Voice Quality for Service Provider Hybrid TDM-IP Networks

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Agenda

- Recognizing Voice Quality Issues
- Classifying Voice Quality Attributes to Root Cause
- Address Voice Quality by Implementing Quality of Service (QoS)
- Proactive Planning—IP Service Level Agreement (SLA)
- Echo
- Proactive Approach to Fix Echo and Voice Distortion Problems—Network Transmission Loss Planning (NTLP)
- Reactive Approach—Troubleshooting

Future of VoIP in South Asia

Internet Telephony is being given the due importance is deserves. While 'unregistered' and 'unlicensed' VoIP termination would still wear the black mask of illegality, the 50+ ISPs of the country (Pakistan) would be offered a slightly different license which will give legal cover to the Internet Telephony operations these ISPs might (and most probably would) like to start.

Tariq Mustafa (APNIC Blog 2003)

Why maintaining Voice Quality in TDM-IP Hybrid Networks is important??

- The VOIP quality issues are further exacerbated when long haul transport is involved specially with the explosive growth of outsourced IP based Contact Centers in South Asian countries.
- VoIP’s growth in Enterprise sector including financial institutions, contact centers (VoIP based), military, and health care makes the compliance to services level agreements important.
- There is a greater need than ever to balance the cost savings promised by VoIP and a network than can deliver business quality voice!!

Recognizing and Categorizing Symptoms of Voice Quality Problems

Categorizing and Defining the Symptoms

- **Noise**
  - Conversation is still Intelligible; presence of static, hum, crosstalk intermittent popping
- **Voice distortion**
  - Problem that affects the voice itself
    - Echoed voice
    - Garbled voice
    - Volume distortion

Noise

Absolute Silence

Cause: Aggressive Voice Activity Detection (VAD)

Clicking

Cause: Clock Slips or Other Digital Errors

Crackling

Cause: Poor Electrical Connection, Electrical Interference

Crosstalk

Cause: Signal Leakage Due to Wires Located in Close Proximity

Noise (Cont.)

Hissing

Cause: VAD

Static

Cause: Codec Mismatch; Enhanced by VAD

Echoed Voice

Listener Echo

Cause: Long Echo Tail; Echo Canceller Is (ECAN) Not Activating

Talker Echo

Cause: Long Echo Tail; ECAN Is Not Activating

Tunnel Voice

Cause: Tight Echo with Some Loss

Garbled Voice

Choppy Voice

Cause: Consecutive Packets Lost or Excessively Delayed Disabling DSP Predictive Insertion Where Silence Is Inserted Instead

Synthetic Voice

Cause: Single Packet Loss or Delay Beyond the Bounds of the De-Jitter Buffer Playout Period

Underwater Voice

Cause: A Common Cause of This Problem Is G729 IETF and Pre-IETF Codec Mismatch

Volume Distortion

Fuzzy Voice

Cause: Too Much Gain on the Signal

Muffled Voice

Cause: Overdriven Signal or Some Other Cause That Eliminates or Reduces Signal Level at Frequencies Inside the Key Range for Voice (Between 440 and 3500)

Soft Voice

Cause: Attenuated Signal

Tinny Voice

Cause: Overdriven Signal that Eliminates or Reduces Signal Level at Frequencies Outside the Key Range for Voice (Between 440Hz and 3500Hz)

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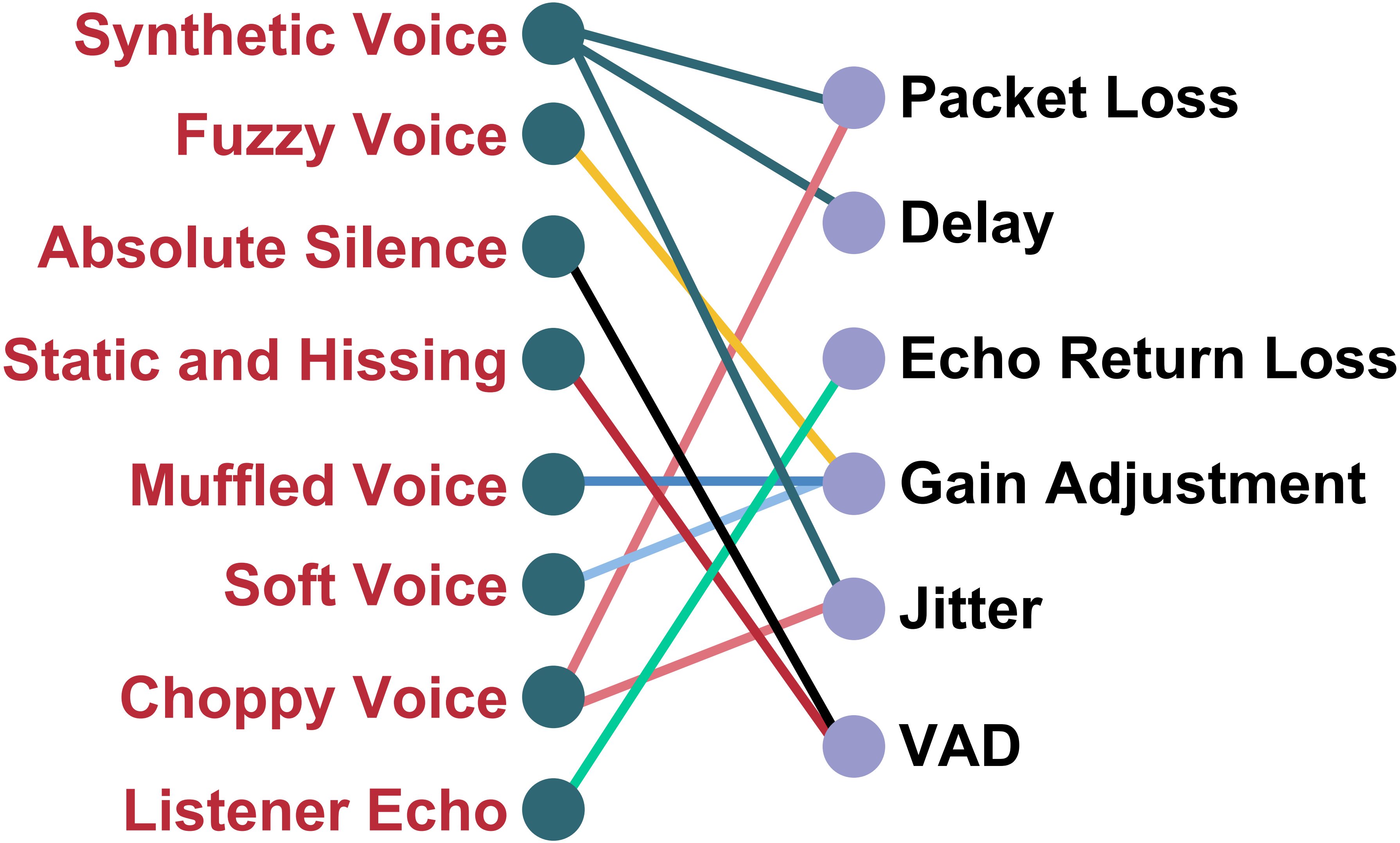
Classifying Voice Quality Attributes to the Root Cause

Problem and Root Cause Association



Classifying Voice Quality Attributes to Root Cause

Quality of Service

- Loss
- Jitter
- Delay

- Synthetic voice
- Robotic voice
- Choppy voice
- Periods of silence

Network Transmission Loss Plan

- Gain adjustment
- ERL

- Talker echo
- Listener echo
- Tunnel voice
- Fuzzy voice
- Muffled voice
- Tinny voice

VAD, Codecs

- Absolute silence
- Clipping
- Static and hissing
- Underwater voice

Synchronization, Cabling

- Crackling
- Clicking
- Crosstalk

Proactive vs. Reactive Approaches

- Proactive approach solves most problems
- Proactive approach does not solve all the problems
- Proactive methodology
  - Planning, Design, Implementation, Operations, Optimization (PDIOO)
  - Network readiness audit, IP SLA, Network Transmission Loss Planning (NTLP), Quality of Service (QoS)
- Reactive approach—too late in the game
- A fix to a specific problem call may adversely effect the entire network
- Reactive tools
  - “show voice call active”
  - Gain adjustments, tail coverage adjustments, VAD tuning, etc.

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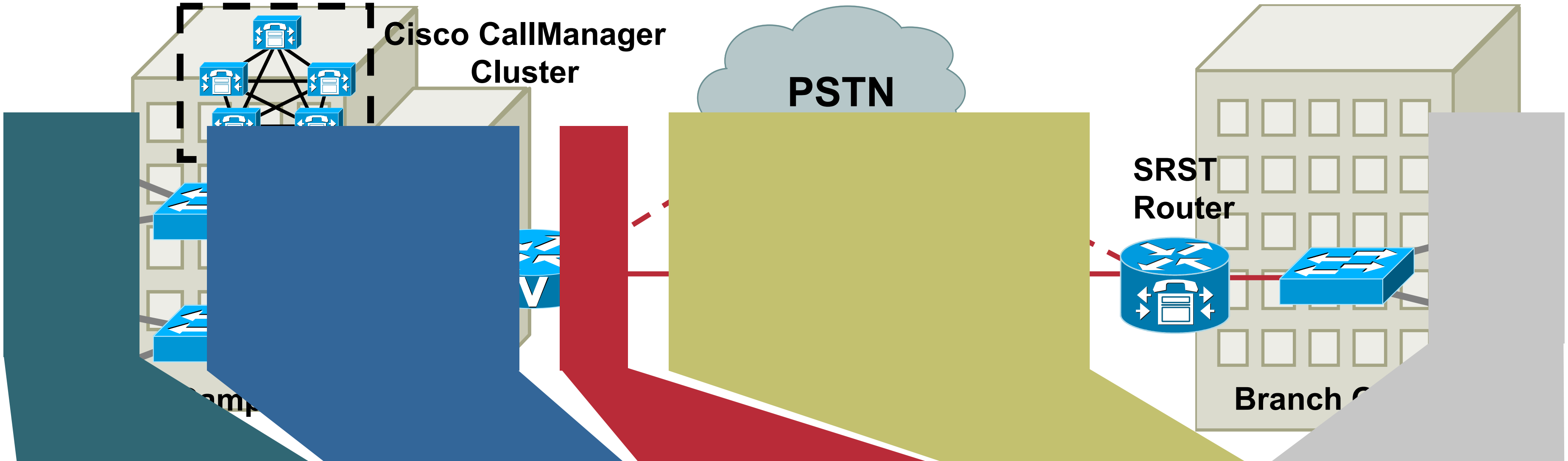
Address Voice Quality by Implementing Proper QoS

Basic Guidelines for of Voice over IP

- **Transmit voice traffic the fastest way possible**
    - Delay is bad (worsens echo, awkward conversations, etc.)
    - Minimize as many sources of delay as possible
    - Goal: keep delay to less than 150ms
- **Transmit VOIP packets as a steady, smooth stream**
    - Any delay should be consistent
    - Inconsistent delay is called "Jitter"
    - Compensating for Jitter creates additional delay
- **Drop any packets received out of order**
    - Voice does not tolerate delays…it's better to drop the packet
    - CODEC logic can compensate for some dropped packets
- **Above all…it's gotta sound good (subjective)**

Elements That Affect End-to-End Delay

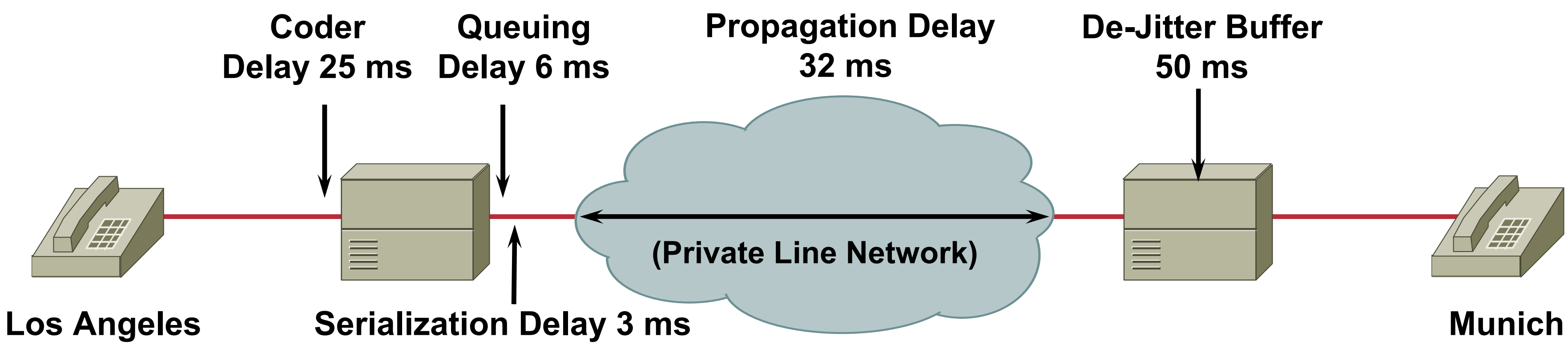


CODEC	Queuing	Serialization	Propagation and Network	Jitter Buffer
G.729A: 25 ms	Variable (Can Be Reduced Using LLQ)	Variable (Can Be Reduced Using LFI)	Network Delay (Variable)	20–50 ms

End-to-End Delay (Should Be < 150 ms)

As per ITU G.114 Recommendations

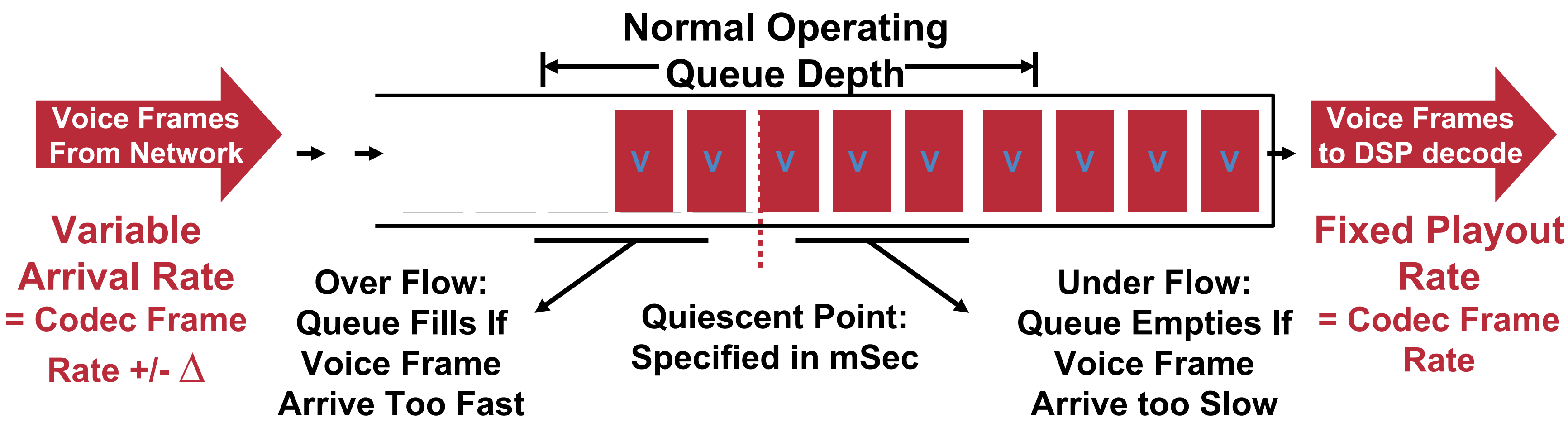
Calculate Delay Budget



	Fixed Delay	Variable Delay
Coder Delay G.729 (5 msec Look Ahead)	5 msec	
Coder Delay G.729 (10 msec per Frame)	20 msec	
Packetization Delay—Included in Coder Delay		
Queuing Delay 64 kbps Trunk		~ 6 msec
Serialization Delay 64 kbps Trunk	3 msec	
Propagation Delay (Private Lines)	32 msec	
Network Delay (e.g.,Public Frame Relay Svc)	N/A—Private Line	
De-Jitter Buffer		~ 50 msec
Total:	~ 116 msec	

Goal:
150mx
Delay

De-Jitter Buffer Operation



- **When voice call starts, the de-jitter buffer fills up to the quiescent point**
  - As voice frames arrive too fast, the queue fills
  - As voice frames arrive too slowly, the queue empties
- **Depth of queue varies with network operation**
  - Over-/-under flow will cause gaps in speech and underwater voice

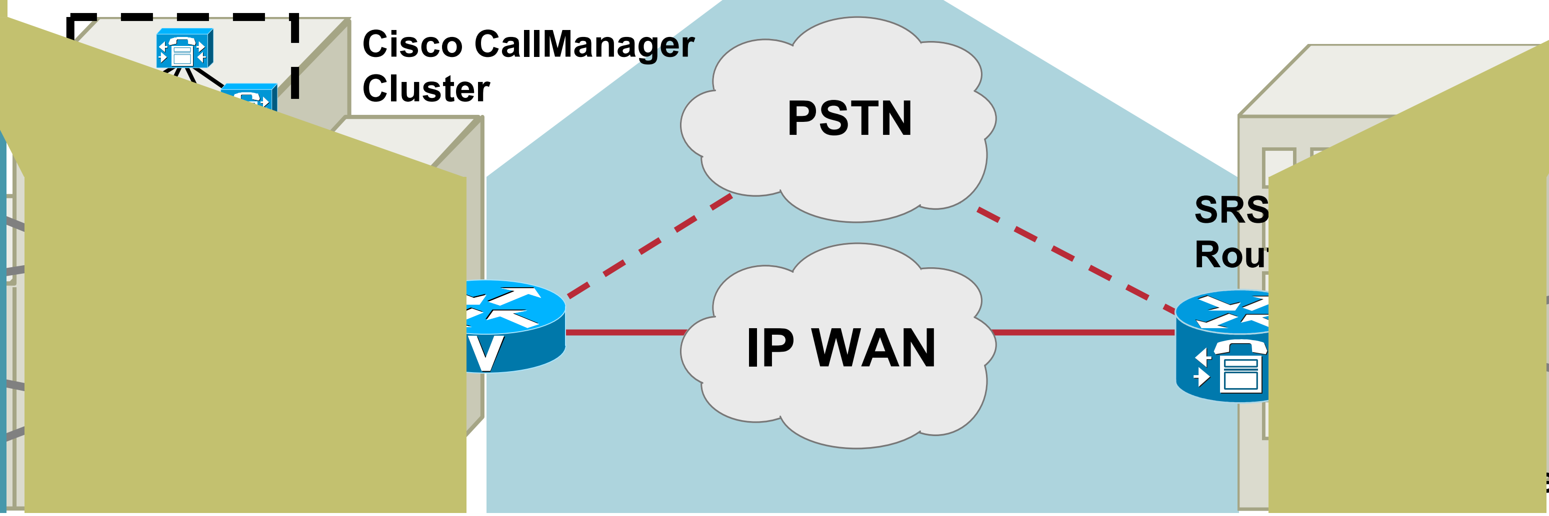
Enabling QoS in the WAN
QoS Approach Summary

Classification: Mark the Packets with a Specific Priority Denoting a Requirement for Class of Service from the Network

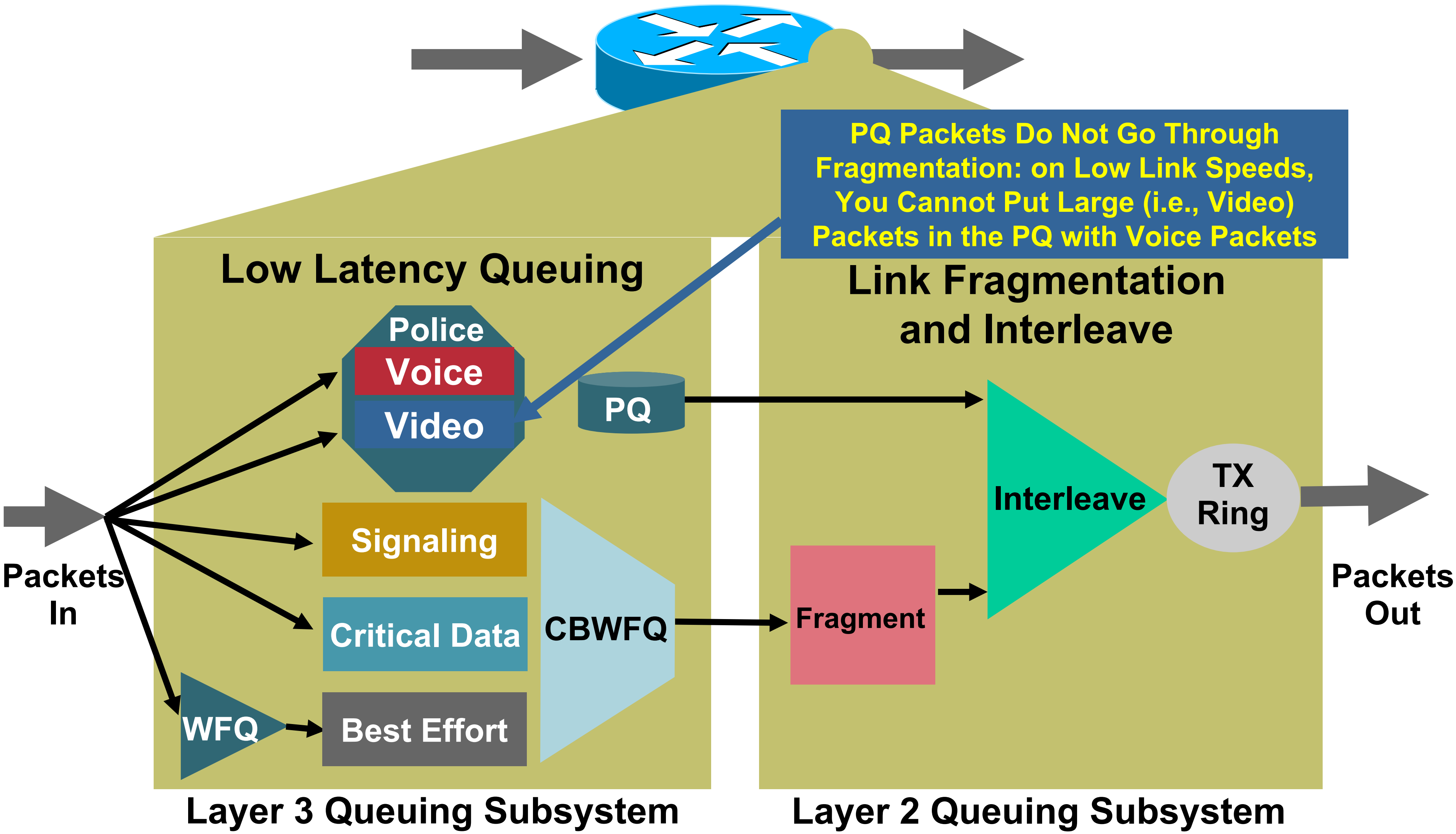
Trust Boundary: Define and Enforce a Trust Boundary at the Network Edge

Scheduling: Assign Packets to One of Multiple Queues (Based on Classification) for Expedited Treatment Through the Network

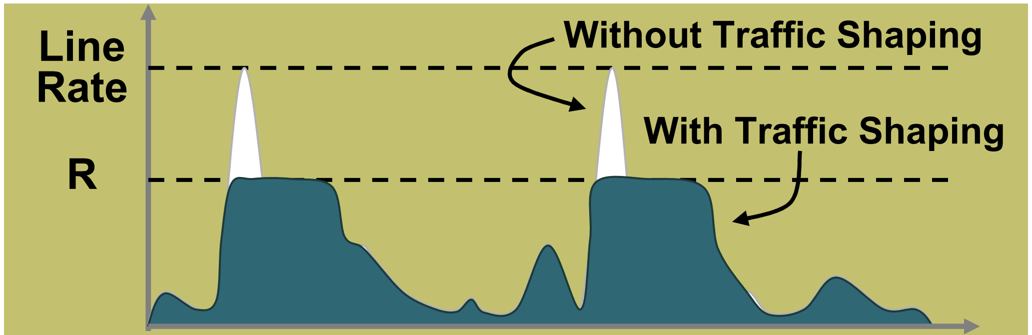
Provisioning: Accurately Calculate the Required Bandwidth for All Applications Plus Element Overhead



Network Infrastructure and QoS
Scheduling in the WAN



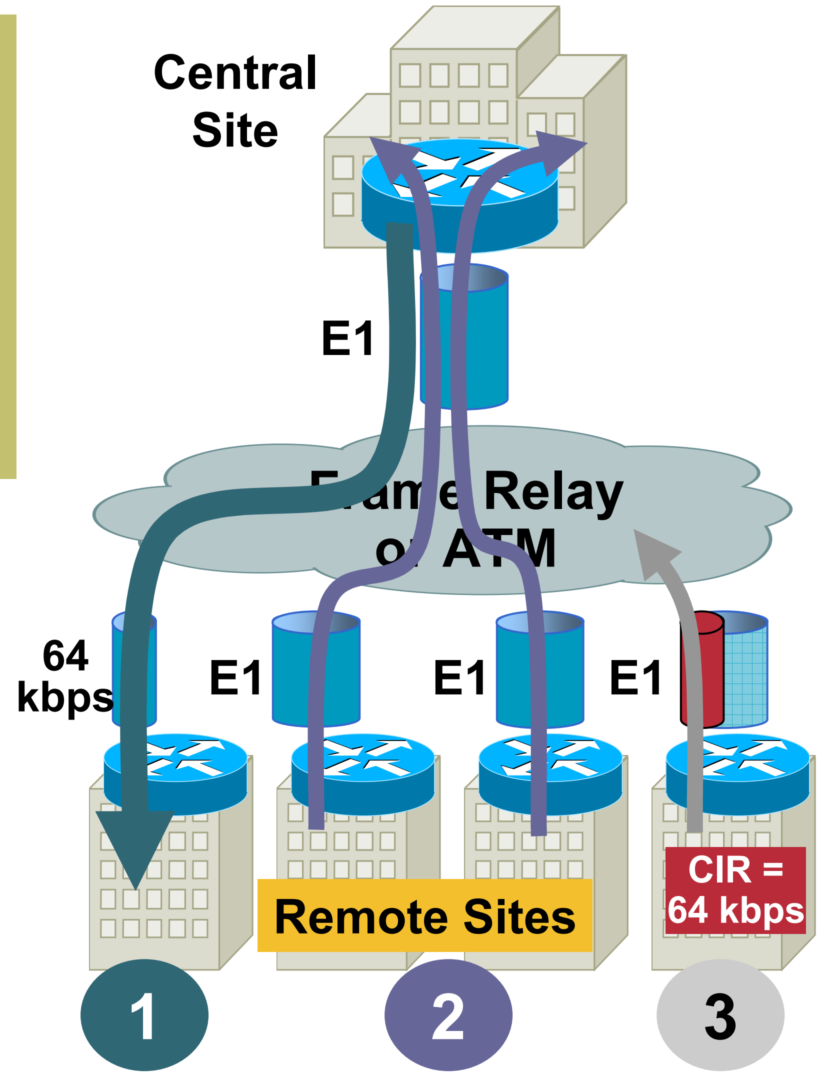
Enabling QoS in the WAN
Traffic Shaping



Traffic Shaping Limits the Transmit Rate to a Value (R) Lower than Line Rate

Why Is It Needed

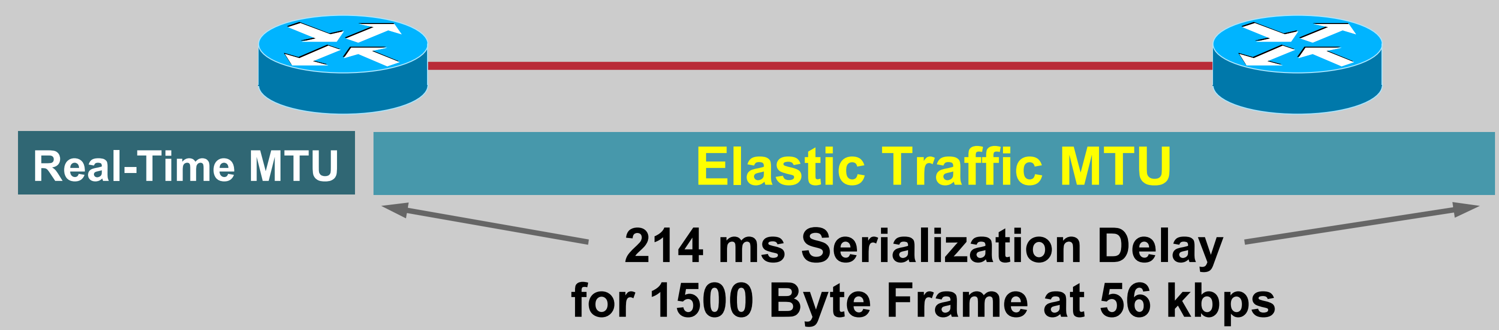
1. Line Speed Mismatch
2. Remote to Central Site Over-Subscription
3. To Prevent Bursting Above Committed Rate (CIR)



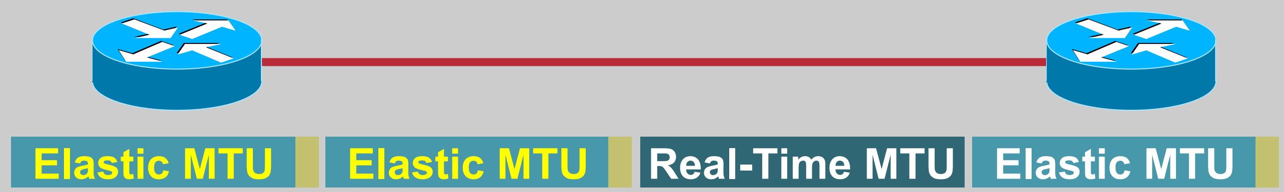
Enabling QoS in the WAN
Link Fragmentation and Interleaving (LFI)

Fragmentation and Interleave Not Needed on Links Greater Than 768 kbps

Before



After



Mechanisms:

Pt to Pt Links:	MLPPP
Frame Relay:	FRF.12
ATM:	MLPPP over ATM
ATM/Frame-Relay SIW:	MLPPP over ATM and FR

Bandwidth Usage—G.729 Example

	IP	UDP	RTP	cRTP	Payload	Total
No cRTP:	20	+ 8	+ 12		+ 20	= 60 Bytes
With cRTP:				2	+ 20	= 22 Bytes

60 bytes/packet x 50 PPS x 8 bits/byte = 24 Kbps
22 bytes/packet x 50 PPS x 8 bits/byte = 8.8 Kbps

Branch Size	VoIP Trunks	RTP	cRTP
Small	3	72 Kbps	26.4 Kbps
Medium	8	192 Kbps	70.4 Kbps
Large	16	398 Kbps	140.8 Kbps

Call Admission Control (CAC) Locations

Why CAC?

Boat Capacity = 5 Persons

When the Sixth Person Climbs Aboard Everybody Gets Wet

Branch Size	Max Calls	Locations BW	Available BW	cRTP
Small	3	72 Kbps	128 Kbps	26.4 Kbps
Medium	8	192 Kbps	256 Kbps	70.4 Kbps
Large	16	398 Kbps ↓	256 Kbps ↑	140.8 Kbps

Enabling QoS in the WAN

Summary

- Use LLQ anytime VoIP over the WAN is involved
- Traffic shaping is a requirement for Frame Relay/ATM environments
- Use LFI techniques for all links below 768Kbps
    - Don't use LFI for any video over IP applications
- TX-ring sizes may require modifications
- Properly provision the WAN bandwidth
- Call admission control is a requirement where VoIP calls can over-subscribe the provisioned BW
- Use cRTP carefully
- Map QoS from L3 (IP Prec or DSCP) to L2 (802.1p) at remote branches if switch is L2 only

More Information on QoS

- **QoS Design Guide: http://www.cisco.com/go/srnd**

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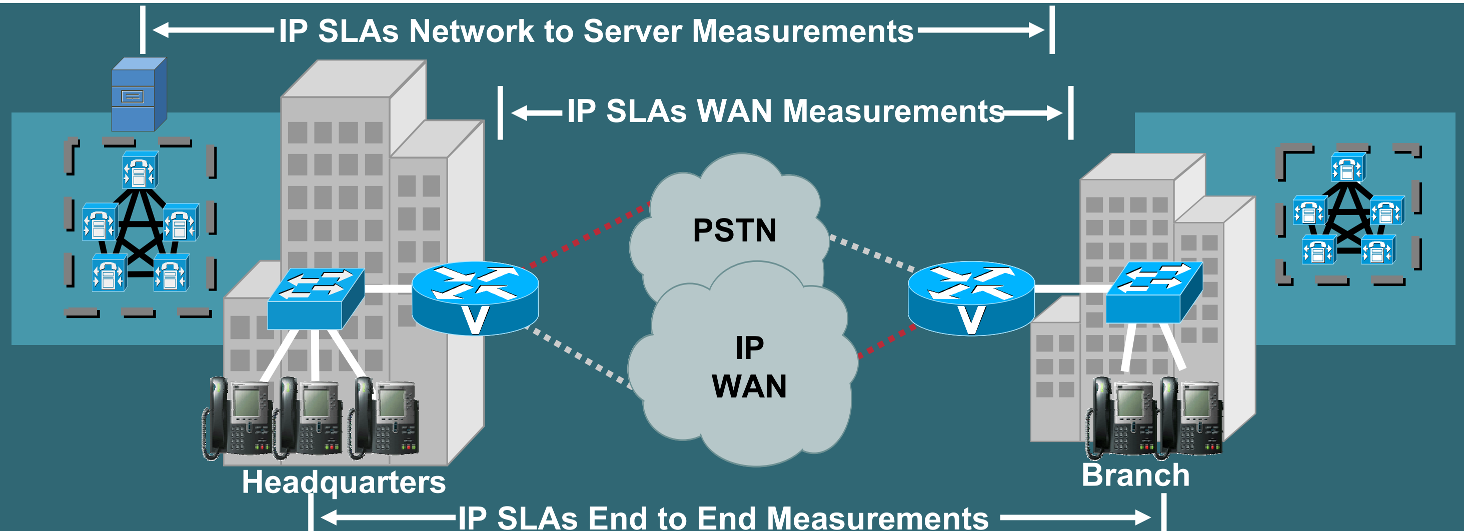
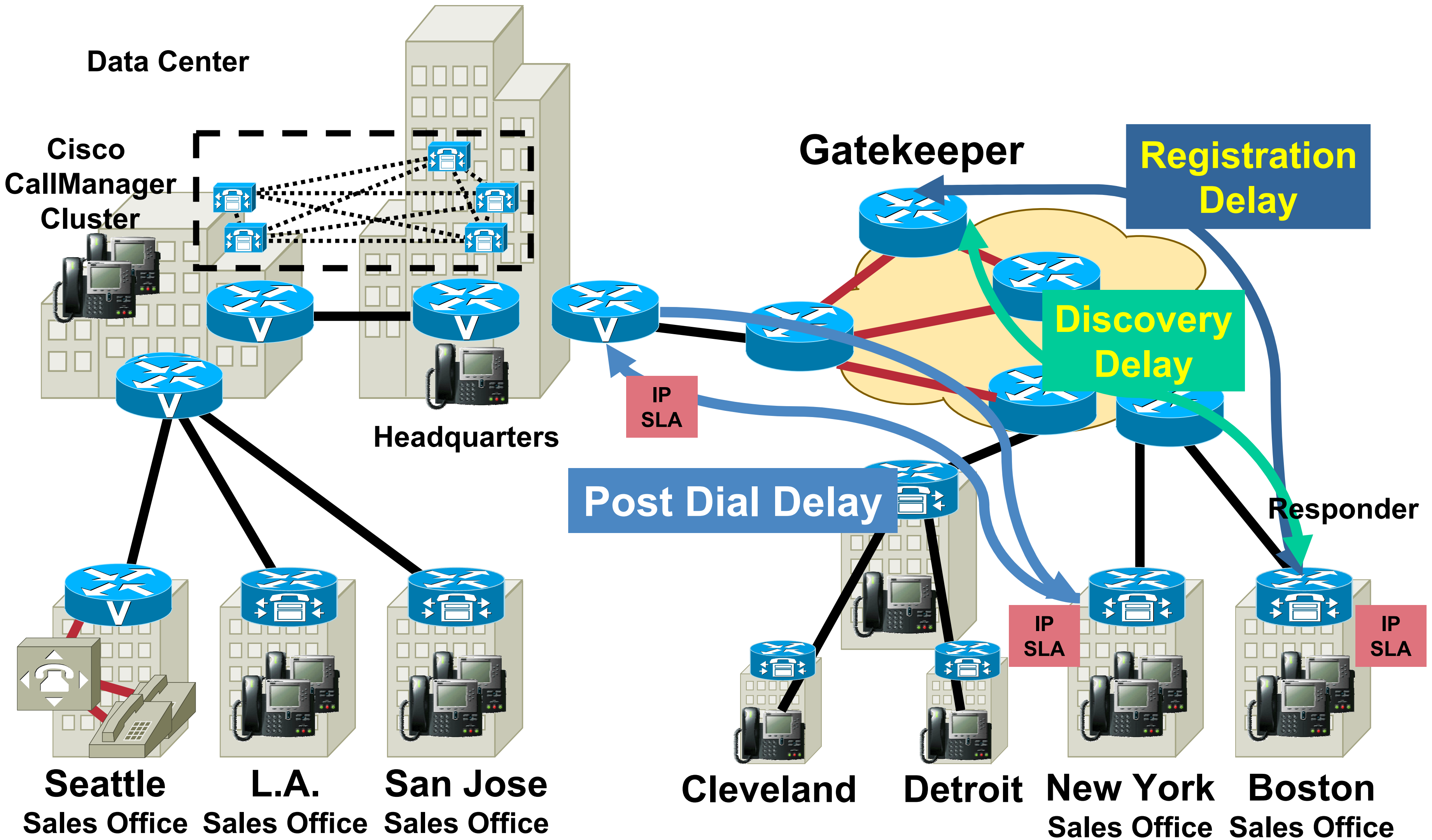
Proactive Planning
IP SLA (a.k.a. SAA)

Cisco IOS IP SLAs for VoIP

- Measurements between any two network points on any path
- Continuous, reliable, predictable performance monitoring
- Cisco IOS® IP SLAs thresholds and hop-by-hop details isolate problems

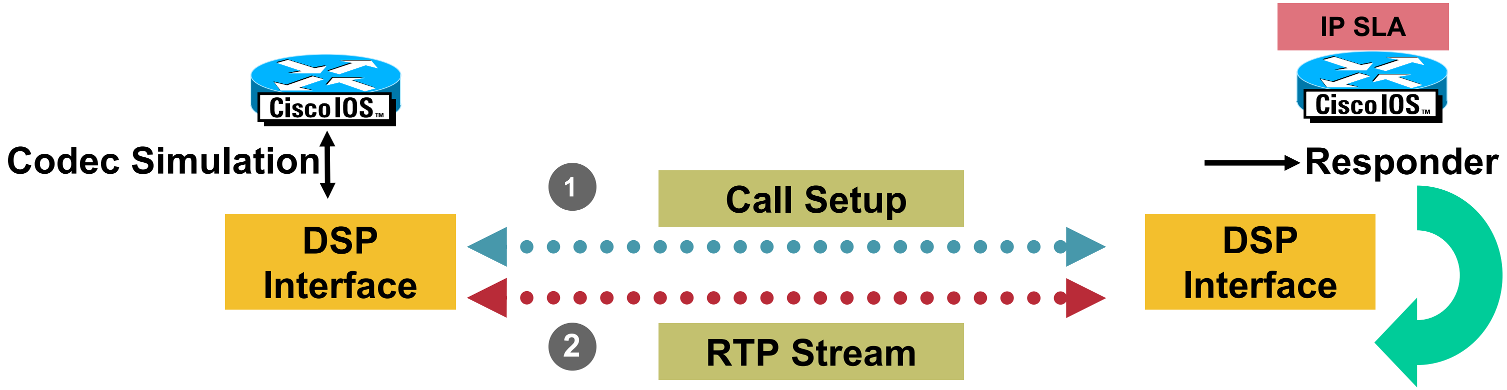


IP SLA VoIP Measurements (Q1 CY'05)

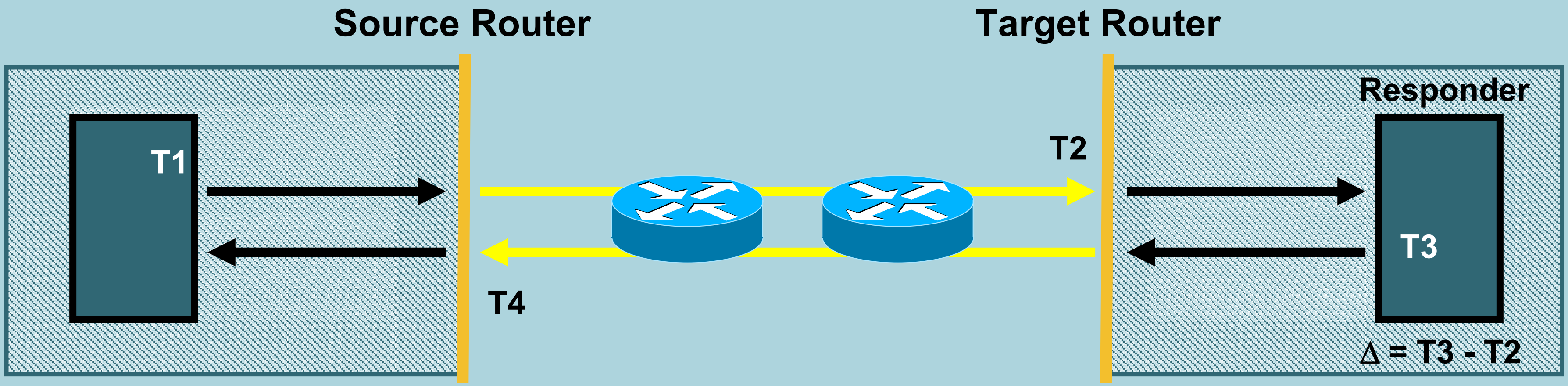


DSP-Based IP SLA Measurements

- Call setup with Session Initiation Protocol (SIP) and establish RTP session between two end points (DSP to DSP)
- Also measure from DSP based source to any Cisco end point
- Measure VoIP statistics from the DSP, so that it can be presented to the user (via command-line interface (CLI) and SNMP)



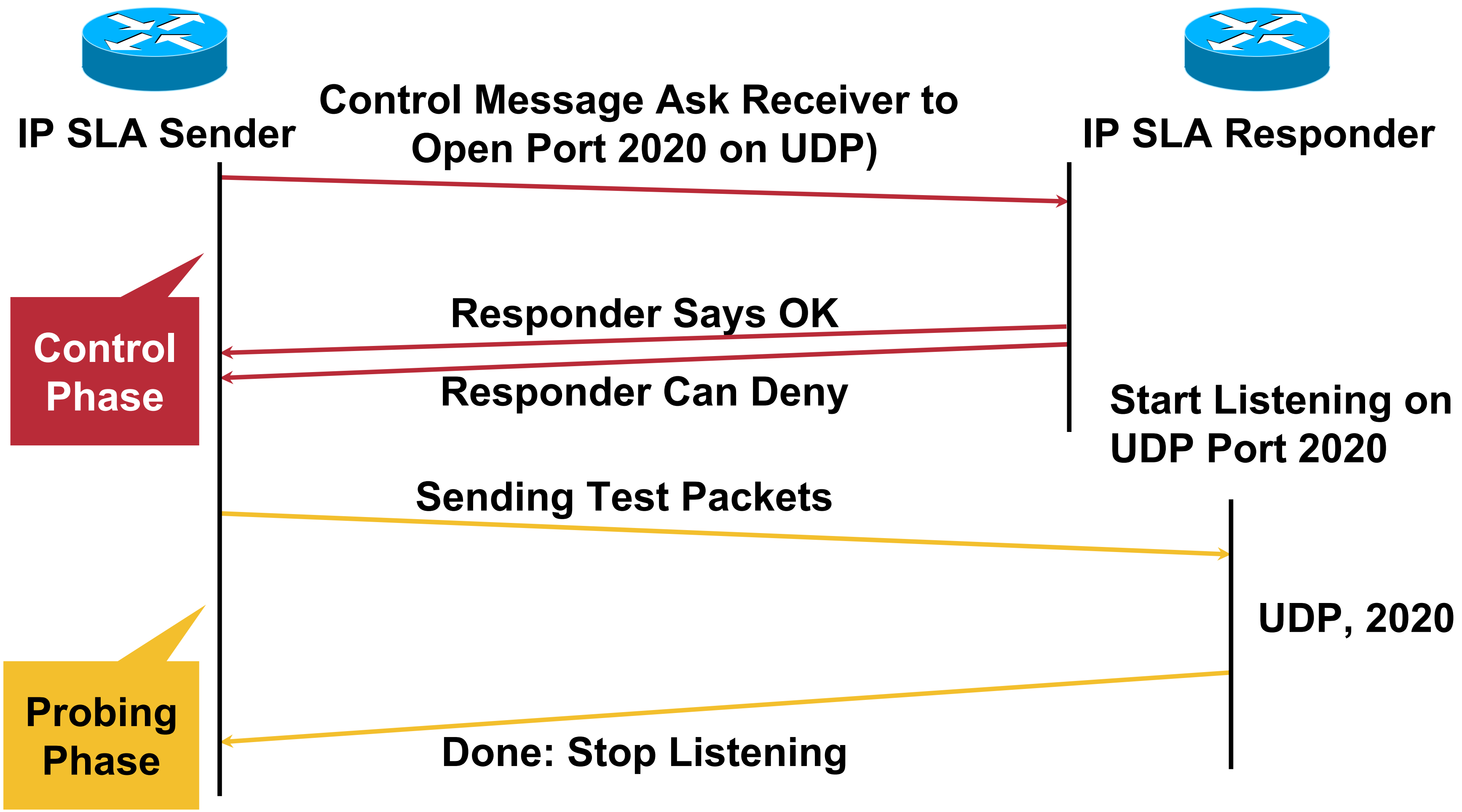
Responder



The Responder Takes Two Timestamps (T2 and T3)

- Responder factors out destination processing time making results highly accurate
- Responder allows for one-way measurements for latency, jitter, packet loss, and MOS

Cisco IP SLA Operation with Responder Control Protocol



UDP Jitter Operation: Output (1/3)

```
Router#sh ip sla monitor op 1
        Current Operational State
Entry Number: 1
Modification Time: 08:22:34.000 PDT Thu Aug 22 2002

                    ry was Reset: Never
Number of Octets in use by this Entry: 1594
Number of Operations Attempted: 1
Current Seconds Left in Life: 574
Operational State of Entry: active
Latest Operation Start Time: 08:2                       22 2002
Latest Oper Sense: ok
RTT Values:
NumOfRTT: 997   RTTSum: 458111  RTTSum2: 238135973
Packet Loss Values:
PacketLossSD: 3 PacketLossDS: 0
PacketOutOfSequence: 0  PacketMIA: 0    PacketLateArrival: 0
InternalError: 0        Busies: 0
(cont…)
```

Three Packets Lost
S → D Out of 1,000 Sent

Average RTT Was
458111/997 = 459ms

UDP Jitter Operation: Output (2/3)

Source to Destination Jitter

Destination to Source Jitter

```
(…cont)
Jitter Values:
MinOfPositivesSD: 1     MaxOfPositivesSD: 249
NumOfPositivesSD: 197   SumOfPositivesSD: 8792  Sum2PositivesSD: 794884
MinOfNegativesSD: 1     MaxOfNegativesSD: 158
NumOfNegativesSD: 761   SumOfNegativesSD: 8811  Sum2NegativesSD: 139299
MinOfPositivesDS: 1     MaxOfPositivesDS: 273
NumOfPositivesDS: 317   SumOfPositivesDS: 7544  Sum2PositivesDS: 581458
MinOfNegativesDS: 1     MaxOfNegativesDS: 183
NumOfNegativesDS: 603   SumOfNegativesDS: 6967  Sum2NegativesDS: 336135
Interarrival jitterout: 16      Interarrival jitterin: 35
One Way Values:
NumOfOW: 0
OWMinSD: 0       OWMaxSD: 0      OWSumSD: 0      OWSum2SD: 0
OWMinDS: 0       OWMaxDS: 0      OWSumDS: 0      OWSum2DS: 0
```

See Next Slide

No Synchro Between Clocks: All Zeroes

VoIP UDP Jitter Operation: Output (3/3)

```
Source# show ip sla monitor operation-state 5
        Current Operational State
…
Voice Scores:
ICPIF Value: 20 MOS score: 3.20
RTT Values:
NumOfRTT: 11     RTTAvg: 2583    RTTMin: 711     RTTMax: 4699
RTTSum: 28422    RTTSum2: 92644272
Packet Loss Values:
PacketLossSD: 0 PacketLossDS: 0
PacketOutOfSequence: 0  PacketMIA: 989  PacketLateArrival: 56
InternalError: 0        Busies: 0
Jitter Values:
MinOfPositivesSD: 1     MaxOfPositivesSD: 249
NumOfPositivesSD: 197   SumOfPositivesSD: 8792  Sum2PositivesSD: 794884
MinOfNegativesSD: 1     MaxOfNegativesSD: 158
NumOfNegativesSD: 761   SumOfNegativesSD: 8811  Sum2NegativesSD: 139299
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Interarrival jitterout: 16      Interarrival jitterin: 35
One Way Values:
NumOfOW: 0
OWMinSD: 0      OWMaxSD: 0      OWSumSD: 0      OWSum2SD: 0
OWMinDS: 0      OWMaxDS: 0      OWSumDS: 0      OWSum2DS: 0
```

SD: Source to Destination
DS: Destination to Source
OW: One-Way Delay

Note: New CLI Shown as Example will Be Available in Release 12.3(pi6)T (Q1 CY'05)

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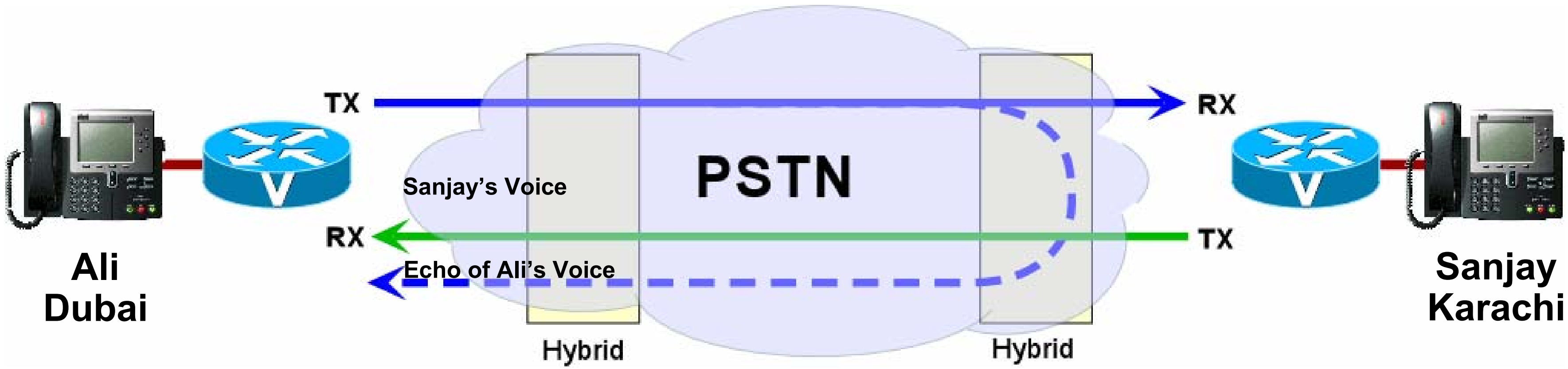
Echo

Talker Echo

Talker Echo (Most Common)

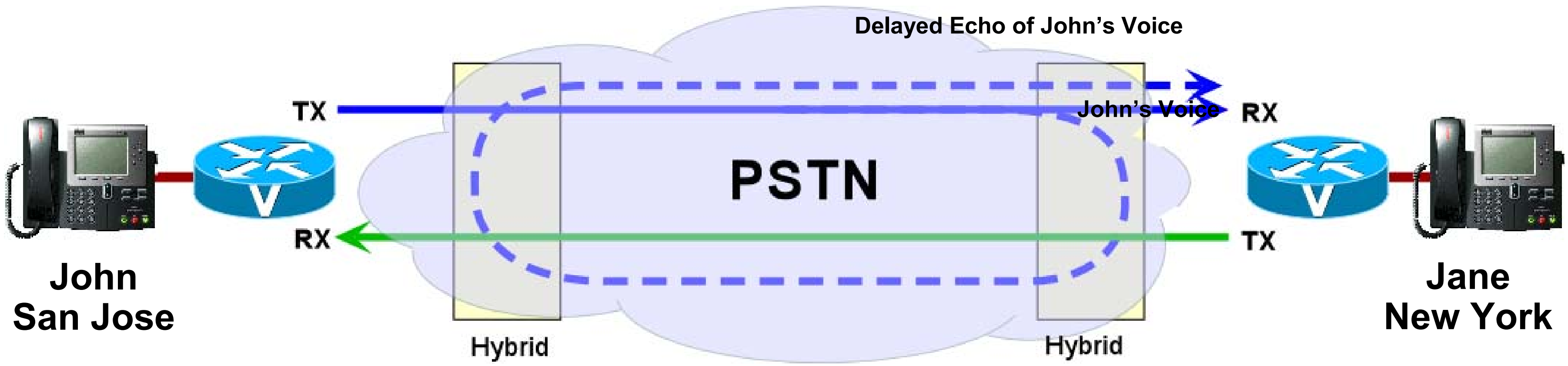
- Talker echo occurs when a talker's speech energy, transmitted down the primary signal path, is coupled into the receive path from the far end; the talker then hears his/her own voice, delayed by the total echo path delay time; if the ‘echoed’ signal has sufficient amplitude and delay, the result can be annoying to the customer and interfere with the normal speech process; talker echo is usually a direct result of the 2-wire to 4-wire conversion that takes place through ‘hybrid’ transformers



Listener Echo

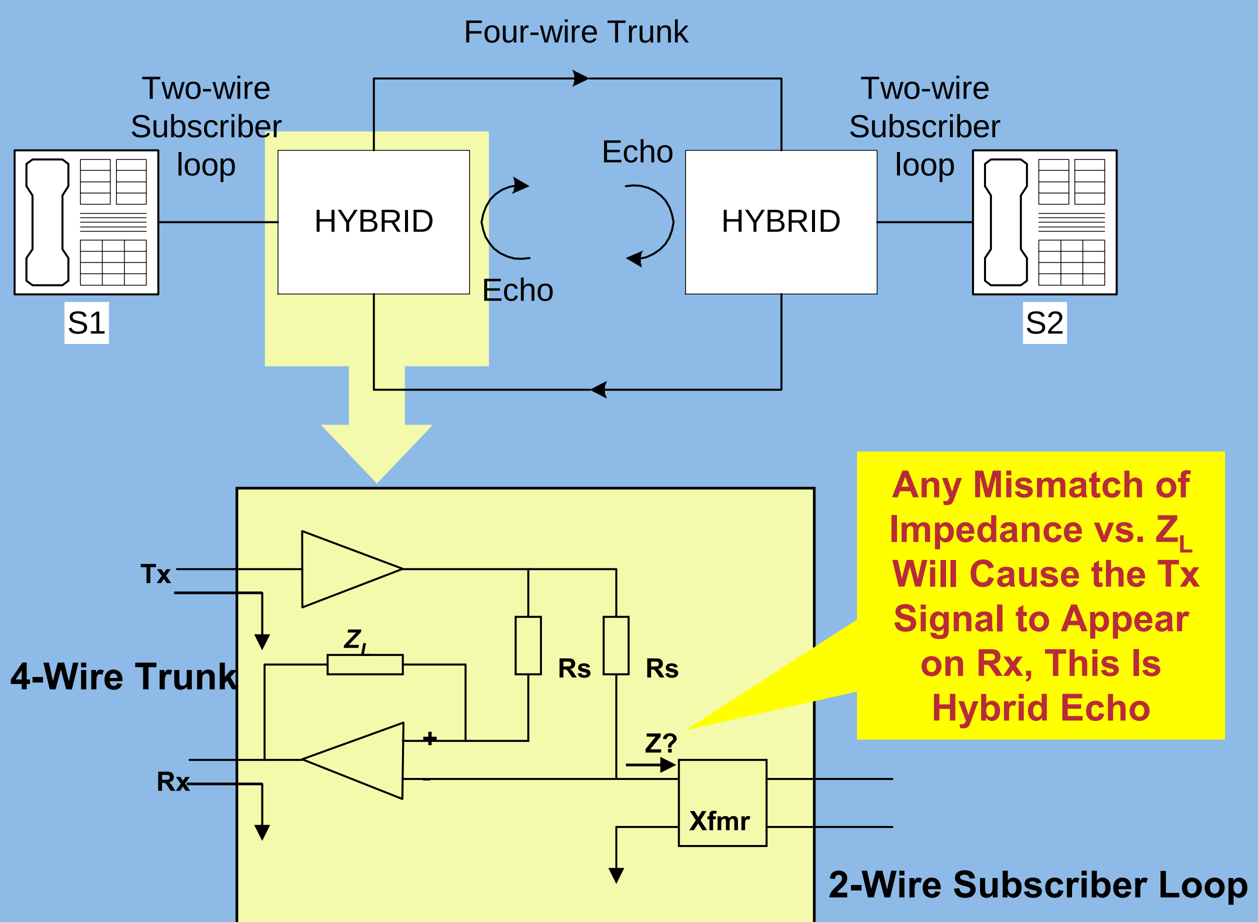
Listener Echo (Less Common)

- **Listener echo occurs at the far-end by circulating voice energy; again, listener echo is generally caused by the 2W/4W 'hybrid' transformers; caused by the "echo being echoed"; the talker's voice is echoed by the far end hybrid and when the echo comes back to the listener, the hybrid on the listener's side echoes the echo back towards the listener; the effect is the person listening hears the talker and an echo of the talker**



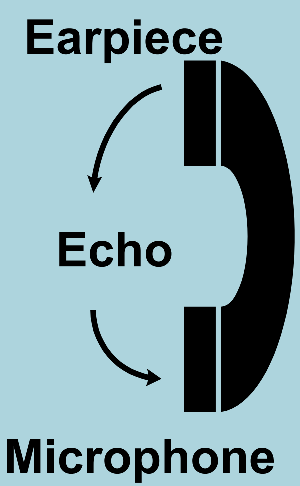
Sources of Echo

Hybrid Echo

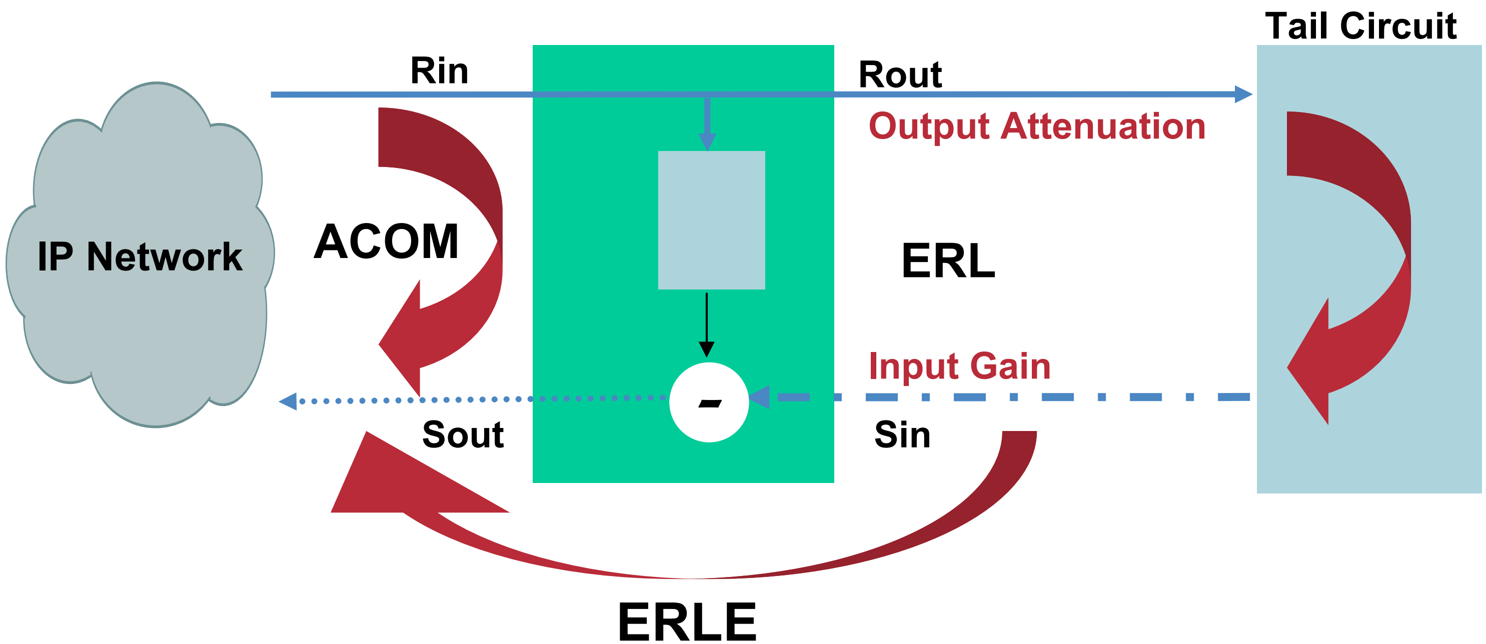


Schematic of a Hybrid Circuit

Acoustic Echo



Eliminating Echo
Leveraging Echo Cancellor



- **So, how do I get rid of echo?**
- **Give the echo canceller enough information to distinguish between echo and normal conversation; the only parameters you have control over are:**
  - **Input level (input gain)**
  - **Output level (output attenuation)**
  - **Echo canceller coverage**

Definitions

- **Output Attenuation** of a signal is performed **after** the echo canceller has 'seen' the original output signal
- **Input Gain** of a signal is performed **before** the echo canceller has 'seen' the echo
- **Echo Cancel Coverage** is the amount of time the Echo Canceller will 'Remember' a signal that has been output; this parameter must be set to a value greater than the time it takes the echo to return back to the gateway
- **Echo Return Loss Enhancement (ERLE)** refers to the additional echo loss obtained through the operation of the echo canceller; an echo canceller is not a perfect device, and the best it can do is attenuate the level of the returning echo; ERLE is a measure of this echo attenuation through the echo canceller; it is the difference is level (in dB) between the signal arriving from the tail circuit at the Sin terminal of the echo canceller and the level of the signal leaving the echo canceller (and entering the network) at the Sout terminal
- **ACOM** (a.k.a. Acombined) is simply the total echo return loss seen across the Rin and Sout terminals of the echo canceller, and is the sum ERL + ERLE; it is the echo return loss seen by the network

  ERL = Echo return loss through tail = Rout - Sin (dB)

  ERLE = Echo return loss enhancement through echo canceller = Sin - Sout (dB)

  ACOM = Combined echo return loss through system = Rin - Sout (dB)

What Makes Echo a Problem?

For Echo to Be a Problem, all of the Following Conditions must Exist:

- **An analog leakage path between analog Tx and Rx paths**
- **Sufficient delay in echo return for echo to be perceived as annoying**
- **Sufficient echo amplitude to be perceived as annoying**

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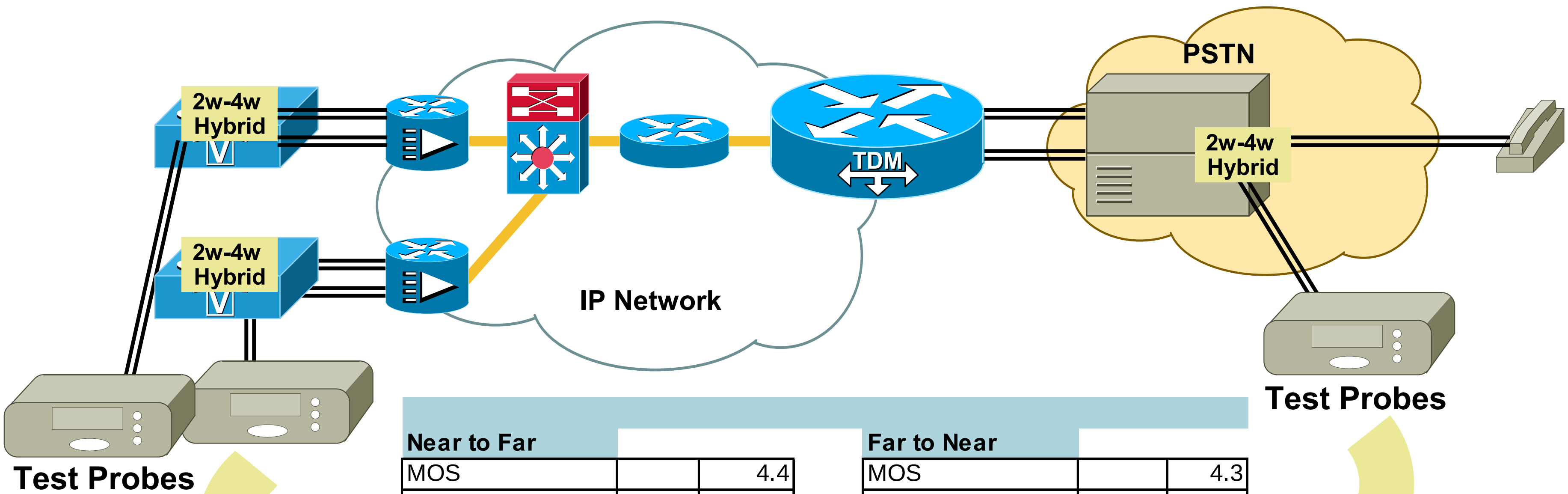
Proactive Approach to Fixing Echo and Voice Distortion

Network Transmission Loss Plan

Network Transmission Loss Planning



Near to Far

MOS		4.4
Noise	dBrnC	14.5
(-)FS	ms	0
(+)FS	ms	0
Gain	dB	3
BW	%	98.1
Codec		PCM
Delay	ms	83
BRL (calculated)	dB	26.5

Far to Near

MOS		4.3
Noise	dBrnC	14.9
(-)FS	ms	0
(+)FS	ms	0
Gain	dB	9.5
BW	%	99.3
Codec		PCM
Delay	ms	83
BRL (calculated)	dB	26.5

ERL	dB	14

Loudness Ratings
Terminology

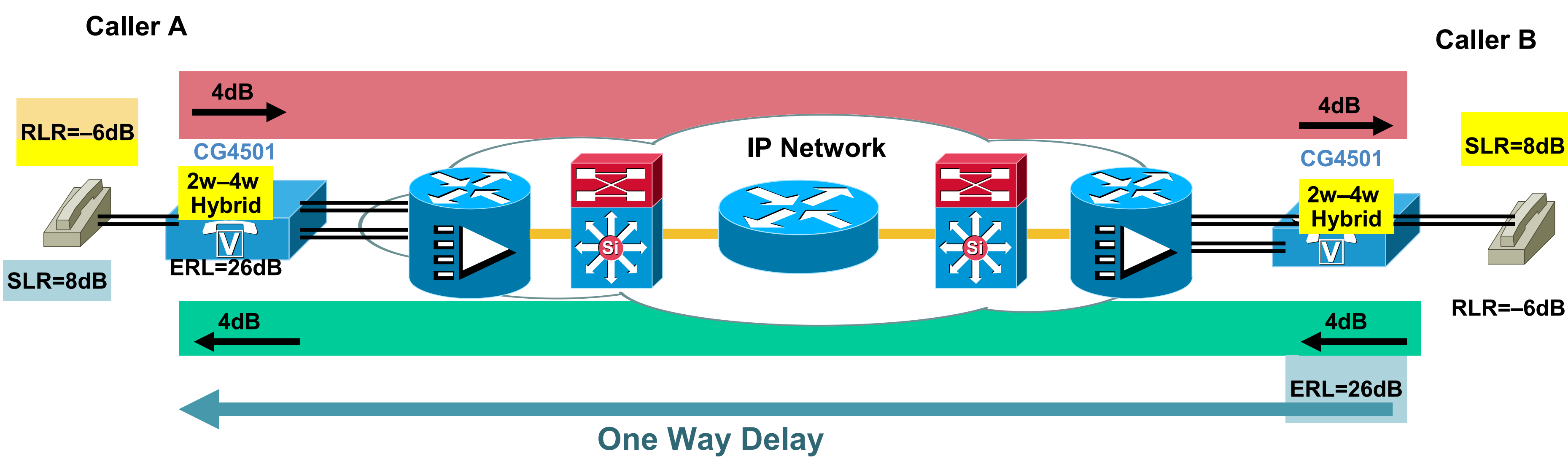
- **Send Loudness Rating (SLR):** the loudness between the Mouth Reference Point (MRP) and the electrical interface
- **Receive Loudness Rating (RLR):** the loudness between the electrical interface and the Ear Reference Point (ERP)
- **Overall Loudness Rating (OLR)**: the total loudness loss between the MRP and ERP in a connection; OLR is calculated as follows:

  OLR = SLRtalker + [sum]attenuations + RLRlistener

- **Talker Echo Loudness Rating (TELR)**: the loudness loss between the talker's mouth and the ear via the echo path. TELR is calculated as follows:

  TELR(A) = SLR(A) + loss in top path +ERL(B) or TCLw(B) + loss in bottom path + RLR(A), where ERL is the echo return loss of the hybrid or echo canceller, and TCLw is the weighted terminal coupling loss of the digital phone set

On-Net to On-Net Call

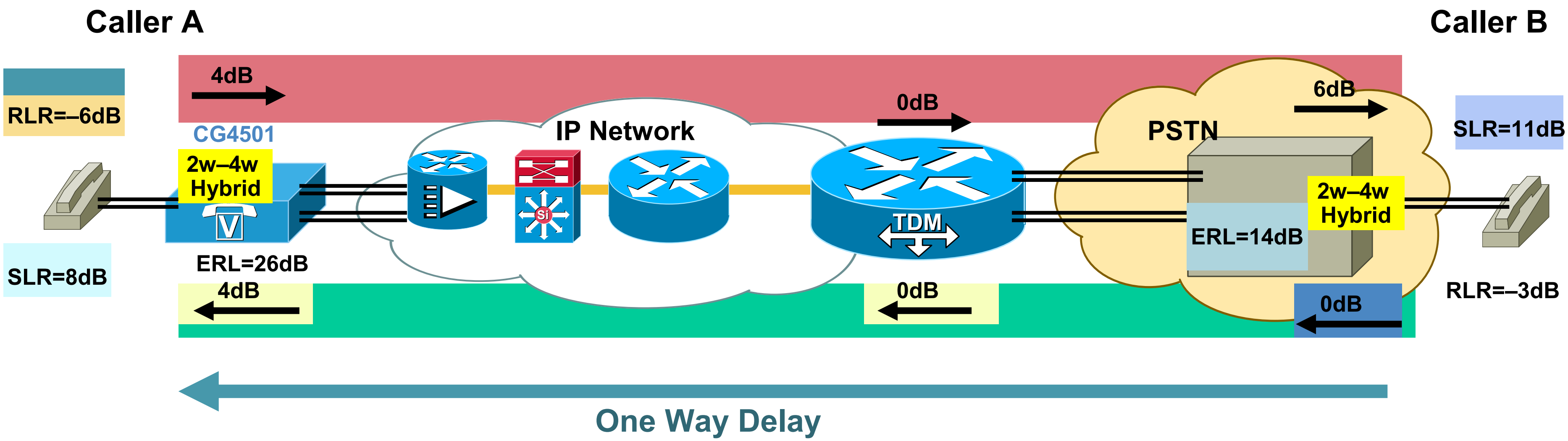


OLR(A) = 8dB + 4dB + 4dB - 6dB = 10dB

TELR (A) = 8dB + 4dB + 4dB + 26dB + 4dB + 4dB - 6dB = 44dB

Talker Echo Loudness Rating (TELR)
Overall Loudness Rating (OLR)

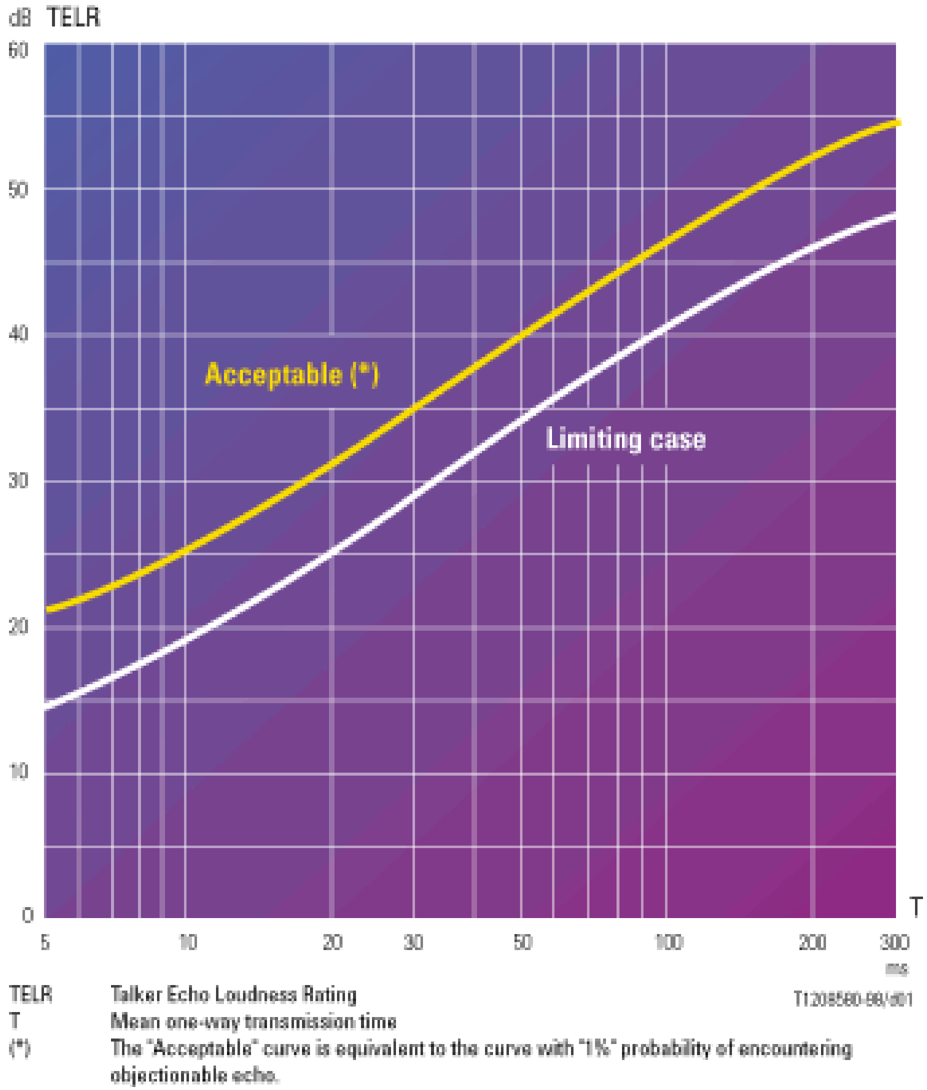
On-Net to Off-Net Call



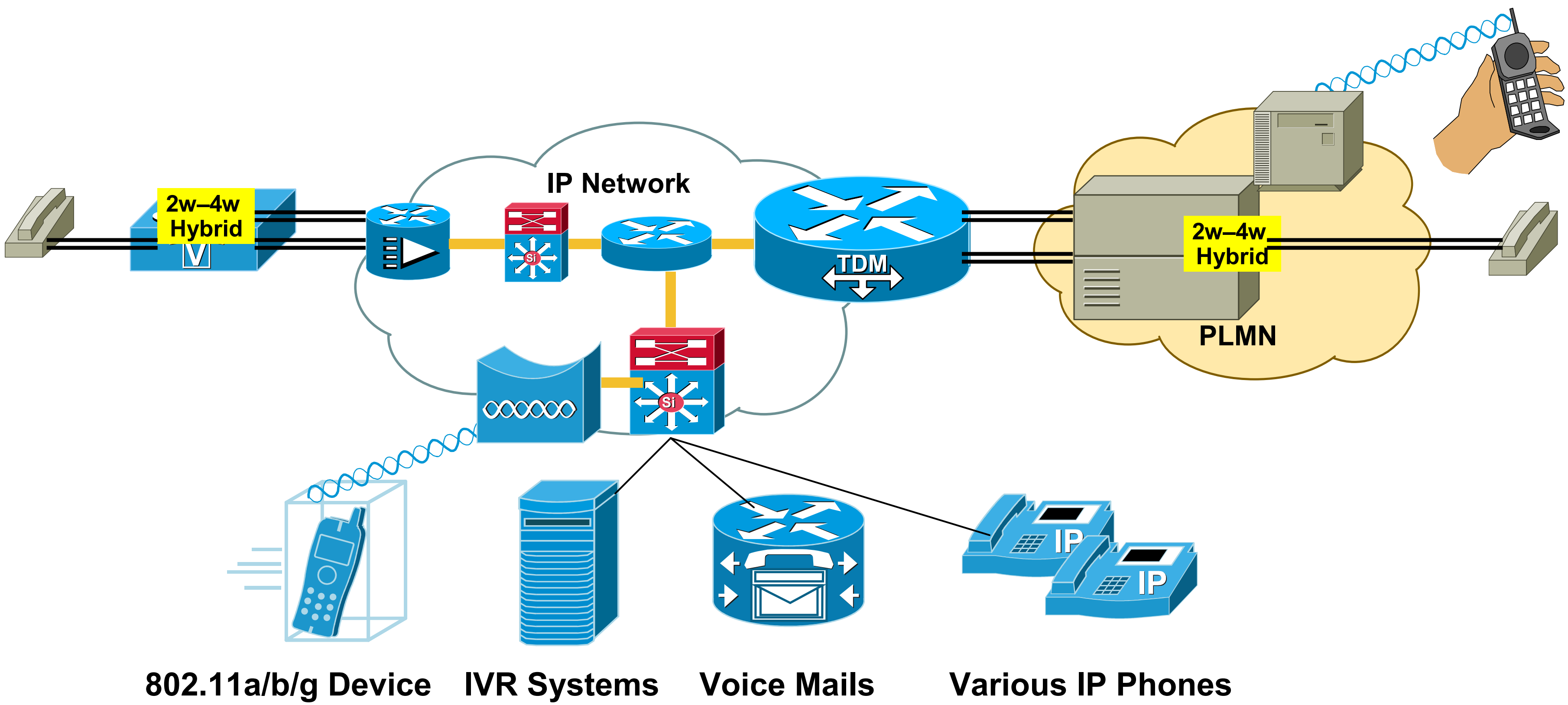
OLR(A) = 11dB + 0dB + 0dB + 4dB - 6dB = 10dB

TELR (A) = 8dB + 4dB + 0db + 6dB + 14dB + 0dB + 0dB + 4dB - 6dB = 30dB

Echo Loudness Rating vs. Delay (ITU G.131)



Test for All the Endpoints

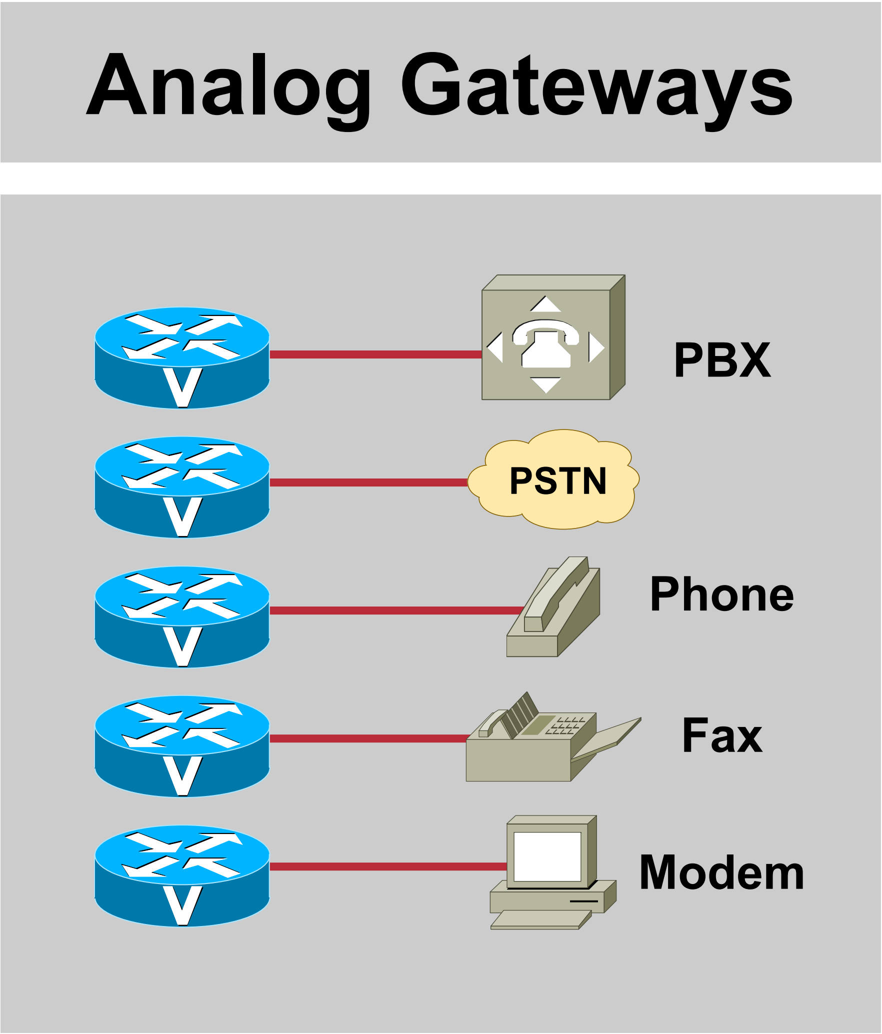


Don't Assume Loss Plan Works Across All Possible Calls

Rules of Thumb

- Echo on one end is typically generated at other end
- Bits don’t leak—Echo is not introduced on digital links
- ERL must be 6dB for ECANs to engage
- Introduced by 2 to 4 wire conversion in hybrid and impedance mismatch or via acoustic feedback
- Be careful setting echo-cancel coverage; longer coverage yields longer convergence time; configure the coverage so that it is long enough to cover the worst-case for your environment, but no higher
- Use **3 on 7960/40 to use the built-in 1004 Hz tone generator
- # or * DTMF tones approximate 1004Hz @ 0dB tones (if test gear is not available)

Cabling



- Cabling is the number one cause of issues in analog connections
- Cabling testing must be a part of implementation plan
- NTLP is a good source for verifying cabling issues

Agenda

- Recognizing Voice Quality Issues
- Classifying Voice Quality Attributes to Root Cause
- Address Voice Quality by Implementing Quality of Service (QoS)
- Proactive Planning—IP Service Level Agreement (SLA)
- Echo
- Proactive Approach to Fix Echo and Voice Distortion Problems—Network Transmission Loss Planning (NTLP)
- Reactive Approach—Troubleshooting

Reactive Approach
Troubleshooting

“show call active voice” Command in Cisco IOS

- Information about POTS and VOIP dial peers
- Information about noise level, output, and input signal levels
- Information about echo (ACOM and ERL)
- Information about jitter, delay, and packet drops
- Information about CODECs and VAD

General Level Adjustment Guidelines

- Map out your network loss plans
- Avoid adding level (gain) on the input side
  - It amplifies noise
- Try to reduce attenuation at the output instead
- To raise an output level
  - First, decrease the attenuation at the output side
  - If you are applying 0 dBm of attenuation, and the signal is still too soft, then go to the input side and increase the gain
  - Working this way avoids over-driving the inputs on the first pass
- To lower an output level
  - Adjust the input side first
  - Then adjust the output side

Measuring Echo in Cisco IOS

If We Configure 1 dB of Attenuation in Each Direction as Follows:

```
voice-port 1/1:23
 input gain -1
 output attenuation 1
```

The Resulting Levels Are as Follows:

```
Gateway# sh call active voice
- snip -
OutSignalLevel=-16
InSignalLevel=-17
ERLLevel=11
- snip -
```

- **Notice the OutSignalLevel is -16 because we attenuated the –15 dB signal by 1 dB; the InSignalLevel is –17 dB due to the input gain of –1**
- **At this point our real ERL is 2dB, however the Echo Canceller still does not acknowledge the input signal as echo**

Adjusting Signal Strength in Cisco IOS

If We Configure 2 dB of Attenuation in Each Direction as Follows:

```
voice-port 1/1:23
 input gain -2
 output attenuation 2
```

The Resulting Levels Are as Follows:

```
Gateway# sh call active voice
- snip -
OutSignalLevel=-17
InSignalLevel=-19
ERLLevel=4
-snip -
```

- **Notice the OutSignalLevel is -17 because we attenuated the –15 dB signal by 2 dB; the InSignalLevel is –19 dB due to the input gain of –2**
- **Our expected ERL of 4dB is now correct**

Measuring Echo in Cisco IOS

If We Configure 2 dB of Attenuation in Each Direction as Follows:

```
voice-port 1/1:23
 input gain -2
 output attenuation 2
```

The Resulting Levels Are as Follows:

```
Gateway#sh call active voice
- snip -
OutSignalLevel=-17
InSignalLevel=-19
ERLLevel=4
-snip -
```

- **Notice the OutSignalLevel is –17 because we attenuated the –15 dB signal by 2 dB; the InSignalLevel is –19 dB due to the input gain of –2**
- **Our expected ERL of 4dB is now correct**

Measuring and Adjusting Echo in VISM

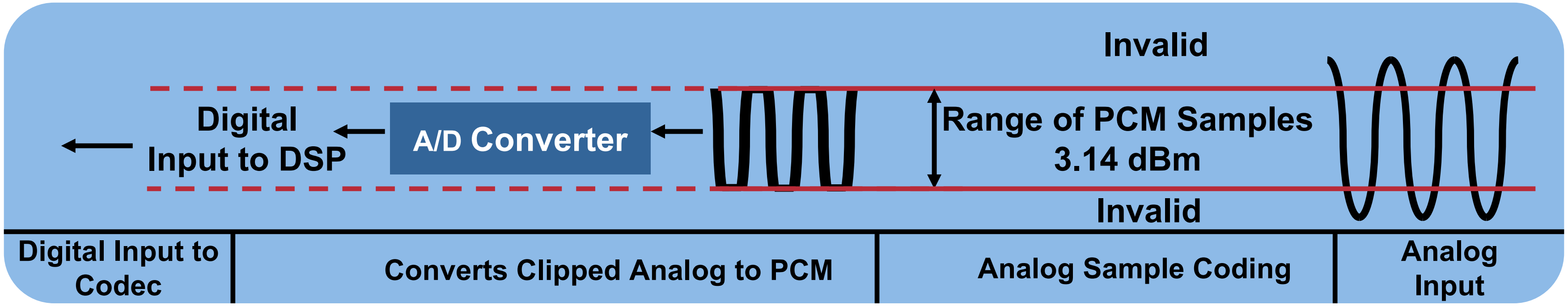
If We Configure 2 dB of Attenuation in Each Direction as Follows:

```
VISM8.a > dsplngain 1
  LineNo/Ds0No        Input Gain      Output Attenuation
  ------------        ----------      ------------------
     1/ 1                 0                    0
     1/ 2                 0                    0
```

```
VISM8.a > cnflngain
ERR : incorrect number of parameters (not enough)
Syntax : cnflngain "line_number input_gain output_attenuation"
         line_number  -- values: 1 - 8.
         input_gain -- Value: -6..14 (dB)
         output_attenuation -- Value: 0..14 (dB)
```

```
VISM8.a > cnfecantail
ERR : incorrect number of parameters (not enough)
Syntax : cnfecantail "lineNum maximumTail"
         line_number  -- values: 1 - 8.
         Maximum TAIL -- Values: 24, 32, 48, 64, 80, 96, 112 and 128 millisecs
        possible errors are :
        a) Incorrect number of parameters
        b) Illegal
```

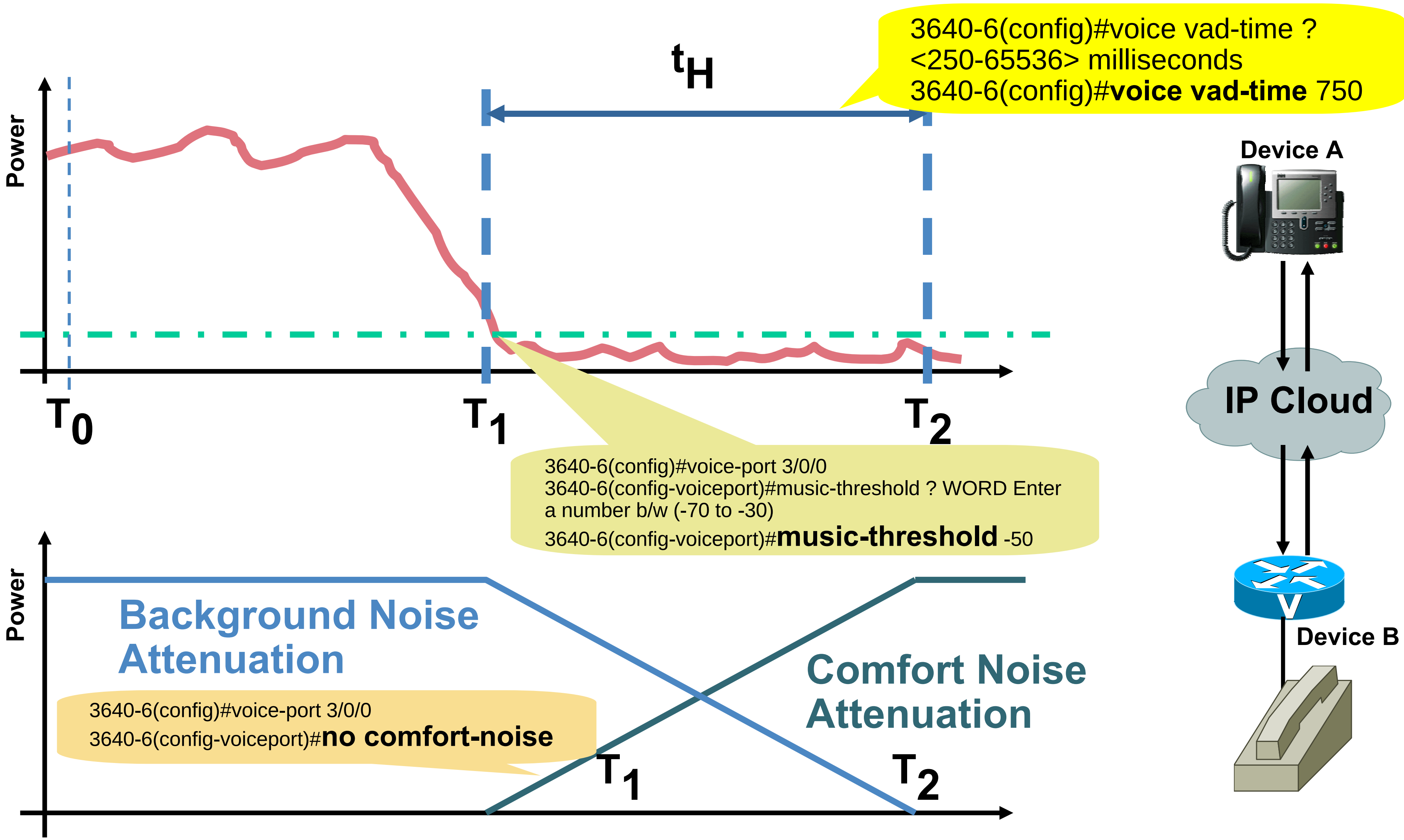
If the Input Gain Is Too High



- If the gain is to high, the analog sample is out of the acceptable PCM range, the results are unpredictable:
  - Nailed to the rail
  - Original sample
  - Silence code
- This will result in both
  - Confusion in the voice coder
  - Distortion at the receiving end
  - Sounds like fuzzy, distorted, clipped syllables

Comfort Noise and VAD

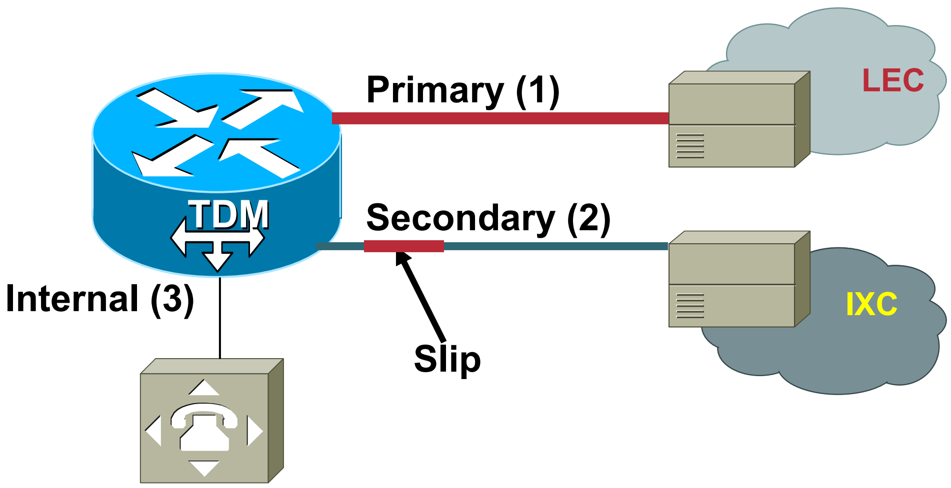
Troubleshooting Hissing, Static, Clipping



Synchronization

- Not all gateways have independent PLL circuitry
- PBX integration requires clock relay
- L2 parameters must match with SP

Synchronization
Troubleshooting Clicking Sound




```
Branch2-3745# show controller t1 3/0
T1 3/0 is up.
  Applique type is Channelized T1
  Cablelength is long gain36 0db
  Transmitter is sending remote alarm.
  Receiver has loss of frame.
  alarm-trigger is not set
  Version info Firmware: 20040202, FPGA: 11
  Framing is ESF, Line Code is B8ZS, Clock Source is Line.
   Current port master clock:local osc on this network module
  Data in current interval (103 seconds elapsed):
     0 Line Code Violations, 0 Path Code Violations
     398 Slip Secs, 0 Fr Loss Secs, 0 Line Err Secs, 0 Degraded Mins
     0 Errored Secs, 0 Bursty Err Secs, 0 Severely Err Secs, 103 Unavail Secs
```

Drop, Delay, and Jitter

Troubleshooting Garbled, Synthetic, Choppy Voice

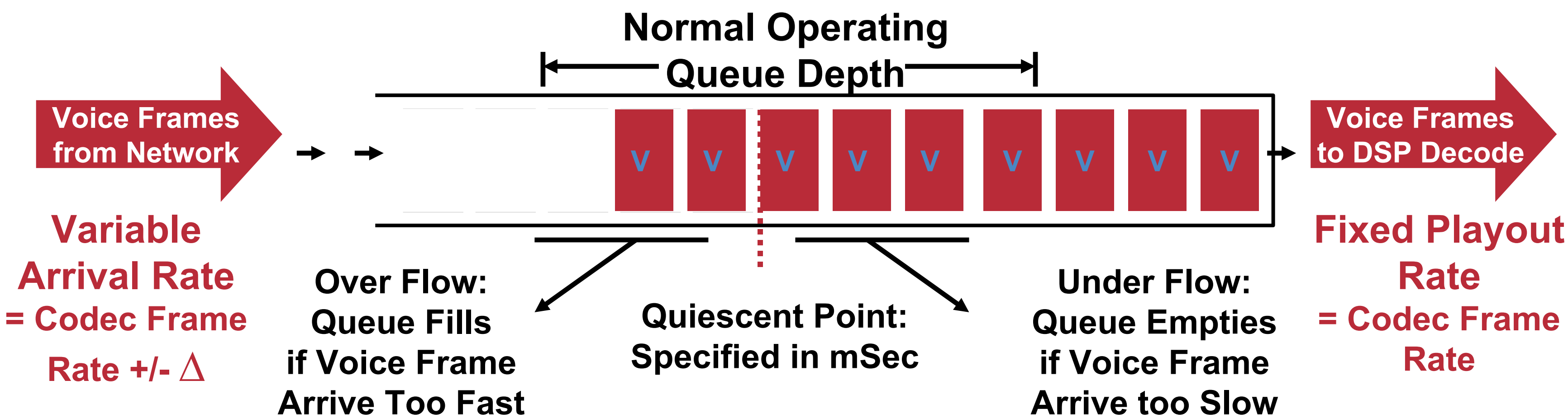
- **LatePackets**: The number of packets arriving outside the de-jitter buffer playback delay period; these packets are discarded
- **LostPackets**: The number of packets that never arrive at the receiving IP phone or gateway
- **GapFillWithPrediction**: The amount of packet prediction in a call; divide this number by the packet sample time to determine the number of packets affected
- **GapFillWithSilence**: Silence is played out in the following situations:

  When a packet is lost and there is no audio sample available to play; for example, when two or more packets are lost in sequence; this situation may result in an audible click or gap being heard by the user

  When the playout buffer is adapting to a larger value by inserting silence between buffered voice packets; this situation does not result in an audible loss in quality
- **HiWaterPlayoutDelay**: First-In, First-Out (FIFO) jitter buffer high mark indicating the maximum depth to which the de-jitter buffer has adapted for this call
- **LoWaterPlayoutDelay**: FIFO jitter buffer low mark indicating the minimum depth to which the de-jitter buffer has adapted for this call
- **ReceiveDelay**: Current playout FIFO delay plus decoder delay for the call

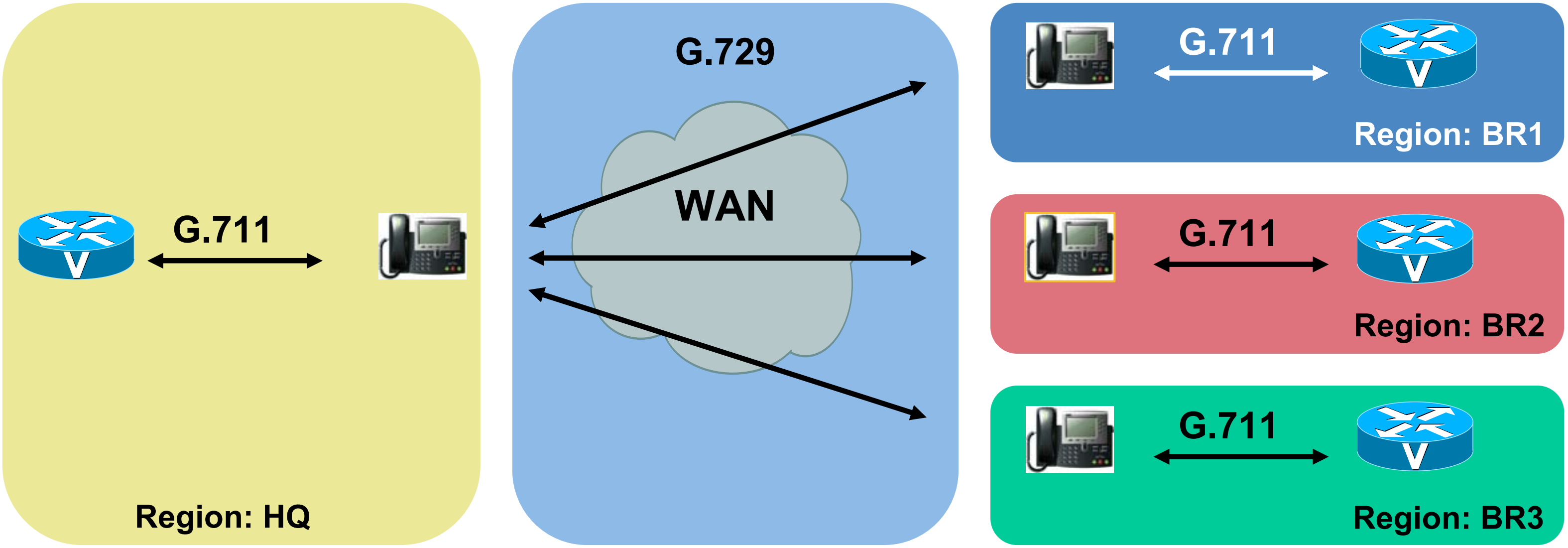
Proper QoS Planning and Implementation Is the Solution

De-Jitter Buffer Adjustment



- **If you're getting underwater voice, or gaps in speech, try**
  - Monitoring the de-jitter buffer for over/under flow
  - Increase the size of the de-jitter buffer
    - Pro: Accommodates larger fluctuations in delay variability
    - Con: Adds to overall end-to-end delay
  - Check network for proper operation/configuration
    - You may have excessive delay in the network due to bursting above CIR and network discards

Compression Methods—Keep Consistency




```
Router# show call active voice
[snip]
LatePackets=0
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
SignalingType=ext-signal
[snip]
```

iLBC (Internet Low Bit Rate) Codec

- Royalty-free
- Low bit-rate
- Narrowband
- Speech quality equal to or better than G.729A
- High complexity codec
- Good robustness in high packet loss environments
- Architecturally designed for packet networks (whereas the CELP codecs such as G.729A and G.723 were designed for circuit networks)

iLBC Codec

- **Standardization**

  iLBC codec: Experimental RFC3951

  SIP: Experimental RFC3952 (RTP Payload format)

  H.323: H245 Version 12, Annex S (compliant with RFC3952)

  CableLabs designated iLBC as mandatory for all PacketCable 1.5 products, recommended for PacketCable 2.0

  Significant momentum in internet telephony for Skype-out applications

- **Modes**

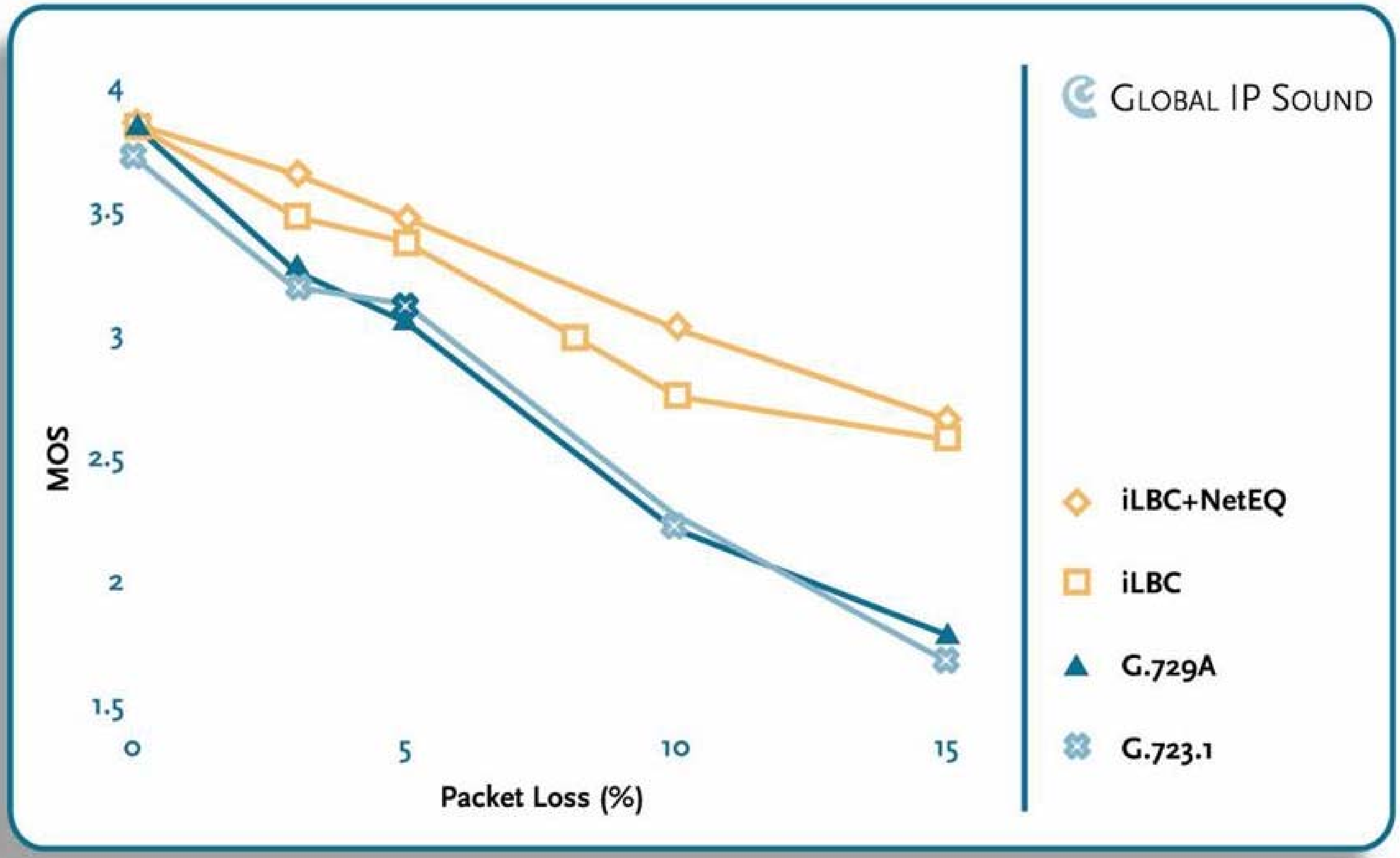
  20ms and 30ms mode

  SIP maxptime is 120ms

  The default dynamic payload type value is 116

iLBC Mode	IP BW	RSVP BW	Payload/Frame	Max Frames per Packet
20ms	15.2K	24K	38 bytes	6
30ms	13.3K	20K	50 bytes	4

iLBC and G.729A Comparison



The tests were performed by Dynstat, Inc., an independent test laboratory.
Score system range: 1 = bad, 2 = poor, 3 = fair, 4 = good, 5 = excellent

Fax Relay Troubleshooting (1/2)

- **Verify normal voice calls complete**
- **Verify correct dial peer is being matched**

  `Show call active voice brief`

  Verify dial peers are correctly configured

  Fax relay is disabled while a low bandwidth codec has been in use

  One side is configured with fax relay but other side is set for T.38 (AS5350/5400 only support T.38) otherwise the negotiation will fail

  Default dial peer is being used inbound on the terminating gateway and these do not match with the outbound dial peer on originating gateway

- **Verify the fax machine works correctly over PSTN lines**
- **Verify error on digital T1/E1 controllers and packet drops over IP network**

  `Show controller  T1/E1`

  `Show interface <interface number>`

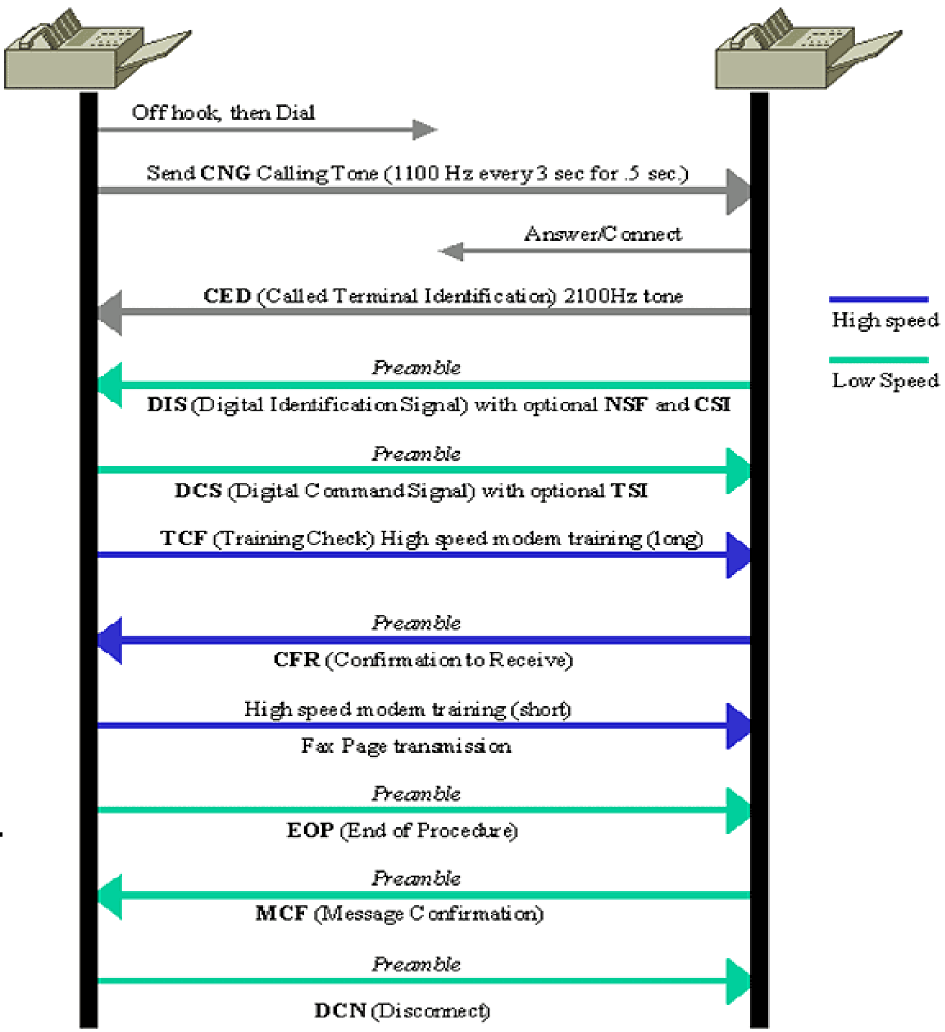
Fax Relay Troubleshooting (2/2)

- **Verify that fax passthrough works**

```
voice-port 2/1:23
  no echo-cancel enable
dial-peer voice 3
  fax rate disbale
  Codec g711ulaw
  no vad
```

- **Troubleshooting**

```
debug fax relay t30 all
```



References

- **Echo**

  http://www.cisco.com/univercd/cc/td/doc/cisintwk/intsolns/voipsol/ea_isd.htm#91601

- **Voice Quality Degradation Symptoms**

  http://www.cisco.com/warp/public/788/voice-qos/symptoms.html#clip

- **Quality of Service**

  http://www.cisco.com/warp/public/732/Tech/qos/

- **IP SLA**

  http://www.cisco.com/en/US/products/sw/iosswrel/ps5207/products_configuration_guide_book09186a00802b2a6c.html

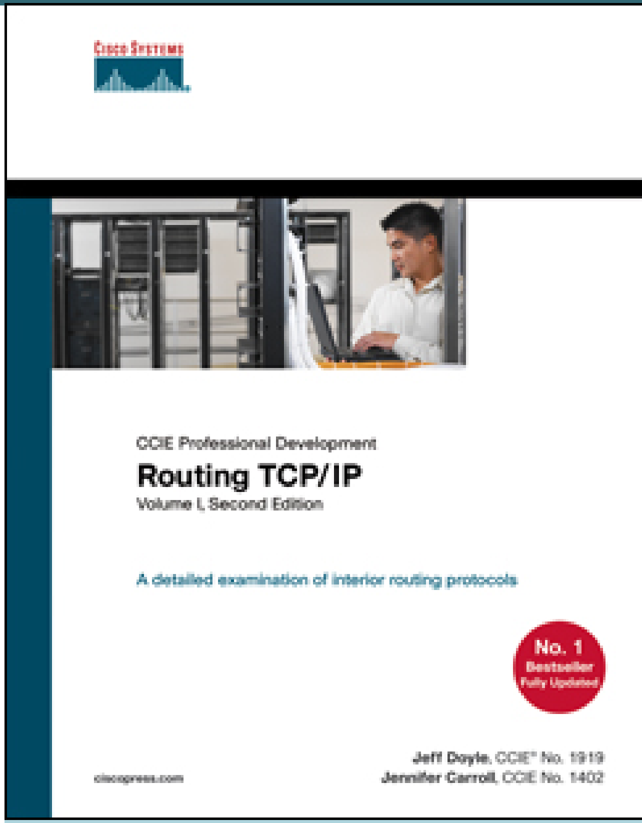
- **VoIP Troubleshooting Using "show call active voice"**

  http://www.cisco.com/warp/public/788/voip/show_call_act_voice.html

Q and A

Recommended Reading

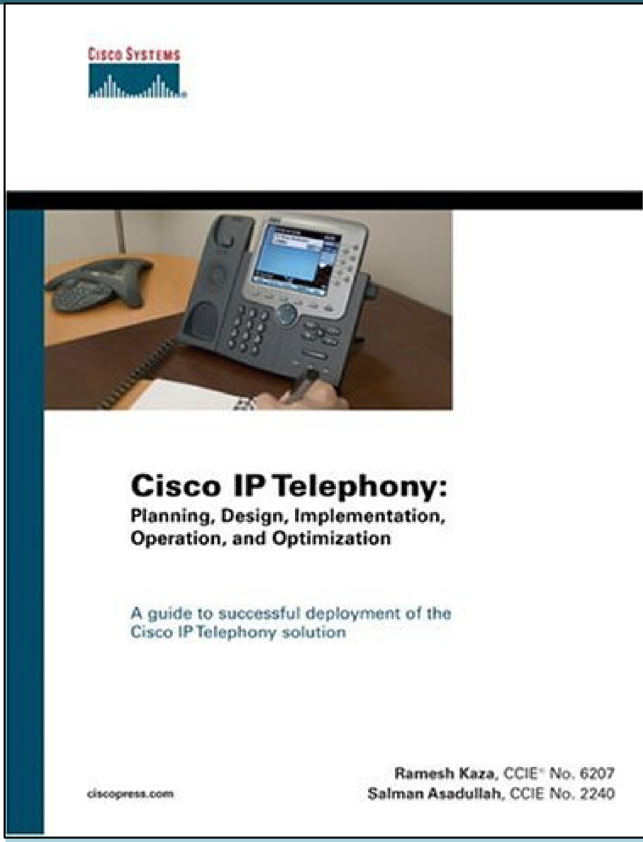
- Continue your Cisco Networkers learning experience with further reading from Cisco Press®
- Check the Recommended Reading flyer for suggested books



Available Onsite at the Cisco Company Store

Recommended Reading

- **Cisco IP Telephony: Planning, Design, Implementation, Operation, and Optimization**
  ISBN: 1587051575



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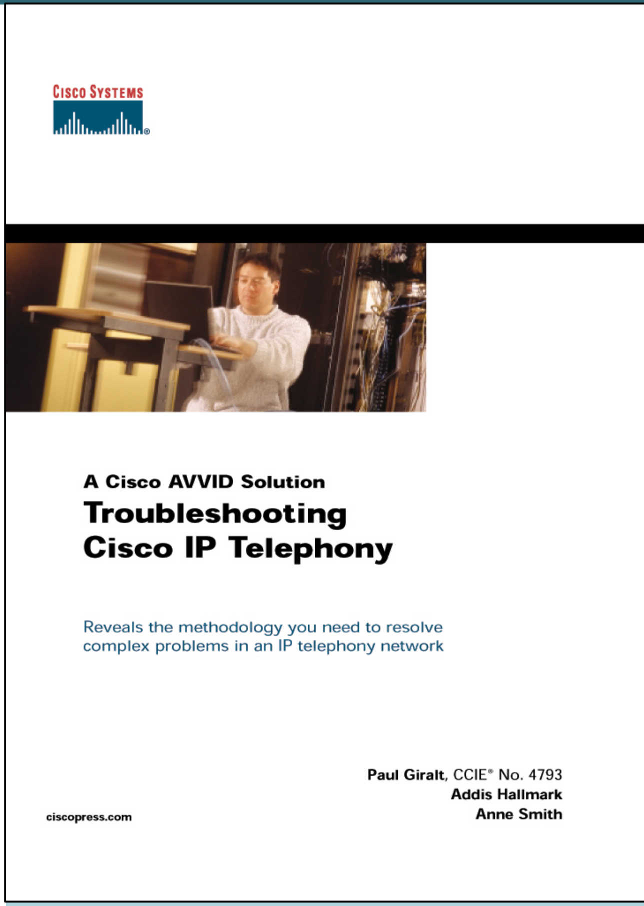
Recommended Reading

- **Troubleshooting Cisco IP Telephony**
  ISBN 1-58705-075-7



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