Common VoIP Service Quality Thresholds

Minacom Reference Chart

Metric	Abbr	Units	Typical Range: Best / Max	Typical Range: Worst /Min	Severity Levels: Critical	Major	Minor	Warning	Excellent
Speech + Call Quality									
Mean Opinion Score (audio)	MOS	-	5	1	<2.6	<3.2	<3.6	<4	>=4
Mean Opinion Score (IP-Based)	MOS	_	5	1	<2.8	<3.4	<3.7	<4.2	<4.2
R Factor	R	-	100	30	<51	<61	<71	<81	>=81
Speech Distortion	SD	%	0	10	>9.2	>7.5	>6.6	>5.7	<=4.6
Speech Power	SP	dBm	-17.5	-50	<-50 or >15	<-40 or >10	<-35 or >5	<-30 or >0	[-30.5]
DTMF Detection Ratio	-	%	100	0	<70	<80	<90	<100	>=100
Impairments									
Calls with Degradation Ratio	-	%	0	100	>5	>4	>3	>2	<=1
Calls with Distortion Ratio	-	%	0	100	>5	>4	>3	>2	<=1
Calls with Echo Ratio	-	%	0	100	>5	>4	>3	>2	<=1
Calls with Low Volume Ratio	-	%	0	100	>5	>4	>3	>2	<=1
Calls with Noise Ratio	-	%	0	100	>5	>4	>3	>2	<=1
Clipping Ratio - Back	-	%	0	100	>4	>3	>2	>1	<=1
Clipping Ratio - Front	-	%	0	100	>4	>3	>2	>1	<=1
Clipping Ratio - In-Between	-	%	0	100	>4	>3	>2	>1	<=1
C-Message Noise	C-MSG	dBrnC	0	90	>50	>40	>30	>25	<=25
C-Notched Noise	C-Notch	dBrnC	10	90	>26	>25	>24	>22	>20
Echo Path Delay	EPD	ms	0	1000	>400	>350	>250	>150	<=150
Echo Path Loss	EPL	dB	65	0	<22	<30	<40	<55	>=55
Frame Muting Ratio	FMR	%	0	100	>5	>3	>2	>1	<=1
Hangover Duration Average	-	ms	0	1000	>350	>300	>250	>200	<=200
K Factor	K	-	87.6	-53	<2.8	<8	<14	<18	>18
Speech Clipping Ratio	%Clip	%	0	100	>4	>3	>2	>1	<=1
Wideband Noise	VBN	dBrn	0	90	>57	>48	>40	>32	<=32
Call Connectivity									
Answer Seizure Ratio*	ASR	%	100	0	<60	<68	<75	<90	>=90
Call Completion Ratio	CCR	%	100	0	<65	<75	<80	<85	>=85
Call Loss Ratio	CLR	%	0	100	>20	>15	>13	>10	<=10
Call Setup Time	CST	sec	0	180	>25	>15	>11	<7	<7
Dial Tone Delay	DTD	ms	0	3000	>300	>1000	>600	>300	<=300
Network Effectiveness Ratio*	NER	%	100	0	<80	<85	<87	<90	>=100
Post Dial Delay	PDD	sec	0	90	>12	>9	>6	>3	<=3
Ring Duration	RD	sec	0	180	>180	>90	>60	>30	<=30
Internet / Data									
DNS Resolution Success Ratio	-	%	100	0	<80	<90	<95	< 98	>=100
DNS Resolution Time	-	ms	-	-	>500	>400	>300	>100	<=100
Jitter Average	-	ms	0	500	>335	>175	>125	>75	<75
Packet Loss Ratio	PLR	%	0	100	>10	>8	>5	>1	<=1
Roundtrip Delay	RTD	ms	0	2000	>600	>400	>250	>100	<=100
Fax Testing									
Calls at Max. Speed Ratio	-	%	100	0	<60	<75	<80	<90	>=90
Fax Connection Speed	-	kbps	-	-	<7.2	<9.6	<12	<14.4	>=14.4
Pages at Max. Speed Ratio	-	%	100	0	<60	<75	<80	<90	>=90

Trunk Testing	Abbr	Units	Best / Max	Worst /Min	FAIL	PASS
Bit Error Ratio (code 108)	BER	-	<1.74E-8	1	>1E-5	>1E-6
Echo Retun Loss (code 105)	ERL	dB	70	0	<22	<38
Errored Second Ratio (code 108)	ESR	-	0	1	>3.2	>1.2
Gain Slope 1004 (code 105)	GS 1004	dBm	-30	5	>15	<-30
Gain Slope 1004 (code 105)	GS 2804	dBm	-30	5	>15	<-30
Gain Slope 404 (code 105)	GS 404	dBm	-30	5	>15	<-30
Loss (code 102, 105)	-	dB	-30	15	>15	<30
Severely Errored Sec. Ratio (code 108)	SESR	-	0	1	>4E-2	>1E-2
Singing Return Loss (code 105)	SRL	dB	70	0	<15	<35

***Note:** values shown represent typical service level class values for active test calls placed by Minacom test systems. They do not reflect performance levels of actual network traffic as normally measured with passive test systems or as typically reported in call data records (CDRs).

● IP/Packet-Based Measurements ● Analog Signal Analysis

Category	Measurement	Type	Abbrev.	Description
Speech + Call Quality	Mean Opinion Score	IP, Analog	MOS	see sidebar at right
	R Factor	IP, Analog	R	The R Factor is the primary output of the E-Model, a figure that aggregates many aspects of telephony transmission performance. R Factor is representative of a user's perceived conversational performance of a voice call, and can be used to calculate MOS.
	Speech Distortion	Analog	SD	Normalized metric - indication of magnitude of "unnatural" sounds not originally spoken by the caller.
	Speech Power	Analog	SP	The average power of a speech signal after inter-syllabic blanks have been removed.
	DTMF Detection Ratio	Analog	-	Percentage of DTMF digits (0,1,2,3,...,*,#, etc.) properly transmitted.
Impairments	Calls with Degradation Ratio	Analog	-	% of calls with poor voice quality, as measured by MOS and the unsatisfied user ratio.
	Calls with Distortion Ratio	Analog	-	% of calls with Speech Distortion classified other than "excellent"
	Calls with Echo Ratio	Analog	-	% of calls with echo not classified as "excellent"
	Calls with Low Volume Ratio	Analog	-	% of calls with volume lower than the "excellent" service level.
	Calls with Noise Ratio	Analog	-	% of calls with noise greater than the "excellent" service level.
	Clipping Ratio - Back	Analog	-	% of speech subject to back-end clipping (end of words clipped). Clipping is defined as the duration of lost speech samples.
	Clipping Ratio - Front	Analog	-	% of speech subject to front-end clipping (beginning of words).
	Clipping Ratio - In-Between	Analog	-	% of speech subject to in-between clipping (between words).
	C-Message Noise	Analog	C-MSG	The noise on an idle channel or circuit (1004 Hz holding tone), measured with C-message weighting.
	C-Notched Noise	Analog	C-Notch	As C-MSG but using a 1010Hz notch filter in tandem; measured at the near end.
	Echo Path Delay	Analog	EPD	The duration between a transmitted signal and its reflection.
	Echo Path Loss	Analog	EPL	The difference in signal strength between a transmitted signal and its reflection (expressed in dB). EPL is EPD dependent.
	Frame Muting Ratio	Analog	FMR	Frame muting rate during speech excluding front-end clipping - i.e. long clipping.
	Hangover Duration Average	Analog	-	Average VAD hang-over time (duration of a sound not transmitted due to VAD delay).
	K Factor	Analog	K	Index of call quality evaluated by the VQES algorithm.
	Speech Clipping Ratio	Analog	%Clip	% of clipping in a call.
	Wideband Noise	Analog	VBN	Noise level on a "silent" wideband channel.
Call Connectivity	Answer Seizure Ratio		ASR	The proportion of the total number of seizures that result in an answer signal (Ref. ITU-T E.425).
	Call Completion Ratio		CCR	Percentage of calls answered vs. total call attempts. Does not include calls terminated by busy or No Answer signals.
	Call Loss Ratio		CLR	Proportion of calls that fail to establish a connection due to network congestion / faults.
	Call Setup Time		CST	The time between circuit seizure and the receipt of a call-connected or service signal.
	Dial Tone Delay		DTD	The time between off-hook and reception of a dial tone (Ref. ITU-T E.431).
	Network Effectiveness Ratio		NER	% of calls successfully completed. Unlike CCR, NER includes calls terminated by Busy and No Answer signals.
	Post Dial Delay		PDD	Time interval between dialing and an appropriate network response.
	Ring Duration		RD	Time between the first Ringback and the Answer signal.
Internet / Data	DNS Resolution Success Ratio	IP	-	% of DNS resolution requests receiving a valid response.
	DNS Resolution Time	IP	-	Delay between the request for IP address for a domain-based Internet address and a valid response from a Domain Name Server.
	Jitter Average	IP	-	Average short-term variations of the significant instants of a packet from their ideal positions in time.
	Packet Loss Ratio	IP	PLR	% of lost packets vs. the total number transmitted.
	Roundtrip Delay	IP	RTD	The sum of the absolute delays on an outgoing and return path.
Fax Testing	Bit Rate Compatibility		BR Cap.	Maximum line rate a fax machine supports.
	Calls at Max. Speed Ratio		-	% of calls a fax machine receives at its (max) bit rate.
	Fax Connection Speed		-	The negotiated connection bit rate of a fax transmission.
	Pages at Max. Speed Ratio		-	% of pages transmitted at max possible speed; speed is renegotiated between pages.
Trunk Testing	Bit Error Ratio (code 108)	Analog	BER	% of erroneous bits.
	Echo Retun Loss (code 105)	Analog	ERL	The frequency weighted average of the return loss over the middle of the voice band.
	Errored Seconds Ratio (code 108)	Analog	ESR	% of Errored Seconds in a test call. Errored Seconds are the number of one second intervals which contain at least one error block. (Ref. ITU G.826).
	Gain Slope 1004 (code 105)	Analog	GS 1004	Difference of levels between a facilities output compared to input power at available at 1004Hz.
	Gain Slope 1004 (code 105)	Analog	GS 2804	Difference between levels at 2804Hz and 1004 Hz measured in (dB) from the transmitting end. Positive values indicate more loss than the 1004 Hz value.
	Gain Slope 404 (code 105)	Analog	GS 404	Difference between levels at 404Hz and 1004 Hz measured in (dB) from the transmitting end.
	Loss (code 102, 105)	Analog	-	Difference in power measured at a facilities output compared to power available to the input.
	Severely Errored Seconds Ratio (code 108)	Analog	SESR	% of Severely Errored Seconds to total test-call duration. Severely Errored Seconds are the number of one second intervals which contain at least 30% of blocks with errors. (Ref. ITU G.826).
	Singing Return Loss (code 105)	Analog	SRL	The rerturn loss of a circuit measured using two separately transmitted signals with a flat spectral distribution between 3dB frequencies of 260Hz and 500Hz, measured at the near end.

MOS Explained

Overview

Mean Opinion Score (MOS) is commonly used to rate phone service speech quality, expressed on a scale from 1 to 5. A score of 4 is normally considered "Toll Quality" for calls placed over PSTN/TDM networks. MOS is a function of many factors, including the type of network and codecs used, wiring and premises equipment, and even the handset used to place the call.

MOS was originally determined using subjective listening tests, where a panel of trained judged recorded speech samples to assign a score. Test equipment calculates MOS using sophisticated algorithms that are designed to closely approximate the results of subjective listening tests.

MOS Algorithms

There are a number of industry-standard MOS calculation algorithms, each originally designed for a specific application. Some algorithms consider only packet-based (IP) statistics, whereas others include analog measurements such as noise, volume, echo, and distortion to enhance accuracy and repeatability. MOS scores calculated using both analog & IP measurements most closely approximate a real caller's perception of speech quality.

Analog / IP MOS Algorithms

PESQ ITU-T P.862

Calculates Listening Quality (LQ) and Conversational Quality (CQ) MOS reflecting the perceived voice quality of a call. PESQ compares an original reference speech sample with a received, degraded speech sample, measuring the effects of one-way speech distortion and noise on speech quality. CQ MOS incorporates the effect of delay and echo on conversations, while LQ MOS does not.

P.563 Listening MOS

P.563 is an extension to the PESQ P.562 algorithm that allows it to calculate MOS non-intrusively from actual speech samples or call recordings, suitable for single-ended testing. Includes the effects of noise, echo, delay, clipping, frame mutes, packet-loss, codec and network type.

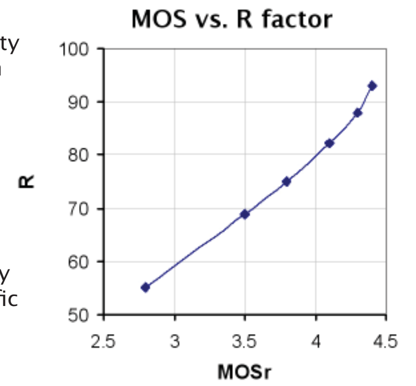
VQES Algorithm

A statistics-based algorithm that estimates end-user satisfaction and perceived voice quality of a call ; includes the effects of low volume (speech power), noise, distortion, echo and delay. Measured probe-to-probe using a predefined natural speech reference sample.

E-Model G.107, 108, 109

The E-Model rates conversational quality with a metric called the R-factor, which can be converted into an approximate MOS rating.

The E-Model was originally designed to measure the effects delay, echo, and distortion created by codecs have on speech quality. Vendor-specific extensions have increased the accuracy of the E-Model by including VoIP-specific IP & analog measurements in R factor calculation.



IP / Packet-Based MOS

RTCP-XR

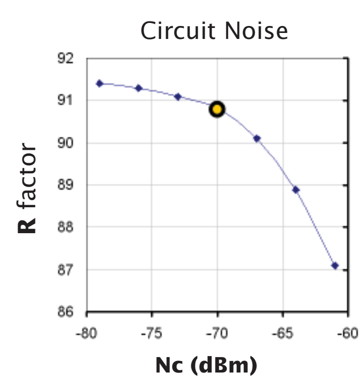
A VoIP Quality measurement standard defined by the IETF in RFC 3611 that estimates MOS and other QoS metrics from packet-loss, jitter and delay.

E-Model (IP-only implementation)

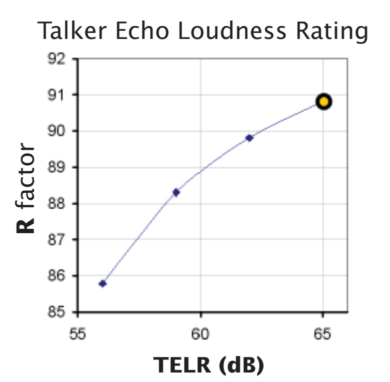
The E-Model algorithm can be usedwith IP only measurements. "Default" values are substituted for the missing analog measurements. The graphs below show how IP/Packet-based MOS can exhibit significant error when actual analog measurements are not available.

Key Analog Measurements Affecting E-Model MOS

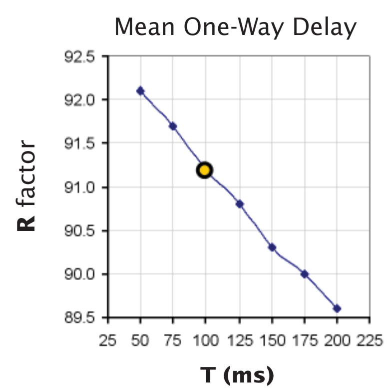
● ITU Default



valid range: -80 to -40 dBm



valid range: 5 to 65 dB



valid range: 0 to 500 ms