CISCO VOICE OVER IP NOTES

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Traditional Telephony

Telephony devices can be grouped into four components to complete the end-to-end calls.

- **Edge Devices** - Analog and Digital Telephones
- **Local loops** - The interface with the telephone company network
- **Private or CO Switches** - This terminates the local loop and handles all the signaling, call routing and call setup and tear down
- **Trunks** - The path between two switches.

Switching and Central Offices
The CO switch includes there components to make the telephone work:

- **Battery** - Power to the circuit
- **Current detector** - The status of the current
- **Dial tone generator** - Generates a dial tone once there has been a request for service
- **Dial register** - This receives the dialed digits
- **Ring generator** - Generates the ringing for the destination phone

PBX

- The PBX is a smaller version of a CO switch used in company's to route calls
- The Key system is a scaled down version.

Call Signaling - the ability to make a request to complete an end-to-end call. Three components make up this function:

- **Supervisory signaling** - On-hook, Off-Hook and ringing
- **Address signaling** - DTMF or Pulse
- **Informational signaling** - All the call progress tones

Digital and analog signaling:

- You must make sure that the signaling methods used by the service provider match the types of equipment and signaling you are supporting in your network.

Packet telephony networks vs. circuit-switch telephony networks benefits include:

- The packet telephony network makes more efficient use of bandwidth by sharing with other applications and logical connections
- The expense of running less cable in the building

Call Control consists of three basic functions:

- **Call setup** - Using gateway control protocols
- **Call maintenance** - Keeping the call going
- **Call teardown** - Sending the disconnect signals

ANALOG INTERFACES:

FXO - an interface that allows an analog connection to the CO of the PSTN
- Uses the RJ-11 connector
- Allow for off-premise connections
- Connects local calls to the telephone company's central office PBX

FXS - an interface that allows connection for normal basic service phones
- Uses the RJ-11 connector
- Connects directly to a telephone handset, fax machine, or similar device
- FXS supplies the following to the handset:
  - **Ring voltage**
  - **Line Power**
  - **Dial tone**

E&M (Ear & Mouth) - an interface that allows connection for PBX trunk lines
- Uses the RJ-48 connector

DIGITAL INTERFACES:

T1 - 24 channels grouped together to form a frame
E1 - 30 channels grouped together to form a frame
BRI - 2 B channels for voice/data and 1 D channel for signaling
PRI – Protocol that uses 23 lines of a T1 to send voice calls, plus 1 line to send supervisory signals

ENTERPRISE CAMPUS DESIGN:
- **Centralized** - contains the CallManager and voice mail servers for all the remote branches
- **Distributed** - each location has its own CallManager and voice messaging system

Service providers need to support the SS7 protocol and be capable of maintaining call records for billing as well as be scalable and have superior performance.

TRUNKS – Analog and Digital:

Local Loop - The loop from your house to the telephone company
Trunks – voice connection between two PBX devices
- Analog or digital
- Connect telephone company and PBX switches
- Switches provide logic and trunks provide the path between switches

Trunk types:
- **Private trunk lines**
- **Central Office trunks**
- **Foreign Exchange (FX) Trunks** - used to talk to FXO's and FXS's. The FXS acts as a central office. It sits at the remote site and looks to the telephone like a switch. The FXO acts like a phone. It's on the switch end of the connection.

DID/DOD - Direct inward/direct outward dialing
- **DID** - one-way trunks that allow users to dial a PBX without operator intervention
- **DOD -** one-way trunks that allow users to connect directly to the CO
  - DOD's are outbound trunks. To access an outside line, user's press "9" and the PBX will forward the call to the CO.

SEIZURE TYPES - used to signal the telephone company when you need a line

Loop-start signaling- allows users or the telephone company to seize a line or trunk when a call is being initiated.

- **Three states:**
  - **On-hook** - Idle
  - **Off-hook** - PBX or telephone seizure
  - **Ringing** - CO seizure
- Loop-start signaling is a poor signaling solution for high-volume trunks because it is possible to seize the trunk simultaneously from both ends. This is known as **glare.**

Ground-start signaling - a modification of loop-start signaling that corrects for the probability of glare. It solves the problem by providing current detection at both ends.

ECHO - due to a reflection of the audio signal. Impedance mismatch at the 2w-4w conversion (hybrid) is the most common reason for echo.

- **Two types of echo:**
  - **Talker echo** - when users hear themselves twice
  - **Listener echo** - hears what the caller says, and then they hear it again

There is always a certain amount of echo on the line. If there is an echo problem, there are two ways to solve the problem:

1. **Echo suppression** - suppresses your voice on the return path and acts as a nose gate, effectively making communications half-duplex
2. **Echo cancellation** - more complex, uses an echo canceller to build a mathematical model of the speech pattern and subtracts it from the transmit path. The echo canceller only removes the echo from one end of the circuit. If the echo is an issue at both ends, another separate echo canceller would need to be applied at the other end.

T1/E1/ISDN/PRI Signaling Types:

CCS – Common Channel Signaling - a signaling method used between computer controlled switching machines that make up the PSTN

- Specified by ITU-T #7
- CCS is commonly know by names including CCS7, SS7, and CCIS
- CCS allows carriers on the PSTN to provide intelligent switching services for the future, allowing trunk terminating equipment to be simpler and less expensive than previous alternatives.

CCS7 network includes three major components:

- **SSP Service switching points** - local exchanges or toll switches that serve as sources or destinations for CCS7 messages
- **STP Signal transfer points** - data switches that translate application addresses and route messages between network nodes and databases.
- **SCP Service control points** - customer profile and routing information databases used to provide advanced network services.

Bit rate calculation:
- A standard frame has 193 bits (1 framing bit + 24 8-bit timeslots)
- Each timeslot is sampled at a rate of 8,000 times per second
- In one second there are 1,536,000 bits of PAYLOAD data transmitted (8000 samples X 8 bits/sample X 24 timeslots)
- Additionally, there are 8,000 framing bits every second
- Combining the 1,536,000 bits of payload and the 8,000 bits of framing yields 1.544 Mbps

CISCO VOICE CONNECTION TYPES:
- **Local** - calls within the same router
- **On-Net**- inter-office calls
- **Off-Net** - connection to outside line (to PSTN)
- **PLAR** (Private Line Automatic Ringdown) – PLAR automatically connects a telephone to a second phone, as soon as the first phone is lifted from the cradle (off-hook)
- **PBX-to-PBX** – tie-line connection between PBX
- **On-Net to Off-Net** - reroutes calls off-net

Converged Networks
- Both data and voice technologies can run over a single medium instead of having voice run on the PSTN and data on a separate IP network. With integration users can have both voice and data running on the data network. Separate networks are inefficient and difficult to administer.

T1/E1 Commands:

To inform the local telephony interface of the type of signaling it should expect to receive from the far-end dial peer:
Router(config-dial-peer)# signal-type (cas | cept | ext-signal | transparent}

Use the framing command to configure the DS1 link framing format. ESF is required for ATM traffic.
Router (config-controller)# framing {sf | esf}

To configure the line encoding format for the DS1 link (b8zs is required for ATM and Frame Relay traffic.
Router (config-controller)# linecode {ami | b8zs}

For E1 controller settings:
Router(config-controller)# framing {crc4 | no-crc4} [name]
Router(config-controller)# linecode {ami | hdb3}

To specify the telephone company's switch that you are attaching to:
Router(config)# isdn switch-type [primary-4ess | primary-5ess | primary-dms100 | primary-net5 | primary-ntt | primary-ts014]

Enter controller configuration mode to specify the controller port to configure:
Router(config)# controller [t1 | e1] [0 | 1 | 2 | 3]

Enter the telephone company's framing and line-code type:
Router(config-controller)# framing [esf | sf | crc4 | nocrc4]
Router(config-controller)# linecode [ami | b8zs | hdb3]

Enter the clock source for the line as well as all channels for ISDN:
Router(config-controller)# clock source [line primary | line secondary | internal]
Router(config-controller)# pri-group timeslots [1-24 | 1-31]

Now configure the ISDN D channels, which carry the control and signaling information for ISDN calls, for each ISDN PRI line:
For T1 interfaces:
Router(config)# interface serial0:23 | interface serial1:23
For E1 interfaces:
Router(config)# interface serial0:15 | interface serial 1:15

Configure all incoming voice calls to go to the modems:
Router (config-if)# isdn incoming-voice {data [56 | 64] voice}

Enter dial-peer configuration mode and specify the method of voice encapsulation:
Router(config)# dial-peer voice <tag> {pots | voip | vofr | voatm}

The codec command specifies the voice coder rate of speech and payload size for a VoFR dial peer:
Router(config-voiceport)# codec {g711alaw | g711ulaw | g726r32 | g729r8} [bytes <payload_size>]

For toll quality, use g711alaw (European) or g711ulaw (North American).

- **g711alaw**: 64,000 bps
- **g711ulaw**: 64,000 bps
- **g729r8**: 8,000 bps (the default codec)
- **g729r32**: 32,000 bps

Voice Dial Plans

- The **connection plar** command specifies a private line automatic ring down (PLAR) connection. PLAR is an autodialing mechanism that permanently associates a voice interface with a far-end voice interface, allowing call completion to a specific telephone number or PBX without dialing.
- The **<string>** you configure for this command is used as the **called number** for all incoming calls over this connection.
- The **destination peer** is determined by the **called number**

- The **connection tie-line** command emulates a temporary tie-line trunk to a PBX. A tie-line connection is automatically set up for each call and torn down when the call ends. Use the **connection tie-line** command when the dial plan requires that additional digits be added in front of any digits dialed by the PBX, and that the combined set of digits be used to route the call onto the network.
- The **connection trunk** <digits> command establishes a two-way, permanent, “nailed up” tie-line connection to a PBX.

This command can be used for E&M-to-E&M trunks, FXO-to-FXS trunks, and FXS-to-FXS trunks.

Configure a POTS dial peer

- Configure basic telephone service or POTS
- Specify the destination for a POTS dial peer
- Specify the port used for incoming calls from a telephony interface to select a dial peer and for VoIP calls to match a port with the selected outgoing dial peer

Router(config) dial-peer voice 1 pots
Router(config-dial-peer)# destination-pattern 5551212
Router(config-dial-peer)# port 0/0

To expand a set of numbers (for example, an extension number) into a destination pattern, use the **num-exp** command. In a VoIP network, by using the **num-exp**, the router expands a particular sequence of dialed numbers into a complete telephone number (destination pattern).
Router (config)# num-exp <extension_number> <expanded_number>

Enable voice activity detection (VAD):
With VAD, silence is not transmitted over the network, only audible speech. If you enable VAD, the sound quality will be slightly degraded but the connection will monopolize much less bandwidth.
Router (config-dial-peer)# vad

The **prefix** command is used to optionally specify a prefix for a specific dial peer. When an outgoing call is initiated to this dial peer, the **prefix** <string> value is sent to the telephony interface first, before the telephone number associated with the dial peer.
Router (config-dial-peer)# prefix <string>

The forward-digits command specifies which digits to forward for voice calls:
Router (config-dial-peer)# forward-digits {<num-digit> | all | extra}

The **destination-pattern** command is used once in dial-peer configuration mode. Use the **destination-pattern** command to specify the full E-164 telephone number (depending on your dial peer) of the destination dial peer.
Router (config-dial-peer)# destination-pattern <pattern>

Use the **session target** command if you are configuring a VoIP dial peer. You must specify a network address for a specified dial peer. Use the **session target** command to specify the network address of the router you are trying to contact.
Router (config-dial-peer)# session target {ipv4:destination_address | dns:[s. | d. | e. | u.] host-name | loopback:rtp | loopback:compressed | loopback:uncompressed}

- **s.** - Indicates that the source destination pattern will be used as part of the domain name.
- **d.** - Indicates that the destination number will be used as part of the domain name.
- **e.** - Indicates that the digits in the called number will be reversed, periods will be added between each digit of the called number, and this string will be used as part of the domain name.
- **u.** - Indicates that the unmatched portion of the destination pattern (such as a defined extension number) will be used as part of the domain name.

Building a Scalable Numbering Plan

There are several applications that can be included in the VoIP network

- **Hoot and Holler** - Permanent 4 wire circuit that stays up so anyone can pick up the phone and listen or talk to the other connected parties. The Cisco IP world can do the same across the packet switched network.
- **Toll ByPass** - The use of the IP WAN reduces or eliminates toll charges.
- **Hospitality Services** - The Cisco product can support the hotel/motel type environment. Additional equipment is needed including the Cisco Building Broadband Service manager, Cisco ICS, Cisco CTE and IP Phones.
- **IP Centrex** - Centrex is normal phone service with all the bells and whistles managed by the carrier, such as hold, conferencing etc. IP Centrex delivers the Centrex application across the packet switch network.
- **Prepaid Calling Card** - The Cisco Voice Infrastructure and Applications (VIA) includes a lot of features to support prepaid calling cards.
- **Multi-Tenant** - Typically found in shared tenant buildings where a central reception area receives calls for all the business in the building and sends the incoming caller to the appropriate business. Multi-dwelling units (MDUs) and Multi-tenant units (MTUs) are typical areas for this application.
- **Computer Telephony Integration (CTI)** - This allows the users to have information coming from the phone system access database information to pull up records about the incoming caller. Some of the Cisco CTI applications include: Cisco IP Phone, Cisco IP AutoAttendant, Cisco WebAttendant, and Personal Assistant.
- **Collaborative Computing** - The sharing or resources between areas at the same time. Microsoft NetMeeting, Video streaming etc.
- **Voice Enabled Web Applications** - VoiceXML allows users to retrieve data via the IP Phone.
- **Call Centers** - IPCC the IP Contact Center along with the ICM Intelligent Call Management software work together to manage the Call Center environment.
- **Unified Messaging** - Cisco Unity Voice Messaging system integrated with the CallManager solutions provide full unified messaging along with Microsoft Exchange server.

DELAY and JITTER

- Packet loss - Voice packets being dropped due to congestion
- Delay - Keep delay to 150ms. This is the time it takes for the packet to reach its destination
- Jitter - Delay times between packets due to buffering

Five key tools to help keep constant throughput:

- **Queuing** - Holding packets and prioritizing them.
- **Congestion avoidance** - Monitor traffic loads and prioritize. WRED is one such avoidance mechanism.
- **Header compression** - RTP header compression is processor intense. But it takes the 40 byte header and compresses it to 2 bytes.
- **RSVP** - Guarantee the bandwidth before the call is established.
- **Fragmentation** - Making sure the data packets are not too large so that the smaller voice packets get stuck behind them.

Setting up gateways for VoIP to the traditional telephony world and the service providers takes planning and consideration. It all depends on what you are integrating with.

Signaling and call control is broken down into two components:

- **Components** - Endpoints, which are typically either terminals or gateways.
- **Services** - Call state information about the call and call control.

Call control consists of the following protocols:

- **H.323** - Multimedia communications such as video conferencing.
- **Session Initiation Protocol (SIP)** - Modifies or terminates multimedia calls.
- **Media Gateway Control Protocol (MGCP)** - Call Control, typically the CallManager contains all the configuration information about the gateway.
- **H.248/Megaco Protocol** - Communication between gateway components.
- **Session Announcement Protocol (SAP)** - Multicast for audio and video.
- **Real Time Streaming Protocol (RTSP)** - On demand real time audio and video.
- **Cisco CallManager ("Skinny")** - Cisco proprietary protocol for call control.

H.323 protocol Stack

- **H.323** is the accepted model for transmitting multimedia (voice) across an IP network that does not guarantee QoS
- H.323 allows for standards-based interoperability with other vendors H.323-compatible equipment
- H.323 describes terminals, equipment, and services for multimedia communication over LANs
- Any H.323-compliant terminal is required to carry voice

H.323 is made up of three components:

- **H.225**
- **H.245**
- **H.323**

H.225- An ITU standard that governs H.225.0 session establishment and packetization.

- H.225.0 is analogous to ISDN Q.931 signaling
- Responsible for initial call-setup

In addition, H.225 actually describes several different protocols:

- **RAS protocol (Registration/Admission/Status)** RAS is used in the H.323 protocol suite for discovering and interacting with a Gatekeeper
- Use of **Q.931**
- Use of **RTP**

H.245- Handles the "capabilities exchange," which encompasses variables like:

- Which codec to use
- Whether to enable VAD or Fax-Relay
- What TCP ports will be used during the audio path

H.323- Handles the audio-path using RTP

RTP and RTCP

RTP - Real-Time Transport Protocol provides end-to-end network functions and delivery services for delay-sensitive, real-time data like voice/video

- In VoIP, RTP works with queuing to prioritize voice traffic over other traffic

RTP services include the following:

- Payload type identification
- Sequence numbering
- Time-stamping
- Delivery monitoring

RTCP - Real-time Transport Control Protocol was established to monitor the quality of the data distribution and provide control information

It provides feedback on current network conditions:

- RTCP provides a mechanism for hosts involved in an RTP session to exchange information about **monitoring and controlling** the session
- **Monitors quality** for such things as packet counts, packets lost, inter-arrival jitter
- RTCP transmits packets as a **percentage of session bandwidth**, but at a specific rate **of at least every 5 seconds**
- RTCP also provides a **separate flow** from RTP for UDP transport use

Media Gateway Control Protocol (MGCP)

The Media Gateway Control Protocol (MGCP) RFC 2705, defines a protocol to control VoIP gateways connected to external call control devices.

MGCP GATEWAYS:

- Trunking gateways that interface the VoIP network to the PSTN
- VoATM gateways that interface to an ATM network.
- Residential gateways providing RJ-11 jacks to the residence and using cable modem/cable set-top boxes, xDSL devices, and broadband wireless
- Access gateways that interface via either analog or digital interfaces to PBX systems or key systems
- Business gateways, connecting a traditional PBX system or a "soft PBX" to the VoIP network
- Network access servers that connect a modem to one of the circuits to provide Internet access
- Circuit switches or packet switches that have an interface to provide access to an external call control device.

MGCP uses a model that includes endpoints and connections.

- Endpoints are sources or destinations for calls.
- Connections may be point-to-point or multipoint.

RSVP - Resource Reservation Protocol

- The Resource Reservation Protocol (RSVP) IETF is a signaling protocol that allows host applications to ask for the bandwidth and other QoS attributes that they require, and then provides a means for the network to determine if the requirements can be met
- Neither RTP nor H.323 provide QoS guarantees through the network
- RSVP is the network control protocol that allows Internet applications to obtain special QoS for their data flows.

RSVP handles:

- Reservation of bandwidth and delay.
- Flows can be signaled by end station or by the router (static reservation).
- RSVP is not a routing protocol, instead it works in conjunction with routing protocols to prioritize traffic.

SAP, SDP, and SIP - used to provide announcements and information abound multicast sessions to users on the network

- Once a user selects a session to view or listen to, the SIP is used to setup and teardown the call
- The SAP packet contains a header containing administrative and control fields as well as a payload consisting of an announcement of a single session using the SDP as the payload

SIP Components

A SIP-based network is made up of several components:

- **SIP user agent** - Any network endpoint that can originate or terminate a SIP session; this might include a SIP-enabled telephone, a SIP PC client (known as a softphone), or a SIP-enabled gateway
- **SIP proxy server** - A call-control device that provides many services such as routing of SIP messages between SIP user agents
- **SIP redirect server** - A call-control device that provides routing information to user agents when requested, giving the user agent an alternate uniform resource identifier (URI) or destination user-agent server (UAS)
- **SIP registrar server** - A device that stores the logical location of user agents within that domain or subdomain; a SIP registrar server stores the location of user agents and dynamically updates its data via REGISTER messages
- **SIP location services** - Additional functionality that can be used by proxy, redirect, and registrar servers to find the identity (with a unique URI) and "logical" location of user agents within the network
- **Back-to-back user agent -** A call-control device that provides routing similar to a proxy server, but allows centralized control of the network call flows; this device allows SIP networks to replicate certain traditional telephony services that require centralized knowledge of device state, such as call park and pickup; this component is always dialog stateful
- **SIP-aware network devices** - Devices that have knowledge of the SIP protocol and allow the network to function more efficiently; this type of device might be a firewall or Network Address Translation (NAT) device that can allow SIP traffic to traverse network borders, or a load-balancing switch that allows requests to SIP servers to be more efficiently handled
- **SIP is complementary to MGCP**:
    - SIP provides session control
    - MGCP provides device control

SIP SHOW COMMANDS:
Router# show sip-ua

Display information and settings for the SIP User Agent (UA)
Router# show sip-ua {retry | statistics | status | timers}

The following example displays a sample of voice call history records showing a local call between two telephones attached to the same Cisco MC3810:
Router# show call history voice record

ConnectionId=[0x2C7AEFDC 0x59830001 0x0 0xB0AAA3]
Media=TELE, TxDuration= 1418 ms
CallingNumber=2001
SetupTime=1157801 x 10ms
ConnectTime=1158046 x 10ms
DisconectTime=1158188 x 10ms
DisconnectText=local onhook

ConnectionId=[0x2C7AEFDC 0x59830001 0x0 0xB0AAA3]
Media=TELE, TxDuration= 1422 ms
CalledNumber=2002
SetupTime=1157802 x 10ms
ConnectTime=1158046 x 10ms
DisconectTime=1158188 x 10ms
DisconnectText=remote onhook

- **ConnectionID** - Global call identifier for this voice call.
- **Media** - Call over the type of media. If the call is over the (telephone) access side, the entry will be TELE. If the call is over the voice network side, the entry will be either ATM, FR (for Frame Relay), or HDLC.
- **LowerIFName** - Physical Lower interface information. Only displays if the Media is either ATM, FR, or HDLC.
- **TxDuration** - The length of the call. Only displays if the Media is TELE.
- **CalledNumber** - The called number.
- **CallingNumber** - The calling number.
- **SetupTime** - Time the call setup started.
- **ConnectTime** - Time the call is connected.
- **DisconnectTime** - Time the call is disconnected.
- **DisconnectText** - Descriptive text explaining the reason for disconnect.

TRAFFIC SHAPING

- Set FRTS on the interface
- Set Bc to 10ms (1/100) of CIR (if adaptive shaping is used)
- Set mincir >= to voice bandwidth
- Shape strictly to CIR one PVC carrying voice, do not burst
- Shape both sides of the VC to prevent egress blocking

RTP Header Compression (cRTP)

- RTP Header Compression (**cRTP**) is used on **a link-by-link basis** to compress the IP/UDP/RTP from **40 bytes to 2 or 4 bytes** most of the time when no UDP checksums are being sent, or to 4 bytes when UDP checksums are used
- cRTP is beneficial when the **RTP payload size is small** (20 and 50 bytes)

Configure cRTP header compression on a serial interface or subinterface if:

- Narrowband links
- Need to conserve bandwidth on your WAN interface

Do not use cRTP if you have high-speed interfaces, generally links greater than 2 Mbps.

Low Latency Queuing (LLQ)

- Low Latency Queuing (**LLQ**) (is actually VIP-based PQ-CBWFQ).
- The CLI **fair-queue** command enables only WFQ.

For PQ-WFQ:

- VoIP you need the **fair-queue** (WFQ) and **ip rtp priority** CLI statements.
- LLQ is basic fair-queue, PQ-WFQ underneath.
- Configuring the ip rtp priority command in conjunction with PQ-WFQ adds a "carpool lane" specifically for voice- this is the "PQ" of PQ-WFQ.

Class-based WFQ (CB-WFQ) add BW guarantees per "class of traffic"- loosely speaking, per queue.

LLQ is really (PQ-CBWFQ). LLQ adds a priority queue (PQ) real-time "carpool lane" to CB-WFQ.

PQ-WFQ (IP RTP Priority)

- **IP RTP Priority** provides a strict priority queuing scheme for delay-sensitive data such as voice
- Voice traffic can be identified by its Real-Time Transport Protocol (RTP) port numbers and classified into a priority queue configured by the ip rtp priority command
- The result is voice is serviced as strict priority in preference to other non-voice traffic. The limit for Priority Queue (PQ) flows is 64. Packets exceeding the allocated BW are dropped
- IP RTP Priority is on the same outgoing interface as WFQ or CB-WFQ, traffic matching the range of ports specified for the priority queue is guaranteed strict priority over other WFQ flows or CB-WFQ classes; and voice packets in the priority queue are always service first
- Configuring WFQ and ip rtp priority actually gives you PQ-WFQ

Configuration Commands
RTP Header Compression (cRTP)
Router(config-if)# ip rtp header-compression [passive]

To specify the total number of RTP header compression connections (default is 16) supported on an interface:
Router(config-if)# ip rtp compression connections <number>

IP RTP Priority
Router(config-if)# ip rtp priority <starting-rtp-port-number> <port-number-range> <bandwidth>

To change the percent of interface bandwidth allocated for LLQ and IP RTP Priority:
Router(config-if)# max-reserved-bandwidth <percent>

To enable WFQ:
Router(config-if)# fair-queue [congestive-discard-threshold [dynamic-queues [reservable queues]]]

IP Precedence

- The **ip precedence** <number> command configures the value set in the IP Precedence field when voice data packets are sent over the IP network.

Use this command if:

- The IP link utilization is high and the quality of service for voice packets needs to have a higher priority than other IP packets.
- RSVP is not enabled and the user would like to give voice packets a higher priority over other IP data traffic.

Instead of requesting **best-effort**, **controlled-load**, or **guaranteed-delay**, IP Precedence forwards packets by prioritizing classes of service

- IP Precedence settings 6 and 7 are reserved for network control information
- IP Precedence for delay-sensitive applications like voice should be set to a higher setting like 5. This matters for voice or if you are going over a non-Cisco network

Precedence levels
Level 7 - Network Control (reserved)
Level 6 - Internetwork Control (reserved)
Level 5 - Critical (best for voice)
Level 4 - Flash-override
Level 3 - Flash
Level 2 - Immediate
Level 1 - Priority
Level 0 - Routine

ITU-T G.114 delay budgets:

- 0-150ms delay is suitable for most delay sensitive applications like voice.
- 150-400ms delay is still acceptable, but starts to influence voice quality. Communication can be annoying if no actions are taken.
- 400ms or more delay is unacceptable for most delay sensitive applications.

CODECS Voice Bandwidth Requirements

- 1 sample = 8 bits = 125ms of voice data(time)
- DSP's encode 10ms of voice data(time) at once
- Cisco Voice Router sends 20ms of voice data per packet by default
- It takes 8 samples (@ 125ms each) to equal 1ms of voice data(time)
- 20ms of voice data(time) = 160 samples

- 8 samples = 1ms voice data(time) = 64 bits or 8 bytes
- 20ms * 8 bytes (per 1ms of voice data (time) = 160 bytes
- Takes 50 IP packets to send 1 second of voice data(time)

Sample Size

- Bytes_per_Sample=(sample size*CODEC Bw) /8
- G.711 = Sample size 20ms and 64000 bw
- x=(.020*64000) /8
- 160 = 1280/8

CODEC BW(bps) Sample (byt es) Packets FR (bps) FR cRTP(bps) Ethernet
(bps) FRw/ VAD(bps)

G.711	64,000	240	33	76,267	76.2k	66,133	79,467	49,573
G.711	64,000	160	50	82,400	82.4k	67,200	87,200	53,560
G.726	32,000	120	33	44,267	44.2k	34,133	47,467	28,773
G.726	32,000	80	50	50,400	50.4k	35,200	55,200	32,760
G.726	24,000	80	25	37,800	37.8k	26,400	41,400	24,570
G.726	24,000	60	33	42,400	42.4k	27,300	47,200	27,560
G.726	16,000	80	25	25,200	25.2k	17,600	27,600	16,380
G.726	16,000	40	50	34,400	34.4k	19,200	39,200	22,360
G.729	8,000	40	25	17,200	17.2k	9,600	19,600	11,180
G.729	8,000	20	50	26,400	26.4k	11,200	31,200	17,160

LAYER 3 OVERHEAD

- 40 bytes IP/UDP/TRP header, assuming uncompressed RTP

DATA LINK OVERHEAD	**TOTAL**				
Ethernet II	18 bytes	6 Src mac	6 Dest mac	2 type	4 for CRC
MLPPP	6 bytes	1 flag,	1 address,	2 control	2 for CRC
Frame-Relay (FRF.12)	6 bytes	2 DLCI	2 FRF.12 header		2 for CRC

Security with Tunneling Overhead

- VPN w/IP Sec adds 50 - 57 bytes overhead
- L2TP/GRE adds 24 bytes overhead
- MLP adds 6 bytes added to each packet
- MPLS adds 4 byte label to each packet

VAD – Voice Activation Detection

- Saves 35% of bandwidth or 17,160(bps) on G.711 CODEC call
- Uses CNG (comfort noice generator) "white noice" instead of silence
- Do not take into account when designing a VoIP network, when servicing 24 calls or less
- Data network can use available bandwidth that VAD saves

Busy Hour

- Amount of traffic/day per 22 business days/month
- Multiply by 15-17% = BUSY HOUR
- 3000 minutes/day * 15% = 450 minutes

Erlangs and GoS values are used to determine the amount of trunks needed

- Erlangs calculator: xxxxxxxxxxxx
- Cisco Voice Bandwidth Calculator: http://tools.cisco.com/support/vbc/do/CodecCalc1.do