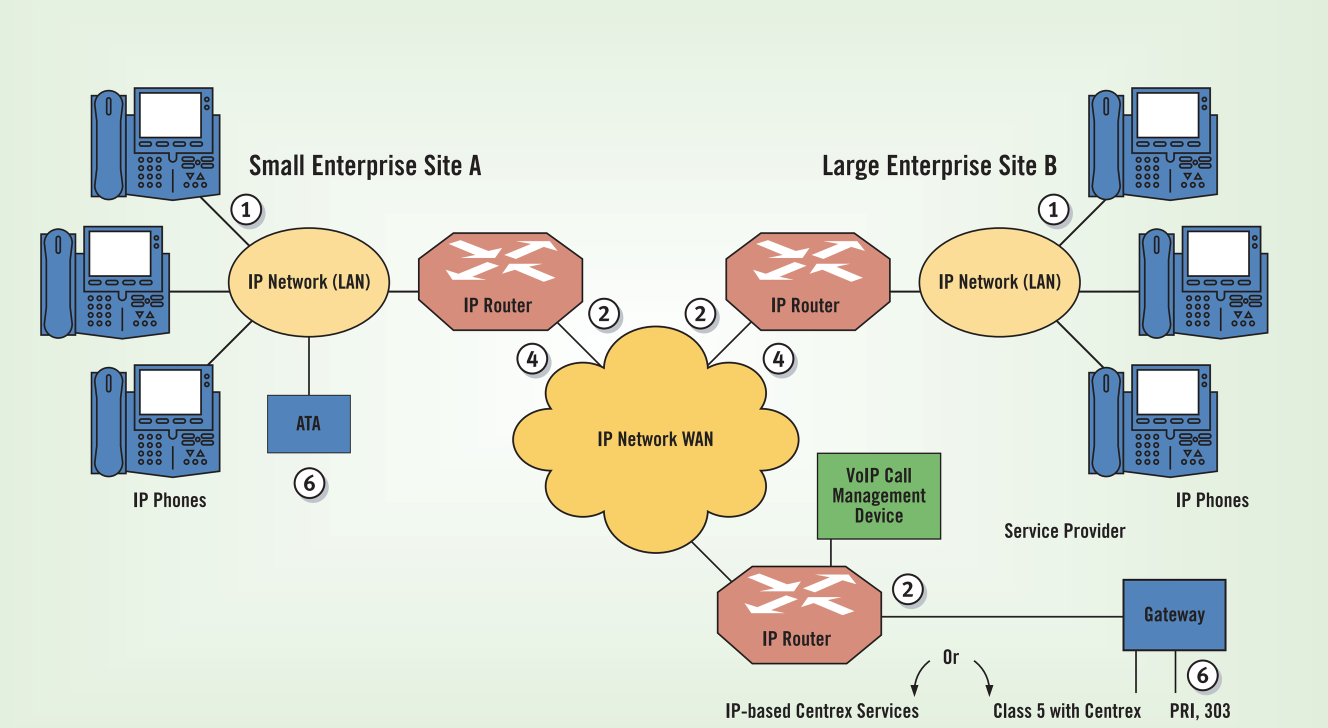
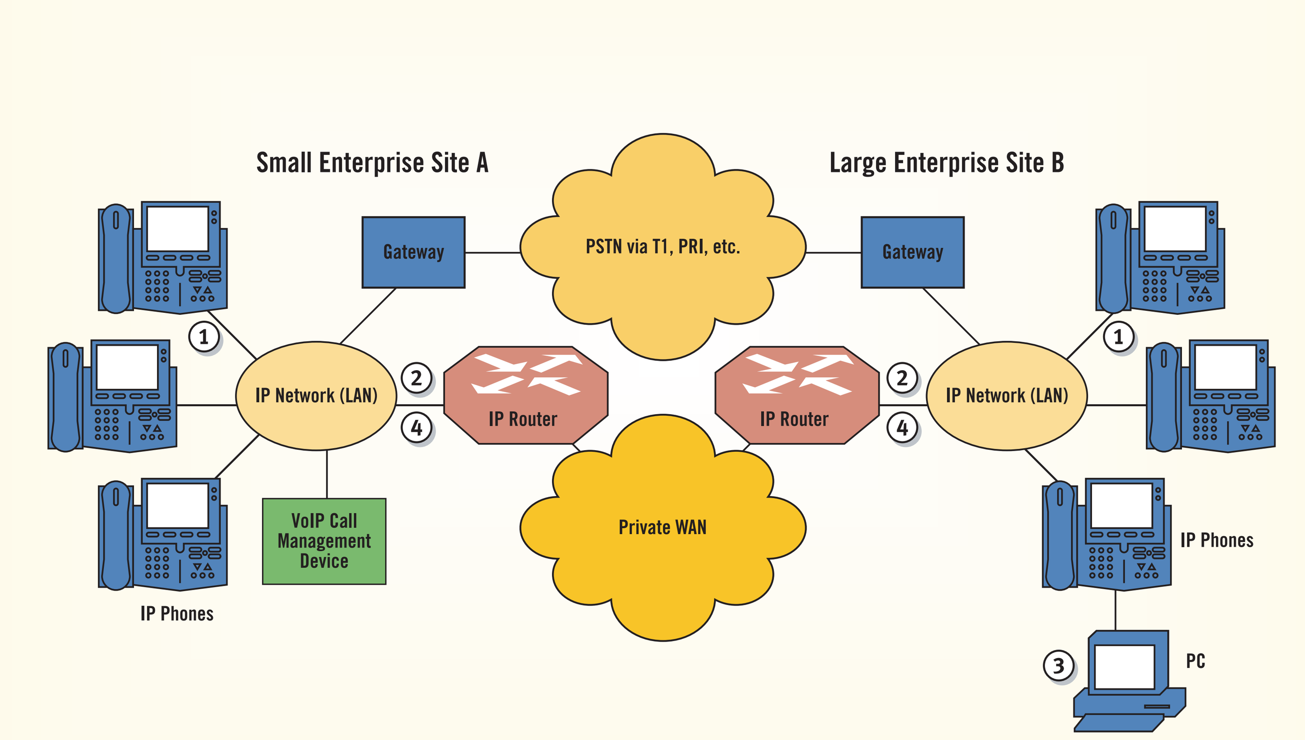
Voice Over IP (VoIP) Technology Reference Chart

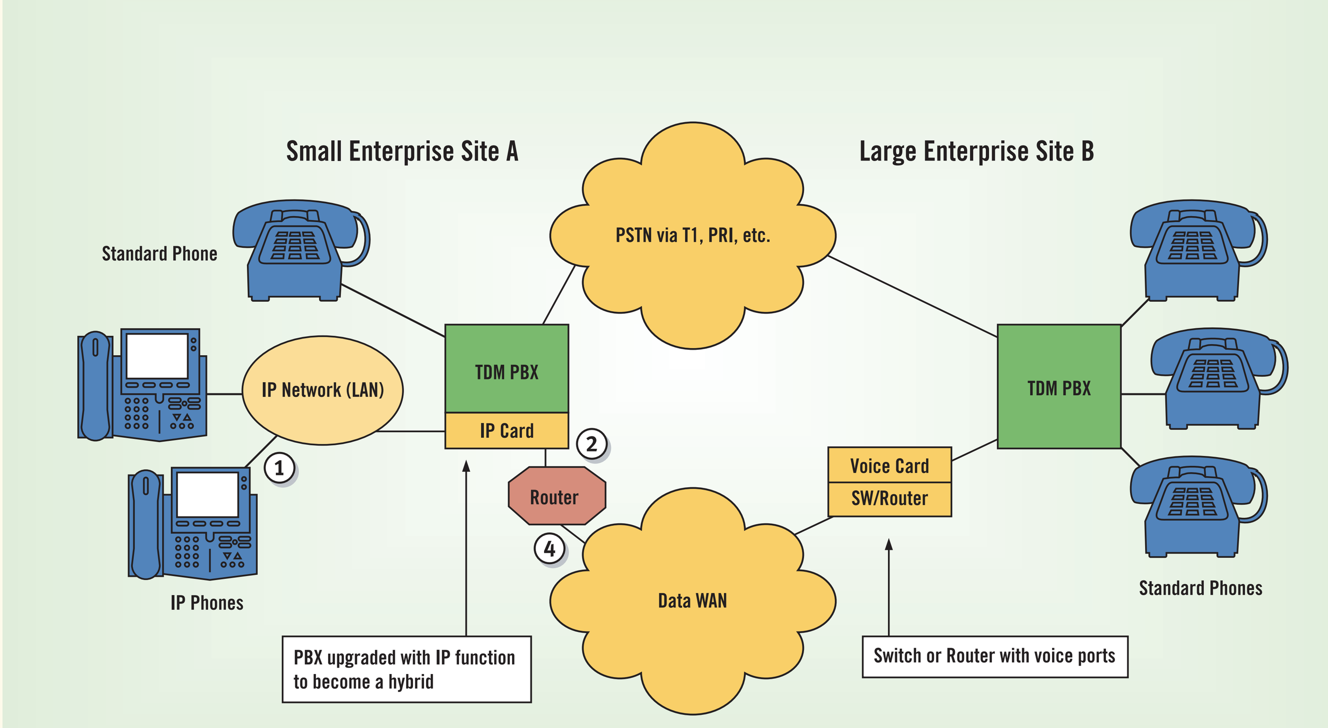
Provider Hosted VoIP Service "IP Centrex"



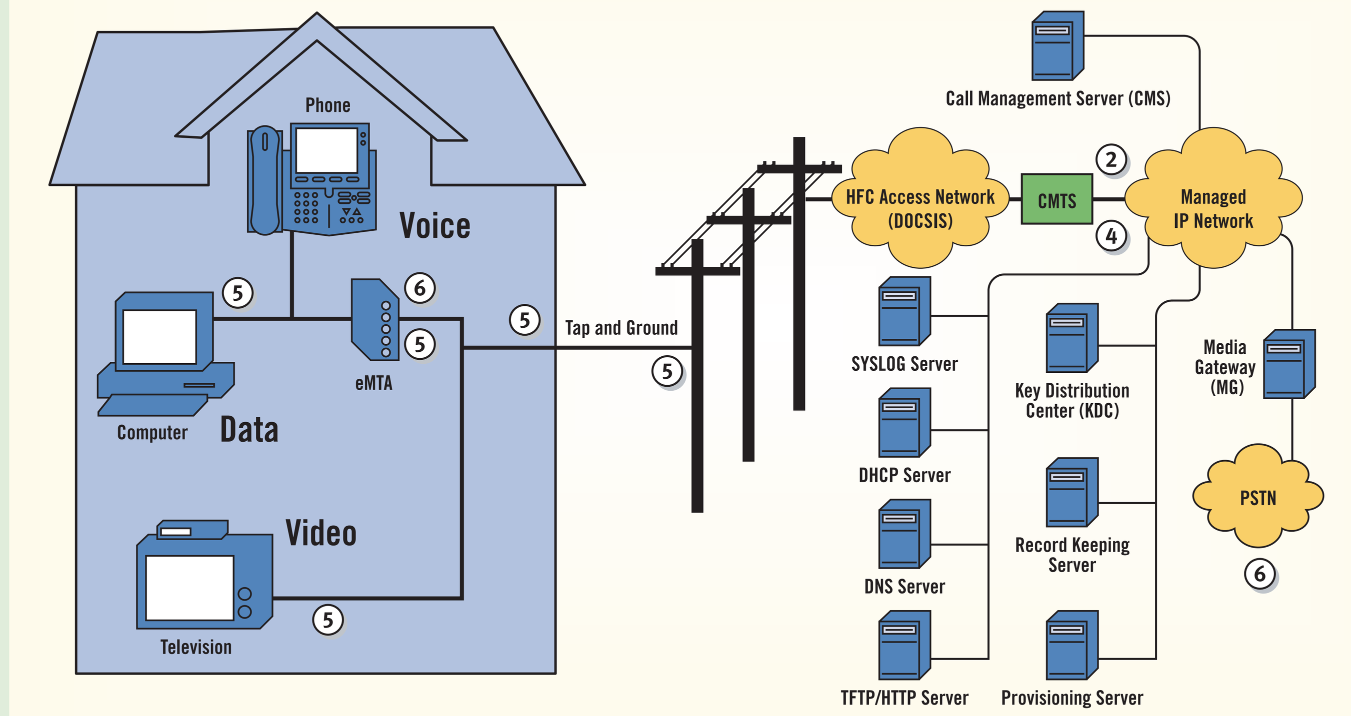
Enterprise VoIP to the Desktop



Enterprise VoIP "Toll Bypass Model"



Residential Voice Over CATV Infrastructure



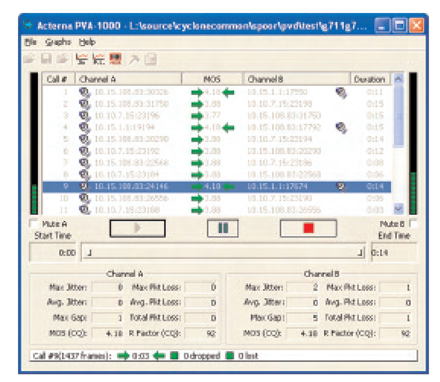
The Communications Test & Measurement Segment of JDSU

① JDSU HST-3000 VoIP

An IP telephony installation field tool for service turn-up and verification. The HST-3000 emulates IP phones, validates VoIP connectivity, feature availability, and end user voice quality.

② JDSU DA-3400

A 7-layer network analyzer for VoIP and Data network monitoring and troubleshooting. The DA-3400 is ideal for VoIP call quality monitoring, signaling analysis, and problem identification and isolation.



③ JDSU PVA-1000

An enterprise and service support software tool for IP telephony problem capture and analysis. PVA-1000 software provides full analysis of VoIP telephone calls including jitter, packet loss, and audio playback.

④ JDSU QT-600

This Ethernet & Triple-Play Probe is a component of the NetComplete™ VoIP Service Assurance Solution, a distributed and scaleable solution for service providers. Capabilities include on-demand call generation, active and passive call quality monitoring, QoS reporting for trouble identification and isolation, performance monitoring, and capacity planning.

⑤ JDSU DSAM

A handheld CATV service and installation meter for VoIP verification and troubleshooting over coax cables. DSAM contains an eMTA and allows for efficient "find and fix" capabilities of both IP and FR issues in an HFC network.

⑥ OPERA - Voice Quality Analysis

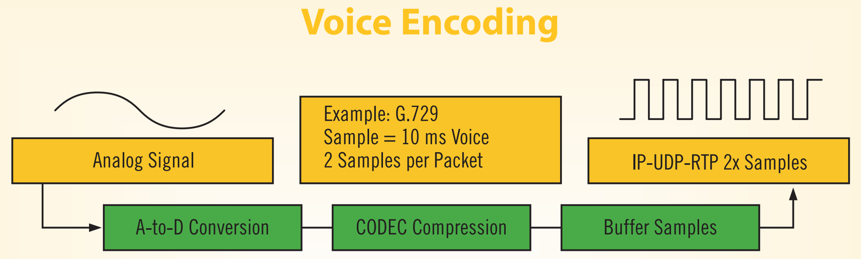
A voice and audio quality analyzer for end-to-end quality testing over any VoIP network, OPERA, measuring to ITU-T standards, objectively evaluates and ensures the quality of compressed speech (P.862 PESQ) and wideband audio signals (BS.1387 PEAQ).

Protocol Stack

Signaling (SIP, H.323, SCCP, NCS, etc.)	Voice Sample/ CODEC G.7XX / RTP	RTCP Y/N
TCP	UDP	
IP		
HDLC (PPP, FR)	ATM	ETHERNET
PHYSICAL (Ethernet, T1, T3, DOCSIS, etc)		

The variables (green) — Constant since this is VoIP (yellow)

Voice Encoding



CODEC	CODEC Bandwidth	IP Bandwidth NO Silence Suppression	IP Bandwidth 30% Silence Suppression
G.711 PCM	64 Kbps	80 Kbps	56 Kbps
G.729 CS-ACELP	8 Kbps	24 Kbps	16.8 Kbps
G.723.1 ACELP	5.6 Kbps	16.27 Kbps	11.39 Kbps
G.723.1 MP-MLQ	6.4 Kbps	17.07 Kbps	11.95 Kbps
G.726	32 Kbps	40 Kbps	29 Kbps

Real-Time Protocol (RTP)

- Defined in RFC 3550 and 3551
- Used by H.323, SIP, MGCP/NCS, MEGACO, and PacketCable
- Transports real-time voice and video content

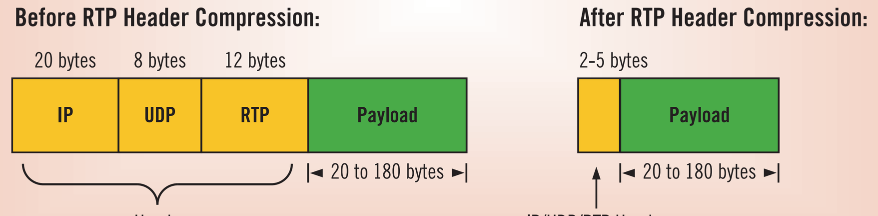
0 1 2 3 4 5 6 7	Octet
V / P / X / CSRC Count	1
M / Payload Type	2
Sequence Number	3
Time Stamp	4
SSRC	5
CSRC	6

RTP Structure

V	This stands for version. It identifies the RTP version.
P	This stands for padding. When set, the packet contains one or more additional padding octets at the end. Padding octets are not part of the payload.
X	This stands for extension bit. When set, the fixed header is followed by exactly one header extension with a defined format.
CSRC Count	This contains the number of CSRC identifiers that follow the fixed header.
M	This stands for marker. The interpretation of the marker is defined by a profile. It is intended to allow for significant events, such as frame boundaries, to be marked in the packet stream.
Payload Type	This identifies the format of the RTP payload (G.729, G.711, etc.) and determines its interpretation by the application. A profile specifies a default static mapping of payload type codes to payload formats. Additional payload type codes may be defined dynamically through non-RTP means.
Sequence Number	This increments by one for each RTP data packet sent. It may may be used by the receiver to detect packet loss and to restore packet sequence.
Time Stamp	This reflects the sampling instant of the first octet in the RTP data packet. The sampling instant must be derived from a clock that increments monotonically and linearly in time to allow for synchronization accuracy and for measuring packet arrival jitter.
SSRC	This identifies the synchronization source. The identifier is chosen randomly with the intent that no two synchronization sources within the same RTP session will have the same SSRC identifiers.
CSRC	This identifies the contributing sources for the payload contained in this packet.

Compressed RTP (CRTP)

- CRTP greatly reduces the overhead for Voice applications over slow links.
- Compresses the IP/UDP/RTP header in a RTP data packet from 40 bytes to approximately 2 to 5 bytes.
- RTP header compression is supported in point-to-point networks (FR, HDLC, PPP, etc.)



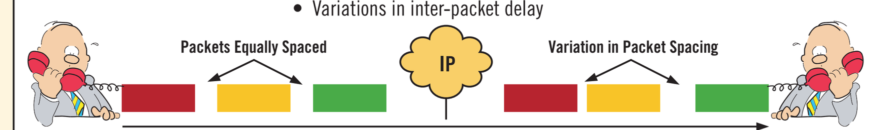
Signaling

H.323	
• ITU-T H.225	Call control signaling for multimedia systems
• ITU-T H.235	Security and encryption for H series terminals
• ITU-T H.245	Media transport channel setup and control
• ITU-T H.450	H.323 supplemental services
• ITU-T T.38	Real-time fax over IP
• IETF RFC 3550	RTP for media transport over IP
SIP	
• IETF RFC 3261	Session Initiation Protocol
• IETF RFC 3550	RTP for media transport over IP
MGCP	
• IETF RFC 3435	Media Gateway Control Protocol
• IETF RFC 3550	RTP for media transport over IP
PacketCable	
• NCS	Network-Based Call Signaling protocol
• IETF RFC 3550	RTP for media transport over IP
MEGACO/H.248	
• ITU-T H.248	H.248 joining of IETF and ITU-T VoIP signaling standards
• IETF RFC 3525	MEGACO joining of IETF and ITU-T VoIP signaling standards
• IETF RFC 3550	Real-time multimedia encapsulation over IP
Proprietary	
• Cisco Skinny Client Control Protocol (SCCP)	
• Nortel UniStim	
• Avaya custom H.323	

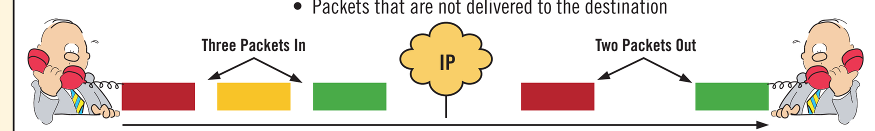
Quality Impairments (Transport-Related)

Delay – Latency
- End-to-end delay from speaker to listener
- CODEC operation
- Lower bit rate = higher delay

Jitter
- Variations in inter-packet delay

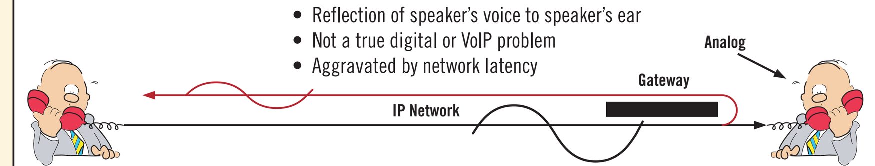


Packet Loss
- Packets that are not delivered to the destination

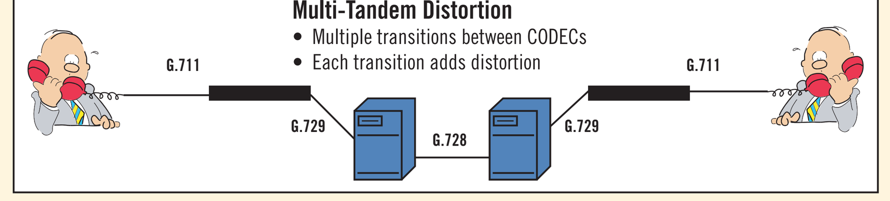


Out-of-Sequence Packets
- Packets that are not delivered in order

Echo
- Reflection of speaker's voice to speaker's ear
- Not a true digital or VoIP problem
- Aggravated by network latency



Multi-Tandem Distortion
- Multiple transitions between CODECs
- Each transition adds distortion



VoIP Test Phases

Equipment and/or Network Design Verification

Equipment manufacturers test their products for full functionality, performance under load, regression, and conformance to specifications. Providers and Enterprises implementing VoIP should repeat many of the same tests in their specific environment before adding or upgrading equipment. Test equipment requirements for this phase include conformance testers, load generators, voice quality analyzers, and protocol analyzers. (Products: DA-3400, Opticom OPERA and PVA-1000)

Network Audit/VoIP Pre-Assessment

Before adding VoIP as an application on an existing LAN/WAN infrastructure, the network should be evaluated for readiness via a detailed baseline and through an active simulation of the deployed VoIP service. Test equipment requirements include network discovery and mapping tools, protocol analysis for statistics collection, and distributed software or hardware VoIP agents for generation and measurement of VoIP performance. (Products: DA-3400, HST-3000 and PVA-1000)

Service Installation/Turn-up

During service installation, the underlying physical and datalink layers should be fully tested for any marginal performance issues followed by a test of the VoIP service itself. Inbound, outbound, on-net and off-net calls should be performed to verify provisioning and performance across the IP/PSTN boundaries. Test equipment requirements include software-based or handheld portable tools suitable for the field technician. (Products: DSAM and HST-3000)

Service Troubleshooting

If the VoIP service is inoperable, intermittent, or degraded, the root cause needs to be identified before corrective action can be taken. Tools for troubleshooting vary widely with the nature and architecture of the VoIP Network, but they can include fixed or portable, passive or active test systems and dispatchless software. Functionality should include VoIP end point emulation, call capture, playback, and QoS scoring via MOS. (Products: DA-3400, DSAM, HST-3000, PVA-1000, and QT-600)

Passive Performance Monitoring

Larger deployments with stringent service expectations warrant a distributed, passive analysis system to monitor and alarm on various performance thresholds. Some of these requirements may be met by monitoring functions in the VoIP equipment itself, but they are often augmented by purpose-built peripherals and software that integrate with operations support systems (OSS) used by the network operations center (NOC). (Products: DA-3400 and QT-600)

Active Performance Testing/Service Assurance

Expansive networks that use VoIP transport on the long haul portion or have strict service level agreements (SLAs) from edge to edge require automated, distributed, active test resources that place calls through the cloud and alarm on any call failures or quality degradation. Some network elements offer capabilities in this realm, but in many cases, they are components of the path and should be verified. This requirement can be satisfied using purpose-built, distributed hardware peripherals and software that integrate with operations support systems (OSS) used by the network operations center (NOC). (Product: QT-600)

Test Methodologies - Signal Degradation Analysis

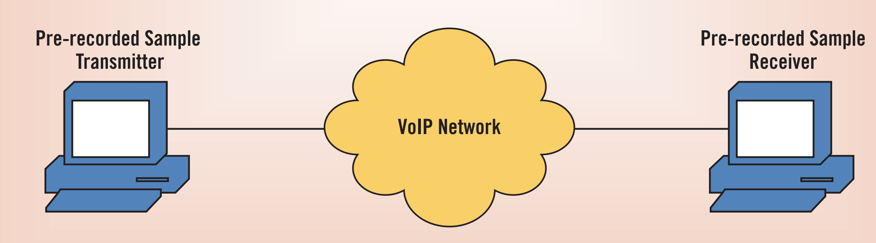
Signal Degradation Analysis

This out-of-service (objective) test evaluates distortion across all network segments from end to end

- Receiver compares received sample against known original sample
- Complex signal analysis identifies network induced distortion
- PESQ MOS quality scale from 1 (lowest) to 4.5 (highest)

PESQ - PAMS
- Uses pre-recorded voice samples
- No network monitoring capability

PESQ (Digital)	P.862 Perceptual Evaluation of Speech
PAMS (Analog)	P.861 Perceptual Analysis and Measurement System



Test Methodologies - Packet Transport Analysis

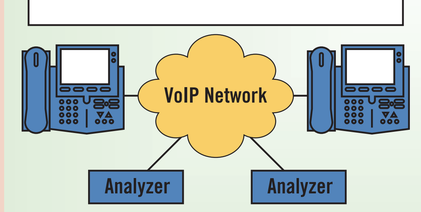
Packet Transport Analysis

This in-service (subjective) test computes the impact of packet transport problems on call quality

- Measures jitter, packet loss, and latency
- Focuses on monitoring and troubleshooting of customer problems
- MOS quality scale from 1 (lowest) to 5 (highest)
- R Factor values from 0 (lowest) to 100 (highest)

MOS – Mean Opinion Score
- In-service monitoring of telephone calls
- Measures packet transport impairments

ITU	ITU-T G.107 MOS Scores for VoIP
ETSI	TS 101 329-5 R Factor Values for VoIP



Acceptable MOS and R Factor Values for Narrowband CODECs

User Option	R Factor	MOS Score
Maximum obtainable for G.711	93	4.4
Very satisfied	90-100	4.3-5.0
Satisfied	80-90	4.0-4.3
Some users satisfied	70-80	3.6-4.0
Many users dissatisfied	60-70	3.1-3.6
Nearly all users dissatisfied	50-60	2.6-3.1
Not recommened	0-50	1.0-2.6

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