Cisco Expo 2007

Managing Voice Quality in Converged IP Networks

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Enable Your Network
Empower Your Business

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

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
- This is typically any noise on the line or in a voicemail message in addition to the voice signal. Noise will typically leave the conversation intelligible but still far from excellent. Static, hum, crosstalk, and intermittent popping tones are examples where the calling and called parties can understand each other, but with some effort. Some noises are so severe that the voice becomes unintelligible. One such example among the samples provided in this document is one described as a motor sound.
- Voice distortion
- This is typically any problem that affects the voice itself. This category was further divided as follows.
  - Echoed voice - Echo is where the voice signal is repeated on the line. It can be heard at either end of the call, in varying degrees and with many combinations of delay and loss within the echoed signal.
  - Garbled voice - A garbled voice signal is one where the actual character of the voice is altered to a significant degree and often has a fluctuating quality. On some occasions the voice becomes unintelligible.
  - Volume distortion - Volume distortion problems are associated with incorrect volume levels, whether constant or in flux.
- **Note:** The categorization of the symptoms is to a large degree dependent on the severity of the symptom, perceptual factors and cultural factors. Therefore, the placement and grouping of symptoms within categories is in many cases arguable. In addition, there can be situations where the categories will overlap, for example static on the line may cause some form of voice distortion. This is a best attempt to give some structure to these terms and define the vocabulary.

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

- **Absolute Silence**
- Symptom—This type of silence between speech can be understood if you have ever had the experience of not knowing whether the other person is still there because there is no sound on the line.
- Cause—A common cause for this problem is Voice Activity Detection (VAD) without comfort noise. To experience this symptom, usually the background noise is loud enough for the silence insertion to be noticeable but soft enough so that VAD will be engaged.

- **Clicking**
- Symptom—Clicking is an external sound similar to a knock that is inserted usually at intervals.
- Cause—A common cause is clock slips or other digital errors.

- **Crackling**
- Symptom—Crackling is an irregular form of very light static, similar to the sound a fire makes.
- Cause—A common cause is poor electrical connections, in particular poor cable connections. Other causes are electrical interference and a defective power supply on the phone.

- **Crosstalk**
- Symptom—Crosstalk is a familiar concept where you can hear someone else's conversation on the line. Commonly the other parties cannot hear you. There are also forms of crosstalk where all parties can hear each other.
- Cause—Wires in close proximity, where the signal of one is induced into the other, is a common cause of this problem.

Noise (Cont.)

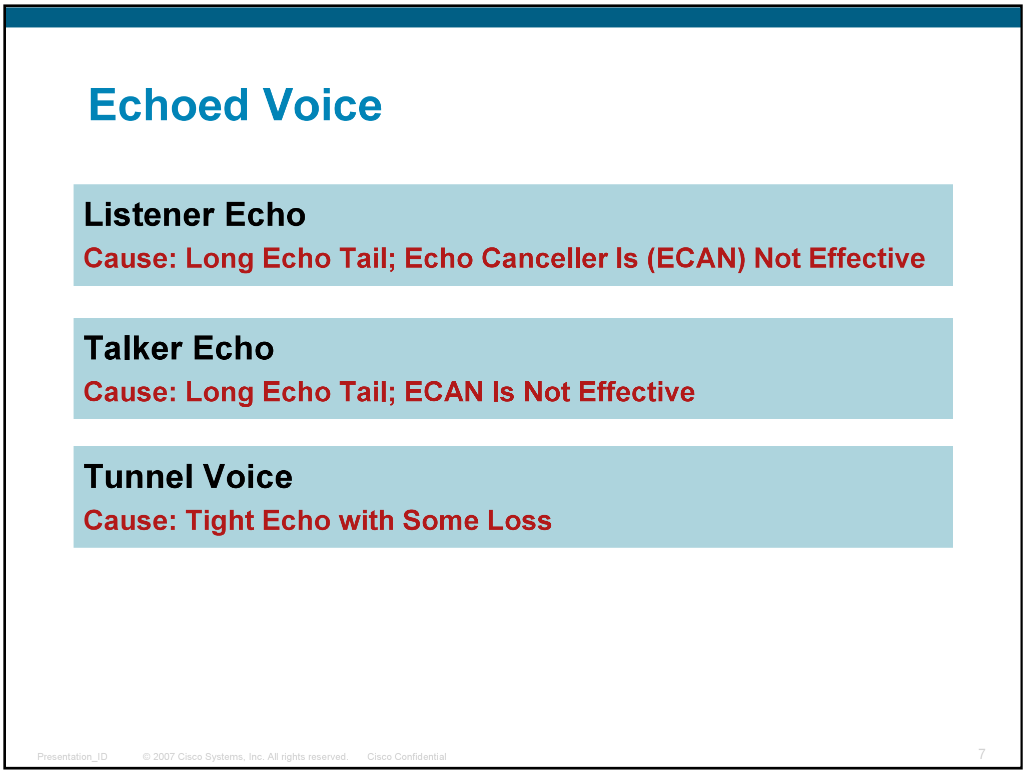
Hissing
Cause: VAD

Static
Cause: Codec Mismatch; Enhanced by VAD

- **Crackling**
- Symptom—Crackling is an irregular form of very light static, similar to the sound a fire makes.
- Cause—A common cause is poor electrical connections, in particular poor cable connections. Other causes are electrical interference and a defective power supply on the phone.

- **Hissing**
- Symptom—Hissing is more driven and constant than static. White noise is a term often associated with strong hissing. Pink noise is a less constant hissing noise and brown noise even less constant still.
- Cause—A common cause of hissing is VAD.

- **Static**
- Symptom—Severe static is an example of static that, in addition to creating background noise, affects the dial and ring tones and the voice itself. Another name for this symptom might be scratchy or gravel voice.
- Cause—A common cause is A-law/Mu-law codec mismatch. For example, Compand-type A-law mistakenly added to an analog voice port.
- http://www.cisco.com/warp/public/788/voice-qos/symptoms.html#clip



- **Listener Echo**
- Symptom—Listener and talker echo sound similar although the signal strength of listener echo may be lower. The essential difference between them is who hears the echo and where it is produced. Listener echo is the component of the talker echo that leaks through the near-end hybrid and returns again to the listener causing a delayed softer echo. The listener hears the talker twice.
- Cause—Common causes are:
- Insufficient loss of the echo signal.
- Long echo tail.
- Echo cancellers in the gateway adjacent to the near-end hybrid not activating.

- **Talker Echo**
- Symptom—Talker echo is the signal which leaks in the far-end hybrid and returns to the sender (talker). The talker hears an echo of his own voice.
- Cause—Common causes are:
- Insufficient loss of the echo signal.
- Echo cancellers in the gateway adjacent to the far-end hybrid not activating.
- Acoustic echo caused by the listener's phone.

- **Tunnel Voice**
- Symptom—Tunnel voice is similar to talking in a tunnel or on a poor quality mobile phone car kit.
- Cause— A common cause is tight echo with some loss. For example, 10 ms delay and 50 percent loss on the echo signal.

Garbled Voice

Choppy Voice

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

Synthetic (Robotic) 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

- **Choppy Voice**
- Symptom—Choppy voice describes the sound when there are gaps in the voice. Syllables appear to be dropped or badly delayed in a start and stop fashion.
- *Note:* Other terms used to describe this sound are "clipped voice" or "broken voice."
- Cause—Common causes are consecutive packets being lost or excessively delayed such that DSP predictive insertion cannot be used and silence is inserted instead. For example, delay inserted into a call through contention caused by a large data packets.

- **Synthetic Voice**
- Symptom—The term "synthetic" means that the sound of the voice is artificial and with a quiver or fuzz. Predictive insertion causes this synthetic sound by replacing the sound lost when a packet is dropped with a best guess from a previous sample. Synthetic and choppy voice commonly occur together.
- Cause—A common cause is single packet loss or delay beyond the bounds of the de-jitter buffer playout period. DSP predictive insertion causes the synthetic quality of the voice. For example when a call was provided insufficient bandwidth (such as G711 codec across 64Kbps).

- **Underwater Voice**
- Symptom—Unintelligible underwater voice describes a distortion that makes it impossible to understand the voice. Descriptions of this sound include the sound of a cassette tape being fast forwarded, a gulping sound, and a wishy-washy sound.
- 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)

- **Fuzzy Voice**
- Symptom—Fuzzy voice sounds similar to the radio being turned up too loud and the voice is shaky. This may only occur at certain signal levels within the sentence depending on the level of gain applied.
- Cause—This is often caused by too much gain on the signal, possibly introduced at one of a number of points in the network. For example, the signal may be overdriven from the PBX or high gain through the Cisco Unity Tag-switched Path (TSP) setting.

- **Muffled Voice**
- Symptom—Muffled voice sounds similar to speaking with your hand over your mouth.
- Cause—A common cause is an 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—Soft voice is usually caused by too much attenuation on signal possibly introduced at one of a number of points in the network (such as voice gateway when trying to reduce echo or Cisco Unity AGC settings for 3.1(3)).

- **Tinny Voice**
- Symptom—Tinny voice is similar to listening to an old-fashioned wireless

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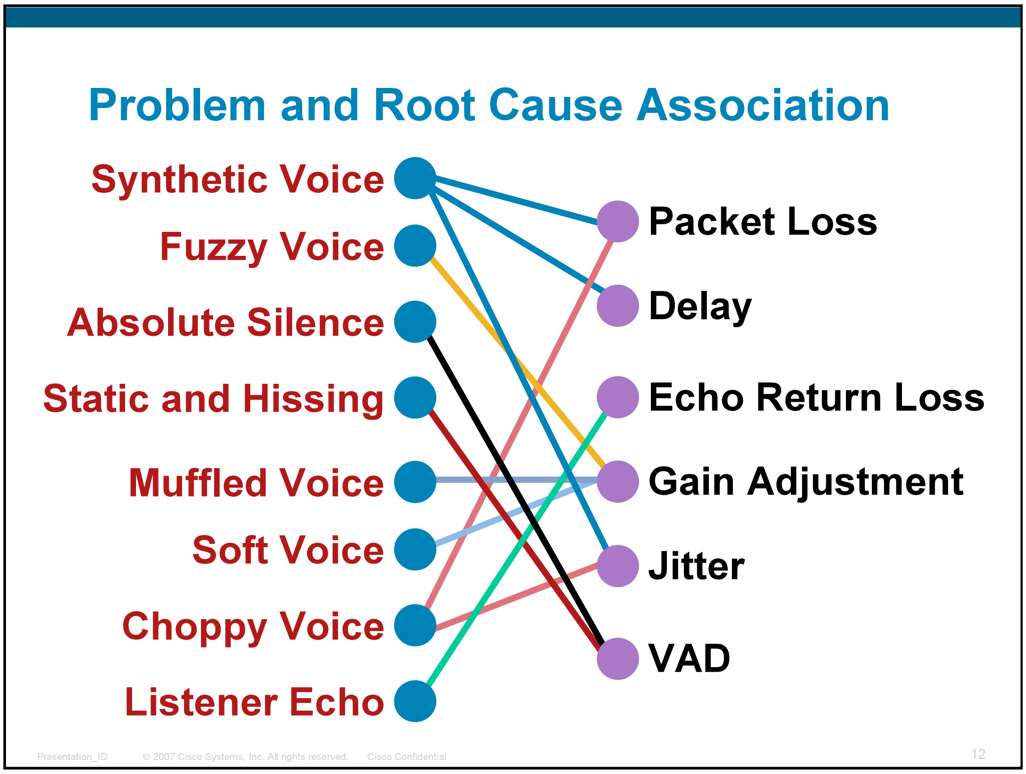
Classifying Voice Quality Attributes to the Root Cause



Problem and Root Cause Association
Synthetic Voice
Fuzzy Voice
Absolute Silence
Static and Hissing
Muffled Voice
Soft Voice
Choppy Voice
Listener Echo
Packet Loss
Delay
Echo Return Loss
Gain Adjustment
Jitter
VAD

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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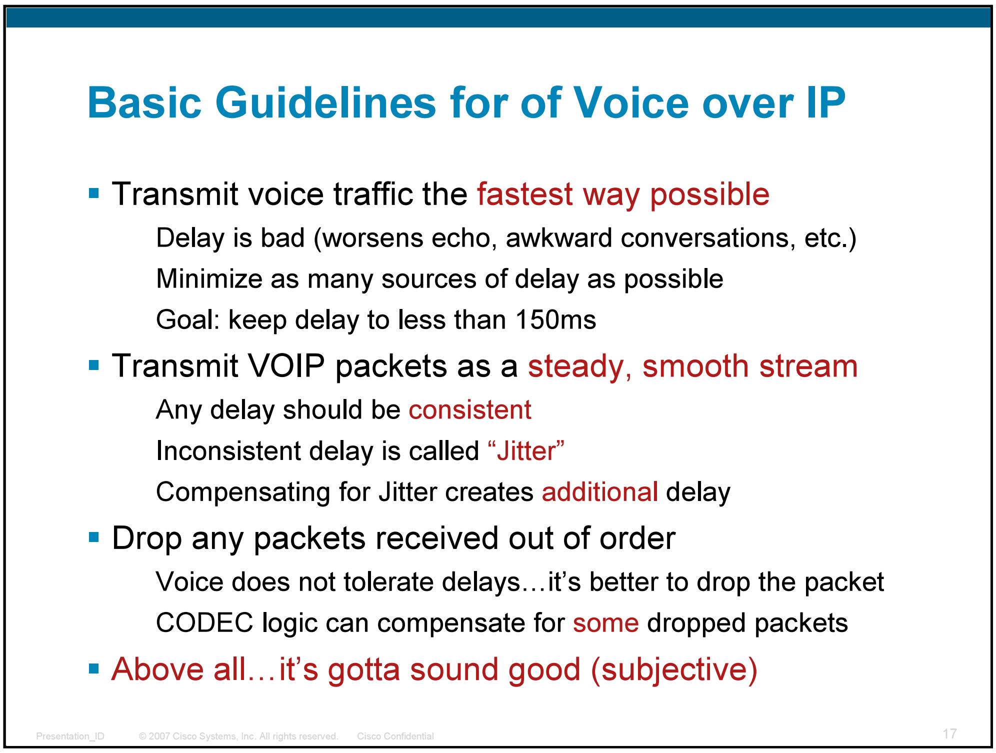
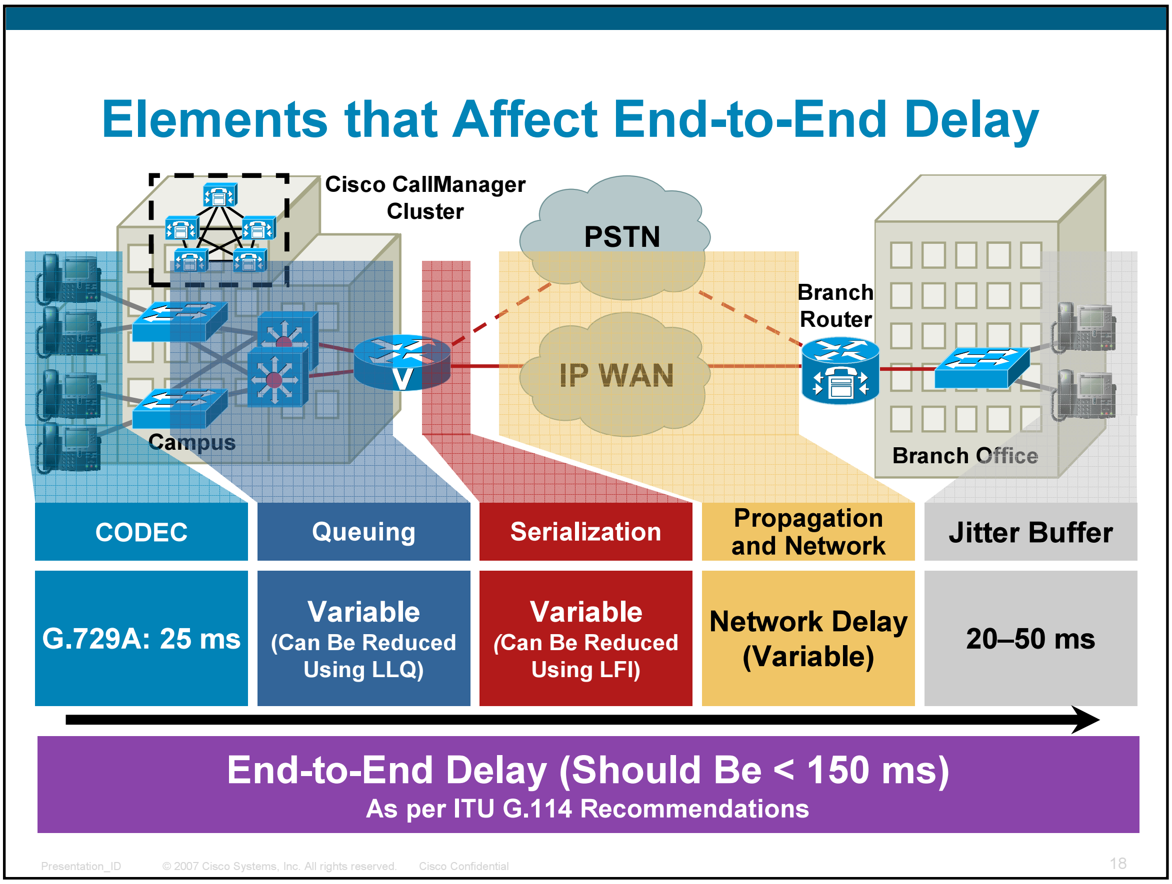
Address Voice Quality by Implementing Proper QoS

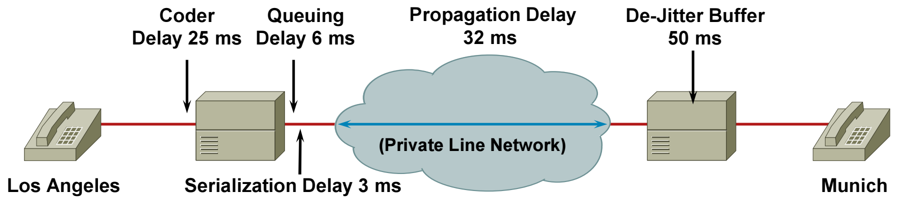


- Loss must be constrained to less than 1% packet loss to keep from affecting quality.



- **6.3 μs/Km + Network Delay (Variable)**

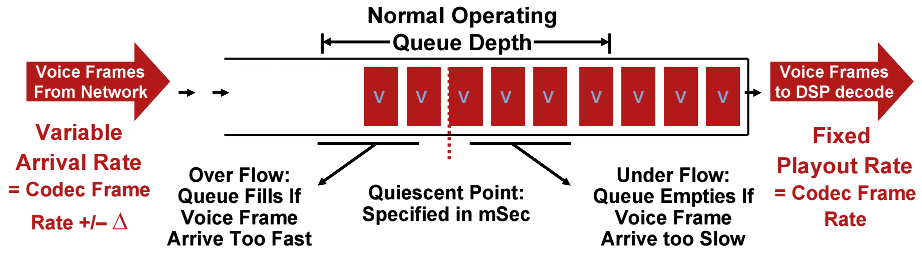
Calculate Delay Budget



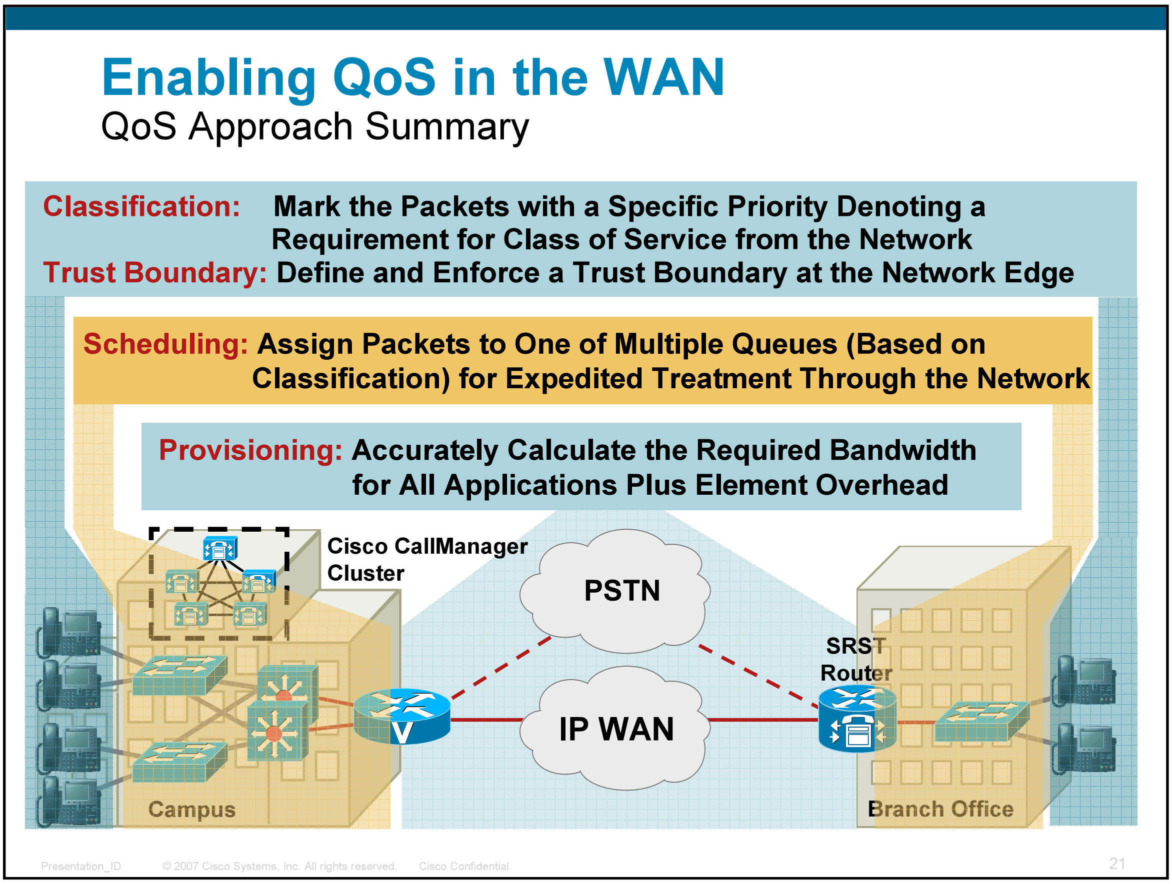
	Fixed Delay	Variable Delay
Coder Delay G.729 (5 msec Look Ahead)	5 msec	
Coder Delay G.729 (10 msec per Frame)	20 msec	
Packetization Delay—Included in Coder Delay		
Queuing Delay 64 kbps Trunk		~ 6 msec
Serialization Delay 64 kbps Trunk	3 msec	
Propagation Delay (Private Lines)	32 msec	
Network Delay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

Goal:
Less than 150 ms 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
Cisco CallManager Cluster
PSTN
IP WAN
SRST Router
Campus
Branch Office
Presentation_ID
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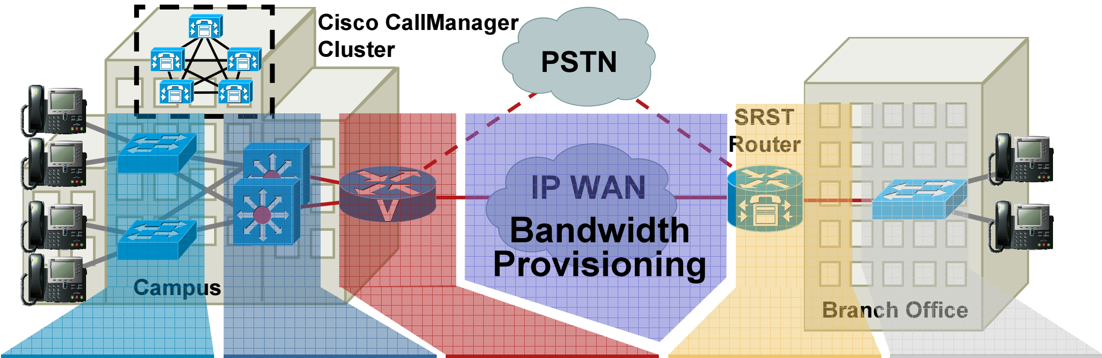
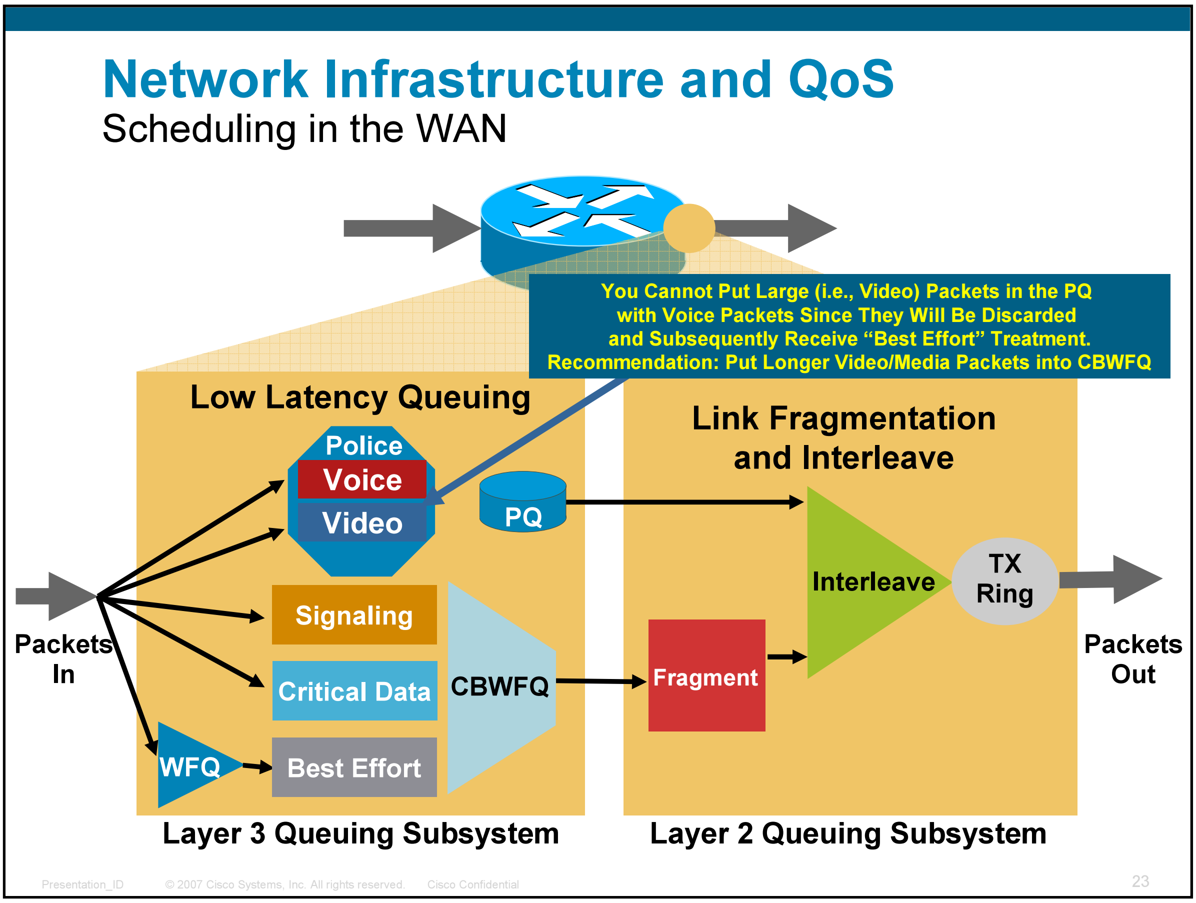
Enabling QoS in the WAN

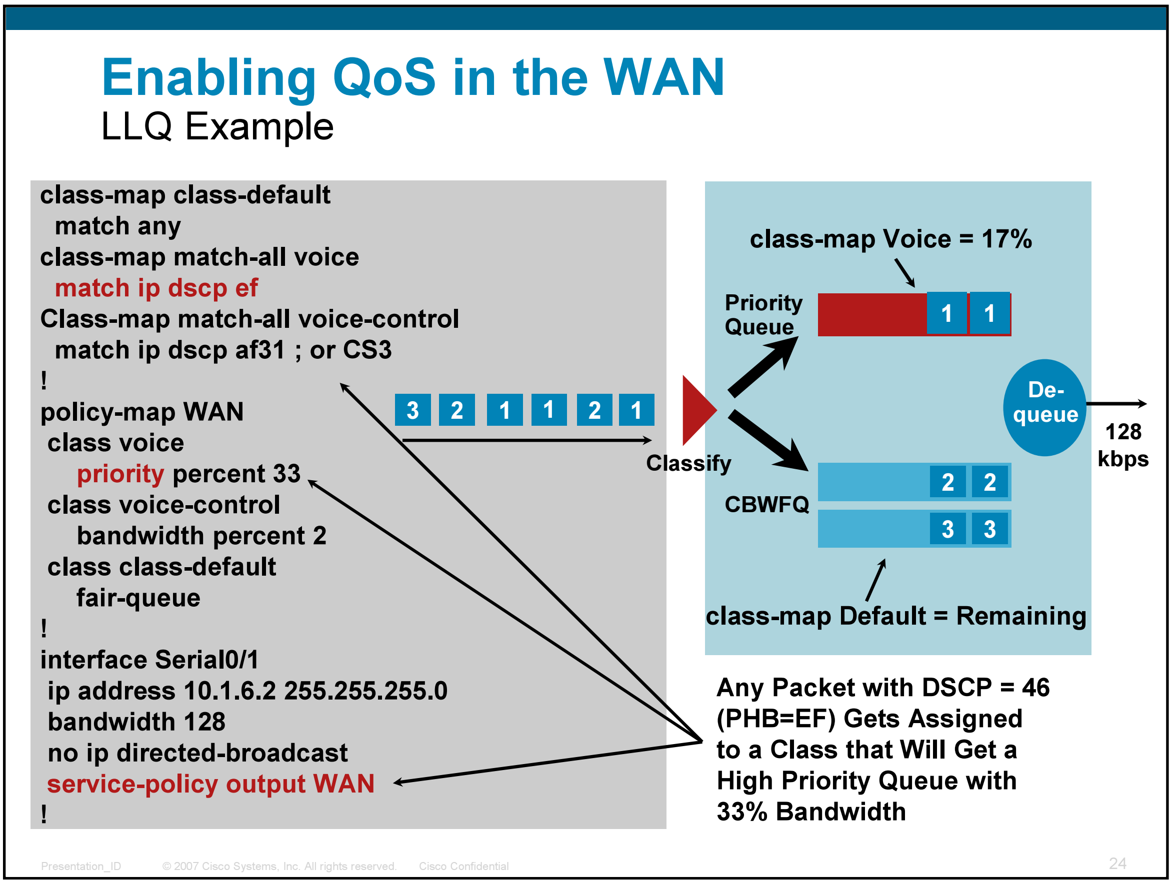
Overall QoS Design Summary



Campus Access	Campus Distribution	WAN Aggregation	Branch Router	Branch Switch
▪ Inline power ▪ Multiple Qs ▪ 802.1p/Q ▪ Fast link convergence	▪ Multiple Qs ▪ 802.1p/Q ▪ Classification ▪ Reclassification	▪ Multiple Qs ▪ 802.1p/Q ▪ Traffic shaping ▪ Link efficiency (LFI, cRTP) ▪ Classification ▪ Reclassification	▪ Multiple Qs ▪ 802.1p/Q ▪ Link efficiency (LFI, cRTP) ▪ Classification ▪ Reclassification	▪ Inline power ▪ Multiple Qs ▪ 802.1p/Q



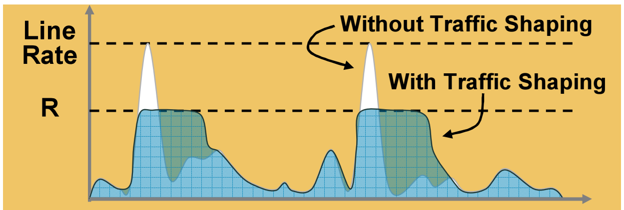
- Misleading! When you have Video, we don't recommend running at below 768Kbps, so there is no need for LFI!!!



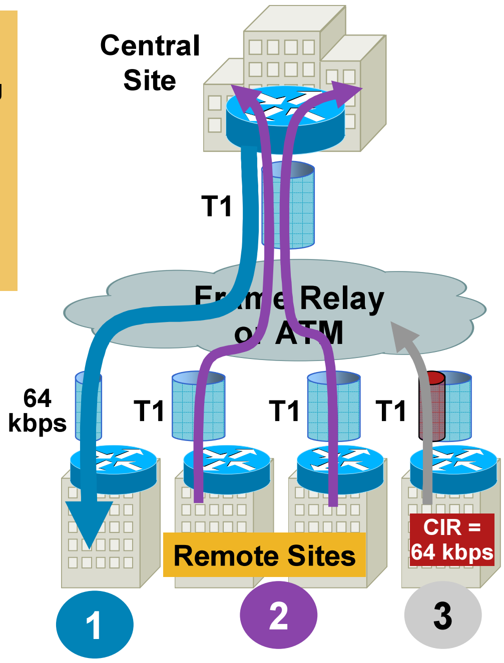
- CQ vs CBWFQ:
- specify the actual rates (Kbps) for CBWFQ; CQ specify the #bytes to send (extra math for config)
- CQ 16 queues; CBWFQ 64
- WRED not supported with CQ; it is with CBWFQ
- RSVP only supported with CBWFQ

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 'sustainable rate'

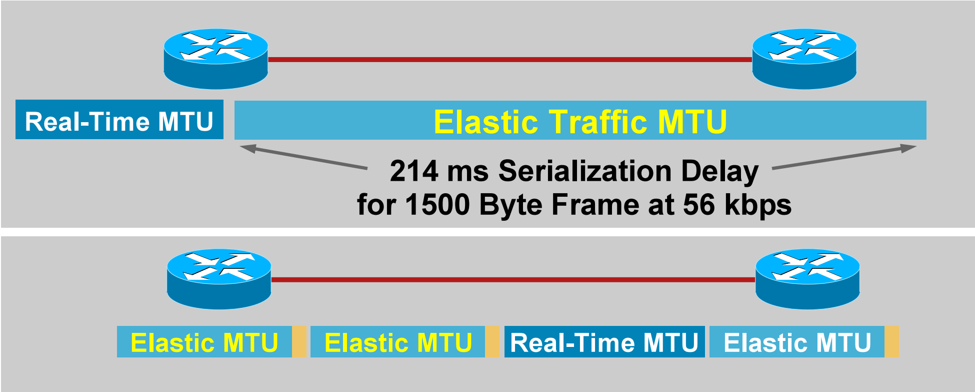
Enabling QoS in the WAN

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

	20 +	8 +	12 +	20	= 60 Bytes
No cRTP:	IP	UDP	RTP	Payload	
With cRTP:			cRTP	Payload	
			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
- 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

- Networkers 2008:

  http://www.cisco.com/web/europe/cisco-networkers/2008/index.html

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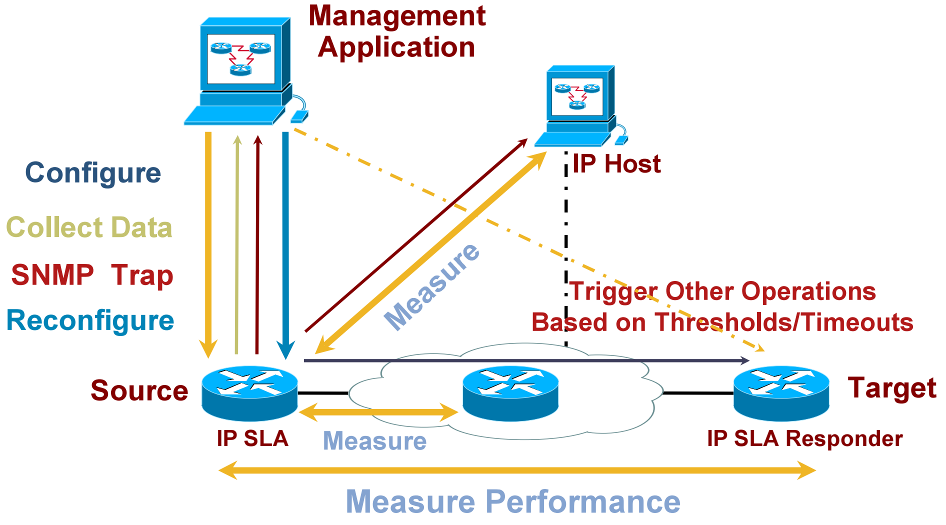
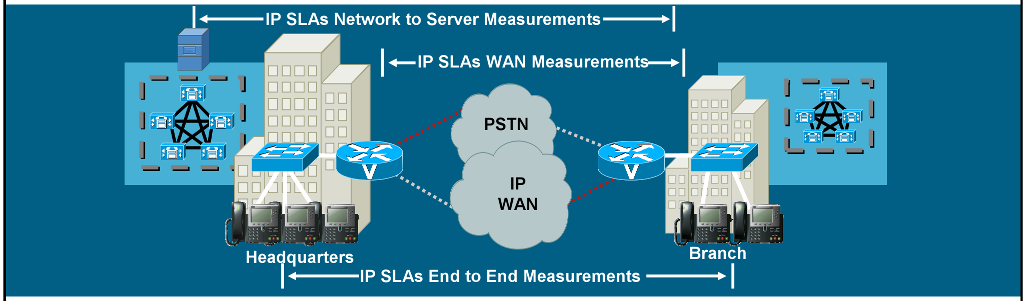
Proactive Planning
IP SLA

How Does Cisco IP SLA Work?

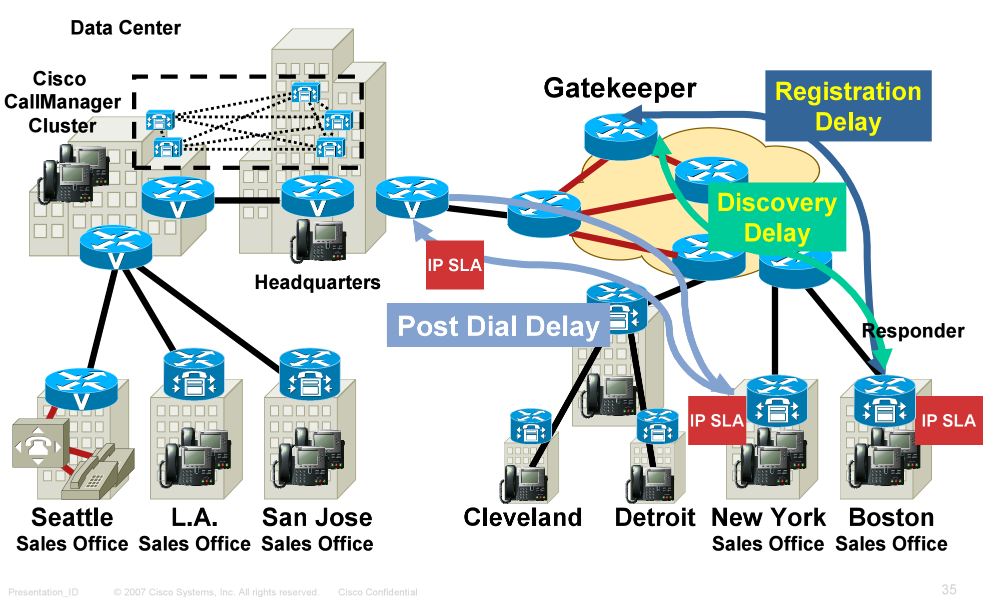


Cisco IOS IP SLA for VoIP

- 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

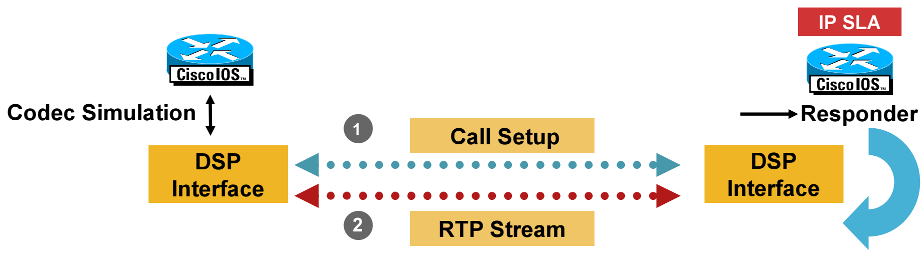


IP SLA VoIP Measurements



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)
- VoIP statistics from Cisco AS5XXX, 2600, 2800, 3600, 3700, 3800, and 7200 Series



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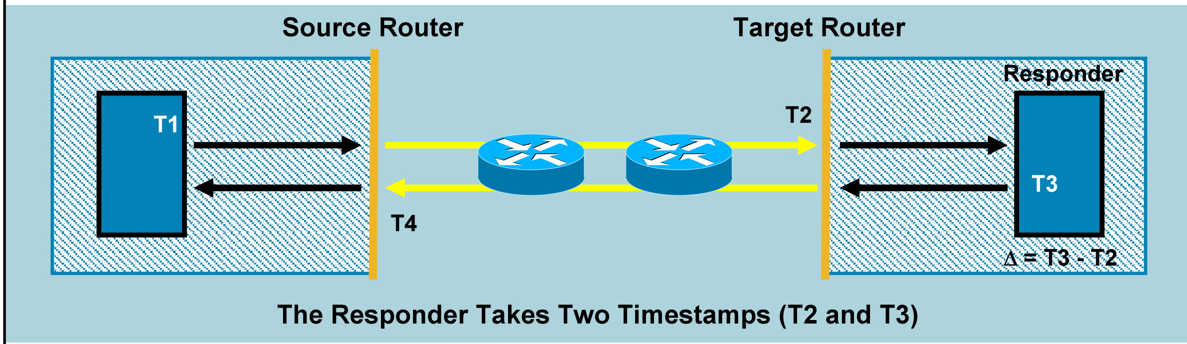
- Key messages:
- A summary of the services we've looked at—and where in the network they would be most useful
- Availability must be end-to-end but NOT necessarily the same functionality in each part of the network.
- Different needs within the box, between ENT and SP, within SP
- Notice certain features span multiple demarcations – fast convergence; NSF-awareness and NSF/SSO work together
- This is particularly what gives us the ability to protect you end-to-end the Network of Networks
- The services that ease operations are also useful across the board, as can be expected

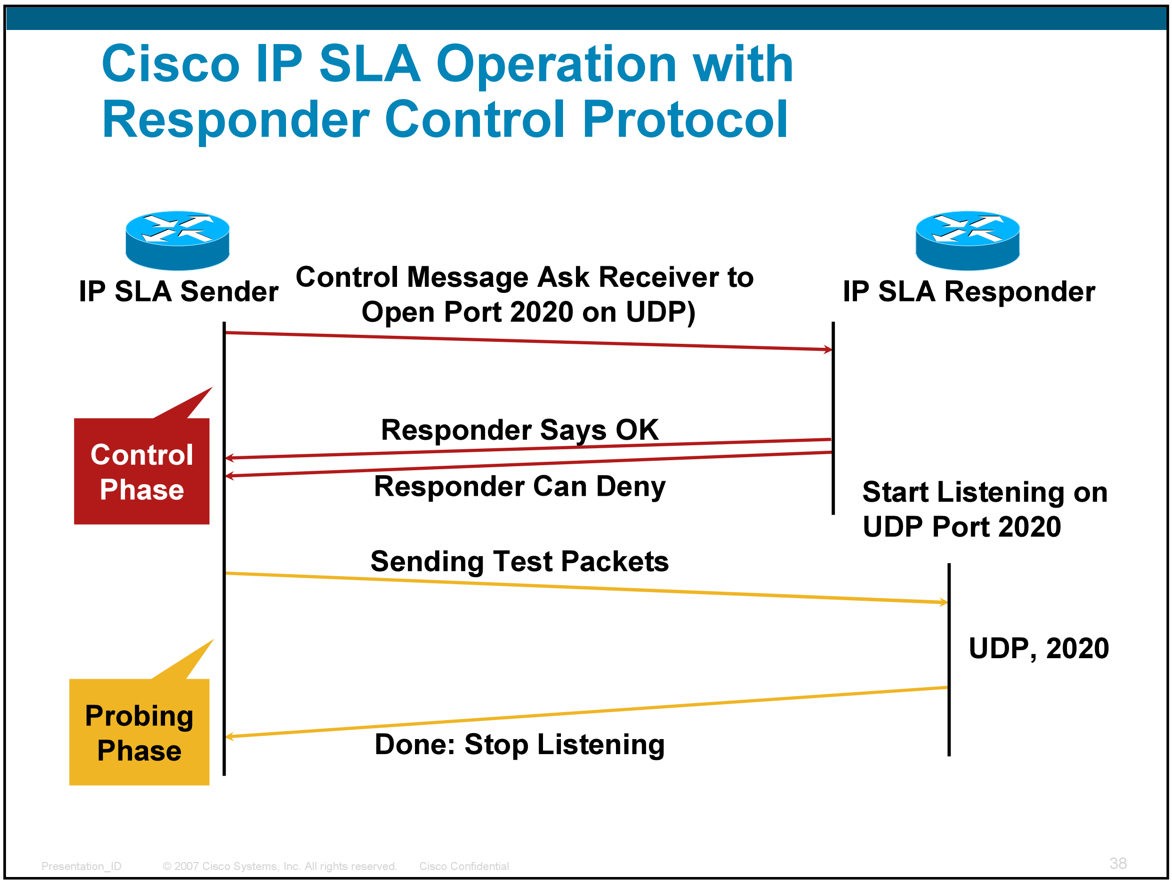
Responder



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

- **Control Sequence Open up UDP ports.**
- IP SLA uses a control message on UDP port 1967 but the responder replies back to the control port on another UDP port number. Don't know what it is but it is a high port number.
- The control process is as follows. IP SLA sends control with type, and udp port and duration of probes, for this cycle.
- Responder Ack control probe but on another port. Don't know what the number is but it is high.
- Responder then opens up udp port for the duration as instructed in control packet.
- Responder received probes and then closes.
- Responder is acknowledging probes as it receives them.

- **IP SLA Response Calculation**
- Source Router IP SLA processing delay =T5-T4
- Responder processing delay = T3-T2
- RTT is T5-T1 minus IP SLA and responder processing delays.
- RTT = T5-(t1-dS+dT)
- dT = Delta on Target Router, dS is delta on source.
- Points to make about this slide.
- IP SLA probe enters output interface queues as any other packet if you set precedent bits is will go to the back of the appropriate queue.



- The responder, based on the type of operation, put timestamps on the return packets for accuracy.
- The Source Router computes all the response time measurements.
- Uses UDP, default port is 1967 but is configurable >

- is the port number, 1967, used in the control protocol between source and
- > target router (responder enabled) possible to configure? if so how?
- No, it's fixed.

- > acl needs to be open for that port in the router,
- > that could be a problem for some customers,

- True - but having the ability to change the control protocol port number
- will not change anything. Access-lists still needs to be open.

- On the other side, you can restrict on the destination router who can
- access that port. This is what I recommended in my presentation.
- E

VoIP UDP Jitter Operation Example

- Simulating G.711 A-Law codec (64 kbps Transmission) VoIP Call

```
Source #
       logging on
       ip sla monitor 10
        type jitter dest-ipaddr 209.165.200.225 dest-port 16384
codec g711alaw advantage-factor 2
        owner admin
        tag jitter-with-voice-scores
       ip sla monitor schedule 10 start-time now
       ip sla monitor reaction-configuration 10 react mos
threshold-type         immediate threshold-value 490 250 action-
type trapOnly
       ip sla monitor logging traps

       snmp-server host 10.10.10.10 version 2c public
       snmp-server enable traps syslog
```

Set Default Values for:
- **codec-numpackets,**
- **codec-size, and**
- **codec-interval**

Enable Specific IP SLA Syslog Messages

To Translate Syslog into Traps

connectionLoss,
jitterAvg,
jitterDSAvg,
jitterSDAvg,
Mos,
PacketLossDS,
PacketLossSD
Rtt,
Timeout,
verifyError

Note: "Logging on" controls (enables or disables) system message logging globally

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
                          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
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
```

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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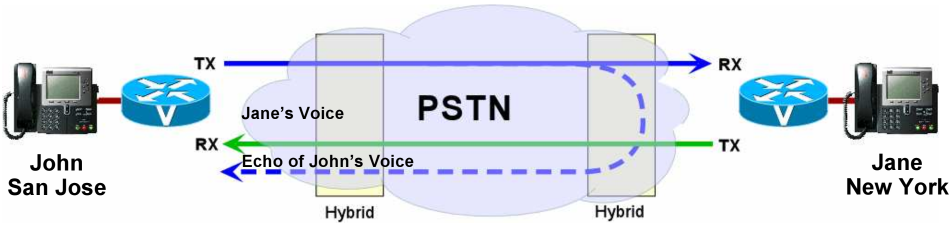
Echo

Talker Echo

Talker Echo (Most Common)

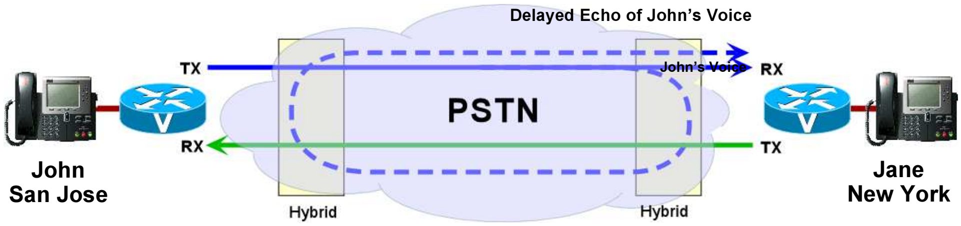
- Talker echo occurs when a talker's speech energy, transmitted down the primary signal path, is coupled into the receive path from the far end; the talker then hears his/her own voice, delayed by the total echo path delay time; if the 'echoed' signal has sufficient amplitude and delay, the result can 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



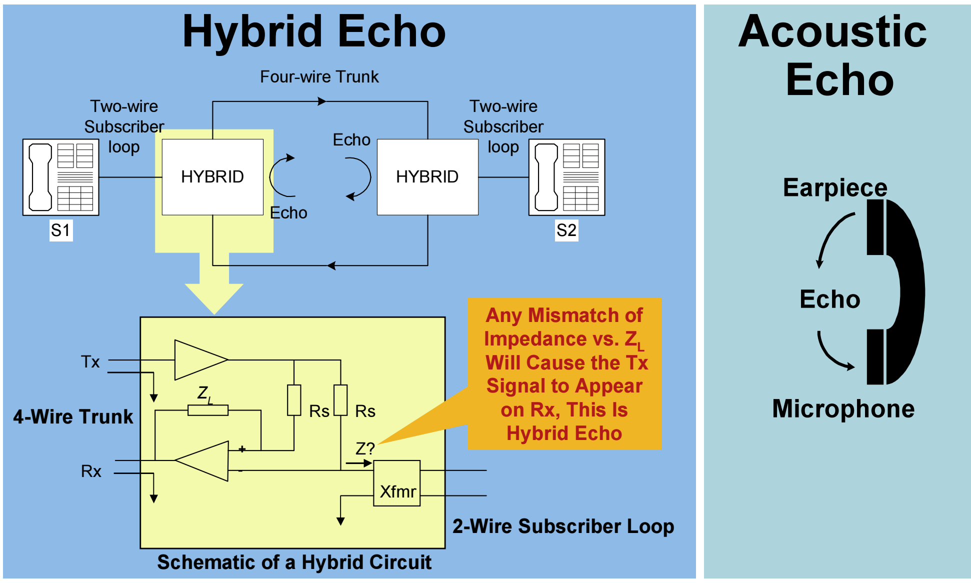
Listener Echo

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

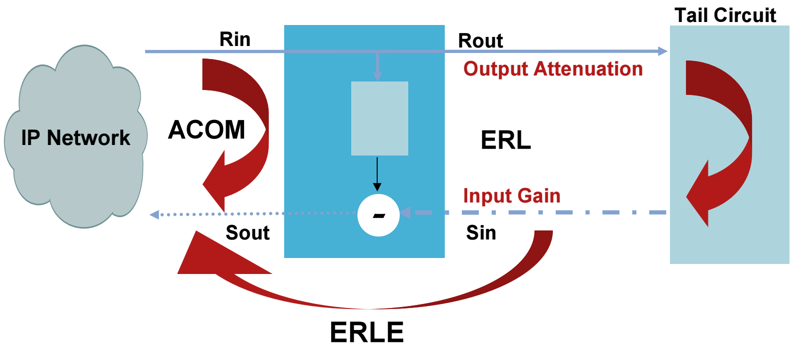


Sources of Echo



Eliminating Echo

Leveraging Echo Canceller



- 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

- **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 (aka 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)**

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
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  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

Agenda

- Recognizing Voice Quality Issues
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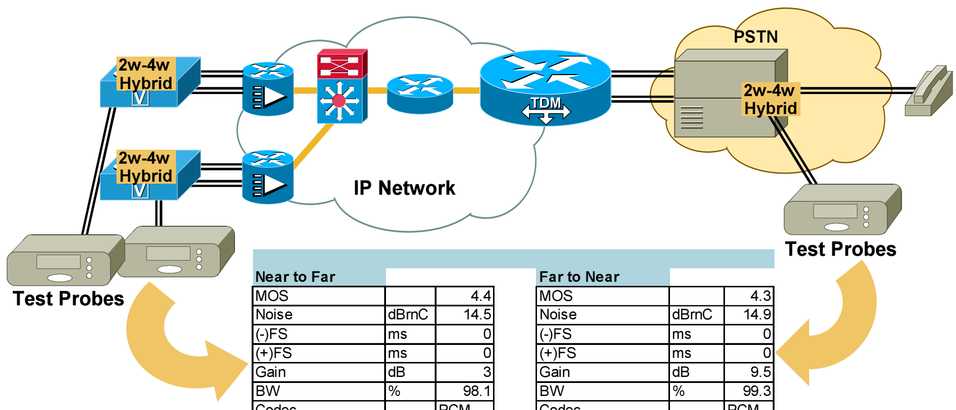
Proactive Approach to Fixing Echo and Voice Distortion

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

- 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)

- ITU Recommendation G.165 - defines characteristics, performance, and tests for echo cancellers.
- ITU Recommendation G.168 - defines characteristics, performance, and tests for echo cancellers in digital networks.
- ITU Recommendation G.131 - control of talker echo.

Loudness Ratings

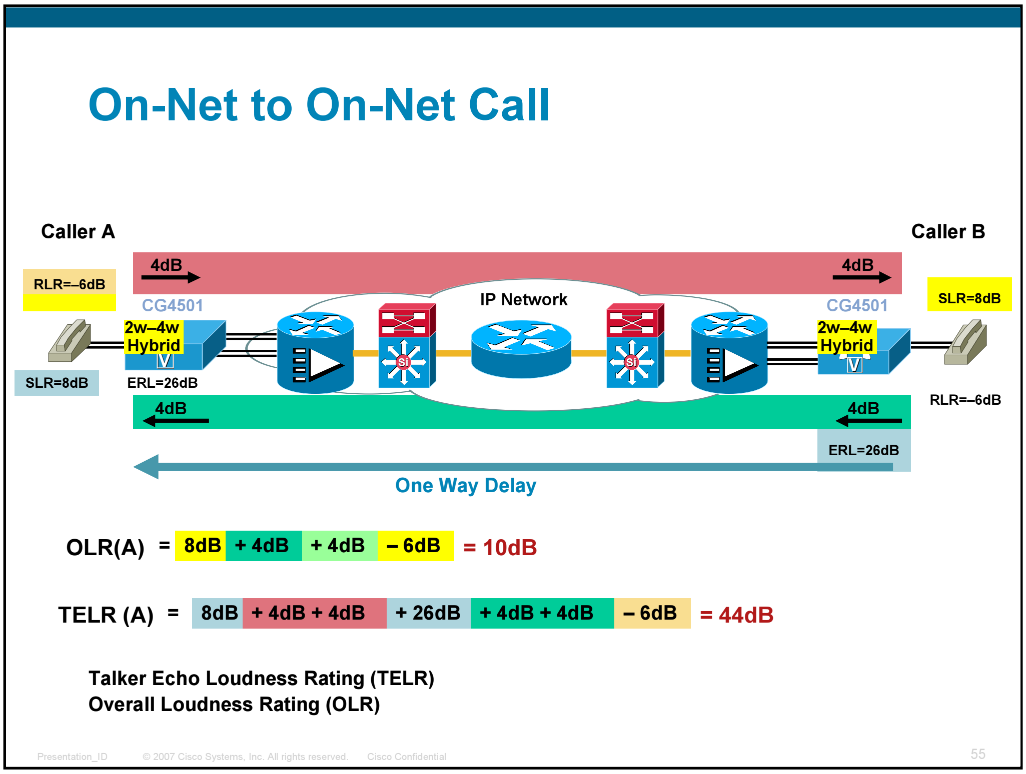
Terminology

- 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)=ERLE, 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



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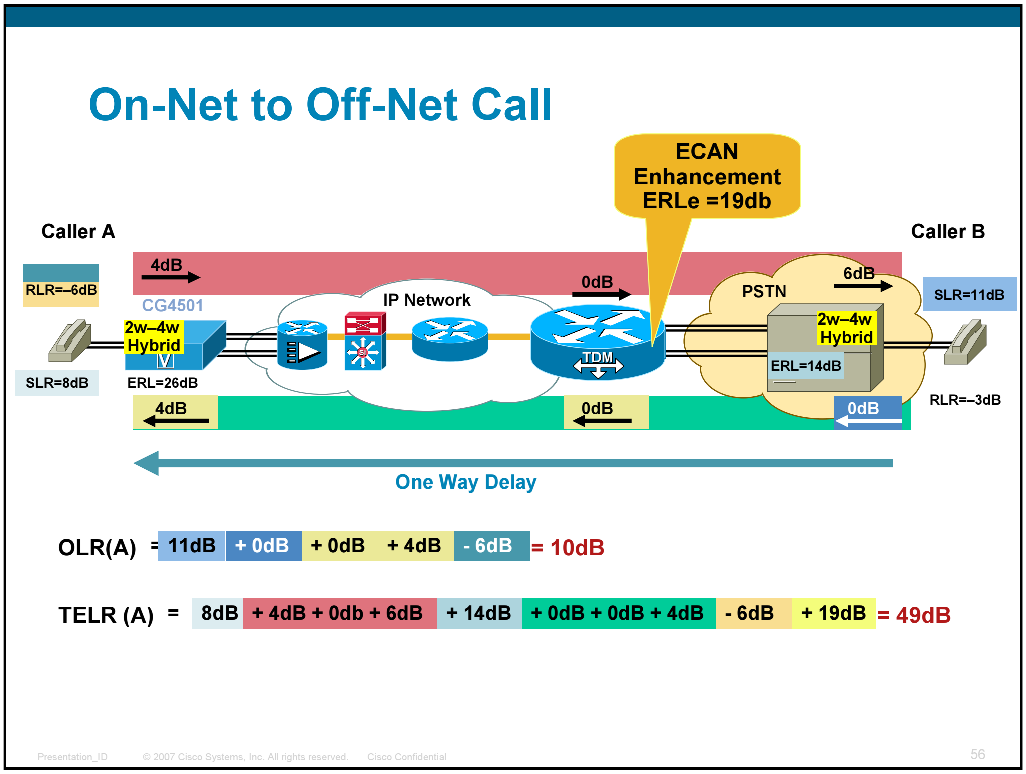
- TELR(A) = SLR(A) + loss in top path +ERL(B) or TCLw(B) + loss in bottom path + RLR(A), where ERL is the echo return loss of the hybrid or echo canceler, and TCLw is the weighted terminal coupling loss of the digital phone set.

- *Note: Standard SLR/RLR for analog phone is 8dB and -6dB respectively. See section 6.4 item 7 of TIA 912 for more details.*



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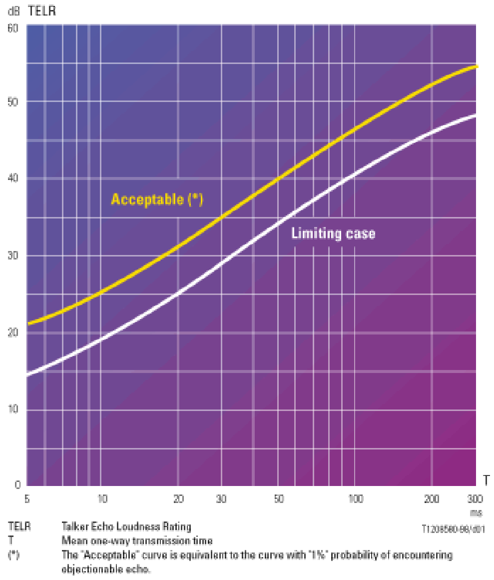
Echo Loudness Rating vs. Delay (ITU G.131)



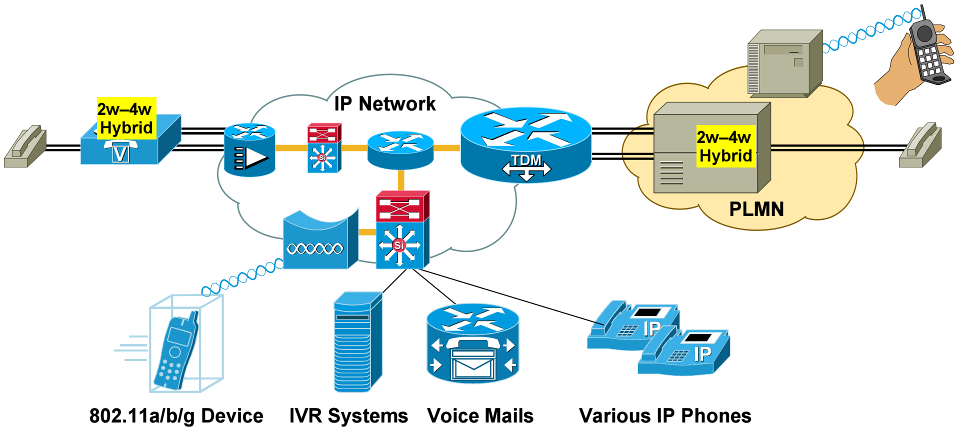
- The long-term goal for TELR is 8-12 dB, but because of the mix of technologies, the short-term goal is 8-21 dB. The difference between OLR in both directions should be no more than 8 dB.

- NTLP (Network Transmission Loss Plan) rule of thumb:
- One way loss = 10-12dB
- About 2/3 of the loss at the RX

- 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.

- The degree of annoyance of talker echo depends both on the amount of delay as well as on the level difference between the voice and echo signals

Test for All the Endpoints



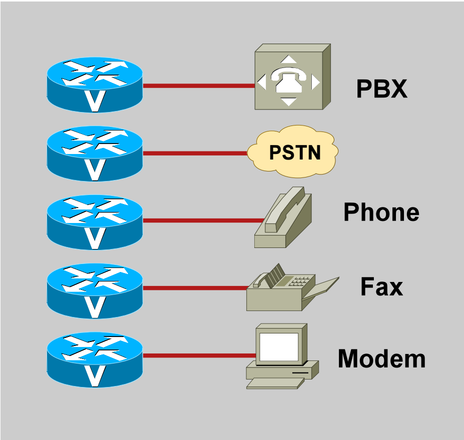
Don't Assume Loss Plan Works Across All Possible Calls

Rules of Thumb

- Echo observed on one end is typically generated at other end
- Bits don't leak—Echo is not introduced on digital links
- ERL must be greater than 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

Analog Gateways



- 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

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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

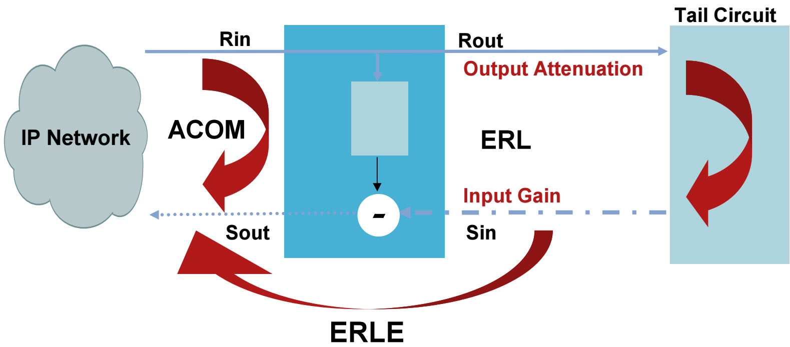
- http://www.cisco.com/warp/public/788/voip/show_call_act_voice.htm

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

Eliminating Echo

Leveraging Echo Canceller (repeated slide)



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

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 and Adjusting Echo in VISM

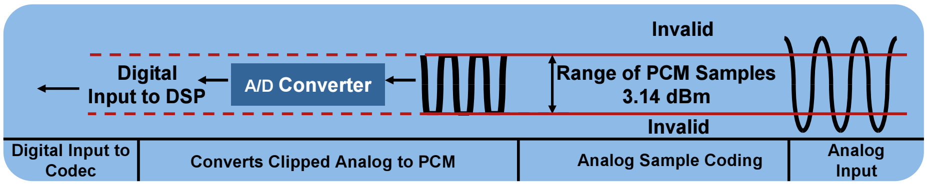
If We Configure 2 dB of Attenuation in Each Direction as Follows:

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

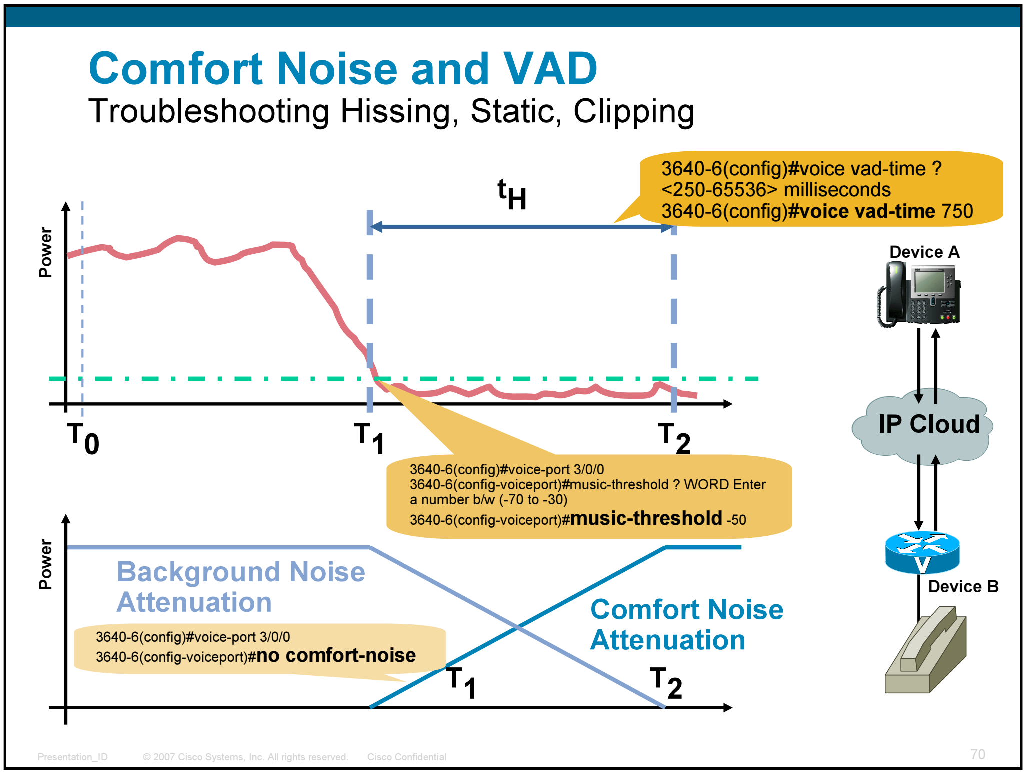
```
VISM8.a > cnflngain
ERR : incorrect number of parameters (not enough)
Syntax : cnflngain "line_number input_gain output_attenuation"
         line_number  -- values: 1 - 8.
         input_gain -- Value: -6..14 (dB)
         output_attenuation -- Value: 0..14 (dB)
```

```
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



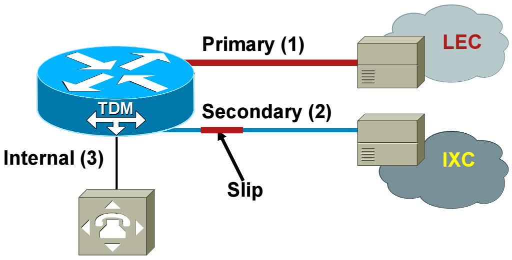
- During a call between device A and device B, device A reaches a silence interval (Figure 2). The voice activity detector uses an algorithm to determine if a silence interval has been reached. Typically, the voice activity detector has a holdover period of time t1 at the end of each speech burst. During this time it continues to send packets to the far-end. This helps to avoid excessive switching and choppy speech. If additional speech is detected during this holdover interval, the voice stream between the devices continues uninterrupted. After a period of length t1 (T2 in Figure 2) has elapsed, IP voice device A stops sending packets if no additional voice is detected.
- At time T1 (Top figure), an indication is sent to device B that alerts it to the fact that the VAD holdover is starting. This also contains the duration of the VAD holdover. When this message is received, device B starts to attenuate down the voice signal that it receives from device A and mixes it with the generated comfort noise that it should attenuate up (as in Bottom figure).
- This attenuation provides a smooth transition between real background noise and generated comfort noise. It makes the transitions from environments where the characteristics of the background noise are much different from those of generated comfort noise smoother and much less noticeable. The length of the VAD holdover interval (th) determines how effective this technique is. Longer intervals result in smoother sounding transitions.
- If the voice signal cuts in before time T2 (top figure), the attenuation is halted immediately and the full scale incoming audio is played. Such a cut-in should be signaled through another indication from device A to device B. Since the voice signal is significantly louder than the background noise, it masks the transition back and is not as noticeable.
- http://www.cisco.com/warp/public/788/voice-qos/hissing.html

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

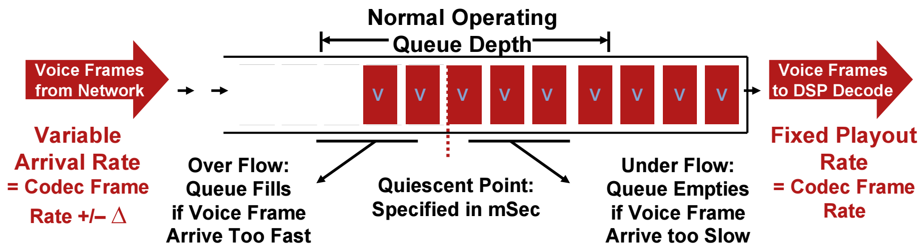
Troubleshooting Garbled, Synthetic, Choppy Voice

- 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

- http://www.cisco.com/warp/public/788/voice-qos/troubleshoot_qos_voice.html

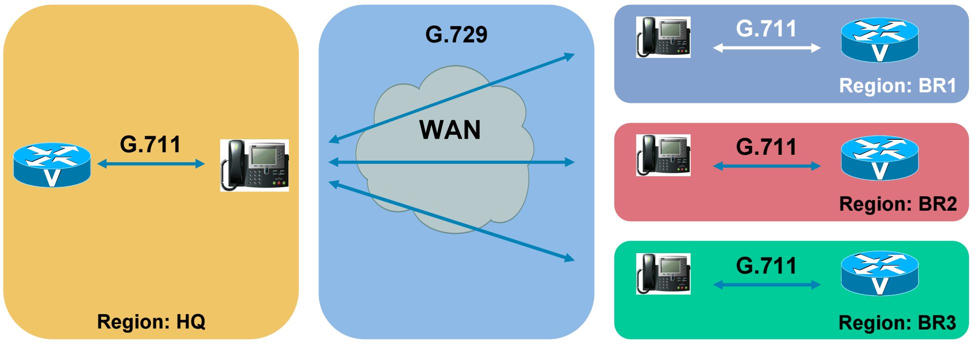
De-Jitter Buffer Adjustment



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]
```


- **For a more comprehensive CODEC table, see the CCO document titled, "Voice Over IP - Per Call Bandwidth Consumption"**

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>`

- **T.30 Message Descriptions**
- **CED** (Called terminal identification) - a 2100 Hz signal that is transmitted by the terminating fax device upon answering a fax call. This signal temporarily disables echo cancellers that are present on the connection to prepare the line for data transmission.

  CFR (Confirmation To Receive) - a response confirming that the previous messaging and training has been completed and that fax page transmission can begin.

  CNG (Calling Tone) - an 1100 Hz tone that is on for half a second and then off for 3 seconds. This signal identifies the fax terminal as being a non-speech device. The signal also indicates that the initiating fax terminal is awaiting the DIS signal from the terminating fax terminal.

  CRP (Command Repeat) - a response that indicates that the previous command was received in error and needs to be repeated. (Optional)

  CSI (Called Subscriber Identification) - may be used to provide the specific identity of the called fax terminal through its international telephone number. (Optional)

  DCN (Disconnect) -ends the fax call and requires no response.
- **DCS** (Digital Command Signal) - the response to the capabilities identified by the DIS signal. This where the calling fax terminal matches its capabilities with the ones provided in the called fax terminal's DIS message.

  DIS (Digital Identification Signal) -identifies the capabilities of the called fax terminal.
- **EOM** (End Of Message) - indicates the end of a complete page of fax information.
- **EOP** (End Of Procedure) - indicates the end of a complete page of fax information and no further pages are to be sent. Proceed to the disconnect phase of the fax call.

  FTT (Failure To Train) - used to reject a training signal and request a retrain (the retrains usually occur at lower modulation speeds).

  MCF (Message Confirmation) - indicates that a message has been satisfactorily received.
- **MPS** (MultiPage Signal) - indicates the end of a complete page of fax information and that the receiver is ready for additional pages.

  NSF (Non-Standard Facilities) - may be used to identify specific capabilities or requirements that are not covered by the T-series specifications. (Optional)
- **NSS** (Non-Standard Facilities Setup) - may be used as a response to the information contained in the NSF signal. (Optional)
- **PPR** (Partial Page Request) - indicates that a previous message has not been satisfactorily received and that the frames specified are to be retransmitted.
- **PPS** (Partial Page Signal) - indicates the end of a partial page or a complete page of facsimile information.
- **RTN** (Retrain Negative) - indicates that a previous message has not been satisfactorily received. Retraining is needed to proceed (usually at a lower modulation speed).

  RTP (Retrain Positive) - indicates that a complete message has been received and that additional messages may follow after retraining.

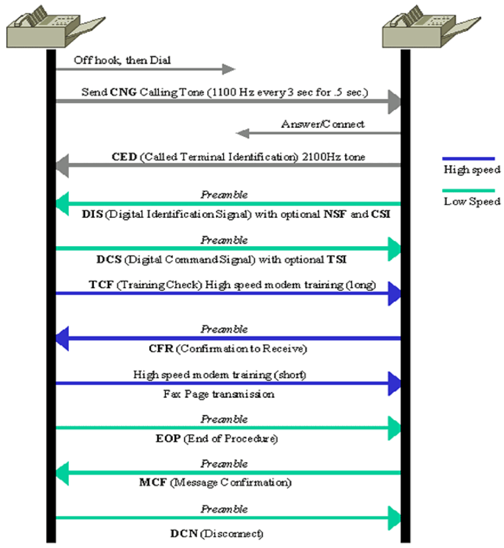
Fax Relay Troubleshooting (2/2)

- 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
```



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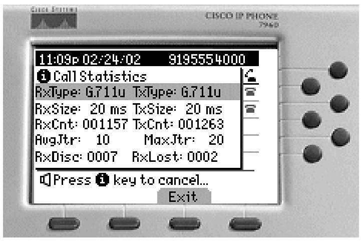
- TRACE OF A GOOD CALL"

```
5400_c#
5400_c#
5400_c#
*Dec 17 09:55:55.158: 7/2:0 (12025) 679738623 fr-entered (20ms)
*Dec 17 09:56:05.562: 7/2:0 (12025) 679796309 state_code: fr-msg-det V21, t30_msg: NSF
*Dec 17 09:56:05.562: 7/2:0 (12025) 679802285 state_code: fr-msg-det V21, t30_msg: CSI
*Dec 17 09:56:05.562: 7/2:0 (12025) 679805269 state_code: fr-msg-det V21, t30_msg: DIS
*Dec 17 09:56:18.578: 7/2:0 (12025) 679901290 state_code: fr-msg-det V21, t30_msg: NSF
*Dec 17 09:56:18.578: 7/2:0 (12025) 679907261 state_code: fr-msg-det V21, t30_msg: CSI
*Dec 17 09:56:18.578: 7/2:0 (12025) 679910251 state_code: fr-msg-det V21, t30_msg: DIS
*Dec 17 09:56:22.578: 7/2:0 (12025) 679933477 state_code: fr-msg-tx V21, t30_msg: TSI
*Dec 17 09:56:22.578: 7/2:0 (12025) 679936365 state_code: fr-msg-tx V21, t30_msg: DCS
*Dec 17 09:56:26.578: 7/2:0 (12025) 679976421 state_code: fr-msg-det V21, t30_msg: FTT
*Dec 17 09:56:30.578: 7/2:0 (12025) 680000211 state_code: fr-msg-tx V21, t30_msg: TSI
*Dec 17 09:56:30.578: 7/2:0 (12025) 680003109 state_code: fr-msg-tx V21, t30_msg: DCS
*Dec 17 09:56:35.578: 7/2:0 (12025) 680043173 state_code: fr-msg-det V21, t30_msg: CFR
*Dec 17 09:57:20.578: 7/2:0 (12025) 680412789 state_code: fr-msg-tx V21, t30_msg: EOP
*Dec 17 09:57:24.578: 7/2:0 (12025) 680431755 state_code: fr-msg-det V21, t30_msg: MCF
*Dec 17 09:57:25.578: 7/2:0 (12025) 680447565 state_code: fr-msg-tx V21, t30_msg: DCN
*Dec 17 09:57:25.646: 7/2:0 (12025) 680462580 fr-end
5400_c#
5400_c#
5400_c#
"TRACE OF A BAD CALL"
5400_c#
5400_c#
5400_c#
*Dec 17 09:57:55.190: 7/2:0 (12027) 681135390 fr-entered (20ms)
*Dec 17 09:58:04.578: 7/2:0 (12027) 681193124 state_code: fr-msg-det V21, t30_msg: NSF
*Dec 17 09:58:04.578: 7/2:0 (12027) 681199101 state_code: fr-msg-det V21, t30_msg: CSI
*Dec 17 09:58:04.578: 7/2:0 (12027) 681202093 state_code: fr-msg-det V21, t30_msg: DIS
*Dec 17 09:58:08.578: 7/2:0 (12027) 681225469 state_code: fr-msg-tx V21, t30_msg: TSI
*Dec 17 09:58:08.578: 7/2:0 (12027) 681228365 state_code: fr-msg-tx V21, t30_msg: DCS
*Dec 17 09:58:13.578: 7/2:0 (12027) 681268493 state_code: fr-msg-det V21, t30_msg: FTT
*Dec 17 09:58:17.578: 7/2:0 (12027) 681292239 state_code: fr-msg-tx V21, t30_msg: TSI
*Dec 17 09:58:17.578: 7/2:0 (12027) 681295095 state_code: fr-msg-tx V21, t30_msg: DCS
*Dec 17 09:58:36.578: 7/2:0 (12027) 681447597 state_code: fr-msg-tx V21, t30_msg: TSI
*Dec 17 09:58:36.578: 7/2:0 (12027) 681450495 state_code: fr-msg-tx V21, t30_msg: DCS
*Dec 17 09:58:51.578: 7/2:0 (12027) 681562381 state_code: fr-msg-tx V21, t30_msg: TSI
*Dec 17 09:58:51.578: 7/2:0 (12027) 681565490 state_code: fr-msg-tx V21, t30_msg: DCS
*Dec 17 09:59:00.578: 7/2:0 (12027) 681641789 state_code: fr-msg-tx V21, t30_msg: TSI
*Dec 17 09:59:00.578: 7/2:0 (12027) 681644692 state_code: fr-msg-tx V21, t30_msg: DCS
*Dec 17 09:59:14.578: 7/2:0 (12027) 681756564 state_code: fr-msg-tx V21, t30_msg: DCN
*Dec 17 09:59:14.594: 7/2:0 (12027) 681770956 fr-end
```

Other Tools

Cisco Catalyst 6608




```
vdtl-Cat6k-PBX1> (enable) sh port voice active 4/8
Port  4/8 :
  Channel #23:
    Remote IP address                        : 172.18.104.74
    Remote UDP Port                          : 24876
    ACOM Level Current                       : 45
    Call State                               : voice
    Codec Type                               : G711 ULAW PCM
    Coder Type Rate                          : 20
    ERL Level                                : 61
    Voice Activity Detection                 : disabled
    Echo Cancellation                        : enabled
[snip]
```


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/

  http://www.cisco.com/go/srnd/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

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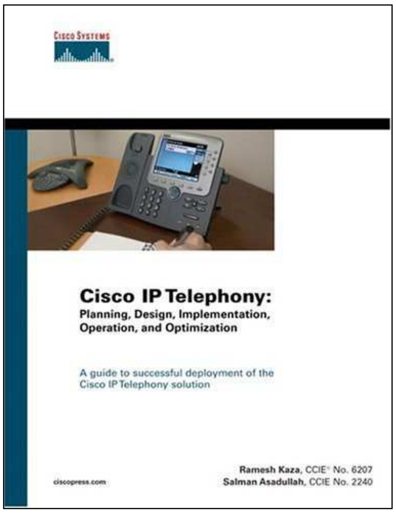
Special thanks to

Goran Obradovic
&
Talal Siddiqui

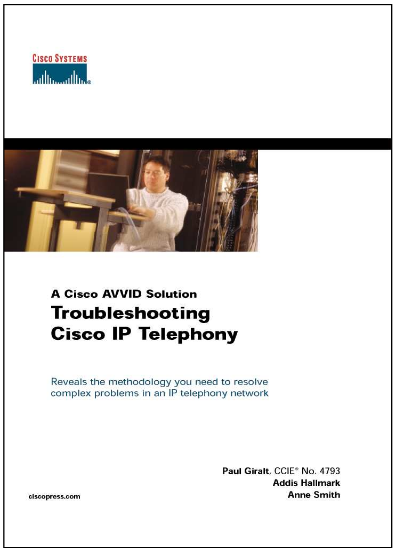
Recommended Reading

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



Recommended Reading

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



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