Technical Specification
for the OPERA™ Objective Perceptual Analyzer
OPR-1XX-XXX-P

Feature Overview

Voice Quality Measurement

- PESQ, perceptual evaluation of speech quality according to ITU-T rec. P.862 [2001]. Combines PSQM with PAMS, optimized for VoIP and hybrid end-to-end applications
- PSQM, perceptual speech quality measure according to ITU-T rec. P.861 [1996]
- PSQM+, advanced perceptual speech quality measure according to ITU-T COM 12-20 [1998]
- PSQM/IP, perceptual speech quality measure according to ITU-T rec. P.861 [1996], incl. advanced delay compensation for end-to-end measurements, developed by OPTICOM
- Echo measurement based on real speech, Echo Return Loss ERLmom, ERLpeak, ERL vs. delay
- One way delay measurement, with PESQ also delay jitter vs.time
- Measurement under double talk conditions
- One way and two way measurements
- Bulk call generation
- H.323 interface functionality using Windows NetMeeting™ and audio interfaces
- Front-end clipping FEC measurement[1] (PESQ)
- Hold-over time HOT measurement[1] (PESQ)
- Background noise measurement[1] (PESQ)
- Utterances detection[1] (PESQ)
- Delay histogram[1] (PESQ)
- Signal-to-noise SNR measurement of speech and silence portions of a test signal
- User definable state machine support for E1 MFC-R2, E1 CAS-R2, T1 CAS
- Official ITU voice samples for testing and conformance verification

Audio Quality Measurement

- PEAQ, perceptual evaluation of audio quality according to ITU-R rec. BS.1387 [1999], Basic and Advanced Model
- Result logging functionality, including user definable log intervals and quality score threshold
- Official ITU audio samples for testing and conformance verification

Common Functionality

- Advanced delay compensation for end-to-end measurements, suitable for constant and variable delays (VoIP), developed by OPTICOM
- Delay measurement with real speech/music. Results in samples as well as ms
- Attenuation measured in dB

[1] Available Q4/2001

- On-line real time acquisition and analysis capability (PEAQ and PSQM)
- Comprehensive scripting interface
- Remote Control via IP
- DDLC™ improved timing accuracy (Dynamic Driver Latency Compensation)
- Play and record functionality for wide band audio and voice signals
- Trigger functionality for targeted analysis of on-line or long duration test signals (PEAQ and PSQM)
- Graphical user interface
- Extensive scripting and command line options for automated testing
- Printing and graph exporting functionality

Graphical Displays

- Time signal displays, zoomable
- FFT Spectra displays, zoomable
- Excitation and various psychoacoustic parameters, like NMR, Masked Threshold and Loudness
- Audible distortion measurement
- Result summary display with MOV, PSQM MOS, PESQ MOS and PEAQ ODG numerical scores and bar graphs
- Delay Jitter vs. Time min/max scores and graph (PESQ)
- Echo Return Loss vs. Delay

Feature Description

PESQ Measurement

Description:	Perceptual measurement of speech quality according to ITU-T rec. P.862. Indicates call clarity. Results can be directly compared to subjective listening tests. Capable of handling varying delays and severe distortions as they occure in VoIP applications.
Algorithm standard:	Based on standard ITU-T rec. P.862
Sample rates:	8 kHz and 16 kHz
Sound file formats:	WAVE-files containing A-law, mu-law, linear PCM (8 or 16 bit)
Max. duration of measurement signals:	As defined by WAVE-format
Gain Compensation:	Maximum gain difference = ±60 dB
Automated operation:	Supported

Available Measurement Results:	Timesignal, Delay vs. Time, PESQ MOS, Jitter, Minimum, maximum and average delay, Front-end clipping FEC[1], Hold-over time HOT[1], Attenuation[1], Background noise[1], Signal-to-noise (SNR) measurement, MOS of silent and speech portions, Detailed analysis of silent and speech portions, Utterances detection[1], Delay histogram[1]
Additional Features:	File-based measurements, Printing and graph exporting functionality, Includes official ITU voice samples for testing, Includes PESQ conformance test samples, Includes PSQM conformance test samples

PSQM Measurement

Description:	Perceptual measurement of speech quality according to ITU-T rec. P.861. Indicates call clarity. Includes highly optimized proprietary mapping from raw PSQM values to P.800 Mean Opinion Score (MOS) scale. Results can be directly compared to subjective listening tests.
Algorithm standard:	Based on standard ITU-T rec. P.861
Sample rates:	8 kHz and 16 kHz
Sound file formats:	WAVE-files containing A-law, mu-law, linear PCM (8 or 16 bit)
Max. duration of measurement signals:	File-based: As defined by WAVE-format On-line: $2^{32} / (fs * (bitspersample/8)$ (e.g. ~12.4 hours at 48 kHz, 16 bit resolution)
Gain Compensation:	Maximum gain difference = ±60 dB
Automatic Delay Compensation:	±1000 ms
Static Delay:	Automatic Delay Compensation ±10 s
Automated operation:	Supported
Available Measurement Results:	Timesignal, Spectrum, Excitation, Percentage of silent intervals during measurement (Silence),

[1] Available Q4/2001

	Percentage of time clipped frames during measurement (Time Clipped), Percentage of severely distorted frames during measurement (Sev. Distorted), PSQM according to P.861, silence weight = 0.0 (PSQM-W0), PSQM according to P.861, silence weight = 0.2 (PSQM-W2), PSQM according to P.861, silence weight = 0.4 (PSQM-W4), PSQM value of the silent intervals (PSQM-Silence), Mean Opinion Score (MOS), silence weight = 0.0 (OMOS-W0), Mean Opinion Score (MOS), silence weight = 0.2 (OMOS-W2), Mean Opinion Score (MOS), silence weight = 0.4 (OMOS-W4), Mean Opinion Score (MOS) vs. Time, Delay between the Reference Signal and the output signal of the device under test, Attenuation of the test signal compared to the reference signal
Additional Features:	File-based measurements, On-line measurements, Trigger functionality, Printing and graph exporting functionality, Includes official ITU voice samples for testing, Includes PESQ conformance test samples, Includes PSQM conformance test samples

PSQM+ Measurement

Description:	Perceptual measurement of speech quality according to ITU-T rec. P.861. Indicates call clarity. Includes highly optimized proprietary mapping from raw PSQM values to P.800 Mean Opinion Score (MOS) scale. Results can be directly compared to subjective listening tests. PSQM+ is optimzed for severe distortions, such as packet loss and time clipping.
Algorithm standard:	Based on standard ITU-T rec. P.861
Sample rates:	8 kHz and 16 kHz
Sound file formats:	WAVE-files containing A-law, mu-law, linear PCM (8 or 16 bit)
Max. duration of measurement signals:	File-based: As defined by WAVE-format On-line: $2^{32} / (fs * (bitspersample/8)$ (e.g. ~12.4 hours at 48 kHz, 16 bit resolution)
Gain Compensation:	Maximum gain difference = ±60 dB

Automatic Delay Compensation:	±1000 ms
Static Delay:	Automatic Delay Compensation ±10 s
Automated operation:	Supported
Available Measurement Results:	Timesignal, Spectrum, Excitation, Percentage of silent intervals during measurement (Silence), Percentage of time clipped frames during measurement (Time Clipped), Percentage of severely distorted frames during measurement (Sev. Distorted), PSQM according to P.861, silence weight = 0.0 (PSQM-W0), PSQM according to P.861, silence weight = 0.2 (PSQM-W2), PSQM according to P.861, silence weight = 0.4 (PSQM-W4), PSQM value of the silent intervals (PSQM-Silence), Mean Opinion Score (MOS), silence weight = 0.0 (OMOS-W0), Mean Opinion Score (MOS), silence weight = 0.2 (OMOS-W2), Mean Opinion Score (MOS), silence weight = 0.4 (OMOS-W4), Mean Opinion Score (MOS) according to PSQM+ (OMOS+), Mean Opinion Score (MOS) vs. Time, Delay between the Reference Signal and the output signal of the device under test, Attenuation of the test signal compared to the reference signal
Additional Features:	File-based measurements, On-line measurements, Trigger functionality, Printing and graph exporting functionality, Includes official ITU voice samples for testing, Includes PESQ conformance test samples, Includes PSQM conformance test samples

PSQM/IP Measurement

Description:	Same as PSQM and PSQM+. Includes advanced time alignment algorithm to cope with static and variable latencies, developed by OPTICOM (Delay Tracking functionality).
Algorithm standard:	Based on standard ITU-T rec. P.861
Sample rates:	8 kHz and 16 kHz

Sound file formats:	WAVE-files containing A-law, mu-law, linear PCM (8 or 16 bit)
Max. duration of measurement signals:	File-based: As defined by WAVE-format On-line: $2^{32} / (fs * (bitspersample/8)$ (e.g. ~12.4 hours at 48 kHz, 16 bit resolution)
Gain Compensation:	Maximum gain difference = ±60 dB
Automatic Delay Compensation:	±1000 ms
Static Delay:	Automatic Delay Compensation ±10 s
Delay Tracking:	±512 samples
Automated operation:	Supported
Available Measurement Results:	Timesignal, Spectrum, Excitation, Percentage of silent intervals during measurement (Silence), Percentage of time clipped frames during measurement (Time Clipped), Percentage of severely distorted frames during measurement (Sev. Distorted), PSQM according to P.861, silence weight = 0.0 (PSQM-W0), PSQM according to P.861, silence weight = 0.2 (PSQM-W2), PSQM according to P.861, silence weight = 0.4 (PSQM-W4), PSQM value of the silent intervals (PSQM-Silence), Mean Opinion Score (MOS), silence weight = 0.0 (OMOS-W0), Mean Opinion Score (MOS), silence weight = 0.2 (OMOS-W2), Mean Opinion Score (MOS), silence weight = 0.4 (OMOS-W4), Mean Opinion Score (MOS) according to PSQM+ (OMOS+), Mean Opinion Score (MOS) vs. Time, Delay between the Reference Signal and the output signal of the device under test, Attenuation of the test signal compared to the reference signal
Additional Features:	File-based measurements, On-line measurements, Trigger functionality, Printing and graph exporting functionality, Includes official ITU voice samples for testing,

	Includes PESQ conformance test samples, Includes PSQM conformance test samples

Echo Algorithm

Description:	Algorithm using real speech as stimulus for calculating echo. Measures the Echo return Loss (ERL) in terms of attenuation and delay values. Includes graphical representation of the ERL vs. Delay.
Sample rates:	8 kHz and 16 kHz
Sound file formats:	WAVE-files containing A-law, mu-law, linear PCM (8 or 16 bit)
Max. duration of measurement signals:	As defined by WAVE-format
Gain Compensation:	Maximum gain difference = ±60 dB
Maximum echo delay:	1000 ms
Frame size:	16 ms at 8 kHz
Averaging window size:	800 ms at 8 kHz
Automated operation:	Supported
Available Measurement Results:	Timesignal, Echo Return Loss (ERL) vs. Delay, Momentary attenuation of the highest echo peak (ERLmom), Momentary delay of the highest echo peak (ERLmom Delay), Attenuation of the highest echo peak during the whole measurement period (ERLpeak), Delay of the highest echo peak during the whole measurement period (ERLpeak Delay)
Additional Features:	File-based measurements, Printing and graph exporting functionality, Includes official ITU voice samples for testing, Includes PESQ conformance test samples, Includes PSQM conformance test samples

PEAQ Algorithm

Description:	Perceptual measurement of wide band audio according to ITU-R rec. BS.1387. Includes the basic version of PEAQ, defined for computational efficiency and the advanced version yielding for highest possible accuracy. Detailed analysis of test signals through BS.1387 model output variables (MOVs).
Algorithm:	Based on standard ITU-R rec. BS.1387
Sample rates:	48 kHz (according to recommendation ITU-R BS.1387), 44.1 kHz

Sound file formats:	WAVE-files containing A-law, mu-law, linear PCM (8 or 16 bit)
Max. duration of measurement signals:	File-based: As defined by WAVE-format On-line: $2^{32} / (fs * (bitspersample/8)$ (e.g. ~12.4 hours at 48 kHz, 16 bit resolution)
Gain Compensation:	Maximum gain difference = ±60 dB
Automatic Delay Compensation:	±1000 ms (File-based measurements), ±500 ms (On-line measurements)
Static Delay:	Automatic Delay Compensation ±10 s
Delay Tracking:	±512 samples
Automated operation:	Supported
Available Measurement Results:	Timesignal, Spectrum, Excitation, Noise-to-Mask Ratio (NMR), averaged, Noise-to-Mask Ratio (NMR) vs. Time, Masked Threshold, Loudness, Objective Difference Grade (ODG), averaged (Advanced and Basic Version), Objective Difference Grade (ODG) vs. Time, Distortion Index (DI), for the Advanced and the Basic Version, Delay between the Reference Signal and the output signal of the device under test, Attenuation of the test signal compared to the reference signal, Modulation of the reference and test signal
BS.1387 intermediate results (MOVs):	Average Bandwidth of the Reference Signal (AvgBwRef), Average Bandwidth of the output signal of the device under test (AvgBwTst), Total Noise-to-Mask Ratio (NMRtotB), Relative fraction of frames for which at last one frequency band contains a significant noise component (RDF), Average Distorted Block (=Frame), taken as the logarithm of the ratio of the total distortion to the total number of severely distorted frames (ADB), Maximum of the Probability of Detection after low pass filtering (MFPD), Harmonic structure of the error over time (EHS), Windowed averaged difference in modulation (envelopes) between Reference Signal and Signal Under Test (WinModDif1B), Averaged modulation difference (AModDif1B),

	Averaged modulation difference with emphasis on introduced modulations and modulation changes where the reference contains little or no modulations (AmodDif2B, RModDifA), RMS value of the averaged noise loudness with emphasis on introduced components (NloudB, NLA), Averaged Linear Distortions (ALD)
Additional Features:	File-based measurements, On-line measurements, Trigger functionality, Result logging option, including user definable log intervals and ODG threshold, Printing and graph exporting functionality, Includes official ITU audio samples for testing, Includes PEAQ conformance test samples

OptiCall™ Application

Description:	Data acquisition tool that supplies test files for file-based measurement analysis with PESQ, PSQM, PSQM+, PSQM/IP, Echo and PEAQ algorithms. Capable of playing/recording WAVE files through all available interfaces, i.e. POTS, E1/T1 and audio interfaces, in any combination. One way and two way measurement, as well as double talk is supported. OptiCall™ controls data acquisition on the local host, as well as on remote OPERA™ systems.
Sample rates:	8 kHz (POTS interface, E1/T1 interface), 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz (audio interface)
Sound file formats:	8 bit G.711 A-law, mu-law (POTS interface, E1/T1 interface), plain PCM (audio interface)
Automated operation:	Supported
Additional Features:	One way and two way data acquisition, Double talk data acquisition, Bulk call generation, H.323 interface functionality using Netmeeting™ and audio interfaces, Remote control via IP, DDLC™ improved timing accuracy (Dynamic Driver Latency Compensation), Play and record functionality for wide band audio and voice signals

Interfaces

POTS Telephony Interface

General:	
Board Capacity:	4 analog 2 wire loop start interfaces.
Power / Environment:	
Power requirements +5V/+12V/-12V:	0.8A (typ. 0.3A)/<0.1A/<0.1A
Operating: temp / humidity (noncondensing):	0°C/+50°C 5%/80%, non-condensing
Storage : temp / humidity (noncondensing):	-20°C/+70°C 5%/80%, non-condensing
Loop Parameters:	
Impedance:	Country specific, linear (600W or 900W) or complex (Europe: according to ETSI TBR 21)
Audio Signal Processing:	
Receive Range:	-50 to +0 dBm (optional configurable AGC above nominal -44 dBm).
Transmit:	Programmable (nominal –12 dBm).
Silence Detection:	Programmable (nominal –44 dBm).
Sampling Rates:	8 ksamples/sec (telephone industry standard).
Speech:	64 kbps m-law or A-law per ITU-T G.711, 16, 24, 32, 40 kbps ADPCM using ITU-T G.726 algorithm, 16, 24, 32 kbps NMS compatible ADPCM, 32 kbps VOX compatible ADPCM, 8, 16 bit PCM 11, 22, 44 kHz, 16 bit mono PCM 8 kHz
Audio Output:	
Frequency:	300-3400 Hz
Impedance:	100 W
Level:	3 dB
Output Connector:	3.5 mm stereo jack
Audio Input:	
Frequency:	300-3400 Hz
Impedance:	47 kW
Level:	2 Vcc peak-to-peak
Output Connector:	3.5 mm stereo jack
Tone Dialling:	
DTMF Digits:	0-9,*,#, and ABCD per ITU-T Q.23 and Q.24
Rate:	Programmable (10 digits/sec nominal)
Dialling Parameters:	Software controllable
Dialling Amplitude:	Network compatible programmable range -33 dBm to 1dBm
Dial Tone Wait:	Wait-for-dial-tone capability

Audio Interface

Analog:	
Interface Type:	Two inputs and two outputs, cross-coupled electronically balanced, XLR connectors on audio cables

Level:	+4dBu nominal / +20dBu max or –10dBV nominal / +6dBV max, 600W load on outputs
Input impedance:	Balanced: 24kW, unbalanced 12kW
Output impedance:	Balanced: 100W, unbalanced 50W
Output drive capability:	600W impedance, 0.16mF capacitance
A/D and D/A converters:	24bit, 128x oversampling, sigma delta
Bit depth:	8, 16, 24 or 32 bit file types
Frequenzy response:	20Hz..20kHz, +0/-0.35dB
Dynamic range:	>103dB, A wtd., analog in to analog out
Signal to Noise Ratio:	>99dB, A wtd., analog in to analog out
Channel crosstalk:	<-103dB, analog in to analog out, 1kHz signal @ -1dBFS
Input THD+N:	0.0022%typ., 1kHz signal @ -1dBFS, 22Hz..22kHz BW, analog in to digital out
Output THD+N:	0.0015%typ., 1kHz signal @ -1dBFS, 22Hz..22kHz BW, digital in to analog out
Digital:	
Interface Type:	One input and one output / AES/EBU or S/P DIF format, transformer coupled, XLR connectors on audio cables
Sample rates:	32kHz, 44.1kHz, 48kHz, 88.2kHz, 96kHz
Bit depth:	8, 16, 24 or 32 bit file types
Clock / Synchronisation:	
Type:	External BNC input and output, internal clock or board to board synchronisation through internal header cable
External: Level / impedance:	TTl / 75W
External: Input frequency range:	25kHz to 27MHz

E1/T1 Trunk Interface

Capacity:	Up to 60 independent digital voice ports (Dual-port version), Up to 120 independent digital voice ports (Quad-port version).
Voice Compression:	G.723.1 MP-MLQ @ 6.3 kbps / ACELP @ 5.3 kbps, G.729A CS-ACELP @ 8 kbps, G.711 PCM @ 64 kbps µ-law/A-law, G.726/G.727 16 to 40 kbps ADPCM and E-ADPCM, NetCoder®@ 4.8 to 9.6 kbps, 800 bps increments, GSM 6.10 @ 13 kbps
Silence Suppression:	G.729 Annex B; G.723.1 Annex A; NetCoder® PCM and ADPCM – Proprietary (AudioCodes) VAD and CNG, GSM 6.10
Echo Cancellation:	G.168, 30 msec
Gain Control:	Programmable
VoIP IA Compliance:	DTMF: mute, transfer in coder or in RTP payload RTP/RTCP per RFC 1889/1890

In-band Signaling	DTMF (TIA 464B), MF R1/R2, MFC, AC15, SS-4, SS-5, Call progress tones generation and detection
E1/T1 Interface Trunk:	RJ-48 Connector, 120 Ohm
Supported PSTN Protocols:	European ISDN PRI, North American ISDN, E1 CAS/MFR2 Protocols, T1 CAS/Robbed Bit Protocols.
Packet interface:	On-board NIC or PCI; Selectable per port
Ethernet:	10/100 Base-T, RJ-45 (used by card for RTP/RTCP)
Control Processor:	Motorola PowerQUICC MPC860
Core DSP:	AudioCodes AC48105A-C VoIP DSP
Maximum Power:	3.0 A @ 5V without E1/T1, 3.6 A with E1/T1
Physical:	Full length, 32-bit PCI card, Rev 2.1, slave, 33 MHz

System Hardware Description

OPERA Portable PC System

DC Power Supply:

Wattage:	400 W
Voltage:	90 to 135 V at 60 Hz; 180 to 265 V at 50 Hz Autoranging 90 to 265 V

Backup Power Supply:

Backup battery:	3-V CR2032 coin cell

Temperature:

Operating:	10° to 35°C°* (50° to 95°F)
Storage:	-40° to 65°C (-40° to 149°F)
Relative humidity:	20% to 80% (noncondensing)

Regulatory:

This device complies with part 15 of the FCC Rules.