642-642

Implementing Cisco
Quality of Service

Study Guide
DEMO Version

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LIST OF ACRONYMS

ACL	Access Control List
AF	Assured Forwarding
ASIC	Application Specific Integrated Circuits
AVBO	Advanced Busy Out Monitor
BA	Behavior Aggregate
BECN	Backward Explicit Congestion Notification
BRI	Basic Rate ISDN
CAC	Call Admission Control
CAR	Committed Access Rate
CAS	Channel Associated Signaling
CB Marking	Class Based Marking
CB Policing	Class Based Policing
CBWFQ	Class Based Weighted Fair Queuing
CCS	Common Channel Signaling
CDT	Congestive Discard Threshold
CEF	Cisco Express Forwarding
CIR	Committed Information Rate
CLP	Cell Loss Priority
CoS	Class of Service
CQ	Custom Queuing
cRTP	Compressed RTP
CSA	Compression Service Adapter
CS-ACELP	Conjugate Structure Algebraic Code Excited Linear Prediction
DE	Discard Eligibility
DiffServ	Differentiated Services
DLCI	Data Link Connection Identifier
DSCP	Differentiated Services Code Point
DSBM	Designated Subnetwork Bandwidth Manager

DSP	Digital Signal Processor
DTE	Data Terminal Equipment
DTS	Distributed Traffic Shaping
dWFQ	Distributed Weighted Fair Queing
EF	Expedited Forwarding
EIGRP	Enhanced Interior Gateway Routing Protocol
ESR	Edge Services Router
FECN	Forward Explicit Congestion Notification
FIFO	First In, First Out
FRED	Flow Based Random Early Detection
FRF	Frame Relay Fragmentation
FRF.12	Frame Relay Forum Implementation Agreement 12
FRTS	Frame Relay Traffic Shaping
FTP	File Transfer Protocol
GSR	Gigabit Switch Router
GTS/DTS	Generic Traffic Shaping/Distributed Traffic Shaping
HDLC	High Level Data Link Control
ICCP	Intra Cluster Communications Protocol
ICPIF	Calculated Planning Impairment Factor
ICMP	Internet Control Message Protocol
IntServ	Integrated Services
IPM	Internetwork Performance Monitor
ISL	Inter Switch Link
ISP	Internet Service Provider
Kbps	kilobits per second (bandwidth)
LAN	Local Area Network

LAPB	Link Access Procedure Balanced
LFI	Link Fragmentation and Interleaving
LLQ	Low Latency Queuing
LVBO	Local Voice Busy-Out
MGCP	Media Gateway Control Protocol
MDDR	Modified Deficit Round Robin
MF	Multi-field
MIR	Minimum Information Rate
MLPPP LFI	Multilink PPP Fragmentation and Interleaving
MPD	Mark Probability Denominator
MPEG	Moving Pictures Experts Group
MPPC	Microsoft Point to Point Compression
MQC	Modular QoS Command Line Interface
MS	Milliseconds
MTU	Maximum Transmit Unit
NBAR	Network Based Application Recognition
NFS	Network File System
OOS	Out Of Service
OS	Operating System
PBR	Policy Based Routing
PCM	Pulse Code Modulation
PDLM	Packet Description Language Module
PHB	Per Hop Behavior
POTS	Plain Old Telephone Service
PPP	Point to Point Protocol
PQ	Priority Queuing
PRI	Primary Rate ISDN
PSTN	Public Switched Telephone Network

QDM	QoS Device Manager
QoS	Quality of Service
QPM	QoS Policy Manager
RAI	Resource Availability Indicator
RDT	RealNetworks Data Transport
RED	Random Early Detection
RSVP	Resource Reservation Protocol
RTCP	Real-Time Control Protocol
RTP	Real-Time Transport Protocol
RTSP	Real-Time Streaming Protocol
SAA	Service Assurance Agent
SBM	Subnet Bandwidth Management
SSCP	Skinny Station Control Protocol
SGCP	Simple Gateway Control Protocol
SML	Service Level Manager
SMS	Service Management Solution
SN	Sequence Number
SNMP	Simple Network Management Protocol
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
TDM	Time Division Multiplexing
TFTP	Trivial File Transfer Protocol
ToS	Type of Service
UDP	User Datagram Protocol
VAD	Voice Activation Detection
VC	Virtual Circuit
VIP	Versatile Interface Processor
VoATM	Voice over ATM

VoFR	Voice over Frame Relay
VoIP	Voice over IP
WAN	Wide Area Network
WFQ	Weighted Fair Queuing
WRED	Weighted Random Early Detection
XML	Extensible Markup Language
XAUTH	eXtended Authentication

Implementing Cisco Quality of Service (QOS)

Exam Code: 642-642

Certifications:

Cisco Certified Internetwork Professional (CCIP)	**Core**
Cisco IP Telephony Design Specialist	**Core**
Cisco IP Telephony Express Specialist	**Core**
Cisco IP Telephony Operations Specialist	**Core**
Cisco IP Telephony Support Specialist	**Core**

Prerequisites: The minimum requirement is attending the ICND and BSCI courses. It would be beneficial if you attended the CVOICE and BCMSN courses as well.

About This Study Guide
This Study Guide is based on the exam questions for the 642-642 CCIP: Implementing Cisco Quality of Service (QOS) exam. It presents all the information necessary to pass the 642-642 CCIP: Implementing Cisco Quality of Service (QOS) exam, and is detailed on the particular proficiencies examined in the exam. The QOS exam is designed to test your knowledge on QoS, defining QoS, the various QoS tools and implementing these. The information provided in this Study Guide is specific to the QOS examination, and does not symbolize a complete reference work on the subject of Qality of Service.

The following topics are covered in this Study Guide: Understanding Bandwidth and QoS Tools which Influence Bandwidth; Using the Bandwidth Command and Clock Rate Command; Understanding Delay; Understanfing the Various Types of Delay, including Serialization Delay, Propagation Delay, Queuing Delay, Forwarding Delay, Shaping Delay, and Network Delay; Implementing QoS Tools which Influence Delay; Understanding Jitter and QoS Tools which Influence Jitter; Understanding Packet Loss and QoS Tools which Influence Packet Loss; Understanding Voice Traffic; Identifying Voice Bandwidth QoS Factors; Identifying Voice Delay QoS Factors; Identifying Voice Jitter QoS Factors; Identifying Voice Loss QoS Factors; Understanding Video Traffic; Identifying Video Bandwidth QoS Factors; Understanding Data Traffic; Understanding Quality of Service Tools; Understanding Classification and Marking QoS Tools; Understanding Queuing QoS Tools; Understanding Congestion Avoidance QoS Tools; Understanding Traffic Shaping and Traffic Policing QoS Tools; Understanding Call Admission Control (CAC) and Resource Reservation Protocol (RSVP) QoS Tools; Understanding Link Efficiency QoS Tools; Implementing Tools Used for Managing QoS; QoS Models and Architecture; Applying QoS; Identifying Flow Based QoS Tools and Class Based QoS Tools; Understanding the Differentiated Services (DiffServ) Model; Understanding Class Selector Per Hop Behavior (PHB); Understanding Assured Forwarding (AF) Per Hop Behavior (PHB); Understanding Expedited Forwarding (EF) Per Hop Behavior (PHB); Understanding DiffServ Classifiers and Traffic Conditioners; Understanding the Integrated Services (IntServ) Model; Understanding Classification Theories and Principles; Understanding Marking Theories and Principles; Implementing Classification and Marking Qos Tools; Understanding Class Based Marking (CB Marking); Understanding Network-Based Application Recognition (NBAR); Understanding Committed

Access Rate (CAR); Understanding Policy Based Routing (PBR); Understanding the VoIP Dial Peer; Understanding and Configuring QoS Policy Propagation with BGP (QPPB); Understanding Queuing Theories and Principles; Understanding Transmit Queues (TX Queues) / Transmit Rings (TX Ring); Understanding Sub-interface and Virtual Circuit (VC) Queuing; Understanding and Configuring Priority Queuing (PQ); Understanding and Configuring Custom Queuing (CQ); Understanding and Configuring Weighted Fair Queuing (WFQ), including Class Based WFQ (CBWFQ), Low Latency Queuing (LLQ), and IP RTP Priority; Understanding and Configuring Distributed Weighted Fair Queuing (dWFQ), including Flow-Based dWFQ, TOS-Based dWFQ, and Distributed QoS Group–Based WFQ; Understanding and Configuring Modified Deficit Round-Robin (MDRR); Understanding Congestion Avoidance Theories and Principles; Understanding Random Early Detection (RED); Understanding Weighted RED (WRED); Understanding Flow Based WRED (FRED); Understanding Traffic Policing and Traffic Shaping Theories and Principles; Understanding the Functions and Operations Associated with Traffic Shaping; Understanding the Functions and Operations Associated with Traffic Policing; Implementing Traffic Shaping QoS Tools, including Generic Traffic Shaping (GTS), Class Based Traffic Shaping, Distributed Traffic Shaping (DTS), and Frame Relay Traffic Shaping (FRTS); Implementing Traffic Policing QoS Tools, including CB Policing and Committed Access Rate (CAR); Understanding Compression and Link Fragmentation and Interleaving (LFI); Understanding Payload and Header Compression Concepts and Theories; Understanding Configuring Payload Compression, and TCP and RTP Header Compression; Understanding Link Fragmentation and Interleaving (LFI) Concepts and Theories; Configuring Multilink PPP Interleaving and FRF.12 Interleaving; Understanding Call Admission Control (CAC) and QoS Signaling; Understanding Local Voice CAC Mechanisms, including the Physical DS0 Limitation CAC Method, the Max-Connections CAC Method, the Trunk Conditioning CAC Feature, the Voice over Frame Relay CAC Feature, and the Local Voice Busy Out (LVBO) Feature; ; Understanding Measurement Based Voice CAC Mechanisms, including Advanced Voice Busy Out (AVBO) and PSTN Fallback (VoIP Fallback); Understanding Resource Based Voice CAC Mechanisms, including the Cisco CallManager Resource Based CAC Mechanism, the Gatekeeper Zone Bandwidth CAC Method, the Resource Reservation Protocol (RSVP); Understanding QoS Design Process; Understanding QoS Design Suggestions for Voice and Video Traffic; Implementing QoS Management Tools, including the QoS Device Manager (QDM), the QoS Policy Manager (QPM), the Cisco Service Assurance Agent (SAA) Probes, the Cisco Internetwork Performance Monitor (IPM); and Understanding Cisco Service Management Solutions (SMS).

Intended Audience

This Study Guide is targeted specifically for those people who wish to take the Cisco 642-642 CCIP: Implementing Cisco Quality of Service (QOS) exam. The information in this Study Guide is specific to that exam. Although our Study Guides are aimed at new comers to the world of IT, the concepts dealt with in this Study Guide are complex. Knowledge on networking would be advantageous.

How To Use This Study Guide

To benefit from this Study Guide we recommend that you:

- Study each chapter carefully until you fully understand the information. This will require regular and disciplined work. Where possible, attempt to implement the information in a lab setup.
- Be sure that you have studied and understand the entire Study Guide before you take the exam.

Good luck!

1. A Synopsis on Quality of Service (QoS)

QoS entails the collection of tools and mechanisms that can be used to influence a packet's access to certain network services. Queuing features, traffic policing, compression, traffic shaping, and signaling features all influence QoS. QoS tools used to enhance the performance of some flows can impair the performance of another flow. With QoS, the performance of one type of packet is enhanced over a different type of packet. Therefore implementing QoS is **managed fairness** or **managed unfairness**.

There are various types of traffic on a network. Each type of traffic has a different performance attribute on the network. An interactive application could require constant response time, where as a file transfer application could have throughput as a performance requirement. QoS endeavors to put an end to network traffic performance issues experienced. The multiple servers', one queue method could enhance the standard wait time. The QoS tools have an effect on **bandwidth**, **delay**, **jitter** and **packet loss**. These four concepts are all attributes of traffic and are influenced by QoS tools.

1.1 Bandwidth

Bandwidth is used to describe the quantity or number of **bits per second** that can practically be perceived to be effectively conveyed over a channel. Bandwidth in instances is equivalent to the **clock rate** or **link speed** of an interface, and in other instances, it could be slighter than the link speed of the interface.

WAN bandwidth in a **point-to-point network** is equivalent to the physical link speed, or clock rate, of the interface. Bandwidth in this network is apparent because the amount of traffic that will be relayed is always equal to the link speed, nothing more or less.

In a **Frame Relay network**, a contracted quantity of bandwidth is available. The **Committed Information Rate (CIR)** identifies the quantity of bandwidth that is ensured to move across the network between the **Data Terminal Equipment (DTE)** at every **Virtual Circuit (VC)** end. A Frame Relay network provider entrusts to providing a network that can hold the CIRs of its combined VCs. The bandwidth for each VC is equivalent to the CIR of every VC correspondingly. In multi-access networks, like Frame Relay and ATM, the frames are dropped and need to be resent, when the switches queues are full. The actual bandwidth in a Frame Relay network could turn out to be less that the CIR.

1.1.1 QoS Tools which Influence Bandwidth

The most favored QoS tool for solving bandwidth problems is adding more bandwidth to enhance the quality of traffic. Certain **link-efficiency** QoS tools enhance bandwidth by decreasing the amount of bits needed to convey data.

With **Compression**, either the **headers** or **payload** are compressed. This in turn decreases the number of bits needed to convey the data and can efficiently double the bandwidth ability of a point to point link in certain instances.

Call Admission Control (CAC) tools are additional QoS tools that have an effect on bandwidth. CAC tools can be used to improve the **effective use** of bandwidth in the network. These tools decreases the total load brought into the network by declining new video and voice calls when there are insufficient network resources for the new call. They determine if the network is permitted to receive these calls by using certain factors that mostly entail measuring bandwidth. When a voice call is declined by CAC, the call can be rerouted via the **Public Switched Telephone Network (PSTN)**. This would however be dependent on the Voice over IP (VoIP) dial plan.

QoS **queuing tools** can be used to set aside minimum quantities of bandwidth for specific packet types. These tools set up many queues. Packets are extracted from these queues are dependant on the queue-servicing algorithm employed.

1.1.2 The Bandwidth Command and Clock Rate Command

With the Cisco Router, the **`clock rate`** and **`bandwidth`** interface commands are associated with bandwidth. The **`clock rate`** command describes the Layer 1 bit rate and is used when the router is connected to another router by means of a serial interface. The default bandwidth setting on the serial interfaces of Cisco routers is T1 speed, irrespective of the actual bandwidth. Any sub-interfaces would acquire the bandwidth setting of the matching physical interface. The **`bandwidth`** command is used to inform a range of Cisco IOS Software functions of the amount of alleged bandwidth that exists on the interface.

1.2 Delay

The majority of QoS tools are associated with delay. Packets are affected by delay at some point in the network. A delay, at certain points in the network, can be overlooked because it is so tiny. At other points, the delay could be considerable. The various types of delay are discussed below.

1.2.1 Types of Delay

1.2.1.1 A Serialization Delay

Serialization delay describes the **time it takes to place all the bits of a frame onto the physical interface**. The bits are placed onto the link more rapidly when the link is fast. When the link is slow, it takes longer to place the bits on the link. When the packet is short, the bits are placed on the link more rapidly than when the packet is longer. Serialization delay takes place any time a frame is sent and is **based on the link speed and size of the packet**. The serialization delay is irrelevant for the majority of applications when dealing with LAN links.

The following formula is used to compute serialization delay for a packet:

$$\frac{\textbf{\#Bits Sent}}{\textbf{Link Speed}} \quad \text{or } \textbf{\#Bits Sent / Link Speed}$$

Assuming that a 125 byte packet, equivalent to 1 000 bits is transmitted to a server over Fast Ethernet to the switch, it would take 1 000 bits / 100 000 000 bits per second (bps), or .01 ms to serialize the packet at Fast Ethernet speed.

Table 1.1 illustrates the serialization delay for a 125 byte frame at various link speeds.

TABLE 1.1: ***Serialization Delay Values for 125 Byte Frame and Various Link Speeds***

Link Clock Rate	Serialization Delay (milliseconds)
56 kbps	17.85
128 kbps	7.8
512 kbps	2
1.544 Mbps	.65
100 Mbps	.01

The serialization delay over Fast Ethernet is therefore irrelevant. Serialization delay turns into a more sizeable number on the lower speed serial links.

1.2.1.2 A Propagation Delay

Propagation delay describes the **time it takes a single bit to cross the link from one end to the other end**. This delay occurs on each physical link as the bits move across it, and is relatively insignificant on LAN links and the shorter WAN links. A delay evidently takes place when an optical or electrical signal is put on a cable. Energy does not instantly spread to the other end of a cable without some delay. The **length of the link has an effect on propagation delay**, and not the link speed or size of the packet, as is the case with serialization delay.

The following formulae are used to compute propagation delay:

Length of Link (in meters) / the speed of light (3.1 x 10^8 meters/second)

or

Length of Link (in meters) / the speed of light copper or optical media (2.1 x 10^8 meters/second)

1.2.1.3 A Queuing Delay

Queuing delay is the time that **packets spend in mainly the output queues in a router, while waiting to be forwarded or sent.** The time spent in input queues is insignificant in a router. Using the single queue method is uncomplicated and packets are dealt with on a first in, first out (FIFO) basis. Queuing prevents the latter packets from being discarded. However, while the primary packet in a queue could experience no delay or an insignificant delay, the delay experienced by the latter and last packet added to the queue could be significant. The last packet would have to wait on the initial and second packet before being forwarded.

1.2.1.4 A Forwarding Delay

Forwarding delay is the **time that it takes to forward the packet, and excludes any queuing delay**. It is the time needed from when the packet is received and examined on the input interface, to the time that the packet is queued for sending on the output interface. A forwarding delay is normally experienced on LAN switches, ATM switches, Frame Relay switches and routers. Fortunately, the forwarding delay experienced on these switching devices is normally an irrelevant factor, and can be disregarded when determining total delay computations.

1.2.1.5 A Shaping Delay

Shaping delays **occur only when traffic shaping is configured** and is normally relevant on a router. Shaping in essence **delays the sending of packets to ultimately prevent the packet from being dropped** or lost in the centre of an ATM or Frame Relay network. Frames and packets can be dropped for a number of reasons. To avoid packets from being dropped, shaping introduces an additional delay.

1.2.1.6 A Network Delay

Network delay occurs **within a service provider's network** and is the delay brought about by elements of service provider's network. The delay in this instance is dependent on network congestion, the situation of the network links and the service provider.

1.2.2 QoS Tools which Influence Delay

- Additional **bandwidth** reduces serialization and queuing delays because packets can exit faster with faster bandwidth.
- **Traffic shaping** can minimize packet loss within an ATM or Frame Relay network. It does however increase delay.
- With **compression**, either the headers or payload are compressed. This decreases the bandwidth requirement and in turn, decreases the size of the queues. Compression also decreases serialization delay. However, compression can also add a processing delay when it compresses and decompresses a packet.
- A **queuing** QoS tool enables packets to be placed in different queues. Multiple queues are created that each has their own scheduling method. Packets that are sensitive to delay can be enabled to leave its queue prior to a packet that is not delay sensitive. However, delay will be increased for packets that are not delay sensitive.
- **Link Fragmentation and Interleaving (LFI)** enables a small delay sensitive packet to be inserted and sent on a link without it waiting for a large packet to be serialized. With LFI, the larger packets are broken into smaller fragments and then sent on a link. After the smaller fragment is sent, the delay sensitive packet can just be inserted and sent on the link.

1.3 Jitter

Jitter occurs when **successive packets that are sent in the same way have various levels of delay**. Jitter is always experienced in a packet network. Data applications are not degraded by jitter, whereas voice and video area. Interactive applications could be negatively affected by large differences in jitter.

1.3.1 QoS Tools which Influence Jitter

Because jitter is a variation of delay, **additional bandwidth** will assist in this delay related issue. Additional bandwidth reduces serialization delay, and in turn jitter. The remainder of the QoS tools, namely **Queuing**, **LFI**, **Compression** and **Traffic Shaping**, listed under Delay also affects jitter.

1.4 Packet Loss

Many QoS tools cannot do anything for the cause of packet loss that occurs when routers drop or lose packets. QoS tools can only be used to minimize the effect of packets that are lost due to full queues.

1.4.1 QoS Tools which Influence Packet Loss

Additional **bandwidth** can assist in preventing queues from becoming full and packets from being tail dropped. Tail drop refers to the situation when a router drops a packet while trying to place it at the tail of a queue.

Queuing can be used to assist in preventing packet loss although it does increase delay.

Random Early Detection (RED) drops packets at random before a queue is full and tail dropping occurs. RED can be implemented to control the end of the queue, while queuing controls the beginning of the queue. RED operates on the concept that certain TCP connections can shrink their windows prior to the output queues becoming full. RED therefore slows down the TCP connections. The load of the total amount of packets arriving in the network is smaller. This in turn, prevents the queues from being filled up. Users' response times can be affected by RED.

In other words, QoS enables a network to supply the correct quantities of QoS resources for applications and their differing bandwidth, delay, jitter and packet loss needs. Table 1.2 illustrates the QoS needs of various applications. Fragile as an application refers to certain applications that are not delay tolerant at all.

TABLE 1.2: *Applications and their Related QoS Requirements*

Application	Bandwidth	Delay	Jitter	Packet Loss
Batch	High	High	High	Low
Interactive	Low	Low	Medium to High	Low
Voice	Low	Low	Low	Low
Interactive Video	High	Low	Low	Low
One Way Video	High	Medium to High	Low	Low
Fragile	Low	Low	Medium to High	N/A

1.5 The Attributes of Voice Traffic

In networks that have no QoS tools, voice traffic tends to disintegrate quite swiftly. Voice over data comprises of Voice over Frame Relay (VoFR), Voice over ATM (VoATM) and Voice over IP (VoIP). Cisco IP Phones uses VoIP and not the other two.

When the voice call occurs between analogue phones, the router accumulates the analogue voice and encodes the voice by means of **voice coders/decoders (codec)**. Codecs translates the incoming analogue signal to a digital signal or binary value. The binary values used to signify the voice would differ and is dependent on the codecs being used. The encoded voice is then placed into the payload field.

When dealing with IP Phones, the **voice signaling** process uses **Skinny Station Control Protocol (SSCP)** that runs among every phone and the Cisco CallManager server. **Voice calls** utilize **Real-Time Transport Protocol (RTP)**. Once the signaling process has ended, an RTP stream has taken place between the phones. In essence, a variety of voice signaling protocols create an RTP stream between the two phones when the caller pushes digits on the phone. The RTP streams in fact convey voice between the two phones.

The following protocols are used for voice signaling and voice payload:

- H.323/H.225
- H.323/H.245
- H.323/H.225 RAS
- Intra Cluster Communications Protocols (ICCP)
- Media Gateway Control Protocol (MGCP)
- Real-Time Transport Protocol (RTP)
- Real-Time Control Protocol (RTCP)
- Skinny
- Simple Gateway Control Protocol (SGCP)

QoS tools can handle voice signaling and voice payload in different manners. The QoS tool first refers to a field within the packet that classifies the packet as voice payload or voice signaling. Table 1.3 illustrates the QoS needs for voice payload and voice signaling.

TABLE 1.3: ***Voice Signaling and Voice Payload QoS Requirements***

	Bandwidth	Delay	Jitter	Packet Loss
Voice Signaling	Low	Low	Medium	Medium
Voice Payload	Low	Low	Low	Low

1.5.1 Voice Bandwidth QoS Factors

Voice calls are isochronous or take place at equal periods of time. These calls produce a stream that contain a fixed data rate, and equally spaced packets. Cisco IOS Software puts 20ms of encoded voice into every packet. By default, voice payload packets are therefore transmitted every 20ms. The bandwidth requirement for the voice payload call is dependent on:

- **Compression**
- **Codec**
- **Data link framing, the actual links being used**

- **Packet Overhead: IP, UDP, RTP header size**

Table 1.4 depicts the bandwidth needs in relation to various voice calls. The IP/UDP/RTP header size in each instance is 40 bytes.

TABLE 1.4: *Bandwidth Needs in relation to Various Voice Calls*

Header Size	Header Type	Codec	Payload Bandwidth	Full Bandwidth
14	Ethernet	G.711	64kbps	85.6
14	Ethernet	G.729	8kbps	29.6
6	MLPPP/FR	G.711	64kbps	82.4
6	MLPPP/FR	G.729	8kbps	26.4

The IP/UDP/RTP header sizes are compressed when using **Compressed RTP (cRTP).** When cRTP is used with the low bit rate codecs, there is a significant reduction in the bandwidth requirement. However, cRTP can add to delay because of the compressing and decompressing processes.

Another component that can have an effect on bandwidth is **Voice Activity Detection (VAD)**. With VAD, voice packets are not transmitted by the sender of the packets when the speaker is quiet. This can reduce actual bandwidth by roughly 60 percent. The actual reduction in bandwidth cannot though be forecasted for every call because factors like a noisy surrounding can overthrow VAD. ATM can place a considerable amount of data link overhead on voice packets because the last ATM cell often contains a lot of wasted space. In addition to this, every ATM cell has 5 bytes of overhead.

1.5.2 Voice Delay QoS Factors

When delay is fairly large, the quality of the voice call is affected. Too much delay can even result in the call being dropped. Voice traffic too experiences the delays that packets do. The types of delay were discussed earlier on in this study guide. In addition to these delays, voice traffic experiences the following delays:

- **Packetization delay**: When dealing with packet telephony, packetization delay and codec delay happen together and overlap in the interval between when a speaker speaks and the IP/UDP/RTP payload packet is sent. This occurs because sound is translated first to electrical signals, and then to digital signals. After this, the actual delay is experienced between when a person speaks and the IP/UDP/RTP payload packet is sent. With a voice gateway or IP Phone, 20ms of voice payload must be accumulated by a voice gateway or IP Phone prior to it being placed into a RTP packet. The speaker must speak for 20ms for the packet to be created. This is the default setting for G.711 and G.729 on IP Phones and Cisco Software gateways.
- **Codec delay**: The codec delay can vary and is affected by the codec and processing load. The following two elements affect codec delay:
  - **The time needed to deal with an incoming analogue signal and to change it to a digital signal**: This time related delay takes place with each and every codecs because any conversion from a voice signal to an equal digital signal consumes some time. The actual

time used is determined by the codecs algorithm employed. The particular codecs algorithm can give rise to more delay that is caused by the look-ahead feature.

- **Look-ahead**: Look-ahead takes place when the codec algorithm uses a smaller number of bits to encode a voice. This is based on the knowledge that a person's voice cannot immediately differ from a particular sound to a completely other sound. The algorithm then proceeds to conclude that the following next ms of voice sound should not drastically differ. This is one method used to convert a 64 kbps G.711 call to an 8 kbps G.729 call. The processing of this predictive portion of the codecs algorithm needs the codecs to process the voice signal to be converted, as well as the following few ms of voice.

- **De-jitter buffer delay, initial playout delay**: This delay element can be configured for a different value, or is variable, and takes place in data networks. The de-jitter buffer accumulates a few voice packets prior to playing out so that any next packet that experiences jitter and is delayed, does not interrupt the consistency of the voice. The initial play out delay comes into existence because the de-jitter buffer has to be filled so that play out can start.

1.5.3 Voice Jitter QoS Factors

Jitter occurs in packet networks and is brought about by variable delay factors, of which network delay and queuing delay is the most common. Queuing and fragmentation QoS tools can be used to cut jitter down to minimal figures. This will assist in ensuring that the de-jitter buffer operates efficiently. Using a queuing QoS tool that processes voice packets without much delay will reduce queuing delays. Lastly, Frame Relay and ATM networks can be designed to cut down on jitter and network delay.

1.5.4 Voice Loss QoS Factors

Packets are discarded by routers for a number of reasons of which the following two are the most common:

- **Bit errors**
- **Insufficient queuing space**

Unfortunately, QoS tools cannot assist a great deal with packet loss due bit errors but can assist with insufficient queuing space.

Queuing methods can be employed that put voice packets in a separate voice queue from those of data packets. Queuing and Link fragmentation and Interleaving (LFI) QoS tools can be used to ensure that the router processes the voice queue as soon as possible to reduce delay for voice packets. This will assist in the router not tail dropping voice packets. Link Fragmentation and Interleaving (LFI) assists in reducing the time that a second voice packet must hold on for any initial packet to be serialized. Call admission control (CAC), can be used to avoid packet loss when too much voice packets are being received. The voice queue can be configured with the maximum amount of bandwidth that it is allowed to be used.

Remember from an earlier discussion that G.729 codecs compresses the voice payload by predicting the following bit of ms. The same logic is used at the receiving end of a voice call when changing the digital signal to an analogue one. At the receiving end of the line, the feature is known as the **autofill algorithm**. Autofill assumes what the following ms of sound could have been when a subsequent voice packet is lost. The algorithm can fill in a maximum of 30 ms of voice that is lost.

1.6 The Attributes of Video Traffic

QoS assists in preventing video quality from degrading. IP packet video can be broken into the following groups:

- **Interactive Video**: This group consists of H.323-compliant video conferencing systems that utilize the Real-Time Transport Protocol (RTP) protocol for the transmission of voice and audio payload. The Cisco IP/VC 3500 product series and NetMeeting desktop videoconferencing of Microsoft are H.323-compliant video conferencing systems.
- **Non-interactive Video**: This group consists of e-learning video services and streaming media. The Cisco IP/TV product, RealNetworks and Microsoft Windows Media Technologies products fall into this group. Certain non-interactive video are H.323-compliant video conferencing systems while others are not.

To enable the commencement of a video conference, a user normally first specifies the host name to the H.323 application for the conference. Next, the VC units execute the relevant H.323/H.225 call setup messages. Lastly, a RTP stream is created for **audio**, and another for **video**. With voice, one stream is created in each direction.

Video codecs too change analogue audio and video to the packetized state. G.711 and G.729 voice codecs convert the audio stream while the video signals uses, among others, the Moving Pictures Experts Group (MPEG) and ITU H.261 codecs. The audio stream is normally transmitted as a flow on its own, apart from the video signal flow.

The QoS needs for voice and video are exactly the same for delay, jitter, and loss. Refer to Table 1.3 in Section 1.5. Interactive packet video has the same delay factors as voice. Streaming packet video on the other hand can bear quite a considerable quantity of delay. QoS tools for reducing delay with interactive two-way video should be handled in the same manner as voice QoS tools that reduce delay. To prevent loss, video should also be put into a separate larger queue.

The only exception is the bandwidth requirement for video payload and video signaling. In the instance of video, it is high, as opposed voice's low requirement. Codec delay, packetization delay, and de-jitter initial play out delay are applicable to video as well. QoS tools for video must be able to check a field in the packet to determine if the packet is video signaling, or video payload.

The following protocols are used for video signaling and video payload:

- H.323/H.225
- H.323/H.245
- H.323/H.225 RAS
- Real-Time Transport Protocol (RTP)
- Real-Time Streaming Protocol (RTSP)

1.6.1 Video Bandwidth QoS Factors

Video uses a number of **packet sizes** and **packets rates** for one video stream. With voice, the packet sizes are fixed and are based on the codecs, and the packet rates are constant. A video queue could require a larger queue size than that needed for voice traffic. The majority of video codecs uses an algorithm that initially transmits a large encoded video frame, and then a number of vectors that hold information pertaining to the prior frame. This algorithm assists in drastically reducing the bandwidth requirement.

The common video codecs and their related bandwidth needs are listed below:

- H.261: 100 – 400 kbps
- MPEG 1: 500 – 1 500 kbps
- MPEG 2: 1.5 – 10 Mbps
- MPEG 4: 28.8 – 400 kbps

1.7 The Attributes of Data Traffic

Applications typically use either **Transmission Control Protocol (TCP) or User Datagram Protocol (UDP)** TCP/IP transport layer protocols. UDP does not execute any error recovery processes. TCP on the other hand does. Because TCP performs error recovery, the protocol is regarded as reliable. The TCP and UDP headers also contain source and destination port numbers that are used by QoS tools to categorize the packet. Common applications like FTP, TFTP, Telnet and web applications utilize a renowned port that simplifies the classifying of traffic for QoS tools. When two applications each utilize TCP, **multiplexing** occurs. Multiplexing is TCP's ability to establish which applications should receive the data for every packet.

- **Data Bandwidth QoS Factors**: Data applications have one **bidirectional traffic flow** with packet rates and sizes that **differ** vastly. Data bandwidth requirements tend to differ tremendously and are based on the application in use. While some applications like non-interactive applications typically consume bandwidth, others like interactive applications usually do not. Web interactive applications that have web pages with large volumes of graphics, on the other hand, have a fairly large bandwidth requirement.
- **Data Delay QoS Factors**: Data applications tend to endure **delay** much better than **voice and video**. Data is also not affected from additional delays initiated by codec, packetization and the de-jitter buffer delays. Key interactive applications should have the lowest delay component. With non-interactive key data applications, the delay can differ largely when bandwidth is provided. Other non-interactive data applications that are not as important could receive bandwidth once all other bandwidth needs are met.
- **Data Jitter QoS Factors**: Non-interactive applications can endure more jitter than interactive applications. However, voice and video can still tolerate less delay and jitter than these applications. QoS tools that are used to improve jitter and delay for voice and video increase jitter and delay for data applications.
- **Data Loss QoS Factors**: Data is not damaged when loss occurs simply because packets can be sent again. When applications utilize TCP, lost packets are transmitted again. This can however increase network congestion. When UDP is used by Network File System (NFS) and Trivial File Transfer Protocol, error recovery is handled within the application. Other applications like those that utilize Simple Network Management Protocol (SNMP) are not bothered when a packet is lost.