Troubleshooting QoS Choppy Voice Issues

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Introduction

For Packet Voice to be a realistic replacement for standard public switched telephone network (PSTN) Telephony services, the received quality of Packet Voice must be comparable to that of basic telephone services. This means consistently high−quality voice transmissions. Like other real−time applications, Packet Voice has a wide bandwidth and is delay sensitive. For voice transmissions to be intelligible (not choppy) to the receiver, voice packets cannot be dropped, excessively delayed, or suffer varying delay (otherwise known as jitter). This document describes various Quality of Service (QoS) considerations that help troubleshoot choppy voice issues. The main reasons for choppy voice problems are lost and delayed voice packets.

Prerequisites

Requirements

Readers of this document should be knowledgeable of these:

- Basic configuration of Packet Voice (VoIP, Voice over Frame Relay (VoFR) or Voice over ATM (VoATM) as per their requirement).
- Basic understanding of voice prioritization, fragmentation, different codecs and their bandwidth requirements.

Components Used

The information in this document applies to all Cisco voice gateways software and hardware versions.

The information presented in this document was created from devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If you are working in a live network, ensure that you understand the potential impact of any command before using it.

Conventions

For more information on document conventions, refer to the Cisco Technical Tips Conventions.

Causes of Choppy Voice

Choppy voice quality is caused by voice packets being either variably delayed or lost in the network. When a voice packet is delayed in reaching its destination, the destination gateway has a loss of real−time information. In this event, the destination gateway must predict what the content of the missed packet can possibly be. The prediction leads to the received voice not having the same characteristics as the transmitted voice. This leads to a received voice that sounds robotic. If a voice packet is delayed beyond the prediction capability of a receiving gateway, the gateway leaves the real−time gap empty. With nothing to fill up that gap at the receiving end, part of the transmitted speech is lost. This results in choppy voice. Many of the choppy voice issues are resolved by making sure that the voice packets are not very delayed (and more than that, not variably delayed). Sometimes, voice activity detection (VAD) adds front−end clipping to a voice conversation. This is another cause of choppy (or clipped) voice.

The various sections in this document show how to minimize the instance of choppy voice. Most of these measures require assuring the introduction of minimum jitter in your voice network.

Bandwidth Requirement

Before you consider applying any measures for minimizing jitter, provision sufficient network bandwidth to support real−time voice traffic. For example, an 80 kbps G.711 VoIP call (64 kbps payload + 16 kbps header) sounds poor over a 64 kbps link because at least 16 kbps of the packets (which is 20 percent) are dropped. The bandwidth requirements vary based on the codec used for compression. Different codecs have different payloads and header requirements. Usage of VAD also affects the bandwidth requirement. If Real Time Protocol (RTP) header compression (cRTP) is used, it can further lower the bandwidth requirement.

For example, the bandwidth required for a voice call using the G.729 codec (default 20 byte payload) with cRTP, is like this:

- Voice payload + compressed (RTP header + User Datagram Protocol (UDP) header + IP header) +Layer 2 header

This is equivalent to:

- 20 bytes + compressed (12 bytes + 8 bytes + 20 bytes) + 4 bytes

This equals:

- 28 bytes, since the header compression reduces the IP RTP header to a maximum of 4 bytes. This yields 11.2 kbps at an 8 kbps codec rate (50 packets per second).

For more information, refer to Voice over IP − Per Call Bandwidth Consumption.

Voice Traffic Priority

There are two important components in prioritizing voice. The first is classifying and marking interesting voice traffic. The second is prioritizing the marked interesting voice traffic. The two subsections here discuss various approaches to classifying, marking, and prioritizing voice.

Classification and Marking

In order to guarantee bandwidth for VoIP packets, a network device must be able to identify the packets in all the IP traffic that flows through it. Network devices use the source and destination IP address in the IP header, or the source and destination UDP port numbers in the UDP header, to identify VoIP packets. This identification and grouping process is called classification. It is the basis for providing any QoS.

Packet classification can be processor intensive. Therefore, classification needs to be done as far out towards the edge of the network as possible. Because every hop still needs to make a determination on the treatment a packet should receive, you need to have a simpler, more efficient classification method in the network core. This simpler classification is achieved through marking or setting the Type of Service (ToS) byte in the IP header. The three most significant bits of the ToS byte are called IP Precedence bits. Most applications and vendors currently support setting and recognizing these three bits. Marking is evolving so that the six most significant bits of the ToS byte, called the Differentiated Services Code Point (DSCP), can be used. Refer to the Request for Comments (RFC).

Differentiated Services (DiffServ) is a new model in which traffic is treated by intermediate systems with relative priorities based on the ToS field. Defined in RFC 2474 and RFC 2475 , the DiffServ standard supersedes the original specification for defining packet priority described in RFC 791 . DiffServ increases the number of definable priority levels by reallocating bits of an IP packet for priority marking. The DiffServ architecture defines the DiffServ field. It supersedes the ToS byte in IP V4 to make Per−Hop Behavior (PHB) decisions about packet classification and traffic conditioning functions such as metering, marking, shaping, and policing. In addition to the previously mentioned RFCs, RFC 2597 defines the Assured Forwarding (AF) classes. This is a breakdown of the DSCP fields. For more information on DSCP, refer to Implementing Quality of Service Policies with DSCP.

ToS Byte − P2 P1 P0 T3 T2 T1 T0 CU

IP precedence: three bits (P2−P0), ToS: four bits (T3−T0), CU: one bit

DiffServ Field − DS5 DS4 DS3 DS2 DS1 DS0 ECN ECN

DSCP: six bits (DS5−DS0), ECN: two bits

XXX00000 Bits 0, 1, 2 (DS5, DS4, DS3) are Precedence bits, where:

- 111 = Network Control = Precedence 7
- 110 = Internetwork Control = Precedence 6
- 101 = CRITIC/ECP = Precedence 5
- 100 = Flash Override = Precedence 4
- 011 = Flash = Precedence 3
- 010 = Immediate = Precedence 2
- 001 = Priority = Precedence 1
- 000 = Routine = Precedence 0

000XXX00 Bits 3, 4, 5 (DS2, DS1, DS0) are Delay, Throughput, and Reliability bits.

- Bit 3 = Delay [D] (0 = Normal; 1 = Low)

- Bit 4 = Throughput [T] (0 = Normal; 1 = High)
- Bit 5 = Reliability [R] (0 = Normal; 1 = High)

000000XX Bits 6, 7: ECN

These two sections discuss two ways in which classification and marking are done.

Voice Dial Peers to Classify and Mark Packets

With Cisco VoIP gateways, you usually use voice dial peers to classify the VoIP packets and mark the IP Precedence bits. This configuration shows how to mark the IP Precedence bits:

```
dial−peer voice 100 voip
destination−pattern 100
session target ipv4:10.10.10.2
ip precedence 5
```

In the example above, any VoIP call that matches the **dial−peer voice 100 voip** command has all of its voice payload packets set with IP Precedence 5. This means that the three most significant bits of the IP ToS byte are set to 101.

```
dial−peer voice 100 voip
destination−pattern 100
session target ipv4:10.10.10.2
ip qos dscp ef media
ip qos dscp af31 signaling
```

In the example above, any VoIP call that matches the **dial−peer voice 100 voip** command has all of its media payload packets (voice packets) set with Expedited Forwarding (EF) bit pattern 101110. All the signaling packets are set with AF bit pattern 011010.

Note: The **ip qos dscp** command is supported since Cisco IOS® Software Release 12.2(2)T. The IP Precedence is no longer available in Cisco IOS Software Release 12.2T. However, the same is achieved by the **ip qos dscp** command. IP precedence 5 (101) maps to IP DSCP 101000. For more information, refer to Classifying VoIP Signaling and Media with DSCP for QoS.

Modular QoS CLI to Classify and Mark Packets

The recommended classification and marking method to use is the modular QoS CLI. This is a template−based configuration method that separates the classification from the policy. This allows multiple QoS features to be configured together for multiple classes. Use a **class−map** command to classify traffic based on various match criteria and a **policy−map** command to determine what needs to happen to each class. Apply the policy to incoming or outgoing traffic on an interface by issuing the **service−policy** command. This configuration example shows how to use modular QoS CLI to classify and mark packets:

```
access−list 100 permit udp any any range 16384 32767
access−list 101 permit tcp any any eq 1720
!
class−map match−all voip
match access−group 100
class−map match−all control
match access−group 101
!
policy−map mqc
class voip
set ip precedence 5
class control
set ip precedence 5
```

```
class class−default
set ip precedence 0
!
interface Ethernet0/0
service−policy input mqc
```

In this configuration example, any traffic that matches Access Control List (ACL) 100 is classified as "class voip" and set with IP Precedence 5. This means that the three most significant bits of the IP ToS byte are set to 101. ACL 100 matches the common UDP ports used by VoIP. Similarly ACL 101 matches H.323 signaling traffic (Transmission Control Protocol (TCP) port 1720). All other traffic is set with IP Precedence 0. The policy is called "mqc". It is applied to incoming traffic on Ethernet 0/0.

Prioritizing

After every hop in the network is able to classify and identify the VoIP packets (either through port/address information or through the ToS byte), those hops then provide each VoIP packet with the required QoS. At that point, configure special techniques to provide priority queuing to make sure that large data packets do not interfere with voice data transmission. This is usually required on slower WAN links where there is a high possibility of congestion. Once all the interesting traffic is placed into QoS classes based on their QoS requirements, provide bandwidth guarantees and priority servicing through an intelligent output queuing mechanism. A priority queue is required for VoIP.

Note: Use any queuing mechanism that effectively gives VoIP high priority. However, Low Latency Queuing (LLQ) is recommended because it is flexible and easy to configure.

LLQ uses the modular QoS CLI configuration method to provide priority to certain classes and to provide guaranteed minimum bandwidth for other classes. During periods of congestion, the priority queue is policed at the configured rate so that the priority traffic does not use up all the available bandwidth. (If the priority traffic monopolizes the bandwidth, it prevents bandwidth guarantees for other classes from being met.) If you provision LLQ correctly, the traffic that goes into the priority queue should never exceed the configured rate.

LLQ also allows queue depths to be specified to determine when the router needs to drop packets if there are too many packets that wait in any particular class queue. There is also a **class−default** command that is used to determine treatment of all traffic not classified by a configured class. The class−default is configured with a **fair−queue** command. This means that each unclassified flow is given an approximately equal share of the remaining bandwidth.

This example shows how to configure LLQ. For more information, refer to Low Latency Queuing:

```
access−list 100 permit udp any any range 16384 32000
access−list 101 permit tcp any any eq 1720
access−list 102 permit tcp any any eq 80
access−list 103 permit tcp any any eq 23
!
class−map match−all voip
match access−group 100
class−map match−all voip−control
natch access−group 101
class−map match−all data1
match access−group 102
class−map match−all data2
match access−group 103
!
policy−map llq
class voip
priority 32
class voip−control
bandwidth 8
class data1
```

```
        bandwidth 64
        class data2
        bandwidth 32
        class class−default
        fair−queue
        !
        interface Serial1/0
        bandwidth 256
        service−policy output llq
```

In this example, any traffic that matches ACL 100 is classified as "class voip" (meaning voice traffic). It is given a high priority up to 32 kbps. ACL 100 matches the common UDP ports used by VoIP. Access−list 101 matches the H.323 signaling traffic (TCP port 1720). Class data1 matches web traffic (TCP port 80 as seen in Access List 102) and guarantees 64 kbps. Class data2 matches Telnet traffic (TCP port 23 as seen in ACL 103) and guarantees 32 kbps. The default class is configured to give an equal share of the remaining bandwidth to unclassified flows. The policy is called "llq". It is applied to outgoing traffic on Serial1/0, which has a total bandwidth of 256 kbps.

Note: By default, the total guaranteed bandwidth and priority bandwidth for all classes needs to be less than 75 percent of the interface bandwidth. Modify this percentage by issuing the **max−reserved bandwidth interface** command.

This table compares different software queuing mechanisms with their respective benefits and limitations.

Software Queuing Mechanism	Description	Benefits	Limitations
First−In−First−Out (FIFO)	Packets arrive and leave the queue in exactly the same order.	Simple configuration and fast operation.	No priority servicing or bandwidth guarantees possible.[1]
Weighted Fair Queuing (WFQ)	A hashing algorithm that flows into separate queues where weights are used to determine how many packets are serviced at a time. You define weights by setting IP Precedence and DSCP values.	Simple configuration. Default on links less than 2 Mbps.	No priority servicing or bandwidth guarantees possible.[1]
Custom Queuing (CQ)	Traffic is classified into multiple queues with configurable	Has been available for a few years. It allows approximate	No priority servicing is possible. Bandwidth guarantees are

	queue limits. The queue limits are calculated based on average packet size, Maximum Transmission Unit (MTU), and the percentage of bandwidth to be allocated. Queue limits (in number of bytes) are de−queued for each queue. Therefore it provides the allocated bandwidth statistically.	bandwidth allocation for different queues.	approximate. There are a limited number of queues. Configuration is relatively difficult.[1]
Priority Queuing (PQ)	Traffic is classified into high, medium, normal, and low priority queues. The high priority traffic is serviced first, followed by medium, normal and low priority traffic.	Has been available for a few years. It provides priority servicing.	Higher priority traffic starves the lower priority queues of bandwidth. No bandwidth guarantees are possible.[2]
Class−Based Weighted Fair Queuing (CBWFQ)	Modular QoS CLI is used to classify traffic. Classified traffic is placed into reserved bandwidth queues or a default unreserved queue. A	Similar to LLQ, except that there is no priority queue. Simple configuration and ability to provide bandwidth guarantees.	No priority queuing is possible.[3]

	scheduler services the queues based on weights so that the bandwidth guarantees are honored.		
Priority Queue Weighted Fair Queuing (PQ−WFQ), also called IP RTP Priority	A single interface command is used to provide priority servicing to all UDP packets destined to even port numbers within a specified range.	Simple, one command configuration. Provides priority servicing to RTP packets.	All other traffic is treated with WFQ. Real−Time Conferencing Protocol (RTCP) traffic is not prioritized. No guaranteed bandwidth capability.[4]
LLQ, previously called Priority Queue Class−Based Weighted Fair Queuing (PQCBWFQ)	Modular QoS CLI with priority queue is used to classify traffic. Classified traffic is placed into a priority queue, reserved bandwidth queues, or a default unreserved queue. A scheduler services the queues based on weights so that the priority traffic is sent first (up to a certain policed limit during congestion) and the	Simple configuration. Ability to provide priority to multiple classes of traffic and give upper bounds on priority bandwidth utilization. You can also configure bandwidth guaranteed classes and a default class.	No mechanism to provide multiple levels of priority yet, all priority traffic is sent through the same priority queue. Separate priority classes can have separate upper priority bandwidth bounds during congestion. However, sharing of priority queue between applications can potentially introduce jitter.[4]

	bandwidth guarantees are met.		

1. Not suitable for voice.
2. Needs bandwidth guaranteed for voice.
3. Needs latency to be taken care of.
4. Sufficient for voice.

Serialization Delay

Even if queuing works at its best and prioritizes voice traffic, there are times when the priority queue is empty and a packet from another class is serviced. Packets from guaranteed bandwidth classes must be serviced based on their configured weight. If a priority voice packet arrives in the output queue while these packets are being serviced, the voice packet can wait a significant amount of time before it is sent. Voice packets experience serialization delay when they have to wait behind larger data packets.

Serialization delay can introduce the worst form of jitter for voice packets. If the voice packets have to wait behind a data packet that is as large as 1500 bytes, on a slower link, this translates to a huge delay. The serialization delay is vastly different if the data packet is 80 bytes, as shown in this example:

- Serialization delay on a 64 kbps link due to a 1500 bytes packet = 1500*8/64000 = 187.5 ms.
- Serialization delay on a 64 kbps link due to a 80bytes packet = 80*8/64000 = 10 ms.

Therefore, a voice packet potentially has to wait up to 187.5 ms before it is sent if it gets stuck behind a single 1500−byte packet on a 64 kbps link. On the other hand, another voice packet has to wait for only 10 ms at the destination gateway. This results into a huge jitter that occurs due to the variance in the inter−packet delay. On the originating gateway, voice packets are usually sent every 20 ms. With an end−to−end delay budget of 150 ms and strict jitter requirements, a gap of more than 180 ms is unacceptable.

Introduce a mechanism of fragmentation that ensures that the size of one transmission unit is less than 10 ms. Any packets that have more than 10 ms serialization delay need to be fragmented into 10 ms chunks. A 10 ms chunk or fragment is the number of bytes that is sent over the link in 10 ms. Calculate the size by using the link speed, as shown in this example:

- Fragmentation size = (0.01 seconds * 64,000 bps) / (8 bits/byte) = 80 bytes

It takes 10 ms to send an 80−byte packet or fragment over a 64 kbps link.

In case of multiple ATM or Frame Relay Permanent Virtual Circuits (PVCs) on a single physical interface, configure fragmentation values (on all PVCs) based on the PVC that has the lowest bandwidth available. For example, if there are three PVCs that have a guaranteed bandwidth of 512 kbps, 128 kbps, and 256 kbps, then configure all three PVCs with a fragment size of 160 bytes (the lowest speed is 128 kbps that requires a 160−byte fragment size). These values are recommended for different link speeds:

Link Speed (kbps)	Fragmentation Size (bytes)
56	70
64	80
128	160
256	320
512	640
768	960
1024	1280
1536	1920

Note: No fragmentation is required if the fragment size is larger than the link MTU size. For example, for a T1 link with a 1500−byte MTU, the fragment size is 1920 bytes. Therefore, no fragmentation is required. The packet fragmentation size should never be lower than the VoIP packet size. Do not fragment VoIP packets. Fragmenting these packets causes numerous call setup and quality issues.

There are currently three link fragmentation and interleaving mechanisms available. For further explanation of various delays introduced in a packet network, refer to Understanding Delay in Packet Voice Networks. This table lists their benefits and limitations:

Link Fragmentation and Interleaving (LFI) Mechanism	**Description**	**Benefits**	**Limitations**
MTU fragmentation with WFQ	Interface level command to change MTU size or IP MTU size. Used to fragment large IP packets to specified MTU size. LFI uses WFQ to interleave real−time packets in between the fragments.	Simple configuration.	Fragments are reassemble only by the receiving application. Therefore, inefficient use of the network. Only IP packets with the Do not Fragment (DF) bit not set can handle fragmentation well. Highly processor intensive. Not recommended.
Multilink Point−to−Point Protocol (MLPPP) LFI	On point−to−point serial links, MLPPP must first be configured, then a fragmentation size must be set in ms. Interleaving must also be enabled on the multilink interface.	Packets are fragmented on one end of the link and reassembled at the other. Several links can be combined to act as a large virtual pipe.	Only available on links configured for PPP. Solutions for PPP over Frame Relay or PPP over ATM are also supported in Cisco IOS Software Release 12.1(5)T or later.
Frame Relay Fragmentation (FRF.12)	On Frame Relay PVCs, the **frame−relay traffic−shaping** command must be enabled and a	Packets are fragmented on one end of the PVC and reassembled at the other.	Only available on Frame Relay PVCs with the **frame−relay traffic−shaping** command

	fragmentation size set under the map−class.		enabled.

VAD

A regular voice conversation consists of several moments of silence. A typical voice conversation consists of 40 to 50 percent silence. Since there is not any voice going through the network for 40 percent of a voice call, some bandwidth can be saved by deploying VAD. With VAD, the gateway looks out for gaps in speech. It replaces those gaps with comfort noise (background noise). Thus, an amount of bandwidth is saved. However, there is a trade−off. There is a small time (in order of milliseconds), before the codecs detect speech activity followed by a period of silence. This small time results in the front−end clipping of received voice. To avoid activation during very short pauses and to compensate for clipping, VAD waits approximately 200 ms after speech stops before it stops transmission. Upon restarting transmission, it includes the previous 5 ms of speech along with the current speech. VAD disables itself on a call automatically if ambient noise prevents it from distinguishing between speech and background noise. However, if the bandwidth is not an issue, turn the VAD off.

Tune VAD Parameters

There are two parameters that dictate the functioning of VAD. These are the **music−threshold** and **voice vad−time** commands.

music−threshold

An initial threshold is decided which governs when VAD needs to become active. This is controlled by defining the **music−threshold** *threshold_value* command on a voice−port, as shown in this example. The range for this is from −70 Decibels Per Milliwatt (dBm) to −30 dBm. The default value for this is −38 dBm. Configuring a lower value (towards −70 dBm) results in VAD becoming active at a much lower signal strength (the volume must drop really low before it is considered as silence). Configuring a higher value (closer to −30 dBm) results in VAD becoming active for even a small drop in voice signal strength. It drives the playout to play comfort noise packets more often. However, this sometimes leads to minor clipping of audio.

```
3640−6#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
3640−6(config)#voice−port 3/0/0
3640−6(config−voiceport)#music−threshold ?
WORD Enter a number b/w (−70 to −30)
3640−6(config−voiceport)#music−threshold −50
3640−6(config−voiceport)#end
3640−6#
3640−6#show run | be voice−portvoice−port 3/0/0 music−threshold −50
```

voice vad−time

Once the VAD becomes active, the component of background noise and comfort noise is controlled by configuring the **voice vad−time** *timer_value* command under the global configuration, as shown in this example. This is the delay time in milliseconds for silence detection and suppression of voice packet transmission. The default value for the holdover time is 250 msec. This means that within 250 msec, comfort noise begins. The range for this timer is 250 msec to 65536 msec. If a high value is configured for this, comfort noise comes into play much later (background noise continues to be played). If this is configured for 65536 msec, then the comfort noise is turned off. A higher value for this timer is desired for smoother transition between background noise and comfort noise. The downside to configuring the **voice vad−time** command to a high level is that it does not achieve the desired 30 to 35 percent bandwidth saving.

```
3640−6#
3640−6#
3640−6#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
3640−6(config)#voice vad−time ?
<250−65536> milliseconds
3640−6(config)#voice vad−time 750
3640−6(config)#end
3640−6#
3640−6#
3640−6#
3640−6#show run | be vad−time voice vad−time 750
```

Typical Configuration Examples for QoS

A typical scenario for setting up VoIP calls is either over a frame−relay link or over a PPP link. These are configuration examples for these scenarios.

VoIPoFR − QoS Configuration Example

In this example (which contains only relevant sections of the configuration), it is assumed that the frame−relay circuit speed is 256 kbps. The guaranteed Committed Information Rate (CIR) on PVC 100 is 64 kbps and on PVC 200 is 192 kbps. PVC 100 is used to carry both data and voice. PVC 200 is used only to carry data. A maximum of four simultaneous voice calls exists at any given time. Configure fragmentation on both PVCs based on the CIR of the lowest−bandwidth−voice−PVC (PVC carrying voice). Based on the examples in this document, that means the fragmentation size is decided based on PVC 100's CIR (which is 64 kbps). As shown in the table of the Serialization Delay section, for a 64 kbps link, a fragmentation size of 80 bytes is required. The same fragmentation size needs to be configured for PVC 200.

For further details about the configuration of VoIP over Frame Relay, refer to VoIP over Frame Relay with Quality of Service (Fragmentation, Traffic Shaping, LLQ / IP RTP Priority).

```
3660−1#show run
Building configuration...
!
class−map match−any voip
match ip rtp 16384 16383
match ip dscp 26 46
class−map match−all voip−control
match access−group 101
!
!
policy−map VoIPoFR
class voip
priority 48
class voip−control
bandwidth 8
class class−default
fair−queue
!
voice call send−alert
voice rtp send−recv
!
!
interface Serial4/0:0
bandwidth 256
no ip address
encapsulation frame−relay
frame−relay traffic−shaping
!
interface Serial4/0:0.1 point−to−point
bandwidth 64
```

```
ip address 10.10.10.10 255.255.255.0
frame−relay ip rtp header−compression
frame−relay interface−dlci 100
 class voice
!
interface Serial4/0:0.2 point−to−point
bandwidth 192
ip address 20.20.20.20 255.255.255.0
frame−relay interface−dlci 200
class data
!
map−class frame−relay data
frame−relay fragment 80
frame−relay adaptive−shaping becn
frame−relay cir 256000
frame−relay bc 32000
frame−relay be 0
frame−relay mincir 192000
frame−relay fair−queue
!
map−class frame−relay voice
frame−relay fragment 80
no frame−relay adaptive−shaping
frame−relay cir 64000
frame−relay bc 640
frame−relay be 0
frame−relay mincir 64000
service−policy output VoIPoFR
!
!
access−list 101 permit tcp any any eq 1720
!
!
voice−port 3/1/0
!
voice−port 3/1/1
!
!
dial−peer voice 10 voip
incoming called−number .
destination−pattern 1408.......
session target ipv4:10.10.10.11
dtmf−relay h245−signal h245−alphanumeric
no vad
!
dial−peer voice 20 pots
destination−pattern 1234
port 3/1/0
!
dial−peer voice 21 pots
destination−pattern 5678
port 3/1/1
```

VoIP Over PPP − QoS Configuration Example

In this example (which contains only relevant sections of the configuration), it is assumed that the QoS needs to be configured for a point−to−point fractional T1 controller (that has twelve channels). A maximum of four simultaneous voice calls exists at any given time. The configuration task involves configuring this serial interface with PPP encapsulation, making it part of a multilink group, creating a multilink interface (that belongs to the same multilink group), and configuring all the QoS on the multilink interface. For further details about the configuration of VoIP over PPP, refer to VoIP over PPP Links with Quality of Service (LLQ / IP RTP Priority, LFI, cRTP).

```
3660−1#show run
```

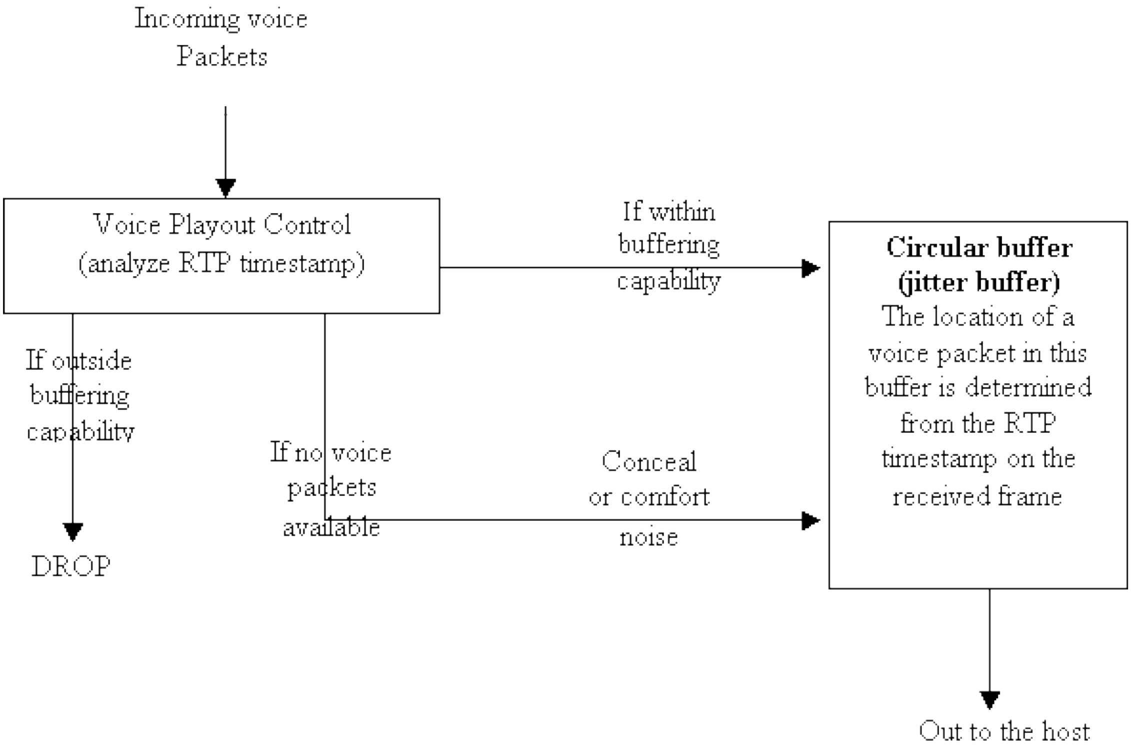
```
Building configuration...
!
class-map match-any voip
match ip rtp 16384 16383
match ip dscp 26 46
class-map match-all voip-control
match access-group 101
!
!
policy-map VoIPoPPP
class voip
priority 48
class voip-control
bandwidth 8
class class-default
fair-queue
!
voice call send-alert
voice rtp send-recv
!
!
interface Multilink7
bandwidth 768
ip address 10.10.10.10 255.255.255.0
ip tcp header-compression iphc-format
service-policy output VoIPoPPP
no cdp enable
ppp multilink
ppp multilink fragment-delay 10
ppp multilink interleave
multilink-group 7
ip rtp header-compression iphc-format
!
!
interface Serial4/0:0
bandwidth 768
no ip address
encapsulation ppp
no fair-queue
ppp multilink
multilink-group 7
!
!
access-list 101 permit tcp any any eq 1720
!
voice-port 3/1/0
!
voice-port 3/1/1
!
!
dial-peer voice 10 voip
incoming called-number .
destination-pattern 1408.......
session target ipv4:10.10.10.11
dtmf-relay h245-signal h245-alphanumeric
no vad
!
dial-peer voice 20 pots
destination-pattern 1234
port 3/1/0
!
dial-peer voice 21 pots
destination-pattern 5678
port 3/1/1
!
```

Jitter and Playout Mechanism

There are always some uncontrolled entities in a network that contribute towards further delays and jitter in the received voice packets. By modifying the jitter buffer on the terminating gateway the uncontrolled jitter is resolved in the voice network.

Playout Mechanism

The jitter buffer is a time buffer. It is provided by the terminating gateway to make the playout mechanism more effective. This is a functional diagram of the playout mechanism:


Incoming voice Packets
Voice Playout Control (analyze RTP timestamp)
If within buffering capability
Circular buffer (jitter buffer)
The location of a voice packet in this buffer is determined from the RTP timestamp on the received frame
If outside buffering capability
DROP
If no voice packets available
Conceal or comfort noise
Out to the host

When the playout control receives a voice packet, it analyzes the RTP timestamp. If the voice packet is delayed beyond the holding capacity of the jitter buffer, then the packet is immediately dropped. If the packet is within the buffering capability, it is placed in the jitter buffer. The location of this packet in the jitter buffer depends on the RTP timestamp calculated for that packet. In the event there is no available voice packet, the playout control tries to conceal it (predicts the missed out packet). If VAD is enabled, comfort noise is played.

The responsibility of the playout control is to handle the events of lost packets, duplicated packets, corrupted packets, and out−of−sequence packets. These events are handled by time aligning the jittered voice packets, playing comfort noise (if VAD is configured), or even regenerating dual tone multifrequency (DTMF) tones to be played out to the host.

The concealment of a voice packet is done by either prediction concealment or by silence concealment. Prediction concealment is based on the previous packet and the next packet (if available). It works best with low bitrate codecs (5 kbps to 16 kbps). Loss of voice packets for a high bitrate codec (32 kbps to 64 kbps) can potentially result in poor prediction concealment. Prediction concealment starts when there are low and infrequent delays or a lesser number of packet loss. Too much prediction concealment can lead to robotic voice quality. Silence concealment is the worst form of prediction concealment. It comes into play when there is no information available to predict. It is simply a background concealment. It starts when there are high delays and more number of packet loss. Too much silence concealment leads to choppy voice quality. The prediction concealment is good for 30 msecs after which the silence conceal comes into play.

Jitter Buffer

The jitter buffer is confined by a high water mark and a low water mark. The high water mark is the upper time limit within which a packet is expected to arrive for on−time playout. Packets that arrive after the high water mark are marked as late packets or lost packets. The low water mark is the minimum time within which a packet is expected to arrive for on−time playout. Packets that arrive before the low water mark are considered early packets (it can still be played out on time).

If the terminating gateway continues to see an increment in the arrival of late packets, it increases the high water mark. This value for high water mark remains the same throughout the duration of the call. This is increased up to a maximum defined in the configuration. In a similar manner, the terminating gateway observes the number of early packets received. If these packets start to frequent the gateway, it reduces the low water mark. This value remains the same throughout the duration of the call. This mode of jitter buffer is referred to as "adaptive mode," where the terminating gateway adapts its jitter buffer based on the traffic pattern. The other mode is "fixed mode." In the fixed mode, there is one initial value for the low water mark and the high water mark. This value is based on the estimated received delay (see the show voice call <port−number> section of this document).

For further details on jitter buffer, refer to Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms).

Identify Delay and Jitter

This section describes how to identify jitter in your network.

show call active voice

The **show call active voice brief** command gives a great deal of information about an ongoing conversation. This output displays some important points that are learned from this command:

```
11E4 : 2170927hs.1 +600 pid:10 Answer 1000 active
dur 00:08:43 tx:26157/522967 rx:7044/139565
Tele 3/0/0:9: tx:151310/755/0ms g729r8 noise:−62 acom:0 i/0:−56/−48 dBm
11E4 : 2171198hs.1 +329 pid:20 Originate 2000 active
dur 00:08:43 tx:7044/139565 rx:26165/523127
IP 30.30.30.29:18682 rtt:51ms pl:148590/290ms lost:0/0/15 delay:65/60/132ms g729r8
```

From the **show call active voice brief** command output, you see that whatever is received on the Telephony leg (rx:7044) is transmitted to the IP leg (tx:7044). The same is true for packets received on the IP legs (26165) that are forwarded to the Telephony leg (26157). The difference in the number of packets received on the IP leg versus the number of packets transmitted on the Telephony leg is contributed to late packets that do not make it in time.

This output of the **show call active voice** command (without the "brief" keyword), points to further details about parameters that directly identify jitter.

```
GapFillWithSilence=850 ms
GapFillWithPrediction=9230 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
```

show voice call <port−number>

The **show voice call** *port−number* command provides useful information. Make sure to be either consoled in the gateway, or if you are Telneted into a gateway, make sure you have issued the **terminal monitor** command from the exec level.

Note: This command is not available on the AS5x00/AS5x50 platforms.

In this output, the value for Rx Delay Est (ms) is 71. This is the current jitter buffer value. A value for the high water mark and low water mark is deduced on this. An average initial value for the high water mark is 70 msec, while that for the low water mark is 60 msec. Once an initial value is set, the gateway keeps track of any early packets or late packets received. As is seen in the output here, the prediction concealment drops are close to 250 ms, while the silence concealment are 30 ms. There is always a higher value for prediction concealment since silence concealment is only a worse case scenario of Prediction concealment. For every Prediction concealment drop, there is an increase in the buffer overflow discard.

If you see buffer discard, it does not necessarily mean that you see an increase in the high water mark. The high water mark is the upper limit of the jitter buffer. It changes only if a trend is observed. In other words, there should be a continuous flow of late packets. This results in an increase of the jitter buffer. In the output here, such a trend is present. Therefore, the high water mark is increased from 70 msec to 161 msec. If this value is not changed (and if you still see 14 late packets), it implies that these are sporadic late packets, not forming a trend.

From the output of the **show call active voice** command, look out for lost packets. For every lost packet, you see two packets that are out of sequence. This is seen on the Rx Non−Seq Pkts output. Since it is not a positive value, it is concluded that there has not been any packet losses either.

```
3640−6# ***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 195, Tx Sig Pkts: 0, Tx Comfort Pkts: 10
Tx Dur(ms): 192070, Tx Vox Dur(ms): 388, Tx Fax Dur(ms): 0
***DSP VOICE RX STATISTICS***
Rx Vox/Fax Pkts: 9604, Rx Signal Pkts: 0, Rx Comfort Pkts: 0
Rx Dur(ms): 192070, Rx Vox Dur(ms): 191560, Rx Fax Dur(ms): 0
Rx Non−seq Pkts: 0, Rx Bad Hdr Pkts: 0
Rx Early Pkts: 0, Rx Late Pkts: 14
***DSP VOICE VP_DELAY STATISTICS***
Clk Offset(ms): 0, Rx Delay Est(ms): 71
Rx Delay Lo Water Mark(ms): 60, Rx Delay Hi Water Mark(ms): 161
***DSP VOICE VP_ERROR STATISTICS***
Predict Conceal(ms): 250, Interpolate Conceal(ms): 0
Silence Conceal(ms): 30, Retroact Mem Update(ms): 0
Buf Overflow Discard(ms): 500, Talkspurt Endpoint Detect Err: 0
***DSP LEVELS***
TDM Bus Levels(dBm0): Rx −49.9 from PBX/Phone, Tx −41.7 to PBX/Phone
TDM ACOM Levels(dBm0): +2.0, TDM ERL Level(dBm0): +11.1
TDM Bgd Levels(dBm0): −58.9, with activity being voice
***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
```

Observe the Tx Comfort Pkts and Rx Comfort Pkts. As from the example outputs, it is concluded that the phone connected to this router mostly keeps quiet since you have lots of Tx Comfort Pkts. At the same time, you have zero Rx Comfort Pkts, which means that the other end continuously speaks.

Compare the output here with the previous command output. There is an increased number of Rx Late Pkts (from 14 to 26). However, there is no increment in the high water mark value. This indicates that the 12 packets are sporadically delayed. The Buffer Overflow discard is increased to 910 msecs. However, since there is no trend observed, the high water mark is not increased.

In the output here, you have a Rx Early Pkts: 3. This means that a packet arrives much before it is expected. As seen from the output here, the Jitter buffer has stretched itself to accommodate for any more early packets by reducing the low water mark from 60 to 51.

```
3640−6# ***DSP VOICE TX STATISTICS***
Tx Vox/Fax Pkts: 209, Tx Sig Pkts: 0, Tx Comfort Pkts: 11
Tx Dur(ms): 337420, Tx Vox Dur(ms): 416, Tx Fax Dur(ms): 0
***DSP VOICE RX STATISTICS***
Rx Vox/Fax Pkts: 16843, Rx Signal Pkts: 0, Rx Comfort Pkts: 1
Rx Dur(ms): 337420, Rx Vox Dur(ms): 335920, Rx Fax Dur(ms): 0
Rx Non−seq Pkts: 0, Rx Bad Hdr Pkts: 0
Rx Early Pkts: 3, Rx Late Pkts: 26
***DSP VOICE VP_DELAY STATISTICS***
Clk Offset(ms): 0, Rx Delay Est(ms): 72
Rx Delay Lo Water Mark(ms): 51, Rx Delay Hi Water Mark(ms): 161
***DSP VOICE VP_ERROR STATISTICS***
Predict Conceal(ms): 510, Interpolate Conceal(ms): 0
Silence Conceal(ms): 70, Retroact Mem Update(ms): 0
Buf Overflow Discard(ms): 910, Talkspurt Endpoint Detect Err: 0
***DSP LEVELS***
TDM Bus Levels(dBm0): Rx −51.5 from PBX/Phone, Tx −44.1 to PBX/Phone
TDM ACOM Levels(dBm0): +2.0, TDM ERL Level(dBm0): +11.9
TDM Bgd Levels(dBm0): −61.3, with activity being voice
***DSP VOICE ERROR STATISTICS***
Rx Pkt Drops(Invalid Header): 0, Tx Pkt Drops(HPI SAM Overflow): 0
```

Configure Jitter Buffer on a Gateway

The QoS guidelines covered in this document take care of the choppy or deteriorated voice quality issue. The configuration of playout−delay buffer is a workaround for improper QoS implementation in the network. Only use this as a stop−gap fix or as a tool for troubleshooting and narrowing down of the jitter problems introduced in the network.

Playout−delay Mode

The jitter buffer is configured for either fixed mode or adaptive mode. Under the adaptive mode, the gateway allows you to configure a minimum value for the jitter buffer, a maximum value, and a nominal value. The jitter buffer expects the packets to arrive within the nominal value range. The nominal value has to be either equal to or more than the minimum, and equal to or less than the maximum. The buffer expands up to the maximum value configured. This can extend up to 1700 msec. One issue with configuring high maximum value is the introduction of end−to−end delay. Choose the value of maximum playout−delay such that it does not introduce unwanted delay in the network. This output is an example of the jitter buffer configured for adaptive mode:

```
3640−6#
3640−6#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
3640−6(config)#voice−port 3/0/0
3640−6(config−voiceport)#playout−delay mode adaptive
3640−6(config−voiceport)#playout−delay maximum 400
3640−6(config−voiceport)#playout−delay nominal 70
3640−6(config−voiceport)#playout−delay minimum low
3640−6(config−voiceport)#^Z
3640−6#
3640−6#
3640−6#show run | begin 3/0/0
voice−port 3/0/0
playout−delay maximum 400
playout−delay nominal 70
playout−delay minimum low
playout−delay mode adaptive
!
```

Under the fixed mode, the gateway looks at the configured value for nominal. Although it allows you to configure the minimum and the maximum value for playout−delay, it is ignored when configured for fixed mode. When in the fixed mode, the high water mark value or the low water mark value always remains constant. It is based on the nominal value and based on Rx Delay Est (ms) value. So it is possible that under the fixed mode, you configure the value as 200 msec. However, if the estimated received delay is close to 100 ms, that is what the high water mark and the low water mark is set to for the entire duration of the call.

```
3640−6#
3640−6#configure terminal
Enter configuration commands, one per line. End with CNTL/Z.
3640−6(config)#voice−port 3/0/0
3640−6(config−voiceport)#playout−delay mode fixed
3640−6(config−voiceport)#playout−delay nominal 70
3640−6(config−voiceport)#^Z
3640−6#
3640−6#
3640−6#show run | begin 3/0/0
voice−port 3/0/0
playout−delay mode fixed
playout−delay nominal 70
!
```

For more details on the playout−delay configuration, refer to Playout Delay Enhancements for Voice over IP.

Related Information

- **Implementing Quality of Service Policies with DSCP**
- **Low Latency Queueing**
- **Comparing the bandwidth and priority Commands of a QoS Service Policy**
- **Configuring Link Fragmentation and Interleaving for Frame Relay and ATM Virtual Circuits**
- **Configuring Link Fragmentation and Interleaving for Multilink PPP**
- **Voice Technology Support**
- **Voice and Unified Communications Product Support**
- **Troubleshooting Cisco IP Telephony**
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