Node	vofr Command to Enter for the Frame Relay DLCI
Cisco 2600/3600	Cisco 2600, Cisco 3600, or Cisco 7200	vofr call-control
Cisco 2600/3600 and Cisco MC3810	Cisco MC3810	vofr cisco
Cisco MC3810	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 doing the following:

- Pick up the handset on a telephone connected to the configuration and verify that you can get a dial tone.
- 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:

- Pick up the telephone and listen to hear the dial tone from the remote PBX.
- 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, you can use the **show dial-peer voice** command to verify that the data configured is correct. On the Cisco MC3810, use the **show dial-peer voice summary** command.

- To show the status of the voice ports, use the **show voice port** command.
- To show the call status for all voice ports, use 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, use 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, use the **show frame-relay vofr [interface [dlci [cid]]]** command. This command is not supported on the Cisco MC3810 when the **vofr cisco** or the **frame-relay interface-dlci dlci voice-encap** command is configured.
- To show the status of Cisco-trunk permanent calls (private-line) on the Cisco MC3810, use 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, serial port, and/or the T1/E1 controller are set to **no shutdown**.
- Be sure to toggle the voice port (by first entering **shutdown**, then **no shutdown**), every time you enter the **connection trunk** or **no connection trunk** commands.

Monitoring and Maintaining Your Voice over Frame Relay Configuration

Use the following commands to monitor and maintain your Voice over Frame Relay configuration:

Command	Purpose
Router# <code>show call active voice [brief]</code>	(Cisco 2600/3600 series only) Displays the active call table.
Router# <code>show call history voice [last number] [brief]</code>	(Cisco 2600/3600 series only) Displays the call history table.
Router# <code>show call history voice record</code>	(Cisco MC3810 only) Displays the call history table.
Router# <code>show dial-peer voice</code>	Displays configuration information and call statistics for dial peers.
Router# <code>show frame-relay fragment</code>	Displays information about the Frame Relay fragmentation taking place in your Cisco router. Note On the Cisco MC3810, if the frame-relay interface-dlci voice-encap command is configured, information about that PVC is not displayed.
Router# <code>show frame-relay pvc</code>	Displays statistics about PVCs for Frame Relay interfaces.
Router# <code>show frame-relay vofr</code>	Displays information about the FRF.11 subchannels being used on VoFR DLCIs. Note On the Cisco MC3810, if the frame-relay interface-dlci voice-encap command is configured, information about that PVC is not displayed.
Router# <code>show interfaces serial</code>	Displays information about a serial interface.
Router# <code>show traffic-shape queue</code>	Displays information about the elements queued at a particular time at the VC (DLCI) level.
Router# <code>show voice call</code>	(Cisco MC3810 only) Displays the call status for all voice ports.
Router# <code>show voice permanent-call</code>	Displays information about the permanent calls on a voice interface.