Voice over Frame Relay

This chapter describes the configuration of Voice over Frame Relay (VoFR) and contains the following sections:

For a description of the VoFR configuration commands using the FRF.11 implementation agreement, refer to the *Cisco IOS Voice Command Reference*, Release 12.3. For additional information about the FRF.12 implementation agreement and wide-area networks (WANs), refer to the *Cisco IOS Wide-Area Networking Configuration Guide* and *Cisco IOS Wide-Area Networking Command Reference*. For information about voice port configurations, refer to the "Configuring Voice Ports" chapter.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and CiscoIOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

VoFR Overview

VoFR enables a router to carry voice traffic (for example, telephone calls and faxes) over a Frame Relay network, using the FRF.11 protocol. This specification defines multiplexed data, voice, fax, dual tone multi frequency (DTMF) digit-relay, and channel-associated signaling (CAS)/robbed-bit signaling frame formats. The Frame Relay backbone must be configured to include the map class and Local Management Interface (LMI).

The Cisco VoFR implementation enables dynamic- and tandem-switched calls and Cisco trunk calls. Dynamic-switched calls have dial-plan information included that processes and routes calls based on the telephone numbers. The dial-plan information is contained within dial-peer entries. For more information, see "Switched Calls" section on page 3.

Tandem-switched calls are switched from incoming VoFR to an outgoing VoFR enabled data-link connection identifier (DLCI) and tandem nodes enable the process. The nodes also switch Cisco trunk calls.

Permanent calls are processed over Cisco private-line trunks and static FRF.11 trunks that specify the frame format and coder types for voice traffic over a Frame Relay network. For more information, see "Permanent Calls" section on page 4.

VoFR connections depend on the hardware platform and type of call. The types of calls are:

- Switched (user dialed or auto-ringdown and tandem)
- Permanent (Cisco trunk or static FRF.11 trunk)

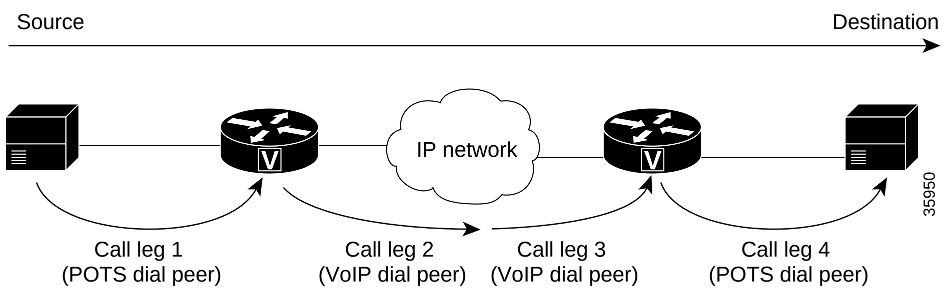
Note Calls to Cisco MC3810 multiservice concentrators running Cisco IOS releases before 12.0(7)XK and 12.1(2)T require specific procedures for VoFR configuration and are described in separate sections.

VoFR Dial Peers

Dial peers are addressable call endpoints that identify the origin and destination of a call. Dial peers define the characteristics applied to each call leg in the call connection. A call leg is a logical connection between two routers or between a router and a telephony device.

A traditional voice call over the Public Switched Telephone Network (PSTN) uses a dedicated 64K circuit end-to-end. In contrast, a voice call over the packet network is made up of call legs. A voice call has four call legs, two from the perspective of the originating router and two from the perspective of the destination router, as shown in Figure 91.

Figure 91 Dial Peer Call Legs



A dial peer is associated with each call leg. Attributes that are defined in a dial peer and applied to the call leg include codec, Quality of Service (QoS), voice activity detection (VAD), and fax rate. To complete a voice call, you must configure a dial peer for each of the four call legs in the call connection.

Two kinds of dial peers are possible in VoFR configurations:

- POTS—Dial peer describing the characteristics of a traditional telephony network connection. POTS dial peers map a dialed string to a specific voice port on the local router, normally the voice port connecting the router to the local PSTN, PBX, or telephone.
- VoFR—Dial peer that is connected between a Frame Relay WAN backbone and a specific voice-network device. VoFR dial peers map a dialed string to the destination router.

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 the sending and receiving of calls are required.

Understanding the the relationship between the destination pattern and the session target is critical to understanding VoFR dial peers. The destination pattern is the telephone number of the voice device attached to the voice port. The session target defines the route to a serial port on the peer router at the other end of the Frame Relay connection.

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

For additional information on POTS dial peers, see the "Configuring Dial Plans, Dial Peers, and Digit Manipulation" chapter.

Switched Calls

The Cisco-switched VoFR protocol handles call setup and parameter negotiation for both endpoints and intermediate nodes within the multihop call path. The call setup mechanism originally implemented in the Cisco MC3810 multiservice concentrator can be used for permanent-switched (Cisco trunk) or dynamic-switched calls. The Cisco VoFR protocol includes forwarding of the called telephone number and supports tandem switching of the call over multiple Frame Relay permanent virtual connection (PVC) hops.

Cisco addresses the lack of end-to-end call parameter negotiation and call setup syntax in FRF.11 by implementing a proprietary Q.931-like session protocol running on a user-configurable channel ID (CID) of an FRF.11-format multiplexed DLCI.

Tandem Switching

Dynamic switching of voice calls between VoFR or VoATM PVCs and subchannels is also called tandem switching (often encountered in multihop VoFR call connection paths). Tandem switching uses nodes that are intermediate router nodes within the Frame Relay call path.

Each node switches the frames from one PVC subchannel to another (from one VoFR dial peer to another VoFR dial peer) as the frames traverse the network. Use of tandem router nodes avoids the need to have complete dial-plan information present on every router.

Dynamic-Switched Calls

Dynamic-switched calls are regular telephone calls in which the switching is performed by the Cisco router. The destination endpoint of the call is selected by the router based on the dialed telephone number and the dial peer configuration entries. This implementation is different from permanent calls (Cisco trunk calls) in which the call endpoints are permanently fixed at configuration time. The dial peer uses the Cisco proprietary session protocol.

Cisco Trunk Calls

A Cisco trunk call is a dynamic-switched call of indefinite duration that uses a fixed-destination telephone number and includes optional transparent end-to-end signaling. The telephone number of the destination endpoint is permanently configured into the router so that it always selects a fixed destination. Once established, at boot-up or when configured, the call stays up until one of the voice ports or network ports is shut down or until a network disruption occurs. The dial peer is configured to invoke the Cisco proprietary session protocol.

Permanent Calls

Permanent calls are transmitted and received on FRF.11 and Cisco trunks. FRF.11 trunk interoperability for standards-based vendors enables specification of the frame format and coder types to be used when sending voice traffic through a Frame Relay network. However, FRF.11 does not have specifications for end-to-end negotiation, call setup process, or any other form of communication between the Frame Relay nodes.

As a result, static FRF.11 trunks are set up by manually configuring each router within the voice trunk path with compatible parameters: a voice port and a specific subchannel on a DLCI are explicitly bound on each end router. Signaling information is packed and sent transparently end-to-end.

The two ends of an FRF.11 call must use the same compatible speech compression codecs. If not, the call exists and voice packets are sent and received, but no usable voice path is created.

When configured, a static FRF.11 trunk remains up until the voice or serial port is shut down or until a network disruption occurs. The FRF.11 specification does not include any standardized methods for performing Operation, Administration, and Maintenance (OAM) functions. There is no standard protocol for detecting faults and providing rerouting of connection paths.

FRF.11 enables up to 255 subchannels to be multiplexed onto a single Frame Relay DLCI. The current implementation supports the multiplexing of a single data channel with many voice channels. However, subchannels from zero to three are reserved and cannot be configured for voice or data.

Frame Relay Fragmentation

Cisco has developed three methods of performing Frame Relay fragmentation that are described in the following sections:

- End-to-End FRF.12 Fragmentation, page 5
- Frame Relay Fragmentation Using FRF.11 Annex C, page 6
- Cisco Proprietary Voice Encapsulation, page 6

FRF.11 can only be used when an end-to-end PVC is available between the voice ports at each end of the connection. At intermediate Frame Relay nodes, the entire PVC must be routed. Because the entire PVC is routed, no prioritization of voice packets is possible at the intermediate Frame Relay. Connection ID-based routing (individual channel-ID switching) is not supported.

FRF.11 specifies that a device can pack multiple FRF.11 subframes within a single Frame Relay frame; however, the Cisco implementation of VoFR currently does not support multiple subframes within a frame. VoFR frames are never fragmented, regardless of size. If fragments arrive out of sequence, packets are dropped. Fragmentation is performed after frames are removed from the weighted fair queuing (WFQ). WFQ at the PVC level is the only queueing strategy that can be used.

Frame Relay Traffic Shaping (FRTS) must be configured to enable Frame Relay fragmentation.

Frame Relay fragmentation can be configured in conjunction with VoFR or independently of it. For additional information regarding FRF.12 fragmentation and the implementation commands, refer to the *Cisco IOS Wide-Area Networking Configuration Guide* and *Cisco IOS Wide-Area Networking Command Reference.*

VoFR provides support for various FRF.11 features depending on the hardware platform used (see Table 30).

Table 30 FRF.11 Forum Features Supported by Hardware Platform

FRF.11 Forum Features	Cisco MC3810 Multiservice Concentrator	Cisco 2600/3600 Series Routers	Cisco 7500 Series Routers with VIP Support
Class 1–Compliance Requirements (sec. 4.1)	Not supported	Not supported	Not supported
Class 2–Compliance Requirements (sec. 4.2)	Supported	Supported	Supported
Annex A–Dialed Digits Transfer Syntax	Supported	Supported	Supported
Annex B–Signaling Bit Transfer Syntax	Supported	Supported	Supported
Annex C–Data Transfer Syntax	Supported	Supported	Supported
Annex D–Fax Relay Transfer Syntax	Supported	Supported	Supported
Annex E–CS-ACELP Transfer Syntax (G.729/G.729A)			
• Sequence Number	Supported	Supported	Supported
• Packing Factor	Supported	Supported	Supported
Annex F–Generic PCM/ADPCM Voice Transfer Syntax	Supported	Supported	Supported
Annex G –G.727 Discard-Eligible E-ADPCM Voice Transfer Syntax	Not supported	Not supported	Not supported
Annex H–G.728 LD-CELP Transfer Syntax	Not supported	Supported	Supported
Annex I–G.723.1 Dual Rate Speech Coder	Not supported	Supported	Supported
Transmission and reception of multiple subframes within a single Frame Relay frame	Not supported	Not supported	Not supported

End-to-End FRF.12 Fragmentation

FRF.12 fragmentation is defined by the FRF.12 standard. The FRF.12 implementation agreement enables long data frames to be fragmented into smaller pieces and interleaved with real-time frames. In this way, real-time voice and nonreal-time data frames can be carried together on lower-speed links without causing excessive delay to the real-time traffic.

Use this fragmentation type when the PVC is not carrying voice, but is sharing the link with other PVCs that are carrying voice. The fragmentation header is included only for frames that are greater than the fragment size configured. FRF.12 is the recommended fragmentation for VoIP packets.

Note VoIP packets should not be fragmented. However, VoIP packets can be interleaved with fragmented packets.

The Cisco 2600 series, 3600 series, and 7200 series routers and the Cisco MC3810 multiservice concentrator support end-to-end fragmentation on a per-PVC basis. Fragmentation is configured through a map class that applies to one or many PVCs, depending on how the class is applied.

When end-to-end FRF.12 fragmentation is used, the VoIP packets do not include the FRF.12 header, provided the size of the VoIP packet is smaller than the fragment size configured. However, when FRF.11 Annex C or Cisco proprietary fragmentations are used, VoIP packets do include the fragmentation header.

Frame Relay Fragmentation Using FRF.11 Annex C

When VoFR and fragmentation are configured on a PVC, the Frame Relay fragments are sent in the FRF.11 Annex C format. FRF.11 fragmentation is used when voice traffic is sent on the PVC, and Annex C format is used for data. With FRF.11, all data packets contain fragmentation headers, regardless of size. This form of fragmentation is not recommended for use with VoIP.

Cisco Proprietary Voice Encapsulation

Cisco proprietary voice encapsulation was implemented for the Cisco MC3810 multiservice concentrator and was used for data packets on a PVC and voice traffic. This fragmentation type is used on data packets on PVCs that carry voice traffic.

When VoFR is configured on a DLCI and fragmentation is enabled on a map class, the Cisco 7500 series router with Versatile Interface Processor (VIP) can interoperate with Cisco 2600 series, 3600 series, 7200 series, and other 7500 series routers as tandem nodes, but it cannot perform call termination with Cisco MC3810 multiservice concentrators running Cisco IOS releases *before* 12.0(3)XG or 12.0(4)T.

Map Classes and Voice Packet Queues

You must create and configure a Frame Relay map class before configuring a Frame Relay DLCI for voice traffic. The map class has configuration information about voice bandwidth, fragmentation size, and traffic shaping attributes. These attributes are required for sending voice traffic on the PVC.

Traffic Shaping

When a Frame Relay PVC is configured to support voice traffic, the carrier must be able to accommodate the traffic rate or profile sent on the PVC. If too much traffic is sent at once, the carrier might discard frames causing disruptions to real-time voice traffic. The carrier might also deal with traffic bursts by queueing up the bursts and delivering them at a metered rate. Excessive queueing also causes disruption to real-time voice traffic. Traffic shaping compensates for this condition and is necessary to prevent the carrier from discarding eligible discard bits on ingress and to prevent excessive burst data from affecting voice quality.

When the outgoing Excess Burst (Be) size is configured, the Committed Burst (Bc) size and the committed information rate (CIR) values must be obtained from the carrier. The configured values on the router must match those of the carrier.

VoFR Configuration Considerations

You must consider certain factors when configuring VoIP to ensure that it runs smoothly over Frame Relay. A public Frame Relay cloud provides no guarantees for QoS. For real-time traffic to be sent in a timely manner, the data rate must not exceed the committed information rate (CIR) or packets may be dropped. In addition, Frame Relay traffic shaping and RSVP are mutually exclusive. Remembering this is particularly important if multiple data link connection identifiers (DLCIs) are carried on a single interface.

For Frame Relay links with slow output rates (less than or equal to 64 kbps) in which data and voice are being sent over the same permanent virtual circuit (PVC), we recommend the following solutions:

- Separate DLCIs for voice and data—By providing a separate subinterface for voice and data, you can use the appropriate QoS tool for each line. For example, with each DLCI using 32 kbps of a 64-kbps line, you could do the following:
  - Apply adaptive traffic shaping to both DLCIs.
  - Use RSVP or IP Precedence to prioritize voice traffic.
  - Use compressed RTP to minimize voice packet size.
  - Use weighted fair queueing to manage voice traffic.
- Lower the maximum transmission unit (MTU) size—Voice packets are generally small. With a lower MTU size (for example, 300 bytes), large data packets can be broken up into smaller data packets that can more easily be interwoven with voice packets.

  Note Some applications do not support a smaller MTU size. If you decide to lower the MTU size, use the **ip mtu** command; this command affects only IP traffic.

  Note Lowering the MTU size affects data throughput speed.

- CIR equal to line rate—Make sure that the data rate does not exceed the CIR. One way you can make sure that the data rate does not exceed the CIR is through generic traffic shaping. For example, you could do the following:
  - Use IP precedence to prioritize voice traffic.
  - Use compressed RTP to minimize voice packet header size.
- Traffic shaping—Use adaptive traffic shaping to throttle back the output rate based on the backward explicit congestion notification (BECN). If the feedback from the switch is ignored, both data and voice packets might be discarded. Because the Frame Relay switch does not distinguish between voice and data packets, voice packets could be discarded, resulting in a deterioration of voice quality. For example, you could do the following:
  - Use compressed RTP, reduced MTU size, and adaptive traffic shaping based on BECN to hold the data rate to the CIR.
  - Use generic traffic shaping to obtain a low interpacket wait time. For example, set Bc to 4000 to obtain an interpacket wait of 125 ms.

VoFR Prerequisite Tasks

Before configuring the router for VoFR, perform the following tasks:

- Complete the company dial plan and establish a working telephony network based on the dial plan:
  - Integrate the dial plan and telephony network into the existing Frame Relay network topology. Make routing or dialing transparent to the user; for example, avoid secondary dial tones from secondary switches, where possible.
  - Contact the PBX vendor for instructions on how to reconfigure the appropriate PBX interfaces.
- Establish a working IP and Frame Relay network. For more information about configuring IP, see the “IP Overview,” “Configuring IP Addressing,” and “Configuring IP Services” chapters in the *Cisco IOS IP Configuration Guide*. For more information about configuring Frame Relay, see the *Cisco IOS Wide-Area Networking Configuration Guide.*
- Configure the required codecs and POTs dial peer configurations in “Configuring Dial Peers, Dial Plans, and Digit Manipulation” chapter.
- Configure voice ports. For more information, see the “Configuring Voice Ports” chapter.
- Configure the clock source interfaces. For more information, refer to the “Configuring Synchronous Clocking” appendix.

VoFR Configuration Task List

This section describes the following tasks:

For information regarding the configuring of voice ports and dial peers, refer to the “Configuring Voice Ports” and “Configuring Voice Dial Peers, Dial Plans, and Digit Manipulation” chapters.

Configuring Frame Relay to Support Voice

To configure Frame Relay to support voice, a map class must be applied to a single DLCI or to a group of DLCIs, depending on how the class has been applied to the virtual circuit. If there is a large number of PVCs to configure, assign the same traffic-shaping properties to the PVCs. The values for each PVC are not statically defined. Multiple map classes with different variables for each map class can also be created.

When the **frame-relay voice bandwidth** command is entered, a special queue is created for voice packets only so that time-sensitive voice packets have preference over data packets.

This section describes the configuration of map classes as follows:

- Configuring a Map Class to Support Voice Traffic, page 9
- Configuring a Map Class for Traffic-Shaping Parameters, page 10

To configure the map class to support FRF.12 fragmentation, refer to the *Cisco IOS Wide-Area Networking Configuration Guide* and *Command Reference* for more information.

Configuring a Map Class to Support Voice Traffic

When you are configuring a Frame Relay map class to support voice traffic, you must reserve the appropriate amount of voice bandwidth. If there is not enough bandwidth reserved, new calls are rejected. When calculating the amount of required voice bandwidth, include the voice packetization overhead and not just the raw compressed speech codec bandwidth.

Remember that there are a six or seven bytes of total overhead per voice packet, including standard Frame Relay headers and flags. For subchannels (CIDs) numbered less than 64, the overhead is 6 bytes. For subchannels numbered greater than or equal to 64, the overhead is 7 bytes. Add one byte if voice sequence numbers are enabled in the voice packets.

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

required_bandwidth = codec_bandwidth * (1 + overhead/payload_size)

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

When 30-ms duration voice packets are used, an approximate general rule is to add 2000 bps overhead to the raw voice compressed speech codec rate. With the 32 kbps G.726 adaptive differential pulse code modulation (ADPCM) speech coder, a 30-ms speech frame uses 120 bytes voice payload plus 6 to 7 bytes overhead, and the overall bandwidth requirement is about 34 kbps for each call.

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

	Command	Purpose
Step 1	`Router(config)#` **`map-class frame-relay`** *`map-class-name`*	Creates a map class name to assign to a group of PVCs and enters map-class configuration mode. A map class name must be unique.
Step 2	`Router(config-map-class)#` **`frame-relay voice bandwidth`** *`bps_reserved`*	Enters the bandwidth in bits per second (bps) and determines the number of voice calls enabled on the DLCIs where the map class is associated. The keywords and arguments are as follows: • *bps_reserved*—Reserved bandwidth. Valid range is from 8,000 to 45,000,000 bps. The default is 0 (disables all voice calls).

Note It is recommended that the bps be no higher than the minimum CIR if the voice quality is impacted when burst is being sent.

Configuring a Map Class for Traffic-Shaping Parameters

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

	Command	Purpose
Step 1	`Router(config-map-class)#` **`frame-relay bc out`** *`bits`*	Configures the outgoing bc size for this group of PVCs. Configure the *bits* value to a minimum of 1000 for voice traffic. Ensure that the bc size matches the carrier to prevent the carrier from discarding DE bits on ingress.
Step 2	`Router(config-map-class)#` **`frame-relay be out`** *`bits`*	Configures the outgoing be size for this group of PVCs. Ensure that the Excess Burst size matches the carrier to prevent the carrier from discarding DE bits on ingress.
Step 3	`Router(config-map-class)#` **`frame-relay min-cir`** `{`**`in`** `|` **`out`**`}` *`bps`*	Configures the minimum acceptable incoming or outgoing CIR for this group of PVCs.
Step 4	`Router(config-map-class)#` **`frame-relay cir out`** *`bits`*	Configures the outgoing excess CIR for this group of PVCs. Configured the CIR size to match your carrier to prevent the carrier from discarding DE bits on ingress.
Step 5	`Router(config-map-class)#` **`frame-relay cir in`** *`bits`*	(Optional) Configures the incoming CIR size for this group of PVCs.
Step 6	`Router(config-map-class)#` **`frame-relay adaptive shaping becn`**	(Optional) Configures the adaptive traffic rate adjustment to support backward explicit congestion notification (BECN) on this group of PVCs.

Configuring VoFR Dial Peers

To configure a VoFR dial peer, you must 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.

Note On the Cisco MC3810 multiservice concentrator, a voice class can be configured to assign idle state and out-of-service (OOS) signaling attributes to a VoFR dial peer. For more information, see the "Configuring Trunk Connections and Conditioning Features" chapter.

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

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voice`** *`number`* **`vofr`**	Defines a VoFR dial peer and enters dial peer configuration mode. All subsequent commands that are entered in dial peer voice configuration mode before exiting apply to this dial peer. The *number* argument identifies the dial peer and must be unique on the router. Do not duplicate a specific tag number.
Step 2	`Router(config-dial-peer)#` **`destination-pattern`** `[`**`+`**`]` *`string`* `[`**`T`**`]`	Configures the dial peer destination pattern. The same restrictions for the string listed in the POTS dial peer configuration also apply to the VoFR destination pattern. Also configures standard VoFR dial peers for switched calls on the tandem routers. • Plus sign (+)—(Optional) Indicates an E.164 standard number. The plus sign (+) is not supported on the Cisco MC3810 multiservice concentrator. • *string*—Specifies the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, the letters A through D, and the following special characters: – Asterisk (*) and pound sign (#) that appear on standard touch-tone dial pads. – Comma (,) inserts a pause between digits. – Period (.) matches any entered digit (this character is used as a wildcard). • **T**—(Optional) Indicates that the destination-pattern value is a variable length dial-string. **Note** Tandem-switched calls are not allowed when the call type is an FRF.11 trunk call. The Cisco 7200 series routers can serve only as tandem nodes in the VoFR network using Cisco IOS Release 12.1. This is the only dial peer procedure supported on the Cisco 7200 series.
Step 3	`Router(config-dial-peer)#` **`session target`** *`interface dlci`* `[`*`cid`*`]`	Configures the Frame Relay session target for the dial peer. **Note** The *cid* argument is required for FRF.11 trunk calls.

	Command	Purpose
Step 4	`Router(config-dial-peer)# session protocol {cisco-switched	frf11-trunk}`	(Optional) Configures the session protocol to support switched calls or FRF.11 trunk calls. If FRF.11 trunk calls are sent over the Frame Relay network, the VoFR dial peers must be statically configured on both sides of the trunk specifically to support FRF.11 trunk calls. FRF.11 trunk calls cannot be used in conjunction with dial plans or be sent through tandem nodes. **Note** The **cisco-switched** keyword is the default.
Step 5	`Router(config-dial-peer)# codec {type} [bytes payload_size]`	Specifies the voice coder rate of speech and payload size for the dial peer. The default dial peer codec is **g729r8**. The keywords and arguments are as follows: • *type*—Specifies the coder rate of speech. The rates are hardware-specific. Refer to the *Cisco IOS Voice, Video, and Fax Command Reference*. • **bytes**—(Optional) Specifies the payload size. Each codec type defaults to a different payload size if a value is not specified. • *payload_size*—(Optional) Specifies the payload size by entering the **bytes** value. Each codec type defaults to a different payload size if a value is not specified. To obtain a list of the default payload sizes, enter the **codec** command and the **bytes** option followed by a question mark (?). **Note** The Cisco MC3810 multiservice concentrator is limited to a maximum of 12 calls when using **g729r8**. Use **g729ar8** to support up to 24 calls on the Cisco MC3810 multiservice concentrator. **Note** If configuring switched voice calls on the Cisco MC3810 multiservice concentrator, configure the codec type on the voice port. **Note** For FRF.11 trunk calls, the **codec** values must be set the same on both sides of the connection.
Step 6	`Router(config-dial-peer)# dtmf-relay`	(Optional) Specifies support for the DTMF relay to improve end-to-end transport of the DTMF tones, if the codec type configured is a low bit-rate codec such as **g729** or **g723**. DTMF tones do not always propagate reliably with low bit-rate codecs. DTMF relay is disabled by default.

	Command	Purpose
Step 7	Router(config-dial-peer)# **signal-type** {**cas** | **cept** | **ext-signal** | **transparent**}	If Cisco trunk permanent calls are being configured, the signal type is required. The signal type defines the ABCD signaling packets that are generated by the voice port and sent to the data network. Use the **cas**, **cept**, **ext-signal**, and **transparent** keywords. To configure FRF.11 calls, use only the **cas** and **ext-signal** keywords. These keywords are optional on Cisco 2600/3600 series routers and configure the signal type on these routers for FXS-FXS trunks. The keywords are as follows: • **cas**—Default signaling type that is North American CAS/robbed-bit signaling. • **cept**—Provides basic E1 ABCD protocol, primarily Conférence Européenne des Postes et des Télécommunications (CEPT) E&M signaling, on the Cisco MC3810 multiservice concentrator. This keyword is used for European voice networks. If the keyword is used with FXS or FXO voice ports, the signaling is equivalent to Mercury Exchange Limited (MEL) CAS. The keyword is not supported on the Cisco 2600/3600 series. • **ext-signal**—Used for required external signaling channels (for example, common channeling signaling), or when no signaling information is sent over a permanent “dumb” voice pipe (for example, carrying audio for a public address system).

	Command	Purpose
		• **transparent**—Used on the Cisco MC3810 multiservice concentrator with *digital* voice ports when the ABCD signaling bits are copied and passed transparently from the T1/E1 interface without interpretation (also known as transparent FRF.11 signaling). The keyword enables the Cisco MC3810 multiservice concentrator to handle or transport unknown signaling protocols. On the Cisco MC3810 multiservice concentrator with *analog* voice ports, the **transparent** keyword does not apply and is equivalent to the **cept** keyword. This keyword is not supported on the Cisco 2600 series and 3600 series in Cisco IOS Release 12.2. **Note** By default, the Cisco MC3810 multiservice concentrator, when configured using **transparent**, operates the voice path in a permanently open state so that voice packets are sent (and network bandwidth consumed) regardless of the state of the call. The signal type must be configured in such a way that the signal type is the same at both ends of the permanent voice call. When a permanent connection is configured between a T1/E1 Cisco MC3810 multiservice concentrator and an analog voice port on a Cisco 2600 or Cisco 3600 series routers, the signal type should be set to **cas**, which is the default.
Step 8	`Router(config-dial-peer)# called-number termination-string`	Required for the Cisco 2600/3600 series routers only. Configures the termination string for FRF.11 trunk calls. This command is required to enable the router to establish an incoming trunk connection. This command applies only when the **session protocol** command is set to **frf11-trunk**. **Note** Although this command is visible on the Cisco MC3810 multiservice concentrator, the command is disabled.
Step 9	`Router(config-dial-peer)# no vad`	(Optional) Disables VAD on the dial peer. This command is enabled by default.
Step 10	`Router(config-dial-peer)# sequence-numbers`	(Optional) Enables the voice sequence number if required for your configuration. This command is disabled by default.

	Command	Purpose
Step 11	`Router(config-dial-peer)#` **`preference`** *`value`*	(Optional) Configures a preference for the VoFR dial peer. The *value* argument is a number from 0 to 10 where the lower the number, the higher the preference in hunt groups.
Step 12	`Router(config-dial-peer)#` **`fax rate`** `{`**`2400`** `|` **`4800`** `|` **`7200`** `|` **`9600`** `|` **`14400`** `|` **`disable`** `|` **`voice`**`}`	(Optional) Configures 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.

To configure another VoFR dial peer, exit dial peer configuration mode and repeat Steps 1 through 10.

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

Configuring Switched Calls

To configure switched calls on Cisco 2600, 3600, and 7200 series routers and Cisco MC3810 multiservice concentrators, use the following commands beginning in interface configuration mode:

	Command	Purpose
Step 1	Router(config-if)# **frame-relay interface-dlci** *dlci*	Enters the DLCI configuration mode.
Step 2	Router(config-fr-dlci)# **vofr** [**data** *cid*] [**call-control**] [*cid*]	Configures the Frame Relay DLCI to support VoFR. When the **vofr** command is used, all subchannels on the DLCI are configured for FRF.11 encapsulation The keywords and arguments are as follows: • **data**—Selects a subchannel (CID) for data other than the default subchannel (CID 4). The recommended setting is **vofr data** *4* **call-control** *5*. • *cid*—Specifies the subchannel to use for data. Valid values are from 4 to 255. The default is 4. If data is specified, a valid CID must be entered. • **call-control**—(Optional) Specifies that a subchannel is reserved for call-control signaling. Call-control is not supported on Cisco MC3810 multiservice concentrators. • *cid*—(Optional) Specifies the subchannel to use for call-control signaling. Valid values are from 4 to 255. The default is 5. If call-control is specified and a CID is not entered, the default CID is used. If the **vofr** command is entered without any keywords or arguments, the data subchannel (*cid*) is 4 and there is no call-control subchannel. **Note** The **vofr** command uses WFQ at the PVC level. If the **vofr cisco** command is used, WFQ cannot be disabled.
	or Router(config-fr-dlci)# **vofr cisco**	Configures the DLCI and the Cisco proprietary voice encapsulation for switched calls to Cisco MC3810 multiservice concentrators. When this command is entered, data CID 4 and call-control CID 5 are automatically assigned. If user-dialed calls are being configured, stop here. If auto-ringdown calls are being configured, continue to the next step.
Step 3	Router(config)# **voice-port**	Identifies the voice port to configure and enters the voice-port configuration mode. **Note** The **voice-port** command is hardware specific. For more information, refer to the *Cisco IOS Voice, Video, and Fax Command Reference*.
Step 4	Router(config-voice-port)# **connection** [**plar** | **tie-line**] *destination-string*	Configures the private-line, auto-ringdown (PLAR) or tie-line connection, specifying the telephone number in the *destination-string*.

Table 31 lists the supported VoFR connections and the appropriate commands to configure switched calls.

Table 31 Supported VoFR Connections for Switched Calls

Switched Calls (User-Dialed or Auto-Dialed)	Data Fragmentation Supported	Frame Relay DLCI Command[1]	Session Protocol Command[2]	Voice Port Command
To routers supporting VoFR	FRF.11 Annex C	**vofr** [**data** *cid*] [**call-control** [*cid*]][3]	**session protocol cisco-switched**[4]	For user-dialed calls: none For auto-ringdown calls: **connection plar** *destination-string*
To a Cisco MC3810 multiservice concentrator running Cisco IOS Releases before 12.1(2)T	Cisco proprietary[5]	**vofr cisco**[6]	**session protocol cisco-switched**	For user-dialed calls: none For auto-ringdown calls: **connection plar** *destination-string*

1. The **voice-encap** option of the **frame-relay interface-dlci** command on the Cisco MC3810 multiservice concentrator is no longer supported.
2. Dial peer configuration mode.
3. The recommended use of this command is **vofr data 4 call-control 5**.
4. The **session protocol cisco-switched** command is the default setting. If the command is not entered, the setting still applies.
5. Cisco proprietary fragmentation is based on an early draft of FRF.12 and is compatible with Cisco MC3810 multiservice concentrators.
6. This command uses data CID 4 and call-control CID 5.

Tandem Switching of Switched Calls

Depending on which router is the end node and which is the tandem node, the correct Frame Relay PVC type must be configured. Table 32 shows the router combinations that can serve as end and tandem nodes and the command that is required to enable VoFR.

Table 32 VoFR End and Tandem Node Combinations

End Node	Tandem Node	Required VoFR Command
Cisco 2600, Cisco 3600, or Cisco 7200 and Cisco MC3810 multiservice concentrator	Cisco 2600, Cisco 3600, or Cisco 7200 and Cisco MC3810 multiservice concentrator	**vofr call-control**
Cisco 2600 or Cisco 3600 and Cisco MC3810 multiservice concentrator	Cisco MC3810 multiservice concentrator running Cisco IOS releases before 12.1(2)T	**vofr cisco**
Cisco MC3810 multiservice concentrator running Cisco IOS releases before 12.1(2)T	Cisco 2600, Cisco 3600, or Cisco 7200	**vofr cisco**

Note When you are creating voice networks with a mixture of router types, the Cisco MC3810 multiservice concentrator must be running Cisco IOS Release 12.0(3)XG, 12.0(4)T, or later releases, to act as a tandem node. For each configured tandem node, two VoFR dial peers must be configured, one for each tandem connection.

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

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voice`** *`number`* **`vofr`**	Defines a VoFR dial peer and enters dial peer configuration mode. All subsequent commands that are entered in dial peer voice configuration mode before exiting apply to this dial peer.
Step 2	`Router(config-dial-peer)#` **`destination-pattern`** `[+]`*`string`*`[T]`	Configures the dial peer destination pattern. The same restrictions for the string listed in the POTS dial peer configuration also apply to the VoFR destination pattern.
Step 3	`Router(config-dial-peer)#` **`session target`** *`interface dlci`*	Configures the Frame Relay session target for the dial peer.
Step 4	`Router(config-dial-peer)#` **`preference`** *`value`*	(Optional) Configures a preference for the VoFR dial peer. The *value* argument is a number from 0 to 10 where the lower the number, the higher the preference in hunt groups.

To configure the next VoFR dial peer, exit dial peer configuration mode by entering **exit**, and repeat Steps 1 through 4. On tandem nodes, at least two VoFR dial peers are required, one for each call leg.

Configuring Cisco Trunk Calls

Before configuring the Cisco trunk calls, consider the following restrictions and recommendations:

- VoFR dial peers must be configured to send Cisco trunk calls over the Frame Relay network. Cisco trunk calls are permanent calls. One critical task is configuring the signal type for the dial peer. It must be the same at both ends of the permanent voice call. See the "Configuring Dial Peers, Dial Plans, and Digit Manipulation" chapter for more information.
- When a permanent connection between a T1/E1 Cisco MC3810 multiservice concentrator and an analog voice port on a Cisco 2600 or Cisco 3600 series routers is configured, the default signal type is **cas**.
- Use of Cisco trunks for permanent calls 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.

Table 33 lists the supported VoFR connections and the commands to enter.

Table 33 VoFR Connections for Cisco Trunk Calls

Cisco Trunk Calls	Data Fragmentation Supported	VoFR Command	Session Protocol Command[1]	Voice Port Command
To routers supporting VoFR	FRF.11 Annex C	**vofr data** *cid* **call-control** *cid*	**session protocol cisco-switched**	**connection trunk** *destination-string* [**answer mode**]
To a Cisco MC3810 multiservice concentrator running Cisco IOS Releases before 12.0(7) XK and 12.1(2)T	Cisco proprietary	**vofr cisco**[2]	**session protocol cisco-switched**	**connection trunk** *destination-string* [**answer mode**]

1. The **session protocol cisco-switched** command, whether entered or not, is the default setting.
2. When the **cisco** keyword is entered, Cisco proprietary data implementation is enabled. This implementation is used only for backward compatibility to earlier releases.

To configure Cisco trunk permanent calls, use the following commands beginning in interface configuration mode:

	Command	Purpose
Step 1	`Router(config-if)#` **`frame-relay interface-dlci`** *`dlci`*	Configures the DLCI to support VoFR. **Note** The **voice-encap** option of the **frame-relay interface-dlci** command on the Cisco MC3810 multiservice concentrator is no longer supported beginning in Cisco IOS 12.2.
Step 2	`Router(config-if)#` **`vofr`** `[[`**`cisco`**`] | [[`**`data`** *`cid`*`] [`**`call-control`**`] [`*`cid`*`]]]]`	Enables VoFR on the DLCI. If the **vofr** command is entered without any keywords or arguments, the data subchannel is CID 4, and there is no call-control subchannel. **Note** When the **vofr** command is used, all subchannels on the DLCI are configured for FRF.11 encapsulation. This configuration uses the standard FRF.11 Annex C fragmentation. The **vofr** command uses WFQ at the PVC level. If the **vofr cisco** command is used, WFQ cannot be disabled. If only tandem calls are being configured, stop here, otherwise proceed to Step 3.
Step 3	`Router(config]#` **`voice-port`**	Identifies the voice port to configure and enters voice-port configuration mode. **Note** The **voice-port** command is hardware specific. See the *Cisco IOS Voice, Video, and Fax Command Reference Guide* for more information.
Step 4	`Router(config-voice-port)#` **`connection trunk`** *`destination-string`* `[`**`answer-mode`**`]`	Configures the trunk connection by specifying the telephone number in *destination-string*. One side must be the call initiator (master) and the other side is the call answerer (slave). By default, the voice port is the master. The **answer-mode** keyword specifies the voice port that operates in slave mode.
Step 5	`Router(config-voice-port)#` **`shutdown`**	Shuts down the voice port.
Step 6	`Router(config-voice-port)#` **`no shutdown`**	Reactivates the voice port to enable the trunk connection.

Note When the **connection trunk** or **no connection trunk** command is entered, the voice port must be toggled by entering **shutdown**, and then **no shutdown** before the changes take effect.

Configuring FRF.11 Trunk Calls

On the Cisco MC3810 multiservice concentrators and Cisco 2600 and 3600 series routers, FRF.11 trunk calls to a second router can be configured, except tandem FRF.11 trunk calls. Configuring FRF.11 trunk calls to a second router requires that the **session protocol** dial peer configuration command be set to **frf11-trunk**.

Table 34 lists the supported VoFR connections and the required commands to configure FRF.11 trunk calls.

Table 34 VoFR Connections for FRF.11 Trunk (Private-Line) Calls

FRF.11 Trunk Calls	Data Fragmentation Supported	VoFR DLCI Command[1]	Session Protocol Command	Voice Port Command
To routers supporting VoFR	FRF.11 Annex C	**vofr** [**data** *cid*] [**call-control** *cid*][2]	**session protocol frf11-trunk**	**connection trunk** *destination-string* [**answer mode**]

1. Dial peer configuration mode.
2. For FRF.11 trunk calls, the call-control option is not required. It is required only if you mix FRF.11 trunk calls with other types of voice calls on the same PVC.

To configure FRF.11 trunk calls, use the following commands beginning in interface configuration mode:

	Command	Purpose
Step 1	`Router(config-if)# frame-relay interface-dlci dlci`	Configures the DLCI and enters DLCI configuration mode.
Step 2	`Router(config-fr-dlci)# vofr [data cid] [call-control cid]`	Configures the DLCI and optionally enters the data and call-control CIDs. When the keywords and arguments are configured, all subchannels on the DLCI are configured for FRF.11 encapsulation except the data subchannel. If no keywords or arguments are entered, the data subchannel is CID 4, and there is no call-control subchannel.
Step 3	`Router(config)# voice-port`	Identifies the voice port to configure and enters voice-port configuration mode. **Note** The **voice-port** command is hardware specific. Refer to the *Cisco IOS Voice, Video, and Fax Command Reference* publication for more information.
Step 4	`Router(config-voice-port)# connection trunk destination-string [answer-mode]`	Configures the trunk connection by specifying the telephone number in *destination-string*. One side of a call must act as the call initiator (master) and the other side as the call answerer (slave). By default, the voice port is the master.
Step 5	`Router(config-voice-port)# shutdown`	Shuts down the voice port.
Step 6	`Router(config-voice-port)# no shutdown`	Reactivates the voice port to enable the trunk connection.

> **Note** When the **connection trunk** or **no connection trunk** command is entered, the voice port must be toggled by entering **shutdown**, and then **no shutdown** before the changes take effect.

Verifying the Voice Connections

To verify switched calls voice connections, perform the following tasks:

- Pick up the telephone handset and verify that there is a dial tone.
- Call from a local telephone to the configured dial peer and verify that the call completes.

To verify the FXO-FXS trunk calls to a remote PBX, perform the following tasks:

- Pick up the telephone and listen for a dial tone from the remote PBX.
- Dial a telephone number, so that the remote PBX routes the call.

To verify the voice connections, perform the following tasks:

- Check the validity of the dial peer and voice port configuration by performing the following tasks:
  - Enter the **show dial-peer voice** command to verify that the data configured is correct.
  - Enter the **show dial-peer voice summary** command to check the validity of the dial peer configurations.
  - Enter the **show voice port** command to show the status of the voice ports.
  - Enter the **show call active voice** with the keyword **brief** to show the call status for all voice ports.
  - Enter the **show voice call** command to check the validity of the voice port configuration.
  - Enter the **show voice dsp** command to show the current status of all DSP voice channels.
  - Enter the **show voice permanent** command to show the status of Cisco trunk permanent calls.
  - Enter the **show call history** command to show the active call table.
- Check the validity of the VoFR configuration on the DLCI by performing the following task:
  - Enter the **show frame-relay vofr** [*interface* [*dlci* [*cid*]]] command to show the VoFR configuration. This command is not supported on the Cisco MC3810 multiservice concentrator when the **vofr cisco** command is configured.

Verifying the Frame Relay Configuration

Check the validity of the configuration by performing the following tasks:

- Enter the **show frame-relay pvc** command to show the status of the PVCs.
- Enter the **show frame-relay vofr** command with the arguments *interface, dlci, and cid* to show statistics and information on the open subchannels. This command does not display if the **vofr cisco** command is entered on the Cisco MC3810 multiservice concentrator.
- Enter the **show frame-relay fragment** command with the arguments *interface number* and *dlci* to show the Frame Relay fragmentation configuration.
- Enter the **show traffic-shape queue** command to display the traffic-shaping information if Frame Relay traffic shaping is configured. The **queue** option displays the queueing statistics.

Troubleshooting Tips

To troubleshoot and resolve configuration issues, perform the following tasks:

- If no calls are going through, ensure that the **frame-relay voice bandwidth** command is configured.
- If VoFR is configured on a PVC and there are problems with data connectivity on that PVC, ensure that the **frame-relay fragment** command has been configured.
- If data is not being transmitted but fragmentation is configured, ensure that Frame Relay traffic shaping is turned on.
- If the problem is with the dial plan or the dial peers, use the **show dial-plan number** command with the argument *dial string* to display which dial peers are being used when a specific number is called.
- If there are problems connecting an FRF.11 trunk call, ensure that the **session protocol** dial peer command is set to **frf11-trunk**.
- If FRF.11 trunk calls on the Cisco 2600 or Cisco 3600 series routers are being configured, 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.
- Ensure that the voice port is set to **no shutdown**.
- Ensure that the serial port or the T1/E1 controller is set to **no shutdown**.
- Toggle the voice port by first entering **shutdown**, and then **no shutdown** every time the **connection trunk** or **no connection trunk** command is entered.

Monitoring and Maintaining the VoFR Configuration

To monitor and maintain the VoFR configuration, use the following commands in EXEC mode as needed:

Command	Purpose
`Router#` **`show call active voice`** `[`**`brief`**`]`	Displays the active call table.
`Router#` **`show call history voice`** `[`**`last`** *`number`*`]	[`**`brief`**`]` or `Router#` **`show call history voice record`**	Displays the call history table.
`Router#` **`show dial-peer voice`**	Displays configuration information and call statistics for dial peers.
`Router#` **`show frame-relay fragment`**	Displays information about the Frame Relay fragmentation taking place in the Cisco router.
`Router#` **`show frame-relay pvc`**	Displays statistics about PVCs for Frame Relay interfaces.
`Router#` **`show frame-relay vofr`**	Displays the FRF.11 subchannels information on VoFR DLCIs.
`Router#` **`show interfaces serial`**	Displays information about a serial interface.
`Router#` **`show traffic-shape queue`**	Displays information about the elements queued at a particular time at the VC (DLCI) level.

Command	Purpose
`Router# show voice call`	Displays the call status for all voice ports on the Cisco MC3810 multiservice concentrators.
`Router# show voice permanent-call`	Displays information about the permanent calls on a voice interface.

VoFR Configuration Examples

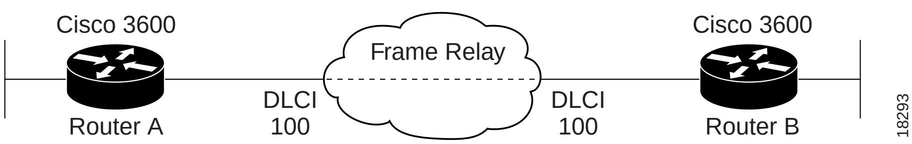
This section provides specific configuration examples for VoFR connections and includes:

- Two Routers Using Frame Relay Fragmentation Example, page 25
- Two Routers Using a VoFR PVC Example, page 26
- Router Using VoFR PVCs Connected to Cisco MC3810s Before 12.1(2)T Example, page 26
- Cisco Trunk Calls Between Two Routers Example, page 27
- FRF.11 Trunk Calls Between Two Routers Example, page 28
- Tandem Configuration Examples, page 29
- Cisco Trunk Call with Hunt Groups Example, page 34

Two Routers Using Frame Relay Fragmentation Example

Figure 92 shows an example of Frame Relay fragmentation between two routers. This configuration uses FRF.12 fragmentation.

Figure 92 Two Routers Using Frame Relay Fragmentation



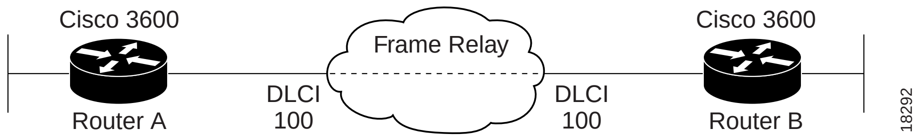
Router A	Router B

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

Two Routers Using a VoFR PVC Example

Figure 93 shows an example of two routers that use FRF.11 Annex C fragmentation with connections using a VoFR PVC.

Figure 93 Two Cisco 3600 Series Routers Using a VoFR PVC



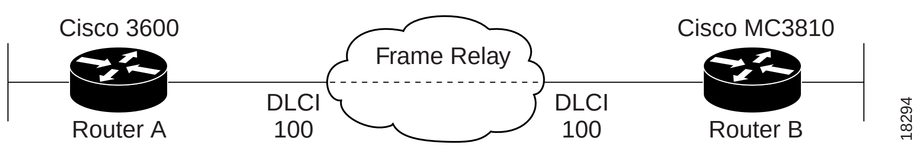
Router A	Router B

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

Router Using VoFR PVCs Connected to Cisco MC3810s Before 12.1(2)T Example

Figure 94 shows an example of a Cisco 3600 series router with connections to a Cisco MC3810 multiservice concentrator running a Cisco IOS release before12.1(2)T. In this example, the VoFR interface on both the Cisco 3600 series router and the Cisco MC3810 multiservice concentrator is configured by using the **vofr cisco** command. This configuration uses FRF.11 Annex C fragmentation.

Figure 94 Router Using VoFR PVCs Connected to a Cisco MC3810 Multiservice Concentrator

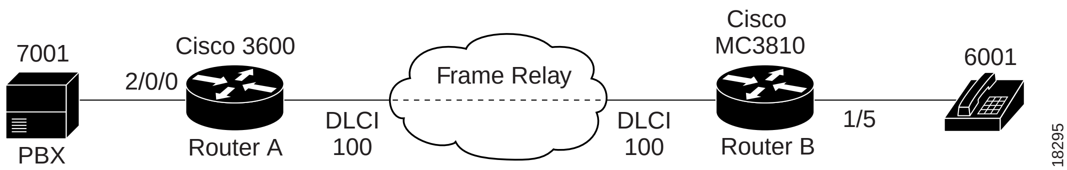


Router A	Router B
interface serial 0/0	interface serial 0
ip address xxx.xxx.xxx 255.255.255.0	ip address xxx.xxx.xxx 255.255.255.0
frame-relay traffic shaping	frame-relay traffic shaping
!	frame-relay class *toto*
frame-relay interface-dlci 100	!
vofr cisco	frame-relay interface-dlci 100
class toto	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

Cisco Trunk Calls Between Two Routers Example

Figure 95 shows an example of VoFR Cisco trunk calls between two routers.

Figure 95 Cisco Trunk (Private-Line) Calls Between Two Routers



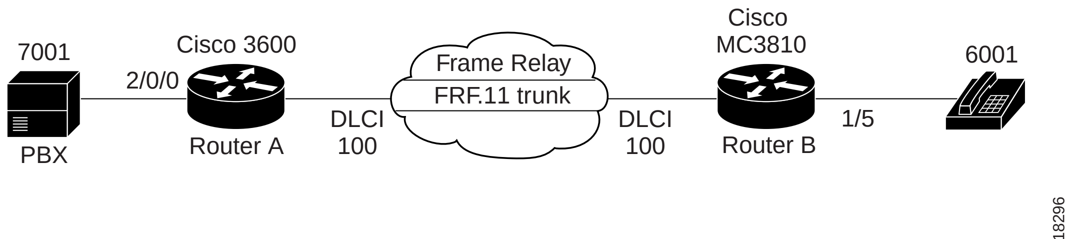
Router A	Router B
interface serial 0/0	interface serial 0
ip address xxx.xxx.xxx 255.255.255.0	ip address xxx.xxx.xxx 255.255.255.0
encapsulation frame-relay	encapsulation frame-relay
frame-relay traffic shaping	frame-relay traffic shaping
frame-relay interface-dlci 100	frame-relay interface-dlci 100
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

Router A	Router B
`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 Example

Figure 96 shows an example of FRF.11 trunk calls configured between two routers.

Figure 96 FRF.11 Trunk Calls Between Two Routers

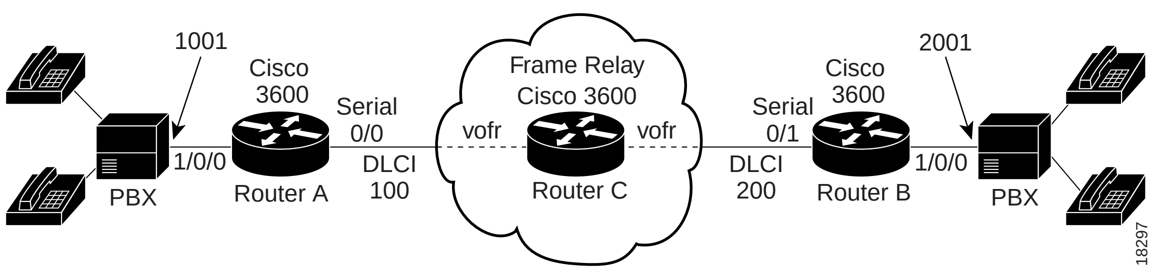


Router A	Router B
hostname 3600A ! interface serial 0/0 ip address xxx.xxx.xxx 255.255.255.0 encapsulation frame-relay frame-relay traffic shaping frame-relay interface-dlci 100 class voice vofr data 4 ! map-class frame-relay voice frame-relay cir s frame-relay min-cir in x frame-relay bc u frame-relay voice bandwidth v frame-relay fragment y ! voice-port 2/0/0 connection trunk 6001 ! dial-peer voice 1 pots destination-pattern 7001 port 2/0/0 ! dial-peer voice 2 vofr codec x bytes y destination-pattern 6001 session protocol frf11-trunk session target Sn 100 d called-number 7001 dtmf-relay vad	hostname mc3810B ! interface serial 0 ip address xxx.xxx.xxx 255.255.255.0 encapsulation frame-relay frame-relay traffic shaping frame-relay interface-dlci 100 class voice vofr data 4 ! map-class frame-relay voice frame-relay cir s frame-relay min-cir in x frame-relay bc u frame-relay voice bandwidth v frame-relay fragment y ! voice-port 1/5 connection trunk 7001 ! dial-peer voice 2 pots destination-pattern 6001 port 1/5 ! dial-peer voice 4 vofr codec x bytes y destination-pattern 7001 session protocol frf11-trunk session target Sn 100 d dtmf-relay vad

Tandem Configuration Examples

Figure 97 shows an example of a tandem configuration with two Cisco 3600 series routers as endpoints and a third Cisco 3600 series router as a tandem node.

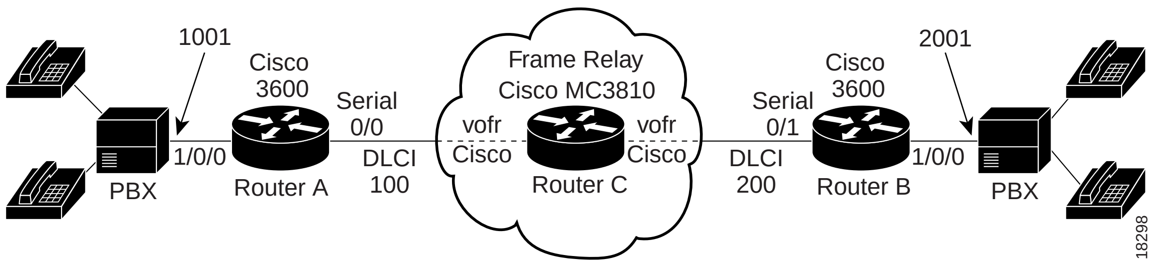
Figure 97 Tandem Configuration with Three Routers for Switched Calls



Router A Endpoint	Router C Tandem Node	Router B Endpoint
hostname 3600A	hostname3600C	hostname3600B
!	!	!
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	!
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 0/0 200
!	dial-peer voice 1 vofr	!
voice-port 1/0/0	destination-pattern 1...	voice-port 1/0/0
	session target serial 0/0 100	
	!	
	dial-peer voice 2 vofr	
	destination-pattern 2...	
	session target serial 0/1 200	

Figure 98 shows an example of a tandem configuration with a Cisco MC3810 multiservice concentrator acting as a tandem node.

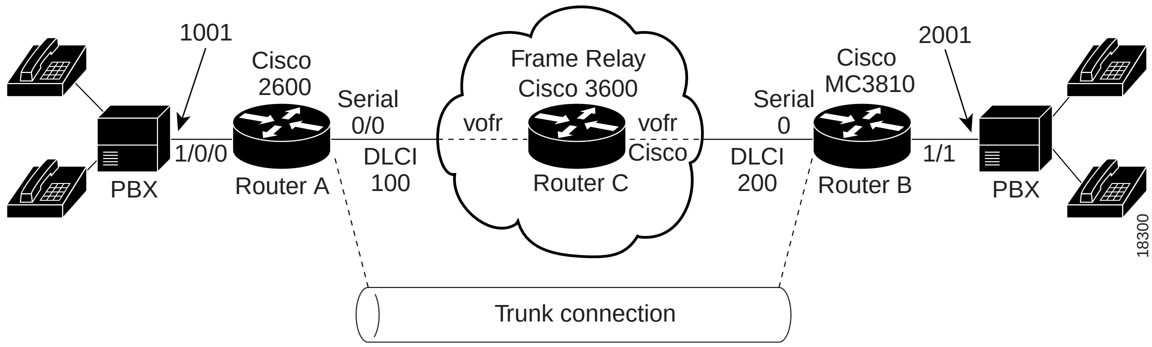
Figure 98 Tandem Configuration with a Cisco MC3810 Multiservice Concentrator Tandem Node for Switched Calls



Router A Endpoint	Router C Tandem Node	Router B 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	!

Figure 99 shows an example of a tandem configuration with a Cisco MC3810 multiservice concentrator acting as an endpoint node for Cisco trunk calls. When a Cisco MC3810 multiservice concentrator is on a VoFR network, the configuration for connections to and from the Cisco MC3810 multiservice concentrator is slightly different than for other routers that support VoFR. The **vofr cisco** command is required for those connections.

Figure 99 Tandem Configuration with a Cisco MC3810 Multiservice Concentrator Endpoint Node

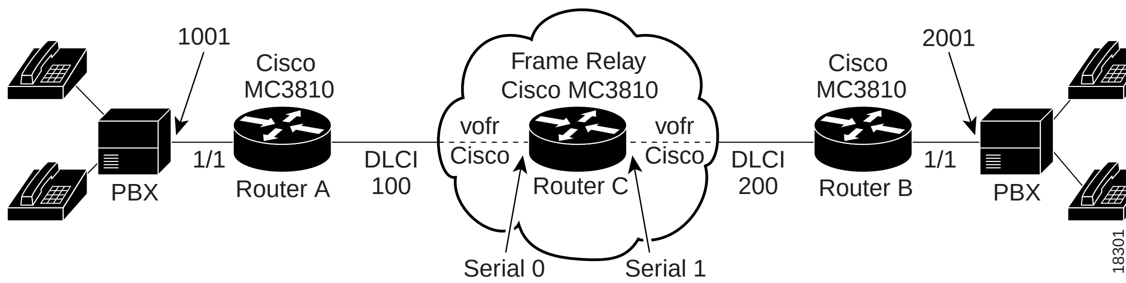


Router A Endpoint	Router C Tandem Node	Router B Endpoint
`interface serial 0/0`	`interface serial 0/0`	`interface serial 0`
`encapsulation frame-relay`	`encapsulation frame-relay`	`encapsulation frame-relay`
`frame-relay traffic-shaping`	`frame-relay traffic-shaping`	`frame-relay traffic-shaping`
`frame-relay 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`
`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`	`!`

Figure 100 shows an example of a tandem configuration with Cisco MC3810 multiservice concentrators as both endpoint and tandem nodes.

Note When a Cisco MC3810 multiservice concentrator running Cisco IOS software releases earlier than 12.1(2)T are used on a VoFR network, the configuration for connections to and from that Cisco MC3810 multiservice concentrator is slightly different from what is used for other routers that support VoFR. The **vofr cisco** command is required for these connections on the Cisco MC3810 multiservice concentrator.

Figure 100 Configuration with All Cisco MC3810 Multiservice Concentrators as Endpoint and Tandem Nodes



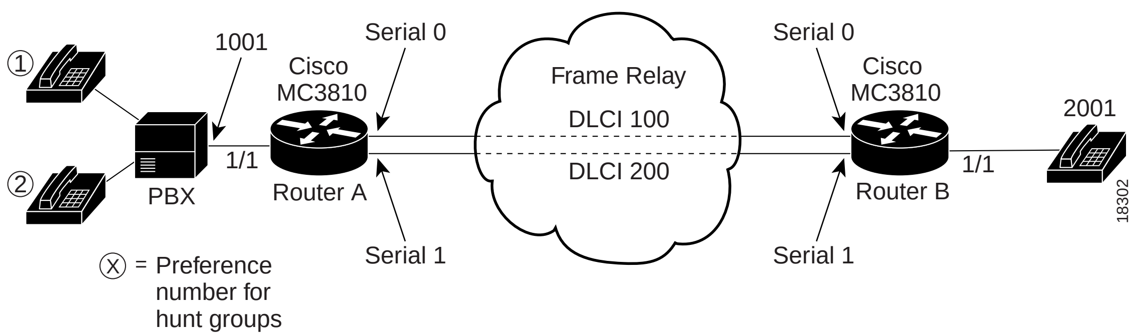
Router A Endpoint	Router C Tandem Node (Cisco IOS Releases Before 12.1(2)T)	Router B Endpoint
interface serial 0	interface serial 0	interface serial 0
encapsulation frame-relay	encapsulation frame-relay	encapsulation frame-relay
frame-relay traffic-shaping	frame-relay traffic-shaping	frame-relay traffic-shaping
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...
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 Example

Figure 101 shows an example of a Cisco trunk call with hunt groups configured. 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 one dial peer has a higher preference, so the call setup is attempted through that dial peer. If the call setup fails, the master can continue attempting call setups 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 101 Cisco Trunk Call with Hunt Groups



Router A	Router B
`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`
`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...`

Router A	Router B
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
!	!
!	!