IP Office Technical Tip

Technical Tip No: **019 VoIP over Frame Relay with FRF12**
Date: **17th July 2003**

Product: IP Office **Revision:** 1.0	**Version:** 1.4(22) **Date:** 17th July 2003

Issue

This document provides a description of the new 1.4 IP Office feature; Voice over IP (VoIP) over Frame Relay using FRF12 fragmentation.

Operating Environment

IP Office 1.4(22), System Monitor 1.4(22), Manager Application 1.4(22), Cisco SW 12.2, Cisco DTE 12.2.

IP Office and VoIP over Frame Relay with FRF 12

Currently, IP Office supports VoIP over Frame Relay only with PPP encapsulation. The new feature described in this document provides for the support of VoIP over Frame Relay using RFC1490 encapsulation and consolidates IP Office existing support for VoIP over Frame Relay. This document assumes the reader has a basic understanding of Frame Relay principles.

Frame Relay Definitions and References

In order to progress the discussion of the new IP Office Frame Relay feature, this section presents a brief overview of the key terms of reference specific to the Frame Relay elements that relate to the new IP Office 1.4 (22) feature; Voice over IP (VoIP) over Frame Relay using RFC1490 encapsulation with FRF12 fragmentation.

The IP Office help contains further background to Frame Relay Specification but it should be noted that the Frame Relay specification is widely discussed in open forums. The reader is encouraged to seek a fuller explanation than is provided here.

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Reference	*Description*
http://www.frforum.com/basicguide/glossary.html	Your Complete Guide to Frame Relay from the Frame Relay Forum.

Frame Relay Terms:

Burst Size (Bc):

In Frame relay networks the user is able to forward data in excess of the agreed rate (CIR) for short periods, provided the average data transfer rate does not exceed the CIR. Bc represents the maximum amount of data during the time period Tc that the network guarantees to carry under normal circumstances.

Excess Burst Size (Be):

Represents the maximum amount of data during the time period Tc by which the user may exceed the committed burst size.

Time period (Tc):

Committed Information Rate (CIR) measurement interval .

Committed Information Rate (CIR):

The CIR rate is the rate at which Frame Relay network agrees to accept data from the user and at which the network commits to transfer under normal conditions. The relationship between CIR the Tc and Bc and is shown in the formula below:

$$CIR = Bc/Tc$$

Access Rate:

The actual clocking rate of the Frame Relay Port.

End Device:

The ultimate source or destination of data flowing through a frame relay network sometime referred to as a Data Terminal Equipment (DTE). As a source device, it sends data to an interface device for encapsulation in a frame relay frame. As a destination device, it receives de-encapsulated data (i.e., the frame relay frame is stripped

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off, leaving only the user's data) from the interface device. Also see DCE.

DLCI-Data Link Connection Identifier:

It is a unique number that is assigned to a PVC end point in a frame relay network. Identifies a particular PVC endpoint within a user's access channel in a frame relay network and has local significance only to that channel.

Permanent virtual Circuit (PVC)

A frame relay logical link, whose endpoints and class of service are defined by network management. Analogous to an X.25 permanent virtual circuit, a PVC (often referred to as PVC) consists of the originating frame relay network element address, originating data link control identifier, terminating frame relay network element address, and termination data link control identifier. Originating refers to the access interface from which the PVC is initiated. Terminating refers to the access interface at which the PVC stops. Many data network customers require a PVC between two points. Data terminating equipment with a need for continuous communication use PVCs. See also Data Link Connection Identifier (DLCI).

Congestion control and reporting FECN, BECN and DE:

FECN=Forward Explicit Congestion notification bit
BECN=Backward Explicit Congestion Notification bit
DE=Discard Eligibility bit

Data Communications Equipment (DCE):

Term defined by both frame relay and X.25 committees that applies to switching equipment and is distinguished from the devices that attach to the network (DTE). Also see DTE.

General Definitions

The following terms will be used in this document:

Non-voice or data:

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All traffic types that is not voice (see voice traffic).

Voice traffic:

UDP/RTP traffic that matches the DSCP value as set in the System/Gatekeeper form.

Signalling or Voice Signalling

The Call setup procedures that is used to establish a VoIP call using H323

Operation of Avaya IP Office Frame Relay QOS

The operation of IP Office VoIP over Frame Relay QOS is described in this section and is divided into the following functional areas:

- Inbound Router Interface
- VPN Line control
- IP RTP Header Compression
- Interface queue
- Traffic Shaping
- IP Office Manager Application
- 3rd Party Frame Relay DTE interoperation

The diagram below illustrates the general operation of IP Office QoS mechanism for Frame Relay links. Refer to this diagram with respect to the functional areas that are discussed in this section.

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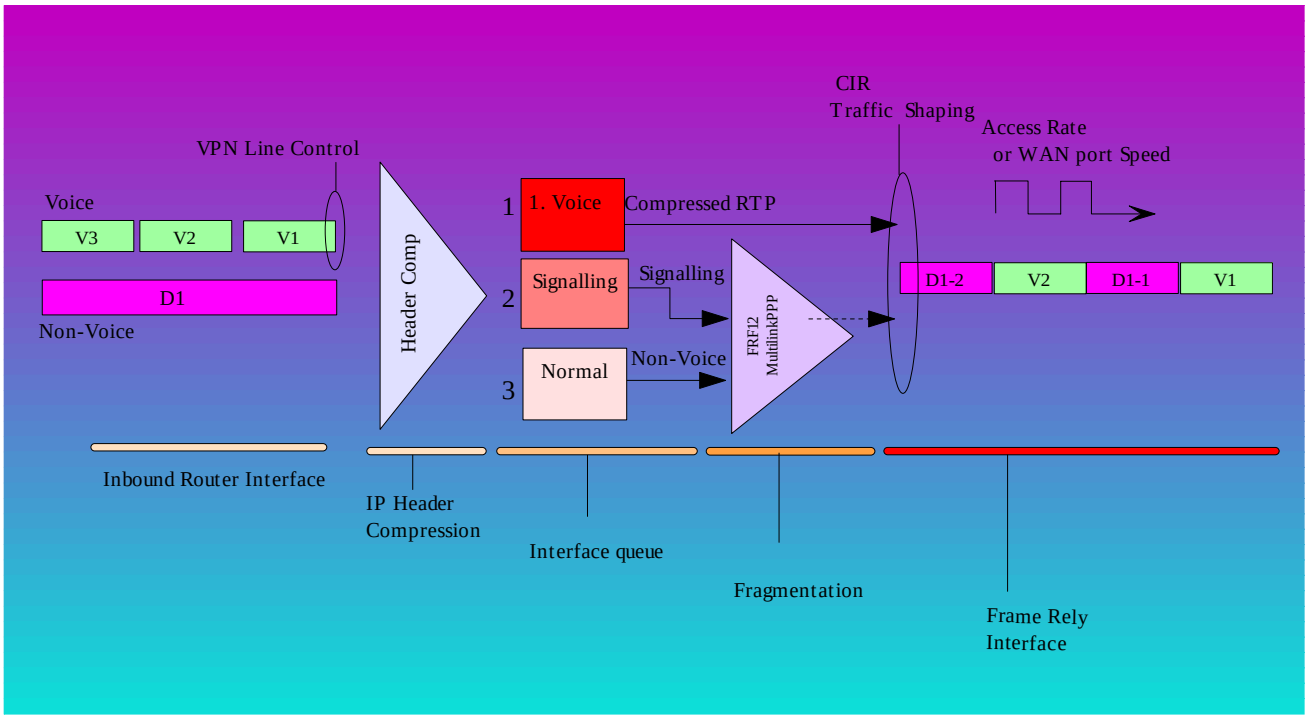

Voice packets are transmitted with a fixed length and intervals and must not be delayed through the interface.

The illustration shows voice packets (V1, V2 and V3) and non-voice packet D1 arriving on the inbound router interface (i.e. LAN, WAN or Local system originated). These packets will now pass through the functional process before they egress the router via the Frame Relay Interface (these functional process will be discussed in the proceeding paragraphs).

The non-voice packet D1 is made to fit into the interval between successive voice packets. This is accomplished by processing large non-voice packets through PPP Multilink. The FRF12 process “fragments” the larger non-voice packet (D1) into smaller components (D1-1 and D1-2) for serialisation to the link. Voice packets are not fragmented in this way. Voice Signalling packets are fragmented in the same way as non-voice packets.

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Inbound Router Interface

Voice and Non-Voice traffic may be originated by IP Office or received on any WAN/LAN interface.

VPN Line control

QoS policies on routers protect delay sensitive VoIP traffic from non-voice traffic on slow speed WAN links. As well as this, voice traffic must be protected from other voice traffic in the case where there is a limited amount of bandwidth. Adding more voice traffic than can be supported on a link has the effect of "cannibalising" other voice currently occupying the link. This means that if a link has a maximum bandwidth for the support of 5 concurrent calls then the placement of a 6th call will affect the other 5 calls already established on the link. Concurrent call load restriction is a function of a VPN Line configuration.

A VPN Line is used to configure VoIP Gateway on IP Office; A VPN Line is distinct from a physical Line type such as T1 or E1 but is configured under the same branch of the Manager Application configuration tree (Line)

The VPN Lines is used to configure the VoIP Gateway for IP Office; it is used to facilitate the VoIP Gateway interaction between two VoIP Gateway endpoints. The term VPN Line control is an explicit reference to the VPN Line parameters that allow for the control of the type, direction and concurrent call load on a VoIP link. The VPN Line control configuration uses the following parameters.

Parameter	***Description***
Number Of Channels:	Defines the number of operational channels that are available on this line.
Outgoing Channels:	Defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.
Data Channels:	The number of channels available for data use.
Voice Channels:	The number of channels available for voice use.

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It is necessary to apply a level of strict access control to the available WAN link for the amount concurrent VoIP calls.

VPN Line control parameters work on the basis of an “allowed number of voice connections”. Other vendors generally apply voice traffic restrictions on the basis of a “maximum voice bandwidth”; usually this is done by apportioning a maximum bandwidth allocation for VoIP traffic under link congestion.

It is important understand the bandwidth requirement and calculations for any VoIP link and must consider whether or not IPHC is to be used on the link. The Bandwidth used by a given compression type for a single VoIP call can be calculated using the formula shown below.

Bandwidth =

(Fr_Header + Encap_type +IP_ UDP_RTP_Header + Payload) X Payload_per_sec X Bit__conversion

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Use the calculation to determine total bandwidth requirement then set the appropriate values (in terms of the number of calls) by using VPN Line control parameters (see following paragraphs)

Variable	***Description***	***Option***		***Values***
FR_Header	The size of a frame relay header is dependant on the DLCI size. Usual the header is 2 bytes but may be 2, 3 or 4	2,3 or 4		***2***
Encap_type;	The method of encapsulation that is used over the Frame Relay network	MultilinkPPP		4
		RFC1490 + FR12		3
IP_ UDP_RTP_Header	The total length of the IP,UDP and RTP header is dependant on whether link is running IPHC	With IPHC		4
		Without IPHC		40
Bit__conversion	Transmission speeds are always specified in Bits Per second. The multiplication factor of 8 (i.e. X 8) is used to convert the calculation to bytes per second.	-		8
Payload & Payload_per_sec	The is sample rate of the codec The sample rate	Type	Payload_per_sec	
		G711	50	160
		G723	33.3	24
		G729	50	20

Examples

(Fr_Header + Encap_type +IP_ UDP_RTP_Header + Payload) X Payload_per_sec X Bit__conversion

Calculation for G729 with IPHC using PPP encapsulation

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((2 + 4 + 4+ 20)* 50 * 8) = 30 * 50 * 8

Bandwidth = 12Kpbs

Calculation for G729 with Header compression using RFC1490+FRF12

((2 + 3 + 4 + 20)* 50 * 8) = 29 * 50 * 8

Bandwidth = 11.6Kbps

Calculation for G729 without Header compression using RFC1490+FRF12

((2 + 3 + 40 + 20)* 50 * 8) = 65* 50 * 8

Bandwidth = 26Kbps

IP Office performs Voice Traffic concurrent call load restriction on a per call basis and does not assume the bandwidth requirement. The actual bandwidth requirement for any given number of VoIP calls must be determined with respect Compression type (i.e. G711 or g729) and the status of IPHC on the link. The table below is included here in order to illustrate this principle

Number of allowed calls	***Compression Type***	***IPHC***	***Minimum Bandwidth Kbps***
1	729	No	*(1 * 26) =26*
1	729	Yes	*(1 * 12)= 12*
10	729	No	*(26 * 10) =260*
15	729	Yes	*(12 * 15) =180*

The table shows, for example, that if IPHC is running and there is a requirement for 15 concurrent G729 calls that a minimum bandwidth or CIR rate of 180Kbps is required. In this case the VPN Line parameter “Number Of Channels” would be set to 15 and the CIR rate would a minimum of 180Kbps

The number of VoIP calls that can occupy a link is directly related to the CIR rate of the link. As General rule when mixing voice and data on the same DLCI it is recommended that no more than 75/80 % of the available bandwidth be used for VoIP traffic; use the VPN Line to restrict the number number of concurrent calls that are allowed to occupy the link.

Summary

- Do not allow VoIP traffic to exceed the CIR rate of the PVC

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- Voice Traffic concurrent call load restriction is performed on a per call basis and does not assume the bandwidth requirement for the call. This means that the bandwidth requirement for the link must be calculated on the basis of the Voice Compression type (i.e. G711 or g729) and the status of IPHC on link.

IP RTP Header Compression

IP Header Compression (IPHC) reduces the IP/UDP/RTP headers from 40 bytes to two bytes for most packets in the case where no UDP checksums are being sent, or four bytes with checksums. IPHC significantly reduces WAN bandwidth requirements.

IP Office applies IPHC to all traffic types i.e. voice, Signalling and non-voice. However it is to be noted that some IP protocols will yield better compression ratio than others.

IPHC imposes process overheads which can become counter-productive at higher WAN speeds. For this reason IP Office will not perform IPHC at CIR rate above 1024Kbps.

IPHC performs compression of the IP headers on a per IP connection basis; a connection is deemed to be an IP stream or flow between two peers denoted by a Source/Destination IP address and Source/Destination port numbers.

Summary

- For PPP encapsulation IPHC is configured on the Service under the PPP tab. For PPP the amount of connection is fixed at 16 for TCP and 60 for UDP, these values can not be changed.
- For RFC1490+FRF12 encapsulation IPHC is configured within the DLCI configuration form of the WANPort tab.
- It is to be noted that when using RFC1490+FRF12 encapsulation parameters on the Service/PPP tab are ignored.
- When using RFC1490+FRF12 with IPHC it is imperative that the number connections (TCP and UDP) are matched at each end for the link.

Interface queue

Traffic outbound to an interface is associated to one of three interface queues which are detailed in the table below.

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Priority	***Queue Name***	***Match Criteria***
1	Voice Traffic	1. The system DSCP value as set on the system/Gatekeeper form 2. Any IP packets which are match using the configured DSCP mask to the system DSCP value
2	Signalling	1. TCP port number 1720 2. AVRIP packets (Voice Networking)
3	Normal.	All other traffic types which is not traffic or Signalling

The Voice Traffic queue has priority over the Signalling queue; both queues have priority over the Normal queue.

IP Office identifies and prioritises voice traffic on the basis of the configured system Differentiated Services Code Point (DSCP) value and is transparent to UDP port numbers (see notes below). Voice Signalling is identified by the TCP port number 1720; this option is not configurable. The DSCP value is configured on the System /Gatekeeper form and is a global parameter.

Differentiated Services (DiffServ) is a new model in which traffic is treated by intermediate systems with relative priorities based on the type of services (ToS) field.

The DiffServ architecture defines the DiffServ field, which supersedes the ToS byte in IP V4 to make per-hop behaviour (PHB) decisions about packet classification and traffic conditioning functions, such as metering, marking, shaping, and policing. DSCP is defined using six bits (D5 to D0 see diagram below) in the ToS byte of an IP V4 header.

D5	D4	D3	D2	D1	D0	ECN	ECN

The six most signification bits (D5 to D0) of the ToS byte are now called the DiffServ field. The last two, Currently Unused (CU) bits, in the DiffServ field were not defined within the DiffServ field architecture; these are now used as Early Congestion Notification (ECN) bits

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IP Office default setting is 46 or B8 (i.e. with the ECN values of “00” the TOS byte would be 0xB8)

The DSCP value is configured on the System /Gatekeeper form and is a global parameter. Whilst the default settings of DSCP 0x46 and mask 0xFC are suitable for IP Office interoperation the System Gatekeeper form provides additional control for 3rd party VoIP interoperation.

The System/ DSCP parameter serves two functions;

1. Sets the value that is to be used by IP Office to mark the TOS field (DiffServ) in the IP Header.
2. The value that is used by IP Office to identify Voice traffic for prioritisation

For the advance users and 3rd party interoperation the mask allows IP Office to identify more than one DiffServ value; the following is an example of how this done.

Two IP Office systems are linked and provide VoIP between two sites; the systems are configured using the default system DSCP value of 46 and mask 0xFC. The two IP Office systems are now required to route an external VoIP stream between two 3rd Party VoIP enabled PABX. The external PABX uses a DSCP value of 40.

Item	***TOS Byte***	***Decimal***	***Binary***
External DSCP	0xA0	40	101000XX
System DSCP	0xB8	46	101110XX
Required DSCP Mask	0xA0	160	10100000

XX= are the ECN bits

The table above shows that the required DSCP mask value for this configuration would be 0xA0 as it provides a match on the 8th and 6th bits of the External DSCP (for this example the mask value could have also been set to 224 or 0xE0, this would match the 8th, 7th and 6th bits).

In addition to the DSCP mask IP Office also provides for the configuration of the DSCP value for Signalling packets. This is useful in the case where an external router requires signalling packets to be marked with a specific DSCP setting for prioritisation.

Care should be taken not to set the Signalling packet DSCP value to the same value as the Voice traffic in the case where IP Office is the VoIP router. Setting the Signalling DSCP value to the same value as voice traffic would

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induce serialisation delay as traffic that is deemed to be voice traffic is not fragmented.

Summary

- For VoIP traffic IP Office uses UDP port range OxC000 to 0xCFFF for voice traffic but these port number are not utilise for Voice traffic prioritisation. The DSCP value is used for the purpose of identifying Voice traffic for prioritisation.

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Fragmentation

Large non-voice packets can adversely affect the smaller voice packets and reduce voice quality. This effect of non-voice packets on voice packet is sometimes referred to as serialisation delay. To minimise the effect of serialisation delay the large non-voice packets are fragmented so as not to delay voice packets; the fragmented non-voice packets are interleaved with the VoIP packets thereby reducing jitter and delay.

New to IP Office 1.4 is FRF12 Fragmentation and Interleave over Frame Relay using RFC1490 encapsulation consolidating IP Office existing functionality of PPP fragmentation and Interleave over Frame Relay. Support for this new feature includes Frame Relay Traffic Shaping which provides for fragmentation on a per PVC basis. IP Office Frame Relay Traffic Shaping forces IP traffic to conform to specified Frame Relay parameters and accurately guarantees the availability of bandwidth for VoIP Traffic.

The table below summaries IP Office 1.4 fragmentation and Traffic Shaping support for the supported encapsulation types for VoIP over Frame Relay

Encapsulation	***Fragmentation***	***Traffic Shaping***	***VoIP Support***
RFC1490 + FR12	FRF12	Yes	Yes
PPP	MultilinkPPP	Yes	Yes
RFC1490	-	No	No

When deploying VoIP over Frame Relay the choice of PPP or RFC1490 + FRF12 encapsulation for Fragmentation Interleave is one of interoperability rather than of functionality. Highlighted in the table below are the differences between PPP and RFR1490 + FRF12

The table below details the Fragmentation Interleave for PPP and RFC1490+FRF12 for IP Office

Item	***PPP***	***RFR1490***
Fragmentation Method	MultilinkPPP	FRF12
Fragment size	Dynamically sized based to the number of established calls on link and the CIR rate of the link.	Fixed, based the CIR rate of the link.

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	(3rd Party routers may not support this option)	
Optional Configuration	IPHC: Configured on the Service/PPP tab	IPHC: Configured on the WANport/DLCI form
Mandatory Configuration Item	Multilink selected on the Service/PPP tab	-
Functionality		

IP Office automatically calculates fragment sizes to minimise the effect of serialisation delay; automatic fragment size calculation are done in one of two ways depending on the Frame Relay encapsulation used.

RFC1490 + FR12 fragments packet to a fixed length based on the configured CIR Rate. The size of non-voice fragments for PPP change with respect to the CIR rate and the number of established VoIP calls occupying the link.

MulticlassPPP allows multiple fragmentation streams. This allows IP Office to reduce signalling packet latency through Frame Relay Interface. MulticlassPPP is use in the case were IP Office begins the fragmentation of a non-voice packets but before the last fragment is forwarded a signalling packet arrives. When this occurs IP Office suspends the fragmentation of the non-voice packets and instead fragments and forwards the signalling packet ahead of the non-voice packet. In this way latency is improved for signalling packet and represent a benefit of using PPP encapsulation over RFR1490+FR12

Whilst between IP Office and IP Office the MulticlassPPP option is always negotiated; it may not be negotiated between IP Office and 3rd party routers.

Summary

- When using FRF12 fragmentation IP Office calculates the size of non-voice packet fragments based on the configured CIR and is fixed for configured CIR.
- For RFc1490+FRf12 or PPP encapsulation only non-voice packets are fragmented.
- To improve Latency for non-Voice traffic IP Office fragments non-voice packets only when voice traffic is present on the link.
- IP Office independently assigns fragment sizes for each PVC using the specified CIR values to determine the fragment size for the Link.

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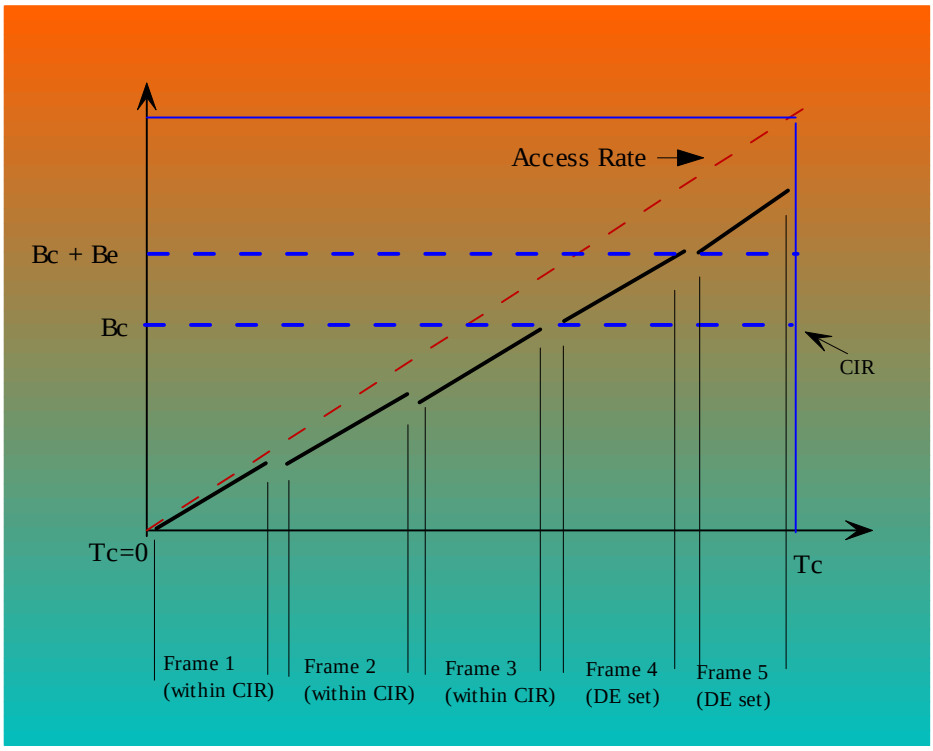
- Selection of the MultilinkPPP on the Service/PPP tab is a mandatory requirement for the correct operation of the QOS when using PPP encapsulation; IPHC is optional (also configured on the Service/PPP tab.

Traffic Shaping

IP Office Frame Relay Traffic Shaping forces IP traffic to conform to specified Frame Relay parameters and accurately guarantees the availability of bandwidth for VoIP Traffic. It is the means by which outbound traffic to an interface is throttled to the rate of the configured CIR. In order to appreciate the operation of the traffic Shaping the relationship between the *CIR, Tc, Bc* and *Be* must be understood.

The CIR rate is the rate at which Frame Relay network agrees to accept data from the user and at which the network commits to transfer under normal conditions. The relationship between CIR the Tc and Bc and is shown in the formula below.

$$CIR = Bc/Tc$$



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The diagram shows that starting at Tc=0 Frame 1 is transmitted at rate equal to the Access Rate (Clocking rate) of the Frame Relay port. Frame 1 is shown to be less than the Bc. The next two frames (Frame 2 and Frame 3) are transmitted at the Access rate but fall beneath the Bc value. Frame 1,2 and 3 together use up the allocated CIR before the time period has expired. When Frame 4 is transmitted it falls above the Bc value and before Tc; this frame falls above the CIR and may be mark Discard Eligible (DE) by the Frame Relay Network.

IP Office doesn't transmit frame which go above the agreed CIR for the Link, so from the explanation above as Frame 4 falls above the CIR it will be discarded by IP Office. The diagram shows that the CIR for the link was met when Frame 3 was transmitted.

IP Office will now wait for a new Tc "window" before transmitting more frames to the network. In this way, IP Office Frame Relay Traffic Shaping ensures that the CIR of the link is not exceeded

Summary

- Do not exceed the CIR on virtual circuits with voice frames (See VPN Line control).
- Be is not supported in 1.4(X) and should be set to 0 (default)
- The sum of all configured CIR must not exceed the Access Rate of the link
- Frame Relay Traffic Shaping allows CIR rate and Tc value to be modified but does not allow the explicit configuration of Bc.
- Frame Relay Traffic Shaping is not supported for the "Frame Link Type" RFR1490 and None (IP Office does not support QOS for VoIP when these options are selected).
- Frames are transmitted across the physical interface at the access rate (clock rate) not the CIR.

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IP Office Manager Application

IP Office Manager Application layout for Frame Relay configuration remains much the same apart from the following changes

1. The Frame relay Link parameters have been moved from the WANport/Frame Relay tab to new WANport/Advance tab.

2. For RFC1490+FRF12, the DLCI tab configuration form now presents configuration for the control (disable/enable) of IPHC and the number of connections that can be supported on the link FRF12 link. When using PPP the IPHC control is on the Service/PPP tab.

3. The DLCI tab now allows the CIR rate for the DLCI to configured on a per PVC basis (formally Bc was configurable on this form)

3rd Party Frame Relay DTE interoperation

IP Office Frame Relay for VoIP is compliant with the following 3rd party implementation:

Implementation	***Description***	***Compliant with IP Office***
Voice Over IP (VoIP) over Frame Relay with RFC1490 encapsulation	Uses IP RTP priority to prioritise Voice over data traffic Uses FRF.12 Fragmentation and RFC149 encapsulation	Yes
Voice Over IP (VoIP) over Frame Relay with PPP encapsulation	Uses IP RTP priority to prioritise Voice over data traffic Uses PPP-Multilink for Fragmentation and PPP (IETF) encapsulation	Yes
Voice Over Frame Relay (VoFR)	Uses FRF.11 or derivative for Fragmentation	No

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Configuration Examples

The following are prerequisites for the configuration of a Frame Relay link between two end points

Frame Relay Service provider settings	*Description*
Frame Management Type	This must match the management type expected by the network provider. The following options are supported: • Q933 AnnexA 0393 • Ansi AnnexD • FRFLMI • None NB the None Management option is useful for back-to-back testing.
DLCI	The DLCI number to be used
CIR Rate	The Agreed CIR rate
Access rate	The Speed of the Link

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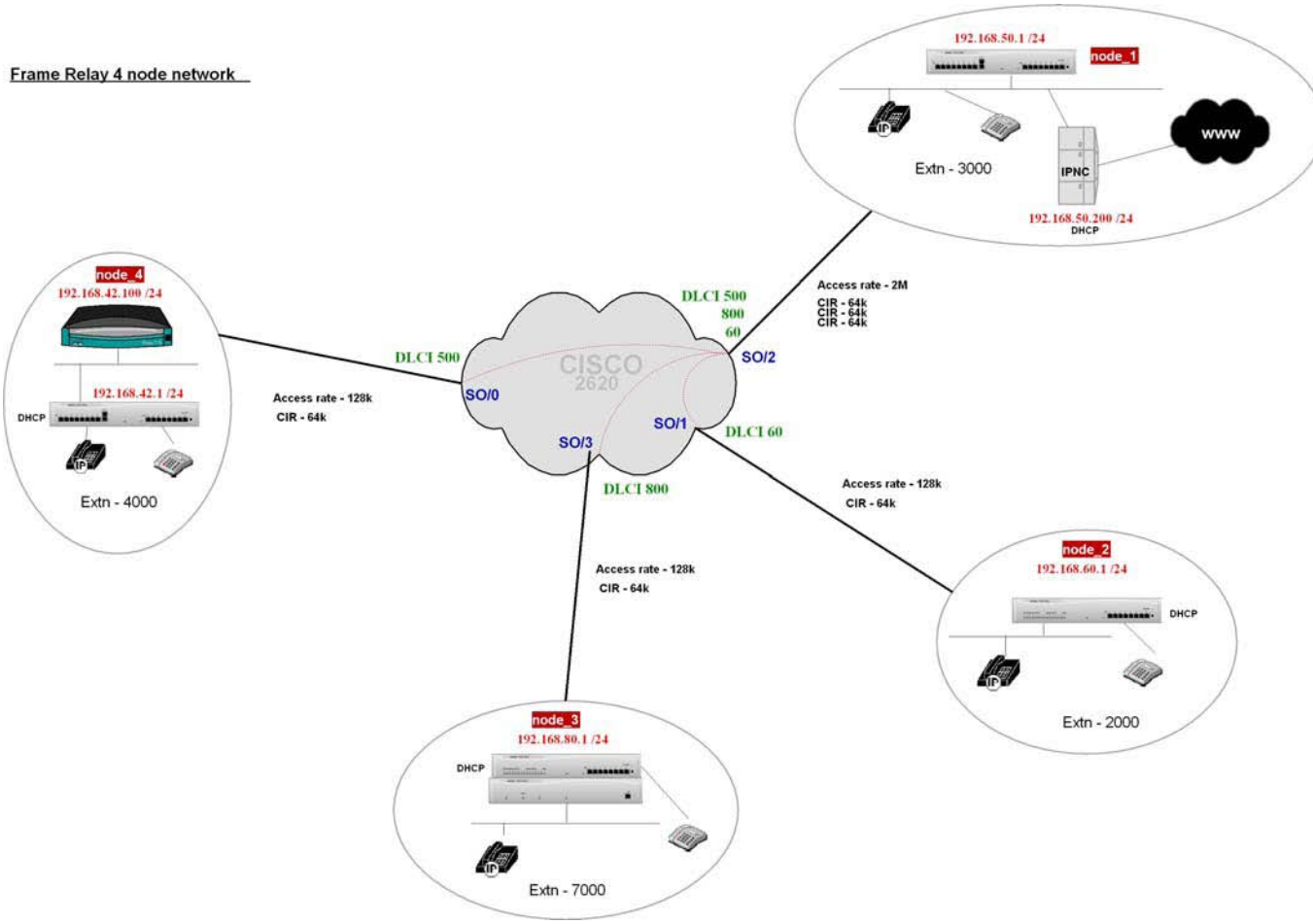
Configuration for IP Office to IP Office

Using the diagram below this section will detail the Configuration of Frame Relay using PPP or RFR1490 + FRF12 between two IP Office and between IP Office and 3rd Party routers.



Configuration Tasks for Node_1 to Node_2

The configuration procedure below details the configuration tasks for Node_1 and Node_2. For 3rd party configuration options, steps 5a (PPP) and 5b (RFC1490+FRF12) stipulate the appropriate 3rd party configuration which are detailed in the Appendix.

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Task	Description
Step 1 For Node_1 and Node_2, attach WAN cable, reboot unit and obtain configuration.	In order to configure the WAN interface IP Office requires the WAN cable to be attached at boot up.
Step 2 For Node_1 and Node_2, create a WAN Service type • Name = FRwan_Link60 • Account Name = FRwan_Link60 • All password fields = <blank>	It is not permissible for a Service to be associated to more than one DLCI; Each DLCI must be assigned a unique Name/Account Name. If running a WAN3 the word "WAN" (upper case) must be added in the Service Dial In tab as a Dial In source number.
Step 3 For Node_1 and Node_2, perform the following tasks on the WANport form • Mode = syncFrameRelay • Speed = <Access Rate>	Set the Speed of the Link to equal Access Rate of the Link; setting the Access Rate allows IP Office to verify the correct configuration of CIR verses Access Rate. *The aggregate of all CIR must not exceed the port speed.*
Step 4 **(**Node_1 and Node_2) WANport/Frame Relay tab set the following • FR Management type =ANSI • Frame Learn Mode = None	All the default parameters (except FR Management type) on this tab are appropriate for a basic Frame Relay connection. FR Management Types Q933 AnnexA 0393, Ansi AnnexD and FRFLMI are supported.
The Next step in this procedure is optional and is dependant on whether PPP or RFC1490 + FRF12 encapsulation is to be used	*Frame Link Type parameter controls the Frame Relay encapsulations type. The encapsulation is transparent to the FR network but must be matched at both ends of the links.*
Step 5a *(Optional for PPP encapsulation)* On the WANport/DLCI tab set the following For (Node_1) and (Node_2) on the WANport/DLCI tab set the following • Frame Link Type = PPP or Frame+FRF12	Frame Link Type parameter controls the Frame relay encapsulations type. The encapsulation is transparent to the FR network but must be matched at both ends of the links. Selection of the PPP Multilink on Service/PPP tab is a mandatory requirement for the correct operation of the QOS when using PPP encapsulation; IPHC is optional

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Task	Description
Frame+FRF12 • DLCI = 1000 • RAS name = FRwan_Link60	***(See Config-1 in Appendix for example of 3rd party configuration in support of IP Office interoperation.***
Step 5b *(Optional for RFC1490 + FRF12 encapsulation)* For (Node_1) and (Node_2) on the WANport/DLCI tab set the following • Frame Link Type = Frame+FRF12 • DLCI = 1000 • RAS name = FRwan_Link60 • Tc= 10 (default) • CIR= 64000 (default) • Be = 0 (default) • IPHC = <selected> • TCP= 16 (default) • UDP/RTP=60 (default) (These parameters are specific to the above diagram)	It is to be noted that when using RFC1490+FRF12 encapsulation parameters on the Service/PPP tab are not used. When using RFC1490+FRF12 it is imperative that all IPHC values are matched at each end of the link ***See Config-2 in Appendix for example of 3rd party configuration in support of IP Office interoperation.***
Step 6 **IPRoute** **(Node_1)** • IP Address = 192.168.60.0 • IP Mask = 255.255.255.0 • Gateway = <Blank> • Destination = FRwan_Link60 **(Node_1)** • IP Address = 192.168.50.0 • IP Mask = 255.255.255.0 • Gateway = <Blank>	A routing entry must be added to allow access between the two networks.

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Task	Description
• Destination = FRwan_Link60	
Step 7 For Node_1 and Node_2 configure the appropriate LAN interface address	It is not necessary to add an IP route in support of any configured LAN interfaces

Debugging IP Office QOS and VoIP

Using the Monitor application options it possible to debug all aspects of the IP Office function and features. Generally outputs are associated directly to the any related standard. PPP monitor outputs will, for example, reference RFC terms relating to the PPP protocol.

The following table describes monitor options that may be useful for debugging QOS issues on IP Office.

Function	Method
MultilinkPPP Fragmentation	This output is taken using the following Monitor option • PPP/LCP Tx • PPP/LCP Rx The highlighted text indicates the PPP Multilink option; the negotiation is successful 97104mS PPP LCP Tx: v=cisco1750 PPP LCP Config-Req(1) id=2 len=33 MagicNum=00102152 Protocol field compression Address and control field compression MRRU=1500 **ShortSeq** **EndPointDiscrim=mac 00e007003f20** **MultiClass=6 Classes=4** 97106mS SERVICE:2002/6/511:30,"cisco1750",0,0 97120mS PPP LCP Rx: v=cisco1750 PPP LCP Config-Req(1) id=196 len=31 AuthProt=c223 Algorithm=5 MagicNum=1de9b16a MRRU=1524

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Function	Method
	EndPointDiscrim=local 636973636f31373530 97121mS PRN: LCPAutomaton::ParseAutomatonPacket 97121mS PPP LCP Tx: v=cisco1750 PPP LCP Config-Ack(2) id=196 len=31 AuthProt=c223 Algorithm=5 MagicNum=1de9b16a MRRU=1524 EndPointDiscrim=local 636973636f31373530 97122mS PPP LCP Rx: v=cisco1750 PPP LCP Config-Rej(4) id=2 len=10 **ShortSeq** **MultiClass=6 Classes=4** 97122mS PRN: LCPAutomaton::ParseAutomatonPacket 97122mS PPP LCP Tx: v=cisco1750 PPP LCP Config-Req(1) id=3 len=27 MagicNum=00102164 Protocol field compression Address and control field compression **MRRU=1500** **EndPointDiscrim=mac 00e007003f20** 97130mS SERVICE:2002/6/511:30,"cisco1750",0,0 97138mS PPP LCP Rx: v=cisco1750 PPP LCP Config-Ack(2) id=3 len=27 MagicNum=00102164 Protocol field compression Address and control field compression **MRRU=1500** **EndPointDiscrim=mac 00e007003f20** 97153mS PRN: LCPAutomaton::UpLink() 97153mS PRN: LCPAutomaton::UpLink() checking link_state 97154mS PRN: ppplink start NetworkControlProtocols 97154mS PRN: stack start NetworkControlProtocols
VPN line	The following options are useful for debugging call setup issue. • Call/Line Send • Call/Line Receive • Call/Targetting The Highlighted text from the CMTARGET output shows the number received on ISDN line (5) is resolved to Line 2 (GROUP 2). The CMLine TX=2 is the resulting call setup on a VPN Line 2.

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Function	***Method***
	These output show the negotiated UDP port numbers 29505mS CMLineRx: v=5 CMSetup Line: type=Q931Line 5 Call: lid=5 id=6 in=2 **Called[8400]** Type=SubscriberNumber (4) Calling[01442404001] Type=SubscriberNumber (4) BC: CMTC=Speech CMTM=Circuit CMTR=64 CMST=Default CMU1=Alaw Bchan: slot=0 chan=2 29506mS CMTARGET: LOOKUP CALL ROUTE:41872 type=4 called_party=8400 sub= calling=01442404001 in=2 complete=0 29507mS CMTARGET: ADD TARGET:41872 number=8400 type=4 depth=1 nobar=1 setorig=1 29508mS CMTARGET: SYS SC:41872 8400 2 400 **sc=type=31 code=8N, num=.** 29508mS CMTARGET: DIAL LINE:41872 **GROUP=2** **SUCCESS=1** 29508mS CMTARGET: LOOKUP CALL ROUTE:41872 returned 1 29511mS **CMLineTx: v=2** CMSetup Line: type=VPN 2 Call: lid=0 id=42875 in=0 Called[8400] Type=SubscriberNumber (4) Calling[01442404001] Type=SubscriberNumber (4) BC: CMTC=Speech CMTM=Circuit CMTR=64 CMST=Default CMU1=Alaw Bchan: slot=250 chan=9868 IE CMIETxChannelAudio (1) comptype=G729A8K (6) pktsize=20 ipaddr=192.168.42.99 port=0 IE CMIERxChannelAudio (2) comptype=G729A8K (6) pktsize=20 ipaddr=192.168.123.98 port=51138 Display [01442404001] Cause=16, Normal Locale: eng 29680mS CMLineRx: v=2 CMProceeding Line: type=VPN 2 Call: lid=0 id=42875 in=0 Bchan: slot=250 chan=9868 IE CMIETxChannelAudio (1) comptype=G729A8K (6) pktsize=20 ipaddr=192.168.42.99 port=49178 IE CMIERxChannelAudio (2) comptype=G729A8K (6)

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Function	Method
	pktsize=20 ipaddr=192.168.123.98 port=51138 Display [400] 29681mS CMLineTx: v=5 CMProceeding Line: type=Q931Line 5 Call: lid=5 id=6 in=2 Called[8400] Type=SubscriberNumber (4) Bchan: slot=0 chan=2 IE CMIETxChannelAudio (1) comptype=G729A8K (6) pktsize=20 ipaddr=192.168.42.99 port=49178 IE CMIERxChannelAudio (2) comptype=G729A8K (6) pktsize=20 ipaddr=192.168.123.98 port=51138 Display [8400] 30785mS CMLineRx: v=2 CMConnect Line: type=VPN 2 Call: lid=0 id=42875 in=0 BC: CMTC=Speech CMTM=Circuit CMTR=64 CMST=Default CMU1=Alaw Display [400] 30787mS CMLineTx: v=5 CMConnect Line: type=Q931Line 5 Call: lid=5 id=6 in=2 Called[8400] Type=SubscriberNumber (4) BC: CMTC=Speech CMTM=Circuit CMTR=64 CMST=Default CMU1=Alaw Bchan: slot=0 chan=2 Display [8400] 30815mS CMLineRx: v=5 CMConnectAck Line: type=Q931Line 5 Call: lid=5 id=6 in=2
Header compression (IPHC)	This output is taken using the following Monitor option • PPP/IPCP Tx • PPP/IPCP Rx The output shows the successful negotiation of an IP address 192.168.168.100 by the remote and the IPHC. The IPHC negotiation is accepted by both local and remote. (Config-Ack received and transmitted) 47458mS PRN: stack start NetworkControlProtocols 47459mS PPP IPCP Tx: v=cisco1750 PPP IPCP Config-Req(1) id=1 len1=20

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Function	Method
	IPHC 0000 00 10 00 10 01 00 00 05 00 a8 01 02 ..... 47473mS PPP IPCP Rx: v=cisco1750 PPP IPCP Config-Req(1) id=14 len1=26 IPHC 0000 00 10 00 14 01 00 00 05 00 a8 01 02 ..... IP-Address 192.168.168.100 47473mS PPP IPCP Tx: v=cisco1750 **PPP IPCP Config-Ack(2) id=14 len1=26** IPHC 0000 00 10 00 14 01 00 00 05 00 a8 01 02 ..... IP-Address 192.168.168.100 47847mS PRN: Wed 5/6/2002 11:09:19 FreeMem=7084108 CMMsg=3 (3) Buff=100 554 500 1392 48337mS PRN: Wed 5/6/2002 11:09:19 FreeMem=7085224 CMMsg=3 (3) Buff=100 555 500 1391 50464mS PPP IPCP Tx: v=cisco1750 PPP IPCP Config-Req(1) id=2 len1=20 IPHC 0000 00 10 00 14 01 00 00 05 00 a8 01 02 ..... 50474mS PPP IPCP Rx: v=cisco1750 **PPP IPCP Config-Ack(2) id=2 len1=20** IPHC 0000 00 10 00 14 01 00 00 05 00 a8 01 02
Frame relay management	(1) EVENT: ev=363,v=0,p1=0,p2=0,p3=0,p4=0,s1=FRlmi: data channel stopped This message occurs when the Management interface is down, this may happen, for example, if there is LMI mismatch between IP Office and the Frame Relay network (2) This output is shown when the Frame Relay/Events and Mgmt events option are selected on the SysMonitor application. It show the DLCI that are active on a link (a PVC will become Inactive if the Remote DTE is not correctly configured or is not connected to the Frame Relay Network. The Full Status Management Packets are shown on if the

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Function	Method
	correct Management type is configured 15812mS FrameRelayMgmt: v=WAN1 dlci=1023 FRM_FRFLMI - Enquiry Type 0 - Full Status RxSeq 76 TxSeq 75 15814mS FrameRelayMgmt: v=WAN1 dlci=1023 FRM_FRFLMI - Response Type 0 - Full Status RxSeq 76 TxSeq 76 DLCI 60 - Active DLCI 500 - Active DLCI 800 – Active
IPHC using FRf13	The RFR1490+FRF12 is using a service name "FR_link302" and RTP is enabled at the remote but not the local end IPHC not enabled on FR_link302, discarding

Appendix

3rd Party Router Configuration

The table below details the Frame Relay Management types equivalence for Cisco interoperation and is specific to the following Cisco 12.2 command.

- frame-relay lmi-type

IP Office	Cisco Equivalent
Q933 AnnexA 0393,	q933a
Ansi AnnexD	ansi
FRFLMI.	Cisco

Notes

Please refer to the Cisco configuration files below for information relating to the encapsulation type.

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Configuration

For the purposes of these tests Cisco's Low Latency Queuing (LLQ) was used. LLQ enables a single strict Priority Queue (PQ) within Class-Based Weighted Fair Queuing (CBWFQ) at the class level. With LLQ, delay-sensitive data (in the PQ) is de-queued and sent first. In a VoIP with LLQ implementation, voice traffic is placed in the strict PQ.

These Configuration files shown here can be used to interoperate with IP Office; the configuration for IP Office is described in earlier paragraphs.

Cisco Configuration 1

Cisco Config- 1 For PPP

Current configuration : 1934 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname cisco
!
enable secret 5 1lj6e$U7rr8a8ynYb
enable password cisco
!
username wan_cisco password 0 passw
username cisco1750 password 0 passw
username wancisco password 0 passwo
memory-size iomem 25
ip subnet-zero
!
!
!
!
class-map match-all voice-signaling
  match access-group 103
class-map match-all voice-traffic
  match access-group 102
!
!
policy-map voice-policy
  class voice-signaling
  bandwidth 8

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Cisco Config- 1 For PPP

```
  class voice-traffic
    priority 48
  class class-default
   fair-queue
!
call rsvp-sync
!
!
!
!
interface FastEthernet0/0
 ip address 192.168.42.100 255.255.
 no keepalive
 speed auto
 half-duplex
!
interface Serial0/0
 description To Carrier Frame Relay
 no ip address
 encapsulation frame-relay IETF
 frame-relay traffic-shaping
!
interface Serial0/0.1 point-to-poin
 frame-relay interface-dlci 500 ppp
  class voice-policy
!
interface Virtual-Template1
 bandwidth 640
 ip address 192.168.168.101 255.255
 ip tcp header-compression iphc-for
 max-reserved-bandwidth 60
 service-policy output voice-policy
 ppp multilink
 ppp multilink fragment-delay 2
 ppp multilink interleave
 ip rtp header-compression iphc-for
!
ip classless
ip route 0.0.0.0 0.0.0.0 192.168.16
no ip http server
ip pim bidir-enable
!
!
map-class frame-relay voice-policy
no  frame-relay adaptive-shaping becn
 frame-relay cir 64000
```

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Cisco Config- 1 For PPP
frame-relay bc 640 frame-relay mincir 64000 access-list 102 permit udp any any access-list 103 permit tcp any eq 1 access-list 103 permit tcp any any ! voice-port 1/0/0 ! voice-port 1/0/1 ! dial-peer cor custom ! ! ! ! line con 0 line aux 0 line vty 0 4 password password login ! end

Cisco Configuration 2

Cisco Config-2 For 1490+ FRF12
version 12.2 service timestamps debug uptime service timestamps log uptime no service password-encryption ! hostname 3620 ! enable password password ! ip subnet-zero ! ! ! !

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```
class-map match-all voice-signaling
  match access-group 103
class-map match-all voice-traffic
  match access-group 102
!
!
policy-map voice-policy
  class voice-signaling
   bandwidth 8
  class voice-traffic
    priority 40
  class class-default
   fair-queue
!
call rsvp-sync
!
voice service voip
 h323 call start slow
 h323 h245 tunnel disable
!
!
interface FastEthernet0/0
 ip address 192.168.42.100 255.255.255.0
 speed 100
 half-duplex
!
interface Serial0/0
 bandwidth 64
 no ip address
 encapsulation frame-relay
 no ip mroute-cache
 frame-relay traffic-shaping
frame-relay lmi-type ansi
!
interface Serial0/0.1 point-to-point
 ip address 192.168.10.1 255.255.255.252
 frame-relay interface-dlci 500
  class VOIP OfficevFR
 frame-relay ip rtp header-compression
 frame-relay ip rtp compression-connections 60
!
ip classless
ip route 0.0.0.0 0.0.0.0 192.168.10.2
ip http server
ip pim bidir-enable
!
!
map-class frame-relay VOIP OfficevFR
```

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```
 no frame-relay adaptive-shaping
 frame-relay cir 64000
 frame-relay bc 640
 frame-relay be 0
 frame-relay mincir 64000
 service-policy output voice-policy
 frame-relay fragment 80
!
access-list 102 permit udp any any dscp ef
access-list 103 permit tcp any eq 1720 any
access-list 103 permit tcp any any eq 1720
!
voice-port 1/0/0
!
voice-port 1/0/1
!
dial-peer cor custom
!
!
!
dial-peer voice 501 voip
 destination-pattern 7...
 session target ipv4:192.168.42.1
 codec g729br8
!
dial-peer voice 402 pots
 destination-pattern 401
 port 1/0/1
!
dial-peer voice 401 pots
 destination-pattern 400
 port 1/0/0
!
dial-peer voice 444 pots
!
dial-peer voice 4545 voip
!
dial-peer voice 701 voip
!
!
line con 0
line aux 0
line vty 0 4
 password password
 login
!
end
```

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The Cisco Frame Relay switch Configuration shown below was used for the as the Frame Relay Network; the links speeds (clockrate) are representative only; these rate were not necessarily used in all tests

Cisco Config- For Frame Relay switching

```
Current configuration : 1481 bytes
!
version 12.2
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname 2620
!
enable password password
!
ip subnet-zero
!
!
no ip domain-lookup
!
frame-relay switching
call rsvp-sync
!
!
!
interface FastEthernet0/0
 ip address 192.168.50.253 255.255.255.0
 no keepalive
 duplex auto
 speed auto
!
interface Serial0/0
 no ip address
 encapsulation frame-relay
 no fair-queue
 clockrate 64000
 frame-relay lmi-type ansi
 frame-relay intf-type dce
 frame-relay route 500 interface Serial0/2 500
 frame-relay lmi-t392dce 30
!
interface Serial0/1
 no ip address
```

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Cisco Config- For Frame Relay switching
encapsulation frame-relay no fair-queue clockrate 256000 frame-relay lmi-type q933a frame-relay intf-type dce frame-relay route 60 interface Serial0/2 60 frame-relay lmi-t392dce 30 ! interface Serial0/2 no ip address encapsulation frame-relay no fair-queue clockrate 768000 frame-relay lmi-type cisco frame-relay intf-type dce frame-relay route 60 interface Serial0/1 60 frame-relay route 500 interface Serial0/0 500 frame-relay route 800 interface Serial0/3 800 frame-relay lmi-t392dce 30 ! interface Serial0/3 no ip address encapsulation frame-relay no fair-queue frame-relay lmi-type ansi frame-relay intf-type dce frame-relay route 800 interface Serial0/2 800 frame-relay lmi-t392dce 30 ! ip classless ip http server ip pim bidir-enable ! ! dial-peer cor custom ! ! ! ! line con 0 line aux 0 line vty 0 4 password password login ! end

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Cisco Config- For Frame Relay switching
2620#

IP Office Comparison with Cisco QOS

The table below illustrates the main differences between IP Office and Cisco implementation of QoS policy management with respect to VoIP.

Description	***IP Office***	***Cisco***
Voice packet matching	• DSCP • TCP port 1720	• DSCP • UDP port number • Host address
Bandwidth Control	Three Queue types 1. Voice traffic 2. signalling 3. Non-voice IP Office makes all available bandwidth accessible to Voice traffic. Care should be taken to ensure the VPN line control to restrict the amount of voice calls that can occupy the link.	Three Queue types 1. Voice traffic 2. signalling 3. Non-voice Low Latency Queue (LLQ). Maximum of 75-80% link bandwidth can be assigned to Voice traffic/signalling class

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Debugging Cisco QOS and VoIP

The follow command was found useful in diagnosing packet loss and the status on the Cisco of Cisco's Low Latency Queuing (LLQ).

!--- LLQ Verification

```
!-----------------------------------

maui-voip-austin#show policy-map int   multilink 1
 Multilink1
 Service-policy output: voice-policy
 Class-map: voice-signaling (match-all)
!—This is the class for the voice signaling traffic
        10 packets, 744 bytes
        5 minute offered rate 0 BPS, drop rate 0 BPS
        Match: access-group 103
        Weighted Fair Queueing
        Output Queue: Conversation 42
        Bandwidth 8 (kbps) Max Threshold 64 (packets)
        (pkts matched/bytes matched) 10/744
        (depth/total drops/no-buffer drops) 0/0/0
 Class-map: voice-traffic (match-all)
!—This is PQ class for the voice traffic
        458 packets, 32064 bytes
        5 minute offered rate 0 BPS, drop rate 0 BPS
        Match: access-group 102
        Weighted Fair Queueing
        Strict Priority
        Output Queue: Conversation 40
        Bandwidth 15 (kbps) Burst 375 (Bytes)
!—Notice that the PQ bandwidth was lowered to force packet drops.
        (pkts matched/bytes matched) 458/29647
        (total drops/bytes drops) 91/5890
!—Some packets were dropped. In a well designed link,
!—there should be no (or few) drops of the PQ class.
```

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