



Identifying the Requirements of Voice Technologies



Identifying Voice Networking Considerations

Voice Quality Considerations

- Examine the possible causes of packet loss and delay in the initial design.
- Use QoS mechanisms as a groundwork for a high-quality voice network.

One-Way Delay (ms)	Description
0–150	Acceptable for most user applications.
151–40	Acceptable provided that organizations are aware of the transmission time impact on the transmission quality of user applications.
401+	Unacceptable for general network planning purposes; however, it is recognized that in some exceptional cases this limit will be exceeded.

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ITU G.114 Recommendation

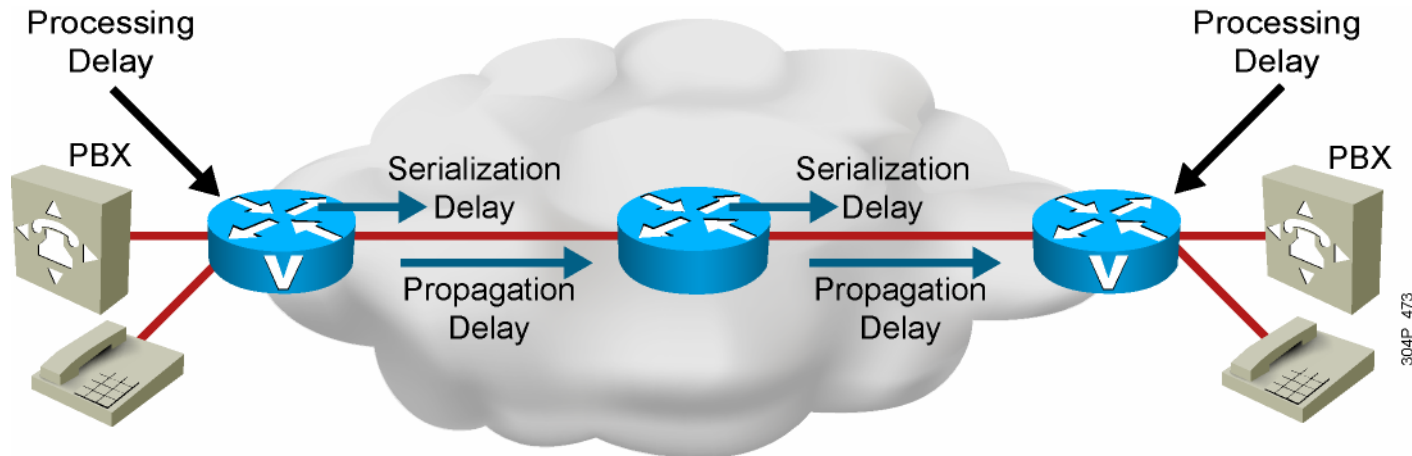
Fixed Network Delay Considerations

Sources of delay:

- Propagation delay: 6 ms per km
- Serialization delay: frame length / bit rate
- Processing delay: depends on codec
 - Coding and compression
 - Packetization

Solutions:

- ➔ None
- ➔ Faster link, smaller packets
- ➔ Hardware DSPs, coding algorithm



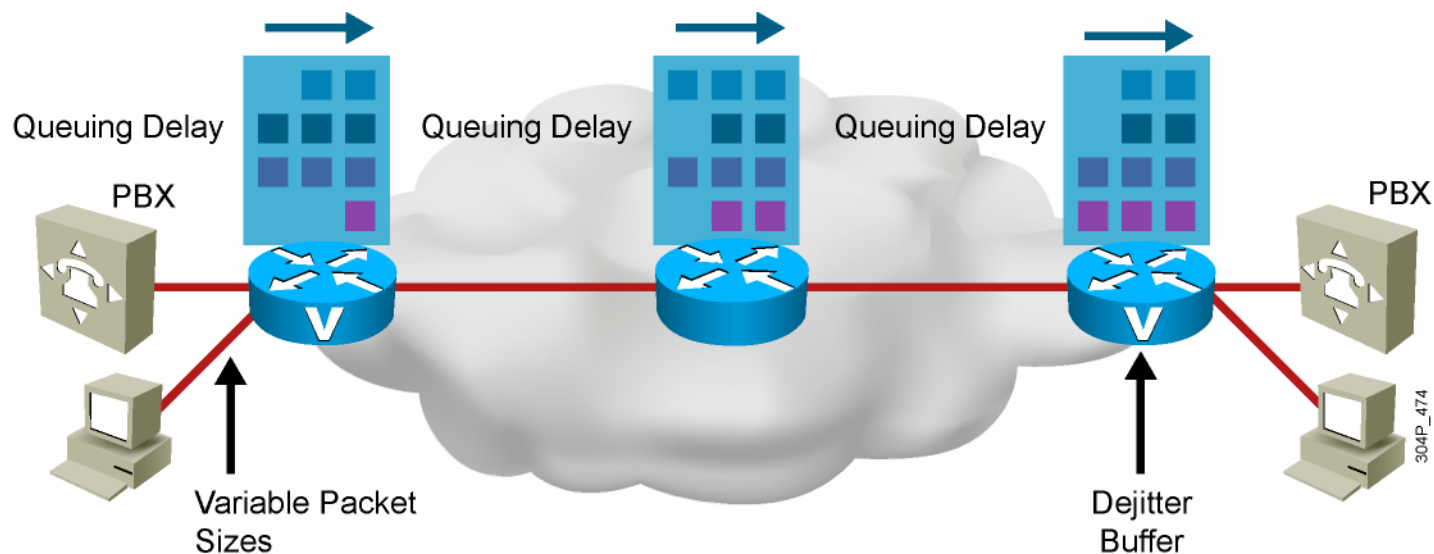
Variable Network Delay Considerations

Sources of delay:

- Queuing delay (variable packet sizes and number of packets)
- Dejitter buffers

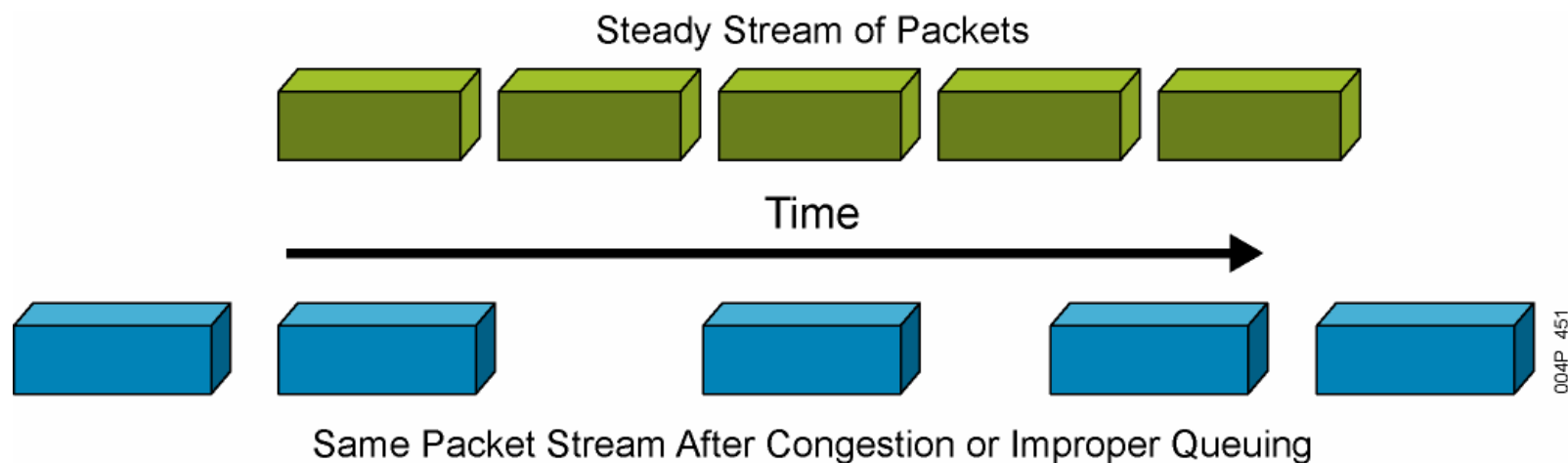
Solutions:

- ➔ Link fragmentation and interleaving
- ➔ Constant delay, uncongested network



Jitter

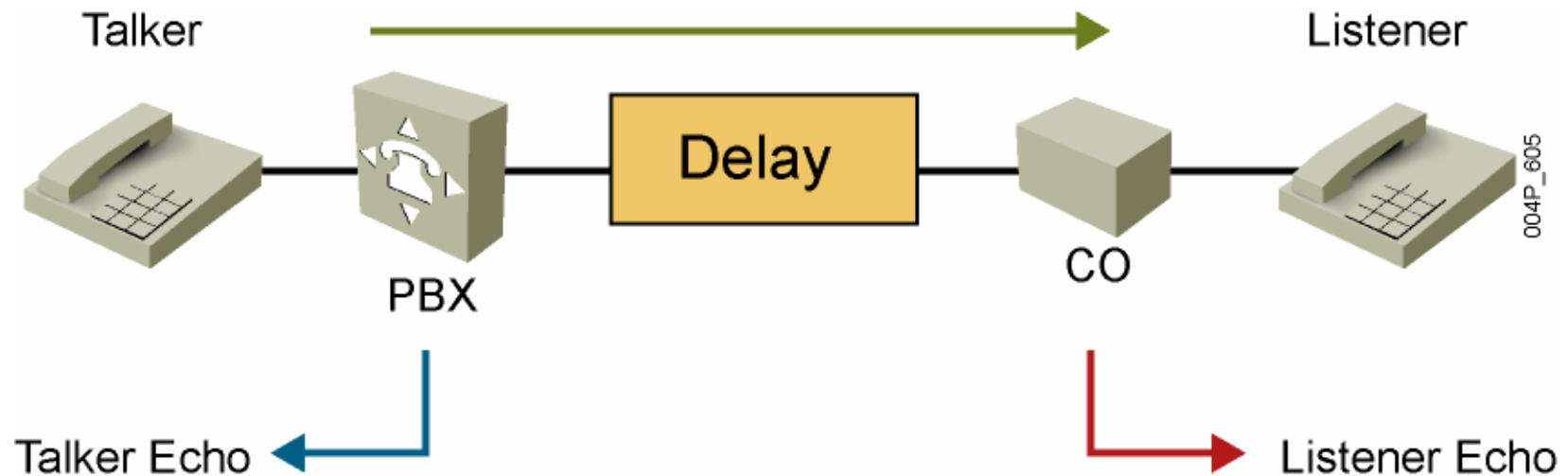
- Variation in the delay of received packets
- Caused by network congestion, improper queuing, or configuration errors



Packet Loss

- Causes voice clipping
- Caused by:
 - Congested links
 - Improper network QoS configuration
 - Bad packet buffer management on the routers
 - Routing problems
- Up to 30 ms of lost voice correctable by DSP using interpolation
- Packet losses up to one packet correctable with no voice quality degradation

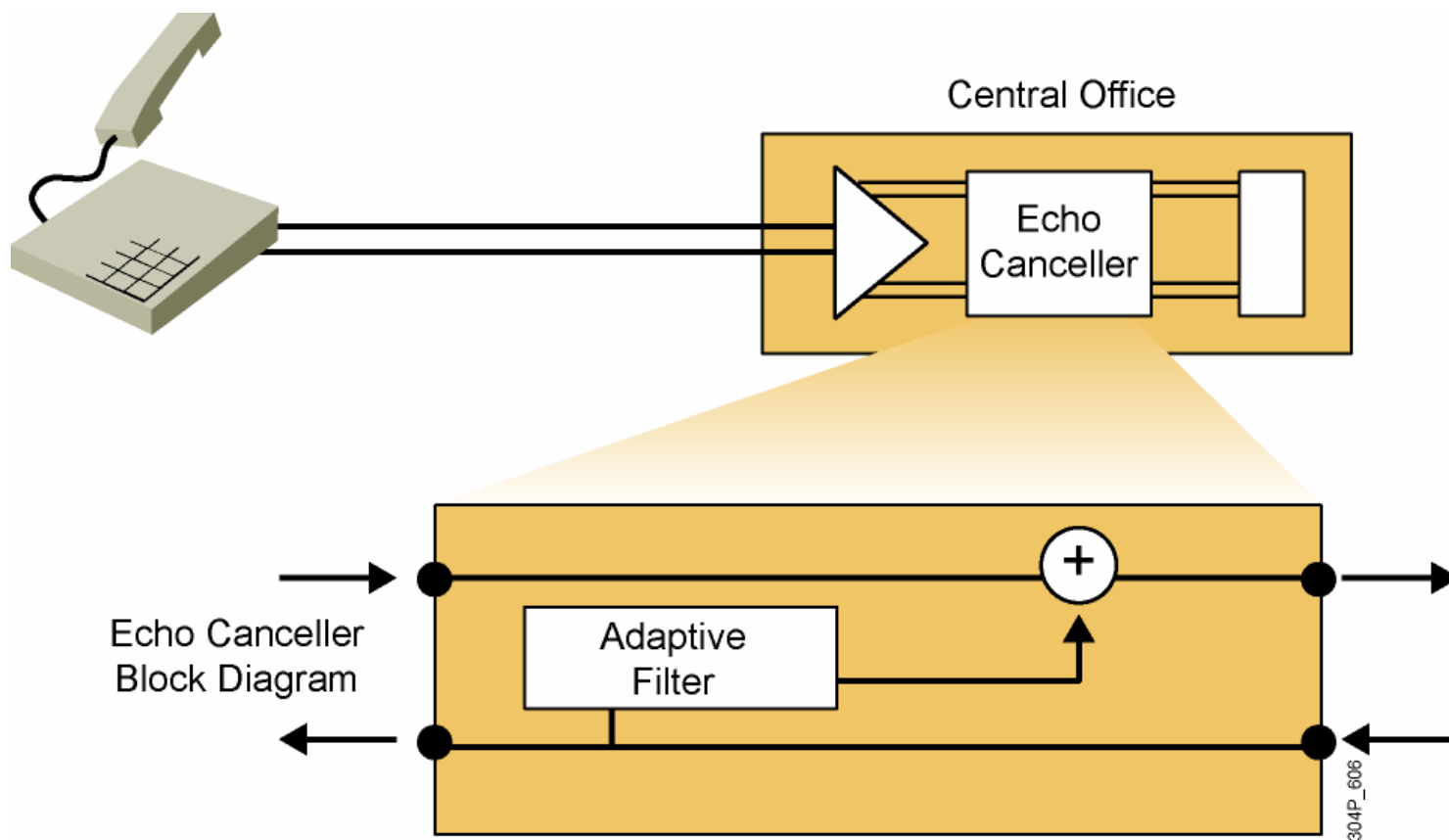
Problem of Echo



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Echo Cancellers

Reduce the Level of Echo



Voice Coding and Compression

- The quality of transmitted speech is a subjective listener response.
- MOS is a common benchmark to define sound quality.
- MOS scales from 1 (bad) to 5 (excellent).

	ITU Standard	Data Rate*	MOS Score
PCM	G.711	64 kbps	4.1
ADPCM	G.726/G.727	16/24/32/40 kbps	3.85 or less
LD-CELP	G.728	16 kbps	3.61
CS-ACELP	G.729	8 kbps	3.92
ACELP/MPMLQ	G.723.1	6.3/5.3 kbps	3.9/3.65

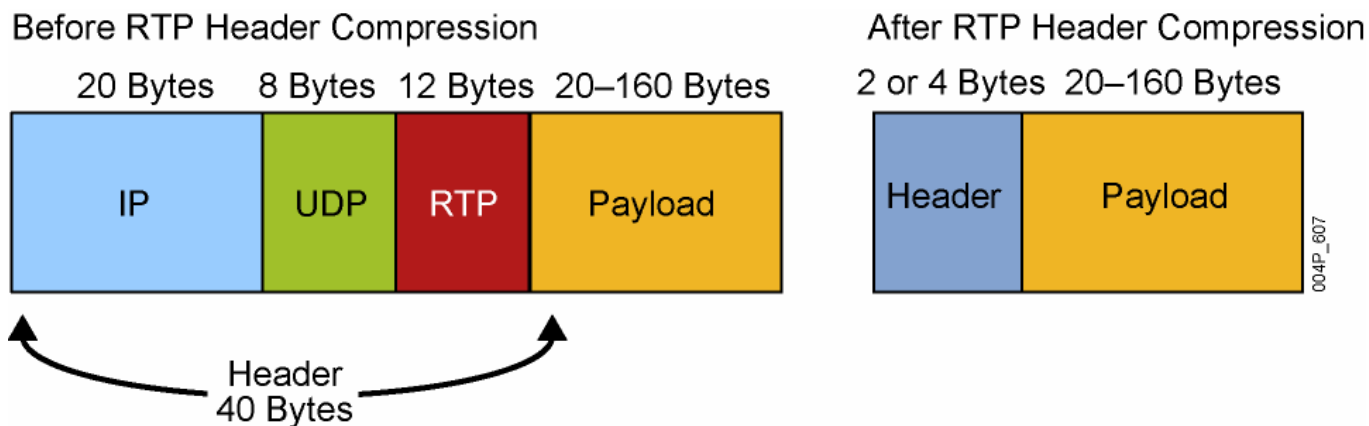
*Note: Data rates shown are for digitized speech only and do not include overhead of RTP, UDP, IP, and Layer 2 headers.

Example: Codec Complexity and Calls per DSP on the Cisco AS54-PVDM2-64 Module

Low Complexity (Maximum 64 Calls)	Medium Complexity (Maximum 32 Calls)	High Complexity (Maximum 24 Calls)
G.711 a-law	G.729a	G.723.1: 5.3K and 6.3K
G.711 mu-law	G.729ab	G.723.1A: 5.3K and 6.3K
Fax passthrough	G.726: 16K, 24K, and 32K	G.728
Modem passthrough	T.38 fax relay	Modem relay
Clear-channel codec	Cisco Fax Relay	AMR-NB: 75K, 5.15K, 5.9K, 6.7K, 7.4K, 7.95K, 10.2K, 12.2K, and silence insertion descriptor

Bandwidth Availability

- Goal: Reduce the amount of traffic per voice call
- Solutions:
 - Use an effective voice coding and compression mechanism.
 - Compress IP headers by using compressed Real-Time Transport Protocol.
 - Suppress packets of silence by using voice activity detection.



Calculating Voice Bandwidth

- Voice packet size = (Layer 2 header) + (IP/UDP/RTP header) + voice payload
- Voice packets per second (pps) = (codec bit rate) / (voice payload size)
- Bandwidth = (voice packet size) * (pps)
- Example for G.729 call with 8-kbps codec bit rate with cRTP and 20 bytes voice payload:
 - Voice packet size = 6 bytes + 2 bytes + 20 bytes = 28 bytes
 - Voice packet size = 28 bytes * 8 bits/byte = 244 bits
 - Voice pps = 8000 bits/sec / 160 bits/packet = 50 pps
 - Bandwidth = 244 bits * 50 pps = 11.2 kbps

Example: Voice Codec Bandwidth Calculator for G.729 Codec

Codec Information		
Codec Bit Rate	8 kbps	= (Codec Sample Size * 8) / (Codec Sample Interval)
Codec Sample Size	10 bytes	size of each individual codec sample
Codec Sample Interval	10 msec	the time it takes for a single sample
Bandwith Per Call (VoIP)		
Voice Packets Per Second	50	(Codec Bit Rate / Voice Payload Size)
Bandwidth Per Call (RTP Only)	11.6 kbps	(Total Packet Size(bits) + Flag(bits)) * (Packets Per Second)
5% Additional Overhead	0.58 kbps	5% additional overhead per call to accomodate bandwidth for signaling (for example: RTCP/H225/H245 messages on H.323 networks).
Bandwith Per Call + 5.0% Additional Overhead	12.18 kbps	Overhead + Bandwidth Per call
Total Bandwith Required (VoIP)		
Bandwidth Used for All Calls (RTP Only)	11.6 kbps	(Bandwidth per Call) * (Number of Calls)
Total Bandwidth (including Overhead)	12.18 kbps	Same as above + 5.0% Overhead
Packet Size Calculation		
Total Packet Size	28 bytes	Excluding Frame Flag
Voice Payload Size	20 bytes	Size of the Codec Samples per packet
Layer2 Overhead	6 bytes	Layer2 Overhead including CRC
Compressed Header	2 bytes	IP/UDP/RTP Compressed header
Frame Flag (7E)	1 byte	Most modern framers can handle a single flag between frames (ie... no beginning flag)

Voice Bandwidth and Codec Standards

Compression	Payload Size	Bandwidth	Bandwidth with cRTP	No. of Calls on a 512-kbps Link (without cRTP/ with cRTP)
G.711 (64 kbps)	160	83	68	6/7
G.726 (32 kbps)	60	57	36	8/14
G.726 (24 kbps)	40	52	29	9/17
G.728 (16 kbps)	40	35	19	14/26
G.729 (8 kbps)	20	26	11	19/46
G.723.1 (6.3 kbps)	24	18	8	28/64
G.723.1 (5.3 kbps)	20	17	7	30/73

Enterprise QoS Mechanisms for Voice

- Traffic classification
- Queuing or scheduling
- Bandwidth provisioning and call admission control

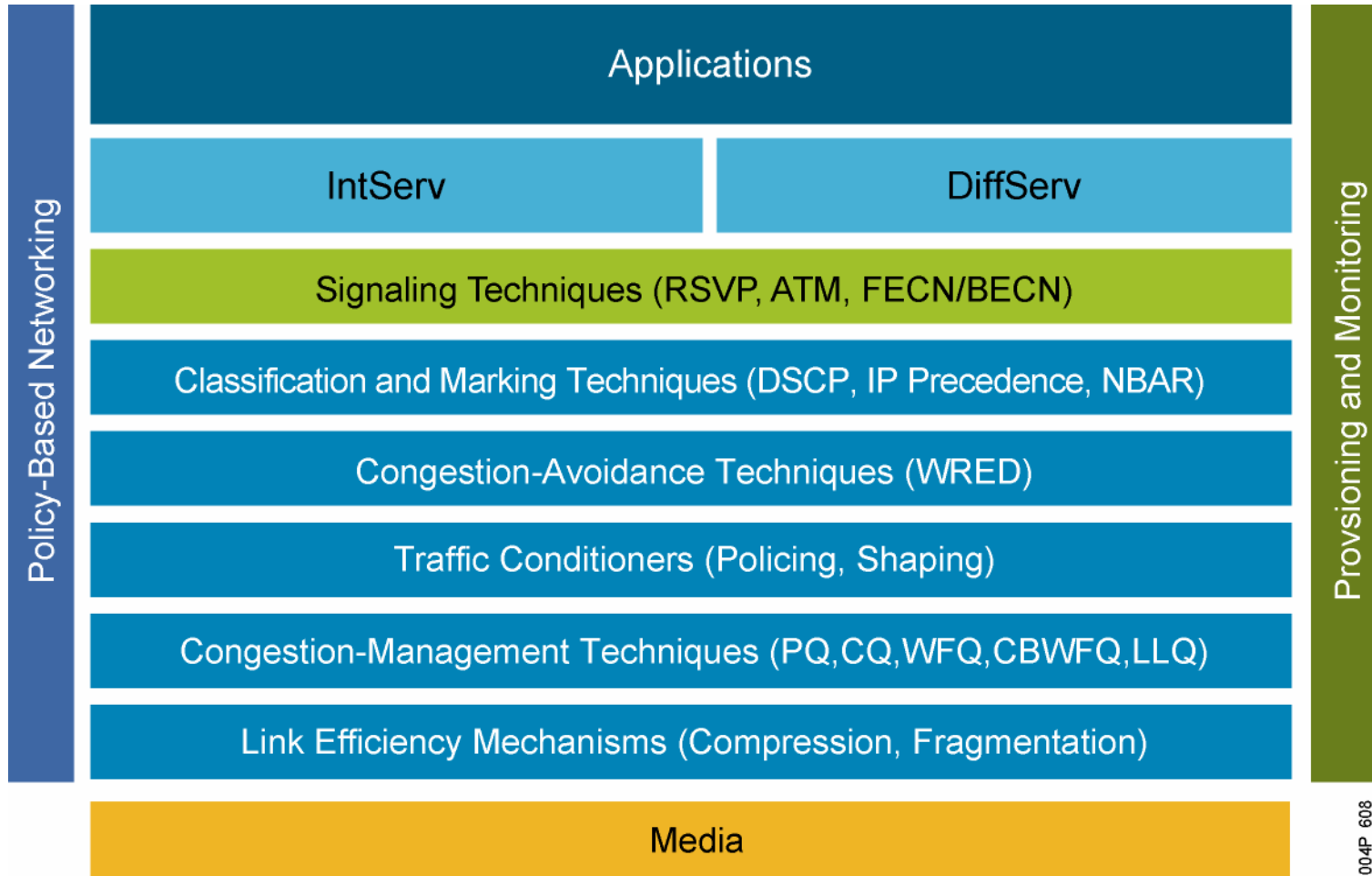
Access Layer QoS Mechanisms for Voice

- 802.1Q trunking and 802.1p
- Multiple egress queues
- Traffic classification and network trust boundary
- Layer 3 awareness and the ability to implement QoS access control lists

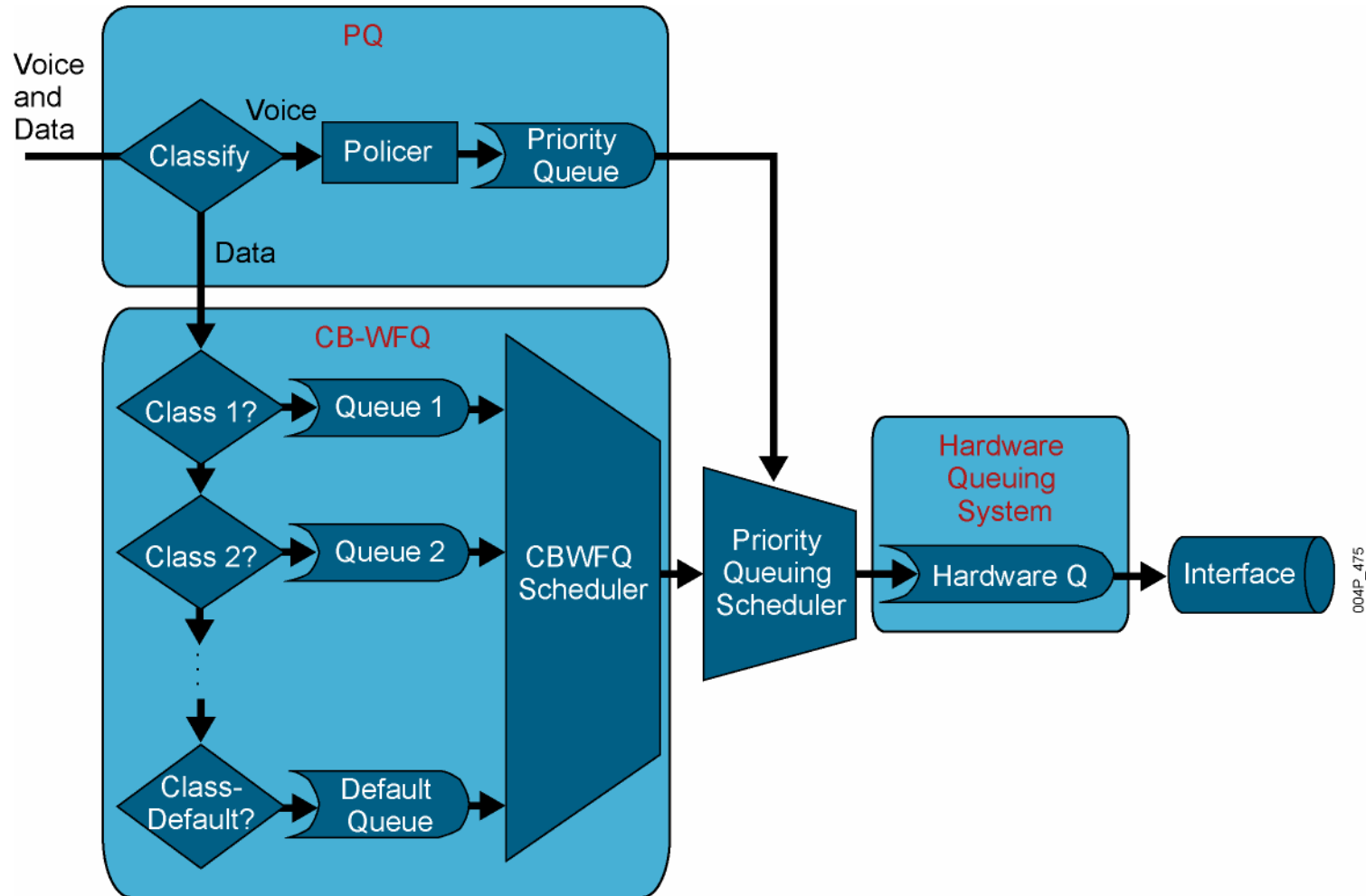
Recommended Practice: Separate Voice and Data VLANs

- Voice device protection from external networks
- QoS trust boundary extension to voice devices
- Protection from malicious network attacks
- Ease of management and configuration

Example: QoS Networking Mechanisms



Example: Low Latency Queuing



QoS Consideration for Voice in the WAN

WAN QoS mechanisms:

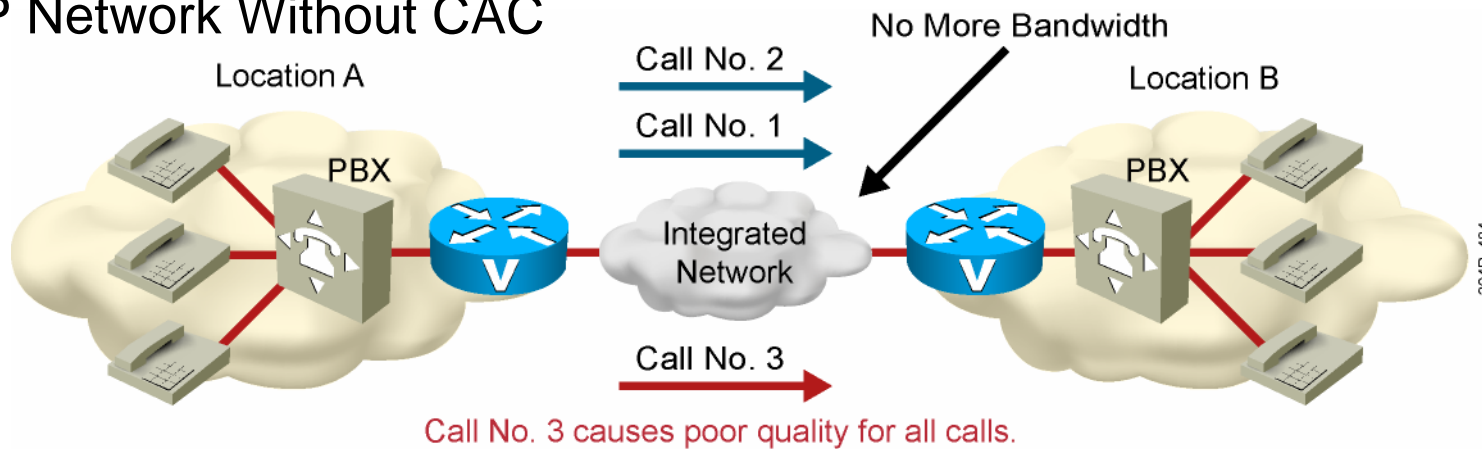
- Bandwidth provisioning
- Traffic classification
- Queuing and scheduling
- Traffic shaping
- Link efficiency techniques
- Call admission control

Call Admission Control

- Protects voice traffic from being negatively affected by other voice traffic
- Keeps excess voice traffic off the network
- Reroutes excess voice traffic in the following scenarios:
 - Call rerouted via an alternate packet network path
 - Call rerouted via the PSTN network path
 - Call returned to the originating TDM switch with the reject cause code

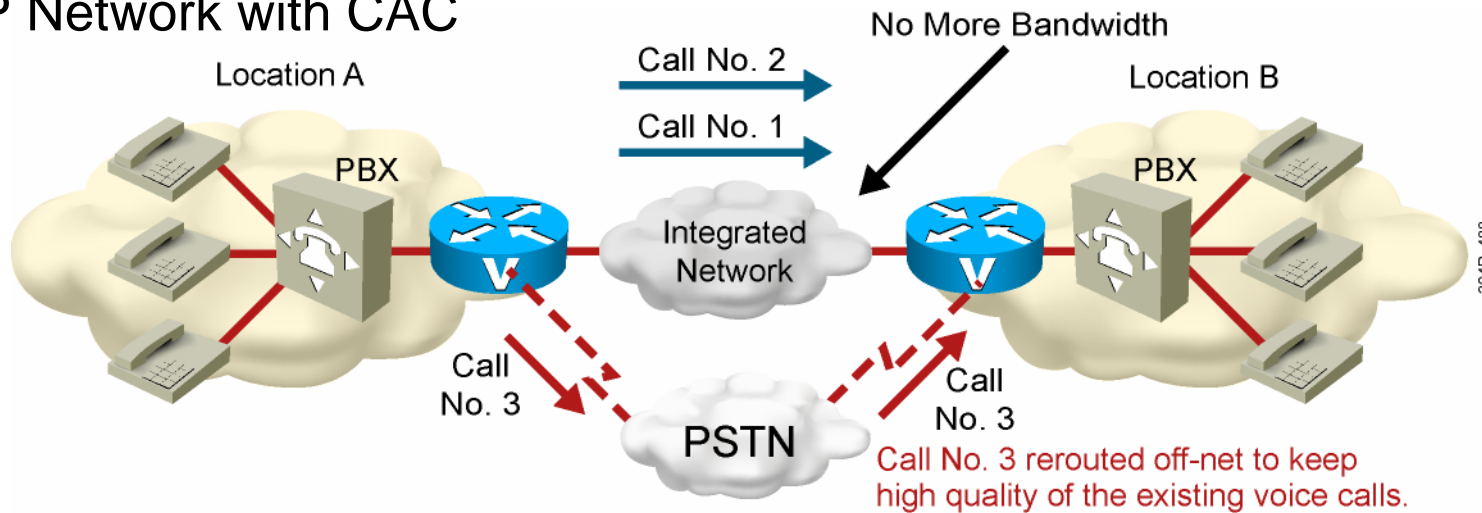
Example: Call Admission Control

VoIP Network Without CAC



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VoIP Network with CAC



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Implementing CAC with RSVP

- RSVP is an industry-standard signaling protocol that enables an application to reserve bandwidth dynamically.
- RSVP signaling messages are exchanged between the source and destination devices.
- RSVP process interacts with the QoS manager on router interfaces to "reserve" bandwidth resources.
- Calls are admitted or rejected based on the outcome of the RSVP reservations.

Traffic Engineering Terms

- Grade of service
- Erlang
- Centum call seconds
- Busy hour
- Busy hour traffic
- Blocking probability
- Call Detail Record

Erlang Tables

- Show erlangs of offered traffic, number of circuits, and grade of service
- Three common erlang tables:
 - Erlang B assumes that calls receiving a busy signal are immediately cleared.
 - Extended Erlang B assumes that a certain percentage of calls receiving a busy signal are redialed.
 - Erlang C assumes that blocked calls are queued.

Example: Erlang B Table

Number of erlangs decreases with the decreased blocking probability.

Number of erlangs increases with the number of simultaneous connections.

Blocking Probability

Number of Circuits	.003	.005	.01	.02	.03	.05
1	.003	.006	.011	.021	0.31	0.053
2	.081	.106	.153	.224	0.282	.382
3	.289	.349	.456	.603	0.716	.900
4	.602	.702	.870	1.093	1.259	1.525
5	.996	1.132	1.361	1.658	1.876	2.219
6	1.447	1.822	1.900	2.278	2.543	2.961
7	1.947	2.158	2.501	2.936	3.250	3.738
8	2.484	2.730	3.128	3.627	3.987	4.543
9	3.053	3.333	3.783	4.345	4.748	5.371
10	3.648	3.961	4.462	5.084	5.530	6.216

Busy hour traffic (BHT) in erlangs

Summary

- Voice quality in an IP network is directly affected by delay, jitter, and packet loss.
- An echo is the audible leak of the voice of the caller into the receive (return) path.
- Voice communication over IP relies on voice that is coded and encapsulated into IP packets.
- A primary WAN issue when network designers are designing voice on IP networks is bandwidth availability.
- QoS mechanisms are important for networks that carry voice.
- Traffic engineering is a science of selecting the right number of lines and the proper types of service to accommodate users.



Integrating Voice in the Network Design

- Define the requirements for voice services.
- Select an IP telephony design model based on the requirements.
- Implement voice support in the infrastructure:
 - Select appropriate call control and transport protocols.
 - Select appropriate coding and compression mechanisms.
 - Provision needed bandwidth.
 - Deploy VoIP components.
 - Implement end-to-end QoS.

Module Summary

- New IP telephony solutions must integrate into existing environments and provide similar functionality.
- Business needs are driving the need for unified networks supporting unified communications networks.
- There are many issues that affect voice traffic, such as delay, jitter, packet loss, congestion, and slow-speed links. Compression techniques, LFI, and QoS mechanisms can alleviate many of these issues.

