PRODUCT DATA

Voice Quality Evaluation System

PULSE™ LabShop Software for Voice Quality Evaluation System BZ-5833

The Voice Quality Evaluation System (V-Quest) provides a new approach to voice testing of communication equipment. This new approach allows, under fully controlled conditions, noise suppression and other speech enhancement techniques to be evaluated using real speech signals and environmental noises as background sound.

Parameters such as PESQ (perceptual evaluation of speech quality), SNRI (signal-to-noise ratio improvement), NPLR (noise power level reduction), Convergence Time and others are calculated from time domain information. These parameters provide useful information regarding the communication equipment's speech and noise processing performance.

V-Quest is a powerful tool in the development process of communication equipment. The PULSE platform, which is the foundation for V-Quest, also provides a wide range of analysis capabilities for the verification of the acoustic design during development process, and is a versatile platform for objective evaluation of specific acoustical components such as receivers and microphones.

Uses and Features

Uses

- Research and development with focus on using real speech and real background noises for voice quality evaluation
- Voice quality evaluation of mobile phones, headsets and hands-free adaptors
- Voice quality evaluation of VoIP phones
- Measurement of the complete transmit signal path

Features

- Pre-programmed test suite for voice quality evaluation
- Test suites that require a minimum of operator interaction
- Supports the use of Head and Torso Simulator (HATS)
- Calculation of PESQ, SNRI, NPLR, Convergence, etc.
- Test of devices in normal operating mode, without turning features on or off
- Embedded library of real speech and sounds
- Automatic equalization and calibration of loudspeakers and HATS mouth
- Allows subjective evaluation by listening to recorded signals
- Automatic report generation

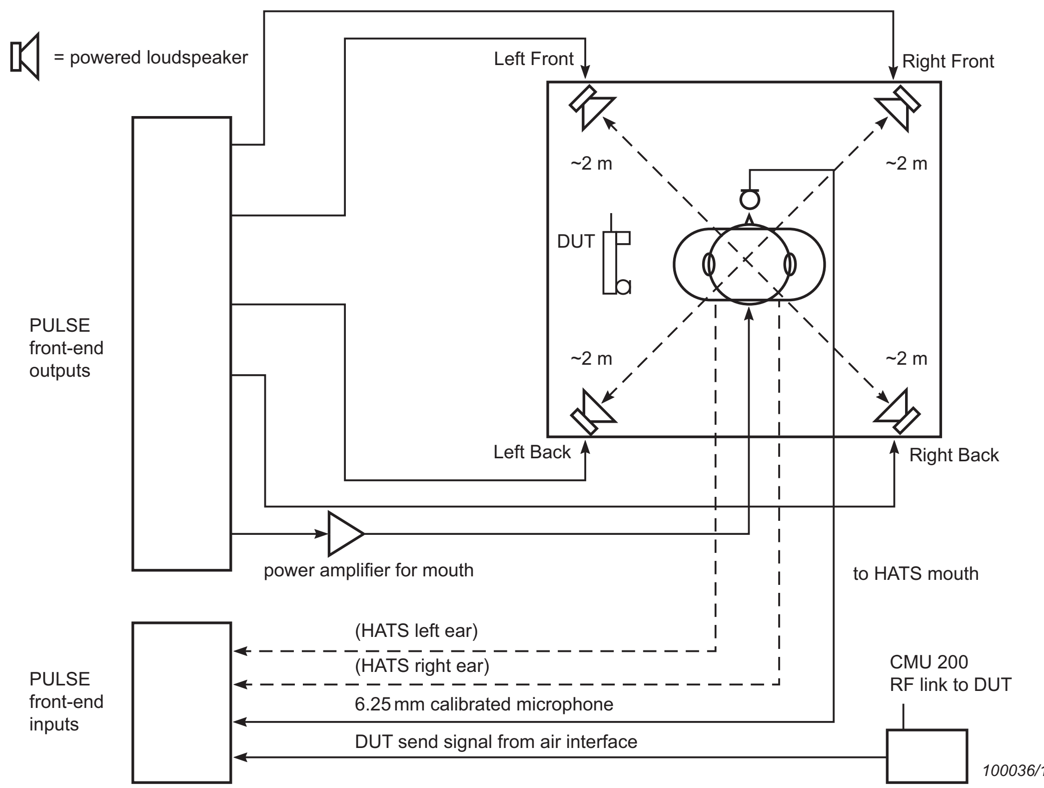
Brüel & Kjær

Introduction

V-Quest is designed for evaluating the acoustical transmission performance of mobile phones in noisy environments. The system can also be used to test hands-free devices or headsets used with mobile phones.

The system shown in Fig. 1 provides an artificial noise field for measuring the noise suppression capabilities of a mobile phone. It also supports a thorough calibration procedure that automatically verifies that the measurement microphones, HATS and the loudspeakers are operating within specifications.

Fig. 1
Overview of Voice Quality Evaluation System



All tasks are easily selected and executed automatically within the software. The software tasks can be customized according to individual need.

Performing Measurements

The software can test a phone as it is, without any control of the noise suppression algorithm. The software does this by making a single measurement which collects signals from two reference microphones and the device tested, all at the same time.

The microphones are placed at the opening of the HATS mouth, the MRP, and near the microphone(s) or the device under test. Measurements made with this setup reflect both the noise suppression features of the device and its position and distance from the mouth.

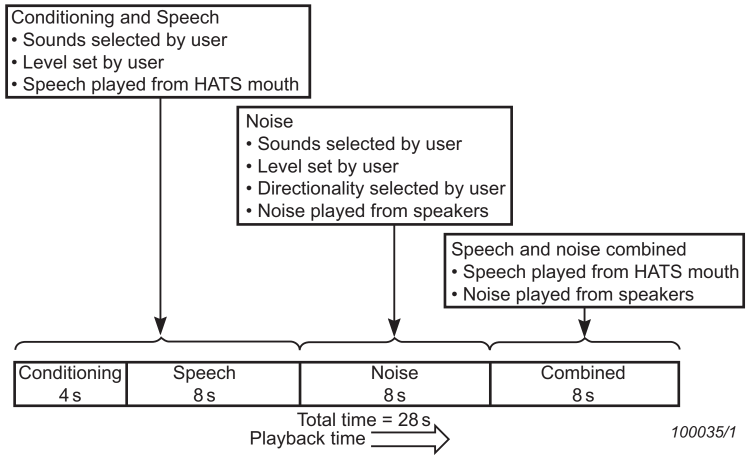
A second method, designed for situations where the noise suppression algorithm can be controlled by the operator, is based on two measurements using one reference microphone – one measurement with the noise suppression off and one with the noise suppression on.

Both methods use the same pattern of excitation signals (see Fig. 2) and enable the same calculations to be performed.

A wide range of speech samples and noise samples are available and can be selected in various combinations. For more specific information on what sound samples are supported, the full lists of sound samples are found in the section “References to Standards and Recommendations”.

PULSE Sequence Player controls the measurement and calibration. It starts and stops mobile phone measurements.

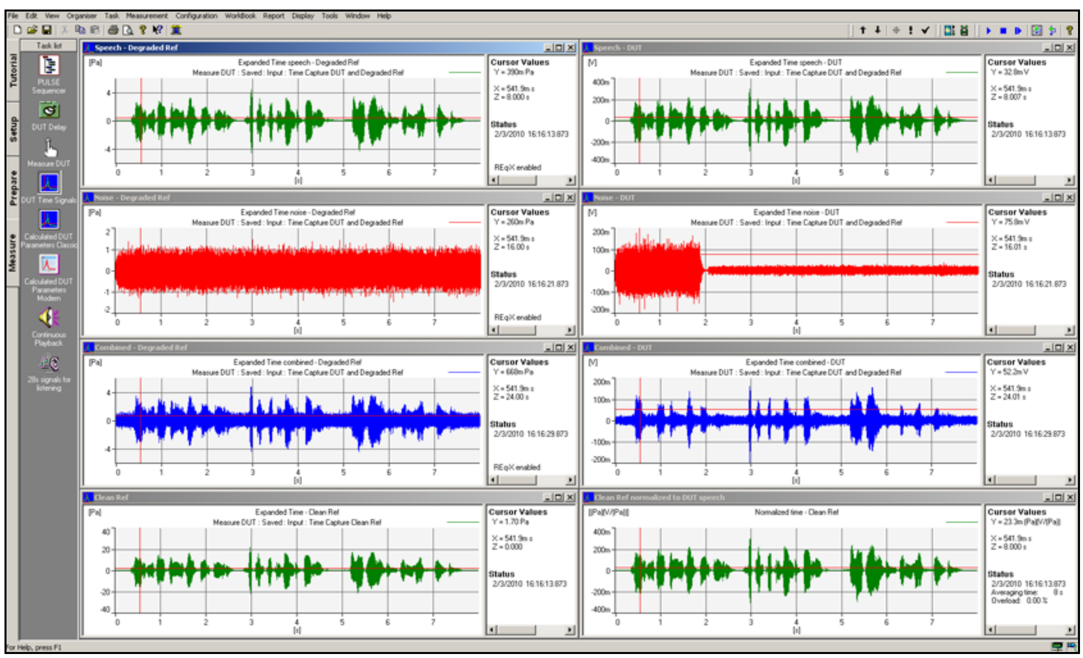
Fig. 2
Sounds used for testing the noise suppressor consist of four parts. The first part is a conditioning part which can be silence, pink noise or speech, then a speech segment, followed by a noise segment, and finally a segment where speech and noise are presented simultaneously



Viewing, Storing and Retrieving Measurements

PULSE LabShop holds a versatile tool for viewing measurements. Fig. 3 shows the actual behaviour of a device under test. From these measurements you can characterize the performance of the noise suppression algorithm or speech enhancement implemented in the device under test. These calculations are described below.

Fig. 3
The measurement results as they unfold in the time domain. Graphs on the left show measurements collected at the second reference microphone ("Degraded Ref") placed either at MRP or near the microphone of the device. Graphs on the right show measurements from the output of the device. The graphs show the signal from the device under test during presentation of speech (green), noise (red), and combined speech and noise (blue).The green graphs at the bottom show reference signals



PESQ – Perceptual Evaluation of Speech Quality

PESQ emulates the process of human perception by employing a measurement technique that compares both a reference signal (i.e., the input signal to the device under test) and a test signal (i.e., the output signal from the device under test). In the end, a total quality figure will be derived, which can be compared to a MOS or Mean Opinion Score resulting from a subjective listening test. By applying a mapping function for the PESQ score, the MOS value (MOS-LQO) can be determined. PESQ can be evaluated for both narrowband as well as wideband applications. The PESQ calculation is performed according to ITU-T Rec. 862.1 and 862.2. Over the years PESQ has become a widely used algorithm for objective speech quality assessment, and has reached a strong market acceptance. PESQ is especially useful to evaluate de-voicing with speech alone or in low noise levels. At higher noise levels it can be used as a relative measurement of quality improvement.

The reference signal is the speech signal captured with a measurement microphone, either at the lip plane of the HATS or close to the microphone in the device being tested, when only the speech signal is present. The test signal is the signal that has been processed in the device being tested, including noise suppression, encoding and any decoding introduced by the air interface, captured during the period where both the speech and noise were present.

A total of three PESQ values are calculated and three PESQ measurements are made. In all cases the reference signal will be the speech signal captured with a measurement microphone, at the lip plane of the HATS, when only the speech signal was present.

The first PESQ value is based on the signal collected from the device under test when noise is not present. This signal has been exposed to the signal processing in the device being tested, noise suppression and encoding as well as any decoding introduced by the air interface. This PESQ result relates to voice quality without any interfering noise.

Test signals are also collected from the second reference microphone, and from the device, during the period when both the speech and noise are present. These two resulting PESQ values relate to the degrading effect of noise at the second reference microphone and from the output of the device. The difference between them relates to the improvement made by the device.

POLQA – Perceptual Objective Listening Quality Analysis

POLQA should be seen as a further development of PESQ and various enhancements have been implemented in the POLQA algorithm. However, POLQA shares the same fundamental capabilities, hence the description provided in the PESQ section should be consulted for further information.

SNRI – Signal-to-Noise Ratio Improvement

SNRI measures the SNR improvement achieved by a noise suppression algorithm. SNRI is calculated separately in four groups of frames that represent power gated constituents of the active speech signal and the noise. Hence, the SNRI measure is calculated separately in frames of high, medium and low power, all compared to the noise frames. These categories are used to characterise the effect of the noise suppression processing on speech, in situations of strong, medium and weak speech. In addition to calculating the SNRI separately in the three categories, they are used to form an overall SNRI measure.

The SNRI calculation is performed according to 3GPP TS 26.077 and uses the signal that has been exposed to the signal processing in the device being tested. In the situation where the noise suppression can be controlled, two measurements may be performed, one where the suppression is off and one where the suppression is on, and the SNRI is calculated based upon these signals. In the situation where the noise suppression cannot be controlled, only one measurement is performed, but this measurement captures the signal close to the microphone located in the device being tested and the signal being captured at the lip plane of the HATS. The SNRI is calculated based on these signals.

The SNRI calculation is performed according to 3GPP TS 26.077. The clean reference signal is collected by the reference microphone at the HATS lip plane. The input signal, without noise suppression, is collected by the second reference microphone, either at MRP or near the microphone of the device. The output signal, with noise suppression, is collected from the output of the phone (or other device).

NPLR – Noise Power Level Reduction

NPLR relates to the capability of the noise suppression algorithm to attenuate the background noise especially at noise levels near and just below those of active speech. The NPLR calculation is performed according to 3GPP TS 26.077. NPLR and SNRI use the same measurement procedure.

TNLR – Total Noise Level Reduction

TNLR relates to the capability of the noise suppression algorithm to attenuate the background noise at all noise levels. The TNLR calculation is otherwise the same as for NPLR.

DSN – NPLR to SNRI Difference

DSN compares NPLR to SNRI. It is a measure of whether speech has been attenuated or amplified as part of the noise-reduction algorithm. It is calculated according to ITU-T G.160, Annex II.

Convergence Time

The convergence time is determined to ensure that a noise suppression algorithm produces the stated amount of noise reduction in response to a step change in noise level, after a maximum allowed time has elapsed. The calculation of Convergence Time is based on 3GPP TS 26. In this case, the signal of interest is the noise signal alone. The threshold of noise reduction which defines convergence can be adjusted.

ASL – Active Speech Level

Active speech level is measured during the time in which the speech is present (called the active time), and then expressing the quotient, proportional to the total energy divided by active time, in decibels relative to the appropriate reference. The calculation of the active speech level is performed according to ITU-T Rec. P. 56 Method B.

Level Distribution and Cumulative Distribution

Level distributions and the cumulative distributions of signals containing speech, noise and both can be calculated, providing a description of test signal dynamics and additional insight into the results.

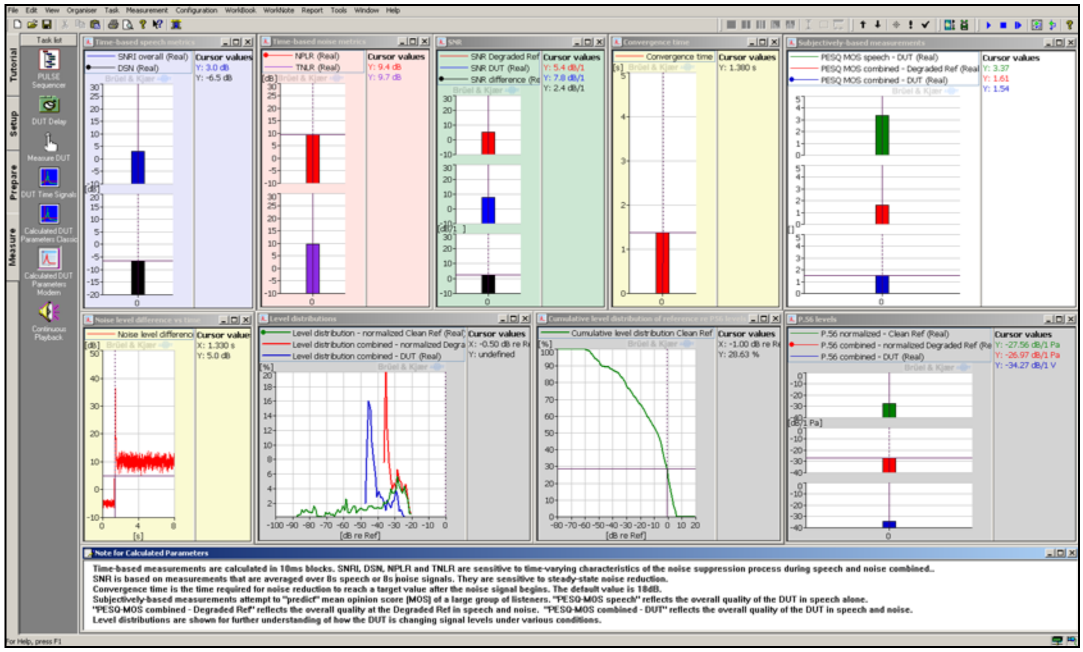


Fig. 4
Calculation results in graphical format. The individual graphs hold SNRI, NPLR, TNLR, DSN, and PESQ, as well as Convergence Time, Average SNR, ASL (P.56), Level Distribution (from both reference mics and the DUT), and Cumulative Level Distribution of the clean reference

MOS – Mean Opinion Scores

With the X-MOS algorithm three individual MOS values are calculated and mapped on a scale from 1 to 5 according to ITU-T Rec. P.835. The three values are:

- Speech MOS or S-MOS for short, evaluates the distortion of the speech
- Noise MOS or N-MOS for short, evaluates the intrusiveness of the background noise
- Global MOS or G-MOS for short, evaluates the overall quality of the speech and background noise

An example of the advanced interim results that are used for the calculation of the three MOS values are shown in Fig. 5.

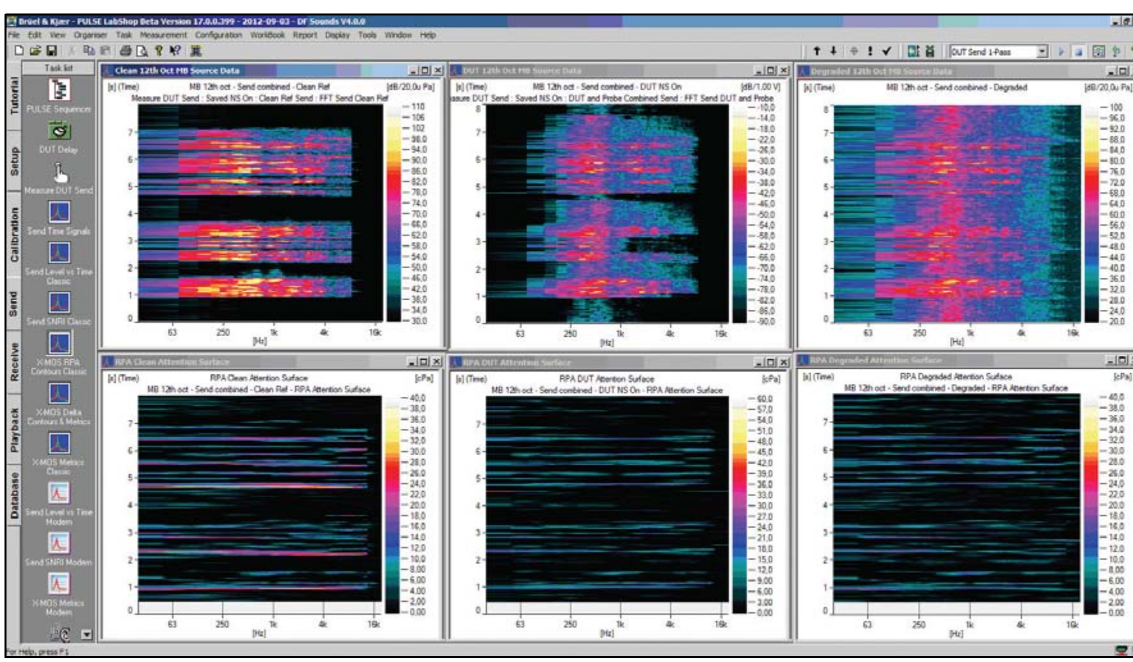


Fig. 5
Result of a measurement conducted during the presentation of both speech and noise (the three upper contour displays), as well as the attention surface calculated by the Relative Physcoacoustic Analysis (the three lower contour displays), which form the basis for calculation of the S-MOS, N-MOS and G-MOS

Documenting the Measurements

The PULSE Report Organiser takes measurement data and converts the results into pre-formatted Microsoft® Word documents. In addition to traditional graphs, the documents can also contain Live Measurement Data, which enables cursor readings, etc. inside the Word document. Once in Microsoft® Word format, the report can be printed, or the page layout modified to suit individual corporate standards.

For customised documentation of measurements, the actual measurement data, etc., can be exported in different file formats. This powerful export facility takes the measurement data and formats them into files that can be imported into a wide variety of standard data-processing programs. Measurement data can even be dynamically linked to enable automatic referencing and updating.

Preparing for Measurements

Before making the actual measurements, the system must be calibrated and the radio link between the air interface and the mobile phone established.

The radio link between the air interface and the mobile phone is controlled manually and established from the front panel of the air interface.

The system needs to be calibrated when the system is first installed, and when changes have been introduced in the acoustical environment. During the calibration procedure, the measurement microphones, mouth simulator and the speakers are measured, ensuring that any deviation can be digitally compensated for during the actual measurements.

Calibration of the speakers utilizes principles similar to those outlined by ETSI EG 202 396-1. V-Quest supports a more appropriate and at the same time simplified speaker configuration, compared to the realization suggested by the guideline. Furthermore, a calibration procedure that ensures a more realistic calibration of the speakers is used. During the calibration process a wide range of sounds (noises) are calibrated in order to assure the correct level when replayed. These noise samples include the noise experienced in a moving vehicle at a constant speed, on a city street, or in public places like restaurants, cafeterias or offices.

System Configurations

V-Quest is a modular system with a simple structure. The system consists of an Acoustic Interface, an Acquisition/Analysis System, and one or more Software Licenses.

Acoustic Interface

To establish a standardised and suitable acoustical coupling between the mobile phone and the Acquisition and Analysis System, Head and Torso Simulator (HATS) Type 4128-D should be used. Type 4128-D is especially suitable for the correct placement of mobile handsets, since it accommodates small handsets, handsets with antennae and non-symmetrical handsets. Type 4128-D with Handset Positioner Type 4606 is a very realistic test setup for handset testing, using either standardised position according to ITU-T recommendations or user defined positions.

Acquisition/Analysis System

The PULSE front-end hardware is the heart of V-Quest and is used for all tests performed by it. It generates the excitation signals, such as the Real Speech Signal and the Artificial Noise Field. Furthermore, it includes CPB (Constant Percentage Bandwidth), FFT (Fast Fourier Transform) and Time Capture Analysis. Alternate configurations are available. For example, a configuration that allows current Type 6712 users to have an easy upgrade path taking advantage of their current hardware platform.

Software Licenses

Software licenses fall into four groups – basic PULSE License for Spectrum Analysis, those required for performing the measurements which are supported by the Voice Quality Evaluation System, those that support testing of mobile phones according to specific standards and those that add analysis capability and features to the PULSE software.

Required licenses for evaluating Voice Quality:
- Type 7700: PULSEFFT & CPB Analysis
- Type 7705: Time Capture
- BZ-5830: Software for calculation of PESQ according to ITU-T P.862
- BZ-5831: Software for calculation of POLQA according to ITU-T P.863
- BZ-5832: Software for calculation of X-MOS according to ETSI EG 202 396-3 (2009-03)
- BZ-5833: Software for Voice Quality Evaluation System

Optional licenses for standards that allow testing according to specific standards:
- BZ-5137-017: EN 300 903 (GSM 03.50) for GSM phones
- BZ-5137-021: 3GPP TS 26.132 for GSM and UMTS phones
- BZ-5137-027: 3GPP TS 51.010 (GSM 11.10) for GSM phones
- BZ-5137-025: CTIA test plan for dual mode AMPS/CDMA phones
- BZ-5137-023: Hands-free based on ITU-T Rec. P.342
- BZ-5137-029: LSTR and ANR based on GSM and 3GPP specifications
- BZ-5137-037: 3GPP2 CS 0056-0 for AMPS and CDMA phones
- BZ-5137-039: 3GPP TS 26.132 for wideband GSM and UMTS phone
- BZ-5137-041: Software for Headset Testing - Generic Requirements
- Type 7909-S1: Voice Testing Software for Hands-free Equipment
- Type 7912-S1: Voice Testing Software for VoIP Terminals

For more information see the separate Product Data BP1683, BP2116 and BP2240.

Optional PULSE licenses that add analysis capability to PULSE:
- Type 7797: Basic Electroacoustics
- Type 7698: Sound Quality Software
- BZ-5265: Zwicker Loudness option for Type 7698
- BZ-5301: Psychoacoustic Test Bench option for Type 7698

For more information see the separate Product Data BP2085 and BP1589.

Optional Accessories
For easy configuration of a complete system, a number of standard system configurations are available. For specific information regarding standard system configurations as well as optional accessories, please see the ordering information.

Using PULSE as a General Research and Development Tool

PULSE provides access to analyzers, post-processing functions and display facilities for many applications within the area of electroacoustic testing. Using the FFT Analyzer, CPB (1/nth octave) Analyzer, Overall Level Analyzer and Signal Generators included with V-Quest, PULSE can be set up to accommodate the vast amount of different measurements typically required for R&D of new electroacoustic devices. Furthermore, PULSE contains a task-oriented user interface that allows the tasks involved in the complete measurement process to be implemented in PULSE as individual tasks to can be performed one after another. All the tasks can be stored together with the actual measurements and can be stored as a PULSE project. For displaying the measurements, PULSE has a large variety of different functions such as 3D waterfall display and contour display. For documenting the measurements PULSE supports the use of either dynamic or static links to Word or Excel® displays.

Additional software that enhances the analysis capability of PULSE and that could be useful during the development of new electroacoustic devices is:

- Type 7797 Basic Electroacoustics for testing of electroacoustic transducers and components. For more information see the separate Product Data BP2085.
- Type 7705 Time Capture for recording of acoustical or electrical signals – recordings that can be exported from PULSE as wave files and then be loaded into the generator and replayed.

For more information on PULSE software and hardware please refer to System Data Sheets BU0229 (PULSE Software) and BU0228 (PULSE Hardware).

Calculations

The calculations implemented are based on the following standards and recommendations:
- 3GPP TS 26.077 V7.0.0 (2007-06) for determination of SNRI, NPLR and convergence time. This method is equivalent with ITU-T Rec. G.160 Appendix II
- ITU-T Rec. G.160 Appendix II for determination of TNLR and DSN
- ITU-T Rec. 862.1 (2003-11) for narrow band PESQ assessment
- ITU-T Rec. 862.2 (2007-11) for wideband PESQ assessment
- ITU-T Rec. 863 (2011-01) for POLQA assessment
- ETSI EG 202 396-3 (2009-03) for calculation of X-MOS

Calibrations

The calibrations implemented are based on the following standards and recommendations:
- HATS mouth calibration according to ITU-T Rec. P.58
- The calibration procedure based on the principles outlined by ETSI EG 202 396-1 (2009-03) is applied for the speakers used for reproduction of the artificial noise field

Speech and Noise Samples

Most of the noise suppressor tests require the addition of noise to the speech material. The following types of noise are available for the system:
- Car Noise: This represents stationary background noise and will be typical of the noise experienced when inside a moving vehicle at a constant speed
- Street Noise: This represents non-stationary noise and will be typical of noise which might be experienced by someone using a mobile on a city street
- Babble Noise: This represents non-stationary noise and will be typical of the background noise encountered in public places: restaurant, cafeteria, open offices

The tables on the next pages show the speech and noise samples which are available for the system. The samples that are supported have been compiled to accommodate the requirements of the V-Quest. Recorded noises have been selected from the ETSI EG 202 396-1 background noise database. Speech samples have been selected from the speech database in ITU-T P.50, Appendix 1.

Table 1 *The list of speech samples available. Speech files are from ITU-T Recommendation P.50, Appendix 1 – Test Signals. Notation such as “m2-4” specifies the talker and sentence number*

Name	Conditioning (4s)	Speech (8s)
Male, conditioning silence	Silence	Male, American English 3 sentences, 3 different talkers (m2-4, m1-2, m5-4)
Male, conditioning pink	Pink noise	
Male, conditioning male	Male, American English, 1 sentence (m5-1)	
Female, conditioning silence	Silence	Female, American English 3 sentences, 3 different talkers (f4-3, f1-1, f8-2)
Female, conditioning pink	Pink noise	
Female, conditioning female	Female, American English, 1 sentence (f5-2)	

Table 2 *The list of simple noise samples available. Mixed noises (MNxx) are shown in Table 3. Environment noise files are from ETSIEG 202 396-1 background noise database*

Environment Noise	Music	Noise	Mixed noises	Voice
Cafeteria noise	Backbeat	Pink noise (diffuse)	MN1	Male speech
Call center	Baroque orchestra	Pink noise (monaural)	MN2	Female speech
Forest and jet	Grunge music	Pink noise – spin	MN3	Male speech – spin
Inside aircraft 1	Orchestra finale		MN4	Female speech – spin
Inside bus	Trumpet jazz		MN5	
Inside train 3			MN6	
Living room			MN7	
Midsize car 2			MN8	
Pub noise			MN9	
Schoolyard noise			MN10	
Traffic crossroads			MN11	
			MN12	

Table 3 *The complete list of mixed noises (MNxx)*

Name	Sound 1	Sound 2	Sound 3	Sound 4	Scenario
MN1	Female	Midsize car 2			GPS female voice in moving car
MN2	Male	Living room			Living room with male voice on TV talk show
MN3	Male	Grunge			Loud male voice at concert
MN4	Male	Midsize car 2	Pub		Male interviewer at restaurant on car radio
MN5	Female	Schoolyard	Forest and jet	Traffic crossroads	Female speech at schoolyard
MN6	Large orchestra	Midsize car 2			Baroque music on car radio
MN7	Grunge	Pub			Grunge music in pub
MN8	Backbeat	Inside aircraft 1	pub		Aircraft during exit
MN9	Baroque	Living room	Forest and jet	Traffic crossroads	Music in living room with open windows
MN10	Male	Trumpet jazz	Pub	Traffic crossroads, forest and jet	Outdoor café with male voice, babble and music
MN11	Inside bus	Pub			Bus with loud babble
MN12	Pub	Traffic crossroads	Forest and jet		Outdoor café, without voice or music

Ordering Information

STANDARD CONFIGURATION FOR VOICE QUALITY EVALUATION SYSTEM

The following items are required for the full system configuration:

3 × Type 3109[*]	Generator, 4/2-ch. Input/Output Module
Type 3560-D-E01	PULSE D-size Front-end incl. Type 2826 and Type 7536
Type 7700-Xy[†‡]	PULSE FFT & CPB Analysis
Type 7705-X[†]	PULSE Time Capture
BZ-5600-X[†]	PULSE Sequencer
BZ-5601-X[†]	PULSE Data Manager for Electroacoustic Applications
BZ-5830	Software for Calculation of PESQ According to ITU-T P.862
BZ-5831	Software for Calculation of POLQA According to ITU-T P.863
BZ-5832	Software for Calculation of X-MOS According to ETSI EG 202 396-3 (2009-03)
BZ-5833	Software for Voice Quality Evaluation System
2 × JJ-0152	BNC Adaptor
2 × AO-0389-013	BNC Cable, 130 mm (5.1")
Type 2716-C	Power Amplifier
WL-1324-D-030	Cable XLR to BNC Connectors 3 m (9.8 ft.)
WL-1325-D-100	Speakon® - Banana Cable, 10 m (33 ft)
Type 4231	Sound Calibrator
Type 4128-D-001	Head and Torso Simulator
Type 4938-A-011	¼″ Pressure-field Microphone, incl. Preamplifier Type 2670 with TEDS
DP-0775	Adaptor for Calibrating ¼″ Microphones on Calibrator Type 4228 or 4231
UA-0801	Lightweight tripod
UA-0587	Heavy Duty Tripod
UA-1588	¼″ Microphone Holder
2 × AO-0414-D-100	Lemo to Lemo Cable, 10 m (33 ft)
4 × WQ-2975	Loudspeakers including Stand and Plate
4 × WL-3443-D-100	Power Cables for Loudspeakers
4 × WL-1324-D-100	Cable XLR to BNC Connectors 10 m (33 ft)

Accessories Required:

One of:

Type 3099-A-X	PULSE LAN-XI and IDA^e/IDA Multiple Module Front-end Driver
Type 3099-A-X1	PULSE LAN-XI Single Module and IDA^e/IDA Systems Any Size Front-end Driver
Type 3099-A-X2	PULSE LAN-XI Dual Module and IDA^e/IDA Systems Any Size Front-end Driver

[*] For system configurations including LAN-XI, Type 3160-A-042 is also available. Please contact your local Brüel& Kjær representative.

[†] License model either N for Node-locked or F for Floating.
y = optional channel count, from 1 (single) to 7 (or 16 for M1 Agreements). No number denotes unlimited channels (channel independent).

[‡] A Type 4939 and AO-0414-D-100 are required if the operator has no means of controlling the noise suppression and Xy should be replaced by X3, otherwise it should be X2.

STANDARD SYSTEM CONFIGURATION FOR UPGRADE OF TYPE 6712-A-S02

If you are upgrading from a Type 6712-A-S02 system to a full system configuration, the following items are required:

2 × Type 3109[*]	Generator, 4/2-ch. Input/Output Module
Type 3560-D-E01	PULSE D-size Front-end incl. Type 2826 and Type 7536
Type 7700-Xy[†‡]	PULSE FFT & CPB Analysis
Type 7705-X[†]	PULSE Time Capture
BZ-5600-X[†]	PULSE Sequencer
BZ-5601-X[†]	PULSE Data Manager for Electroacoustic Applications
BZ-5830	Software for Calculation of PESQ According to ITU-T P.862
BZ-5831	Software for Calculation of POLQA According to ITU-T P.863
BZ-5832	Software for Calculation of X-MOS According to ETSI EG 202 396-3 (2009-03)
BZ-5833	Software for Voice Quality Evaluation System
UA-0801	Light-weight tripod
UA-0587	Tripod heavy duty
UA-1588	¼″ Microphone Holder
2 × AO-0414-D-100	Lemo to Lemo Cable, 10 m (33 ft)
4 × WQ-2975	Loudspeakers including Stand and Plate
UL-0229	5-port Gigabit Ethernet Switch
WL-1325-D-100	Speakon – Banana Cable, 10 m (33 ft)
4 × WL-3443-D-100	Power Cables for Loudspeakers
4 × WL-1324-D-100	Cable XLR to BNC Connectors 10 m (33 ft)

Accessories Required:

One of:

Type 3099-A-X	PULSE LAN-XI and IDA^e/IDA Multiple Module Front-end Driver
Type 3099-A-X1	PULSE LAN-XI Single Module and IDA^e/IDA Systems Any Size Front-end Driver
Type 3099-A-X2	PULSE LAN-XI Dual Module and IDA^e/IDA Systems Any Size Front-end Driver

SOFTWARE MAINTENANCE AND SUPPORT AGREEMENTS

M1-5600-X[†]	Software Maintenance and Support Agreement for PULSE Sequencer
M1-5601-X[†]	Software Maintenance and Support Agreement for PULSE Data Manager for Electroacoustic Applications
M1-5830-X[†]	Software Maintenance and Support Agreement for software for Calculation of PESQ
M1-5831-X[†]	Software Maintenance and Support Agreement for software for Calculation of POLQA
M1-5832-X[†]	Software Maintenance and Support Agreement for software for Calculation of X-MOS
M1-5833-X[†]	Software Maintenance and Support Agreement for software for Voice Quality Evaluation System
M1-7700-Xy[†]	Software Maintenance and Support Agreement for Type 7700
M1-7705-X[†]	Software Maintenance and Support Agreement for Type 7705

Optional Accessories

ACCESSORIES FOR RACK-MOUNTED SYSTEMS

WF-0059	Schroff Minirack 16HE, 553 × 600 mm, incl. Accessories for Telephone Test System

ACCESSORIES FOR REMOTE CONTROL OF AIR INTERFACE

WQ-1270	IEEE-488 Interface Card – PCI-GPIB
WQ-1290	IEEE-488 Interface Card – PCMCIA
WQ-2464	IEEE-488 Interface Card – USB
AO-0265	IEEE-488 Cable (2 m)
WL-1368	Antenna Cable-BNC to open-end
AO-0530	Cable for R&S CMD-55 codec
WL-3162	Cable for R&S CMU-200 codec (GSM)
WL-3162-A	Cable for R&S CMU-200 codec (CDMA)
2 × AO-0087	Cable for HP-8922 codec

The accessories for Remote Control of Air Interface are available for all standard Type 6712 system configurations. Please note that Type 6712 supports Remote Control of Air Interface HP-8922, CMD-55 and CMU-200 whereas V-Quest only supports manual control of the Air Interface

OPTIONAL LICENSES FOR STANDARDS THAT ALLOW TESTING ACCORDING TO SPECIFIC STANDARDS (WINDOWS® XP ONLY)

BZ-5137-017	EN 300 903 (GSM 03.50) for GSM phones
BZ-5137-021	3GPP TS 26.132 for GSM and UMTS phones
BZ-5137-023	Hands-free based on ITU-T Rec. P.342
BZ-5137-025	CTIA test plan for dual mode AMPS/CDMA phones
BZ-5137-027	3GPP TS. 51.010 (GSM 11.10) for GSM phones
BZ-5137-029	LSTR and ANR based on GSM and 3GPP specifications
BZ-5137-037	3GPP2 CS 0056-0 for AMPS and CDMA phones
BZ-5137-039	3GPP TS 26.132 for wideband GSM and UMTS phones
BZ-5137-041	Headset Testing – Generic Requirements

OPTIONAL PULSE LICENSES THAT ADD ANALYSIS CAPABILITY TO PULSE

Type 7797-X*	Basic Electroacoustics
Type 7698-X*	PULSE Sound Quality Software
BZ-5265-X*	Zwicker Loudness option for Type 7698
BZ-5301-X*	Psychoacoustic Test Bench option for Type 7698

* X = License model either N for Node-locked or F for Floating.

SERVICE AND SUPPORT PRODUCTS

2716-TCF	Conformance Test with of Type 2716-C with Certificate
3560-SI1	Installation and Configuration (at Brüel & Kjær)
3560-C-CAF	Portable PULSE Accredited Calibration
3560-C-CAI	Portable PULSE Accredited Initial Calibration
3560-C-EW1	3560-C Extended Warranty, one year extension
3560-C-TCF	Conformance Test of 3560-C with Certificate and Measured Values
3560-HL1	3560 Software and Hardware Support. One year of Helpline Support
4128-CFF	Head and Torso Simulator Type 4128-C (Factory Standard Calibration)
4138-CAI	1/8 Pressure-field Microphone, 6 Hz to 140 kHz, 200 V Polarization, Initial Open Circuit Sensitivity Calibration (DANAK) plus Factory Calibration
4938-CAI	Falcon ¼″ Condenser Microphone 200 V polarization, pressure-field, 4 Hz to 70 kHz, Accredited Initial Calibration

SOFTWARE MAINTENANCE AND SUPPORT AGREEMENTS

M1-5137-017	Software Maintenance and Support Agreement for Type 6712 Software for EN 300 903
M1-5137-021	Software Maintenance and Support Agreement for Type 6712 Software for TS 26.132
M1-5137-023	Software Maintenance and Support Agreement for Type 6712 software for ITU-T Rec. P.342
M1-5137-025	Software Maintenance and Support Agreement for Type 6712 software for CTIA test plan
M1-5137-027	Software Maintenance and Support Agreement for Type 6712 software for TS 51.010
M1-5137-029	Software Maintenance and Support Agreement for Type 6712 software for LSTR and ANR
M1-5137-037	Software Maintenance and Support Agreement for Type 6712 Software for 3GPP2 CS 0056-0
M1-5137-039	Software Maintenance and Support Agreement for Type 6712 Software for 3GPP TS 26.132 Wideband
M1-5137-041	Software Maintenance and Support Agreement for Type 6712 Software for Headset Testing

PULSE SOFTWARE UPDATE OF EXPIRED AGREEMENTS

M3-5137-XXX	Update of Expired Agreement M1-5137-XXX

BP 2391 – 12 2012-12

TRADEMARKS
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