Voice Over IP Technology Protocol Reference

RADCOM

TEST-OF-THE-ART

Standards

Signalling

ITU-T Standards and Recommendations	
H.323 V2	Packet-based multimedia communications systems
H.225.0	Call signalling protocols and media stream packetization for packet-based multimedia (includes Q.931 and RAS)
H.225.0 Annex G	Gatekeeper to gatekeeper (inter-domain) communications
H.245	Control protocol for multimedia communications
H.235	Security and encryption for H-series multimedia terminals
H.450.x	Supplementary services for multimedia: 1. Generic functional protocol for the support of supplementary services in H.323 2. Call transfer 3. Diversion 4. Hold 5. Park & pickup 6. Call waiting 7. Message waiting indication
H.323 Annex D	Real-time fax using T.38
H.323 Annex E	Call connection over UDP
H.323 Annex F	Single-use device
T.38	Procedures for real-time group 3 facsimile communications over IP networks
T.120 series	Data protocols for multimedia conferencing
IETF RFCs and Drafts	
RFC 2543	SIP: Session initiation protocol
RFC 2327	SDP: Session description protocol
Internet Draft	SAP: Session announcement protocol

Gateway Control

ITU	
H.GCP	Proposed recommendation for gateway control protocol
IETF	
Internet Draft	MGCP: Media gateway control protocol
Internet Draft	MEGACO protocol
Draft	SGCP: Simple gateway control protocol
Internet Draft	IPDC: IP device control

Media Transport

IETF	
RFC 1889	RTP: Real-time transport protocol
RFC 1889	RTCP: Real-time transport control protocol
RFC 2324	RTSP: Real-time streaming protocol

Media Encoding

ITU

Voice

Standard	Algorithm	Bit Rate (Kbit/s)	Typical end-to-end delay (ms) (excluding channel delay)	Resultant Voice Quality
G.711	PCM	48, 56, 64	< < 1	Excellent
G.723.1	MPE/ACELP	5.3, 6.3	67-97	Good (6.3), Fair (5.3)
G.728	LD-CELP	16	< < 2	Good
G.729	CS-ACELP	8	25-35	Good
G.729 annex A	CS-ACELP	8	25-35	Good
G.722	Sub-band ADPCM	48, 56, 64	< 2	Good
G.726	ADPCM	16, 24, 32, 40	60	Good (40), Fair (24)
G.727	EADPCM	16, 24, 32, 40	60	Good (40), Fair (24)

Video

Standard	Algorithm	Bit Rate (Kbit/s)	Picture Quality
H.261	Discrete cosine transform (DCT) with motion compensation	p x 64 (p = # of ISDN B channels)	Low
H.263	Improved version of H.261	Various	Medium

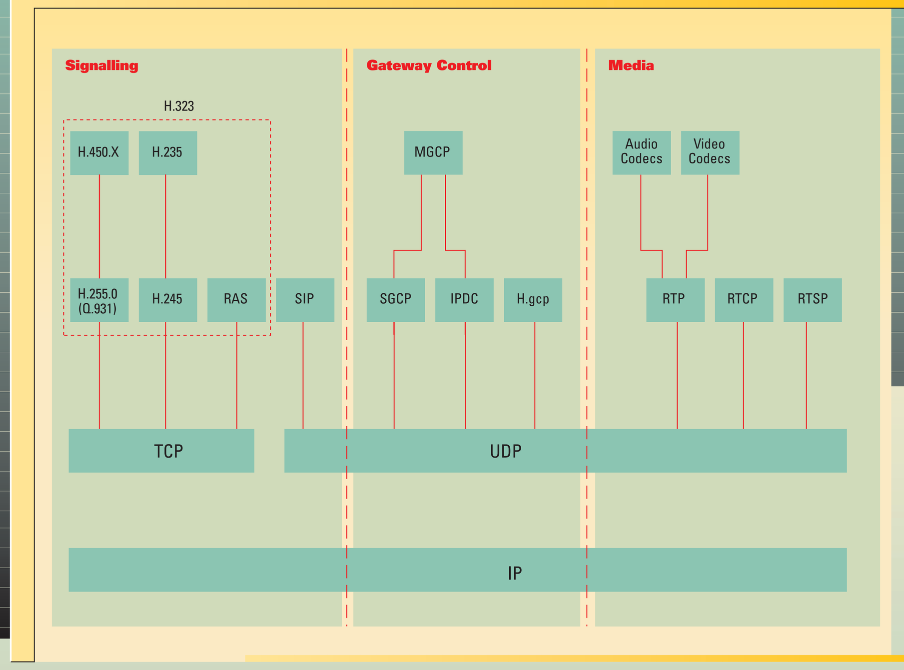
RTP Header

0	1	2	3	4	5	6	7	Octet
	V		P	X	CSRC count			1
M	Payload type							2
Sequence number								3
Timestamp								4
SSRC								5
CSRC								6

RTP structure

V	Version. Identifies the RTP version.
P	Padding. When set, the packet contains one or more additional padding octets at the end, which are not part of the payload.
X	Extension bit. When set, the fixed header is followed by exactly one header extension, with a defined format.
CSRCcount	Contains the number of CSRC identifiers that follow the fixed header.
M	Marker. The interpretation of the marker is defined by a profile. It is intended to allow significant events such as frame boundaries to be marked in the packet stream.
Payload type	Identifies the format of the RTP payload 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	Increments by one for each RTP data packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence.
Timestamp	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 synchronization and jitter calculations. The resolution of the clock must be sufficient for the desired synchronization accuracy and for measuring packet arrival jitter (one tick per video frame is typically insufficient).
SSRC	Identifies the synchronization source. This identifier is chosen randomly, with the intent that no two synchronization sources within the same RTP session will have the same SSRC identifier.
CSRC	Contributing source identifiers list. Identifies the contributing sources for the payload contained in this packet.

Protocol Stack



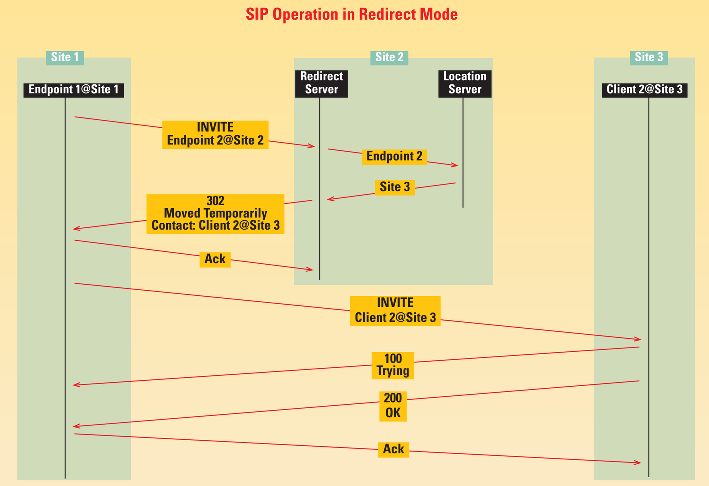
MGCP Commands

MGC ⟶ MG	CreateConnection: Creates a connection between two endpoints; uses SDP to define the receive capabilities of the participating endpoints.
MGC ⟶ MG	ModifyConnection: Modifies the properties of a connection; has nearly the same parameters as the CreateConnection command.
MGC ⟷ MG	DeleteConnection: Terminates a connection and collects statistics on the execution of the connection.
MGC ⟶ MG	Notification Request: Requests the media gateway to send notifications on the occurrence of specified events in an endpoint.
MGC ⟵ MG	Notify: Informs the media gateway controller when observed events occur.
MGC ⟶ MG	AuditEndpoint: Determines the status of an endpoint.
MGC ⟶ MG	AuditConnection: Retrieves the parameters related to a connection.
MGC ⟵ MG	RestartInProgress: Signals that an endpoint or group of endpoints is taken in or out of service.

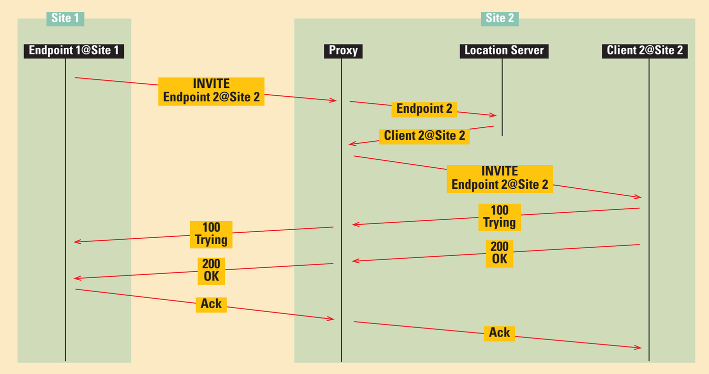
MGC = Media Gateway Controller
MG = Media Gateway

Typical SIP Calls

SIP Operation in Redirect Mode

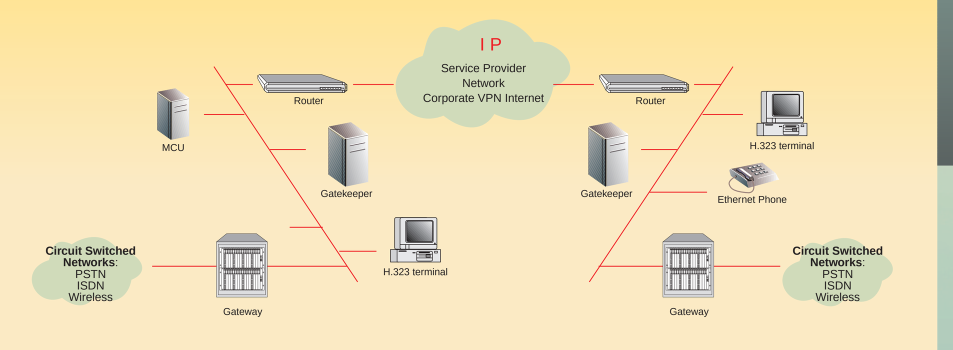


SIP Operation in Proxy Mode



H.323 Architecture

Gatekeeper	Manages a zone (collection of H.323 devices).
Required Functionality	Address translation, admissions control, bandwidth control.
Optional Functionality	Call authorization, bandwidth management, supplementary services, directory services, call management services.
Gateway	Provides interoperability between different networks, converts signalling and media e.g., IP/PSTN gateway.
H.323 Terminal	Endpoint on a LAN. Supports real-time, 2-way communications with another H.323 entity. Must support voice (audio codecs) and signalling (Q.931, H.245, RAS). Optionally supports video and data e.g., PC phone or videophone, Ethernet phone.
MCU	Supports conferences between 3 or more endpoints. Contains multipoint controller (MC) for signalling. May contain multi-point processor (MP) for media stream processing. Can be stand-alone (i.e. PC) or integrated into a gateway, gatekeeper, or terminal.

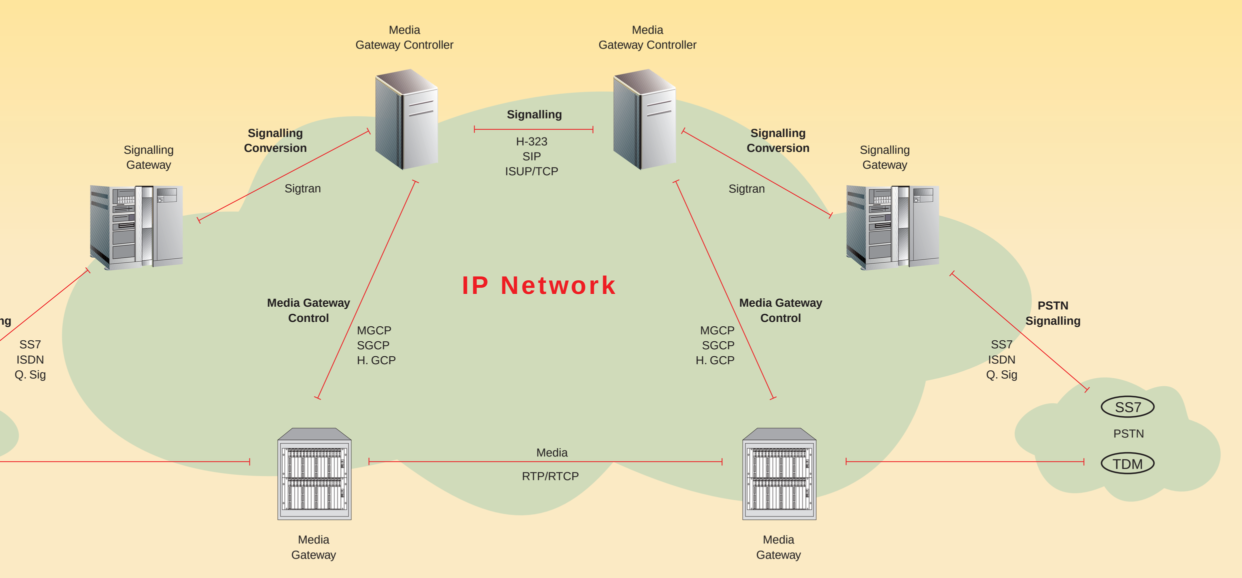


Converged Network Architecture

Media Gateway Controller: Coordinates setup, handling and termination of media flows at the media gateway.

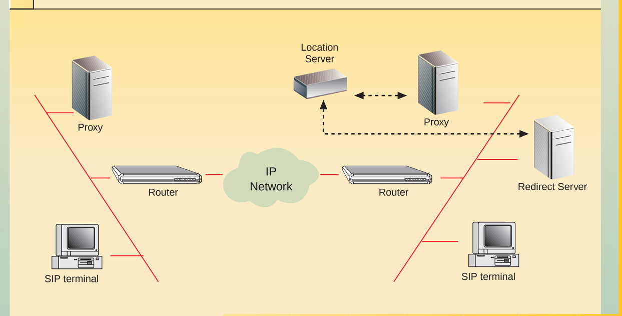
Signalling Gateway: SS7-IP interface, coordinates SS7 view of IP elements and IP view of SS7 elements.

Media Gateway: Terminates PSTN lines and packetizes media streams for IP transport.



SIP Architecture

UAC (user agent client)	Caller application that initiates and sends SIP requests.
UAS (user agent server)	Receives and responds to SIP requests on behalf of clients; accepts, redirects, or refuses calls.
SIP Terminal	Supports real-time, 2-way communication with another SIP entity. Supports both signalling and media, similar to H.323 terminal. Contains UAC.
Proxy	Contacts one or more clients or next-hop servers and passes the call requests further. Contains UAC and UAS.
Redirect Server	Accepts SIP requests, maps the address into zero or more new addresses and returns those addresses to the client. Does not initiate SIP requests or accept calls.
Location Server	Provides information about a caller's possible locations to redirect and proxy servers. May be co-located with a SIP server.



SIP Methods

Command	Function
INVITE	Initiate call
ACK	Confirm final response
BYE	Terminate and transfer call
CANCEL	Cancel searches and "ringing"
OPTIONS	Features support by other side
REGISTER	Register with location service

SIP Response Codes

Response Code Prefix	Function
1xx	Searching, ringing, queuing
2xx	Success
3xx	Forwarding
4xx	Client mistakes
5xx	Server failures
6xx	Busy, refuse, not available anywhere

H.323 Commands

Important H.323 Messages

RAS

Message	Function
RegistrationRequest (RRQ)	Request from a terminal or gateway to register with a gatekeeper. Gatekeeper either confirms or rejects (RCF or RRJ).
AdmissionRequest (ARQ)	Request for access to packet network from terminal to gatekeeper. Gatekeeper either confirms or rejects (ACF or ARJ).
BandwidthRequest (BRQ)	Request for changed bandwidth allocation, from terminal to gatekeeper. Gatekeeper either confirms or rejects (BCF or BRJ).
DisengageRequest (DRQ)	If sent from endpoint to gatekeeper, DRQ informs gatekeeper that endpoint is being dropped; if sent from gatekeeper to endpoint, DRQ forces call to be dropped. Gatekeeper either confirms or rejects (DCF or DRJ). If DRQ sent by gatekeeper, endpoint must reply with DCF.
InfoRequest (IRQ)	Request for status information from gatekeeper to terminal.
InfoRequestResponse (IRR)	Response to IRQ. May be sent unsolicited by terminal to gatekeeper at predetermined intervals.
RAS Timers and Request in Progress (RIP)	Recommended default timeout values for response to RAS messages and subsequent retry counts if response is not received.

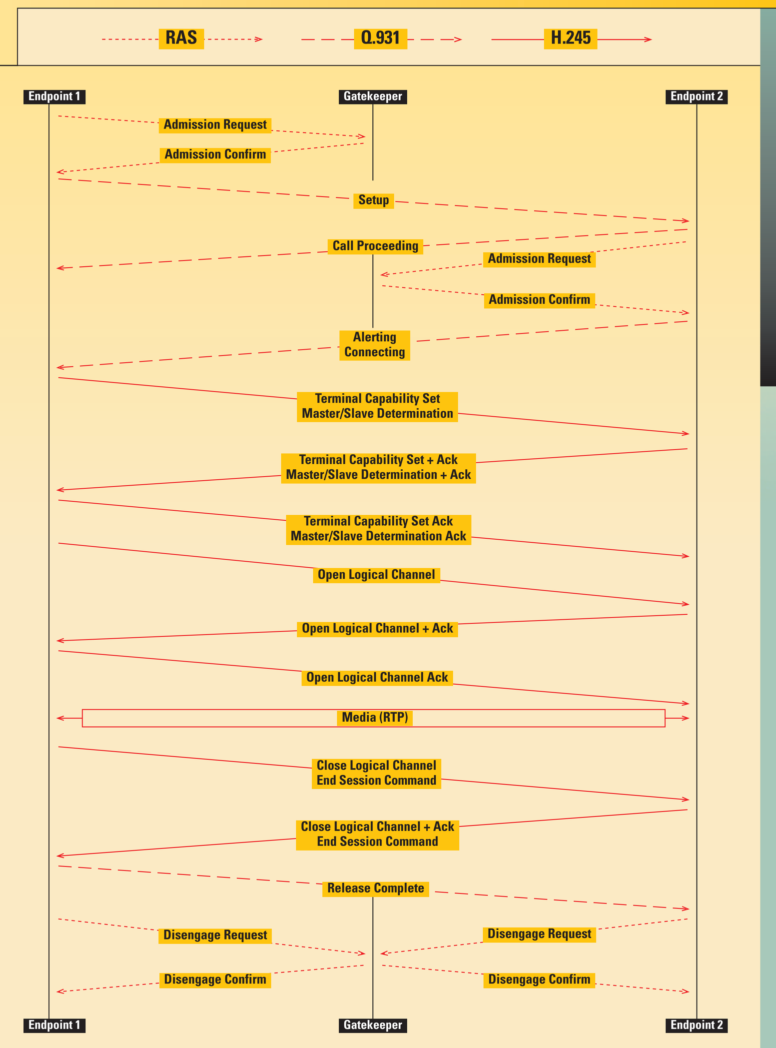
Q.931

Message	Function
Alerting	Called user has been alerted - "phone is ringing". Sent by called user.
Call Proceeding	Requested call establishment has been initiated and no more call establishment information will be accepted. Sent by called user.
Connect	Acceptance of call by called entity. Sent from called entity to calling entity.
Setup	Indicates a calling H.323 entity's desire to set up a connection to the called entity.
Release Complete	Indicates release of call if H.225.0 (Q.931) call signalling channel is open. Afterwards, call reference value can be reused. Sent by a terminal.
Status	Responds to an unknown call signalling message or to a Status Inquiry message. Provides call state information.
Status Inquiry	Requests call status. Can be sent by an endpoint or gatekeeper to another endpoint.

H.245

Message	Function
Master-Slave Determination	Determines which terminal is the master and which is the slave. Possible replies: Acknowledge, Reject, Release (in case of a time out).
Terminal Capability Set	Contains information about a terminal's capability to transmit and receive multimedia streams. Possible replies: Acknowledge, Reject, Release.
Open Logical Channel	Opens a logical channel for transport of audiovisual and data information. Possible replies: Acknowledge, Reject, Confirm.
Close Logical Channel	Closes a logical channel between two endpoints. Possible replies: Acknowledge.
Request Mode	Used by a receive terminal to request particular modes of transmission from a transmit terminal. General mode types include VideoMode, AudioMode, DataMode, and Encryption Mode. Possible replies: Acknowledge, Reject, Release.
Send Terminal Capability Set	Commands the far-end terminal to indicate its transmit and receive capabilities by sending one or more Terminal Capability Sets.
End Session Command	Indicates the end of the H.245 session. After transmission, the terminal will not send any more H.245 messages.

Typical H.323 Call



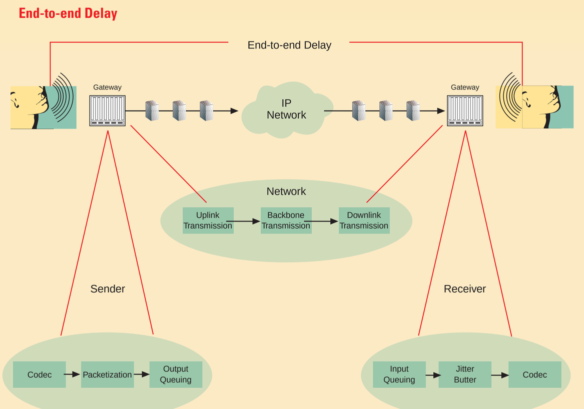
Keys to Testing *Factors Affecting Voice Quality and How to Measure Them*

Delay: Excessive end-to-end delay makes conversation inconvenient and unnatural. Each component in the transmission path - sender, network, and receiver - adds delay. ITU-T G.114 (One-Way Transmission Time) recommends 150 mSec as the maximum desired one-way latency to achieve high-quality voice.

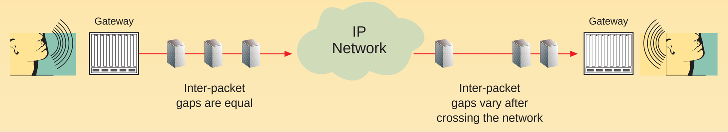
Sample Delay Budget Table

Parameter	Fixed delay	Variable delay
CODEC (G.729)	25 mSec	
Packetization	Included in CODEC	
Queuing delay		Depends on uplink. In the order of a few mSec.
Network delay	50 mSec	Depends on network load.
Jitter buffer	50 mSec	
Total	125 mSec	

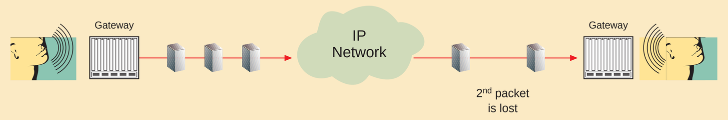
End-to-end Delay



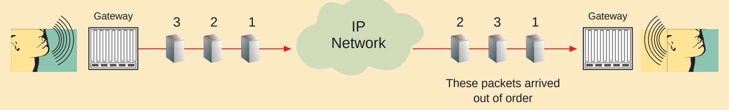
Jitter: Quantifies the effects of network delays on packet arrivals at the receiver. Packets transmitted at equal intervals from the left gateway arrive at the right gateway at irregular intervals. Excessive jitter makes speech choppy and difficult to understand. Jitter is calculated based on the inter-arrival time of successive packets. For high-quality voice, the average inter-arrival time at the receiver should be nearly equal to the inter-packet gaps at the transmitter and the standard deviation should be low. Jitter buffers (packet buffers that hold incoming packets for a specified amount of time) are used to counteract the effects of network fluctuations and create a smooth packet flow at the receiving end.



Packet loss: Typically occurs either in bursts or periodically due to a consistently congested network. Periodic loss in excess of 5-10% of all voice packets transmitted can degrade voice quality significantly. Occasional bursts of packet loss can also make conversation difficult.



Sequence Errors: Congestion in packet switched networks can cause packets to take different routes to reach the same destination. Packets may arrive out of order resulting in garbled speech.



Recommendations for Measuring Voice Quality

ITU-T Recommendation P.800 - Subjective quality test based on Mean Opinion Scores (MOS). Preselected voice samples recorded according to recommendation P.50 are played back to a mixed group of men and women under controlled conditions. The scores given by the group are weighed to give a single MOS score ranging from 1 (worst) to 5 (best). A MOS of 4 is considered "toll-quality" voice.

Mean Opinion Scores (MOS) for Various Voice Quality Tests

Score \ Test Type	Opinion Scale - Conversation Test	Difficulty Scale	Opinion Scale - Listening Test	Listening-Effort Scale	Loudness-Preference Scale
5	Excellent	----	Excellent	Complete relaxation possible, no effort required	Much louder than preferred
4	Good	----	Good	Attention necessary; no appreciable effort required	Louder than preferred
3	Fair	----	Fair	Moderate effort required	Preferred
2	Poor	----	Poor	Considerable effort required	Quieter than preferred
1	Bad	Yes	Bad	No meaning understood with any feasible effort	Much quieter than preferred
0	------	No	---	---	---
Result - average of all participants scores	MOS_C	%D (% difficulty)	MOS	MOS_{LE}	MOS_{LP}

Objective Voice Quality Measurements

ITU-T Recommendation P.861 - Objective Quality Measurement of Telephone Band (300-3400 Hz) Speech Codecs

PAMS - Perceptual Analysis Measurement System (proposal from British Telecom)

Intrusive methods based on comparison of a predefined speech sample before and after transmission through a codec or network. The resulting score approximates MOS scores as would be given by humans under recommendation P.800.

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