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

Feature Overview

The Voice over Frame Relay (VoFR) capabilities that were introduced on the Cisco MC3810 multiservice access concentrator beginning with Cisco IOS Release 11.3 are now extended to the Cisco 2600 series, 3600 series, and 7200 series router platforms.

The following new functionality is supported in Release 12.0(3)XG and 12.0(4)T:

- FRF.11-compliant VoFR trunking
- FRF.12-compliant end-to-end fragmentation

> **Note** This feature supports both FRF.12-compliant fragmentation in addition to the proprietary Cisco fragmentation introduced in Cisco IOS Release 11.3.

- Dynamic call switching and termination
- Permanent trunk connections (private line) via dynamic switched call path

When VoFR is configured on a Cisco router, the router is able to carry voice traffic such as telephone calls and faxes over a Frame Relay network.

This document describes how to configure VoFR on the Cisco routers that support this feature. It is assumed you have already configured your Frame Relay backbone network. As part of your Frame Relay configuration, you need to configure the map class and the Local Management Interface (LMI) among other Frame Relay functionality. For more information about basic Frame Relay configuration, see the *Wide-Area Networking Configuration Guide*.

Benefits

The Cisco implementation of Voice over Frame Relay provides the following benefits to existing Frame Relay networks:

- Enables real-time, delay-sensitive voice traffic to be carried over slow Frame Relay links
- Allows dedicated 64-kbps Time-Division Muliplexing (TDM) telephony circuits to be replaced by more economical Frame Relay permanent virtual circuits
- Allows voice-enabled routers from multiple remote sites to be multiplexed into a central site router through Frame Relay links
- Utilizes voice compression technology that conforms to ITU-T specifications

- Enables Cisco 2600 series, 3600 series, and 7200 series routers and the Cisco MC3810 series multiservice access concentrators to support Frame Relay fragmentation
- Allows intelligent setup of proprietary switched VoFR connections between two VoFR endpoints, saving the extensive configuration overhead associated with pure FRF.11 implementations
- Supports standard FRF.11 functionality, allowing Cisco routers to interconnect with other equipment supporting this specification

Restrictions

The following restrictions and limitations apply to Voice over Frame Relay:

- In order for VoFR on the Cisco 2600 series, 3600 series, and 7200 series routers to interoperate with VoFR on the Cisco MC3810, the Cisco MC3810 must be running Cisco IOS Release 12.0(3)XG or 12.0(4)T or later.

  Note The Cisco MC3810 first supported VoFR in Cisco IOS Release 11.3(1)MA. The Cisco 2600 series, 3600 series, and 7200 series routers can tandem-switch voice calls generated by the Cisco MC3810 running Release 11.3(1)MA or later.

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

Supported Platforms

- Cisco 2600 series routers
- Cisco 3600 series routers
- Cisco MC3810 series multiservice access concentrators
- Cisco 7200 series routers (tandeming only)

Prerequisites

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

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

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

Supported MIBs and RFCs

None

List of Terms and Acronyms

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

ADPCM—Adaptive differential pulse code modulation. A process by which analog voice samples are encoded into high-quality digital signals.

Call leg—A logical connection between the router and either a telephony endpoint over a bearer channel, or another endpoint using a session protocol.

CELP—Code excited linear prediction. A compression algorithm used in low bit-rate voice encoding. CELP is used in ITU-T Recommendations G.728, G.729, and G.723.1.

CEPT—Conférence Européenne des Postes et des Télécommunications. Association of the 26 European PTTs that recommends communication specifications to the ITU-T.

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

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

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

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

CS-ACELP—Conjugate structure algebraic code excited linear prediction. A CELP voice compression algorithm specified in ITU-T Recommendation G.729, providing 8 kbps, or 8:1 compression.

DLCI—Data-link connection identifier.

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

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

DTMF—Dual tone multifrequency. Use of two simultaneous voice-band tones for dial (such as touch tone).

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

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

E&M—Stands for recEive and transMit (or Ear and Mouth). E&M is a trunking arrangement generally used for two-way switch-to-switch or switch-to-network connections. Cisco's analog E&M interface is an RJ-48 connector that allows connections to PBX trunk lines (tie lines). E&M is also available on E1 and T1 digital interfaces.

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

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

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

FRF.11 Annex C—See FRF.12.

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

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

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

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

G.711—Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through PBXs. Described in the ITU-T standard in its G-series recommendations.

G.723.1—Describes a compression technique that can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This CODEC has two bit rates associated with it: 5.3 and 6.3 kbps. The higher bit rate is based on ML-MLQ technology and provides a somewhat higher quality of sound. The lower bit rate is based on CELP and provides system designers with additional flexibility. Described in the ITU-T standard in its G-series recommendations.

G.726—Describes ADPCM coding at 40, 32, 24, and 16 kbps. ADPCM-encoded voice can be interchanged between packet voice, PSTN, and PBX networks if the PBX networks are configured to support ADPCM. Described in the ITU-T standard in its G-series recommendations.

G.728—Describes a 16-kbps low-delay variation of CELP voice compression. CELP voice coding must be translated into a public telephony format for delivery to or through the PSTN. Described in the ITU-T standard in its G-series recommendations.

G.729—Describes CELP compression where voice is coded into 8-kbps streams. There are two variations of this standard (G.729 and G.729 Annex A) that differ mainly in computational complexity; both provide speech quality similar to 32-kbps ADPCM. Described in the ITU-T standard in its G-series recommendations.

hookflash—A short on-hook period usually generated by a telephone-like device during a call to indicate that the telephone wishes to perform dial-tone recall from a PBX. Hookflash is often used to perform call transfer.

LD-CELP—Low-delay CELP. A CELP voice compression algorithm specified in ITU-T Recommendation G.728, providing 16 kbps, or 4:1 compression.

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

OOS—Out of Service signaling.

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

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

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

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

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

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

PVC—Permanent virtual circuit.

SVC—Switched virtual circuit.

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

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

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

UIO—Universal I/O serial port (Cisco router).

VAD—Voice Activity Detection. When VAD is enabled on voice port or a dial peer, silence is not transmitted over the network, only audible speech. When VAD is enabled, the sound quality is slightly degraded but the connection monopolizes much less bandwidth.

VoFR—Voice over Frame Relay.

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

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

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

VoIP—Voice over IP.

Functional Description

Cisco's VoFR implementation allows the following types of VoFR calls:

- Static FRF.11 trunks
- Switched VoFR calls:
  - Dynamic switched calls
  - Cisco-trunk (private line) calls

This section describes the setup of Cisco VoFR calls. In addition, the following functionality is described:

- FRF.11-compliant speech encoding and packetization
- FRF.12-based end-to-end fragmentation under Frame Relay
- Permanent trunks over dynamic switched calls
- Tandem switching of calls from one VoFR dial peer to another VoFR dial peer

Static FRF.11 trunks and permanent switched trunks are used to create fixed point-to-point connections, which are typically used to connect two PBXs. In this case, the VoFR system simply provides transportation of the voice connection channels, but does not provide dial-plan-based telephone call switching. This functionality is sometimes referred to as "tie-line emulation." In this scenario, all telephone call switching is performed by the PBXs.

With dynamic switched calls, the VoFR system includes dial-plan information that is used to process and route the calls based on the telephone numbers dialed by the callers. The dial-plan information is contained within dial-peer entries.

Static FRF.11 Trunks

FRF.11 trunks allow for standards-based vendor interoperability by specifying the frame format and coder types to be used when transmitting voice traffic through a Frame Relay network; however, FRF.11 includes no 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 must be 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.

There is no possibility of automatic enforcement of compatible configuration parameters between the two ends of an FRF.11-based call. For example, it is possible to incorrectly configure the two ends of an FRF.11 call using different and incompatible speech compression CODECs. In this situation, the call will exist and voice packets will be transmitted and received, but no usable voice path will be created.

When configured, a static FRF.11 trunk remains up until the voice port 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 allows 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. (Subchannels 0 to 3 are reserved and cannot be configured either for voice or data.)

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, you must route the entire PVC; connection ID-based routing (individual Channel ID switching) is not supported. Because the entire PVC is routed, no prioritization of voice packets is possible at the intermediate Frame Relay nodes.

The **connection trunk** voice-port configuration command is used to establish a static FRF.11 trunk; the dial peer is configured using the **session protocol frf11-trunk** command, which invokes the FRF.11-compliant session protocol.

Note FRF.11 specifies that a device can pack multiple FRF.11 subframes within a single Frame Relay frame; however, the Cisco implementation of Voice over Frame Relay currently does not support multiple subframes within a frame.

Cisco Switched Voice over Frame Relay

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 CID of an FRF.11-format multiplexed DLCI. The Cisco-switched Voice over Frame Relay 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 is used; this mechanism can be used for either permanent switched (Cisco-trunk) or dynamic switched calls. The Cisco-switched VoFR protocol includes forwarding of the called telephone number and supports tandem switching of the call over multiple Frame Relay PVC hops.

A tandem node is an intermediate router node within the Frame Relay call path. Its purpose is to switch the frames from one PVC subchannel to another (from one VoFR dial peer onto another VoFR dial peer) as the frames traverse the network. Use of tandem router nodes also avoids the need to have complete dial-plan information present on every router.

The Cisco MC3810 also supports Voice over ATM (VoATM) and Voice over HDLC (VoHDLC). The Cisco MC3810 is able to tandem switch voice calls between VoFR, VoATM, and VoHDLC call legs. The Cisco 2600 series and 3600 series routers also support VoIP. It is currently not possible to translate from the VoIP transport protocol to other protocols such as VoFR. As a result, a call coming in on a VoIP connection may not be (tandem) switched to a VoFR connection.

Dynamic Switched Calls

Dynamic switched calls are regular telephone calls in which the dial-plan-based call switching is performed by the Cisco router. The destination endpoint of the call is selected by the router based on the telephone number dialed and the dial-plan information contained in the dial-peer configuration entries. Contrast this implementation with permanent calls (Cisco-trunk calls), where the call endpoints are permanently fixed at configuration time.

The dial peer is configured using the **session protocol cisco-switched** dial-peer configuration command, which uses the Cisco proprietary session protocol.

Cisco-Trunk (Private Line) Calls

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

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

Frame Relay Fragmentation

Cisco has developed the following three methods of performing Frame Relay fragmentation:

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

These Frame Relay fragmentation methods are briefly described in the following sections.

Frame Relay fragmentation can be configured in conjunction with Voice over Frame Relay or independently of it.

End-to-End FRF.12 Fragmentation

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

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 only included for frames that are greater than the fragment size configured. As a result, FRF.12 is the recommended fragmentation to be used by VoIP.

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 access concentrator support end-to-end fragmentation on a per-PVC basis. Fragmentation is configured through a map class, which can apply to one or many PVCs, depending on how the class is applied.

Frame Relay Fragmentation Using FRF.11 Annex C

When Voice over Frame Relay (FRF.11) and fragmentation are both configured on a PVC, the Frame Relay fragments are transmitted in the FRF.11 Annex C format.

This fragmentation is used when FRF.11 voice traffic is transmitted on the PVC and it uses the FRF.11 Annex C format for data.

With FRF.11, all data packets contain fragmentation headers regardless of size. This form of fragmentation is not recommended for use with Voice over IP.

Cisco Proprietary Voice Encapsulation

Cisco proprietary voice fragmentation was implemented on earlier releases of the Cisco MC3810 multiservice access concentrator. This fragmentation type is used on data packets on a PVC that is also used for voice traffic. When the **vofr cisco** command is configured on a DLCI and fragmentation is enabled on a map class, the Cisco 2600 series, 3600 series, and 7200 series routers can interoperate as tandem nodes (but cannot perform call termination) with Cisco MC3810 concentrators running Cisco IOS releases prior to 12.0(3)XG or 12.0(4)T.

On the Cisco 2600, 3600, and 7200 series routers, entering the **vofr cisco** command is the only method for configuring Cisco proprietary voice encapsulation. You must then configure a map class to enable voice traffic on the PVCs.

On the Cisco MC3810, you have two methods for configuring Cisco proprietary voice encapsulation:

- Configuring the map class to support proprietary voice encapsulation by entering the **vofr cisco** command on the DLCI.

  When using this method, you must configure a map class to enable voice traffic.

- Configuring a Frame Relay DLCI to support proprietary voice encapsulation by entering the **frame-relay interface-dlci voice-encap** command.

  When using this method, you can configure either an individual DLCI or configure a map class to enable voice traffic. This command is not supported on the Cisco 2600 and 3600 series routers.

These commands are mutually exclusive on the interface and each provides different advantages. The **vofr cisco** command uses weighted fair queuing for controlling the data flows. The **frame-relay interface-dlci voice-encap** provides only simple FIFO queuing and does not provide prioritization between non-voice (data) flows. The command does provide prioritization for voice flows.

Because the **frame-relay interface-dlci voice-encap** command provides only FIFO queuing, this method has less overhead than the **vofr cisco** command. However, the **vofr cisco** command provides greater control over the queuing mechanism.

Frame Relay Fragmentation Conditions and Restrictions

When Frame Relay fragmentation is configured, the following conditions and restrictions apply:

- Hardware compression is not currently supported.
- Weighted fair queueing at the PVC level is the only queueing strategy that can be used.
- Frame Relay traffic shaping must be configured to enable Frame Relay fragmentation.
- Voice over Frame Relay frames are never fragmented, regardless of size.
- When using end-to-end FRF.12 fragmentation, the VoIP packets will not include the FRF.12 header, provided the size of the VoIP packet is smaller than the fragment size configured. However, when using FRF.11 Annex C or Cisco proprietary fragmentations, VoIP packets will include the fragmentation header.
- If fragments arrive out of sequence, packets are dropped.

Note Fragmentation is performed after frames are removed from the weighted fair queue.

FRF.II Implementation Agreement Support

This feature provides support for different FRF.11 forum features depending on the hardware platform used (see Table 1).

Table 1 FRF.11 Support

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