

Testing Speech Quality of Mobile Phones in a Live Network

Application Note

Products:

- R&S®UPV
- R&S®UPV-K63
- R&S®UPV-K1

This application note provides an application program and instructions for performing speech quality tests and delay measurements in a live network, using the R&S®UPV audio analyzer.

Note:

Please find the most up-to-date document on our homepage

<http://www.rohde-schwarz.com/appnote/1GA68>.

This document is complemented by software. The software may be updated even if the version of the document remains unchanged

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1 Background

1.1 Test methods for voice call functionality of mobile phones

Various kinds of tests are available for the voice call function of mobile phone terminals.

1.1.1 Acoustic test

Acoustic test methods for mobile phone terminals are for example specified in 3GPP TS 26.132. The main purpose of such tests is characterizing the acoustic performance of a mobile phone terminal. Characterized properties are sensitivity and frequency response, idle channel noise, distortion and delay in sending and receiving direction, echo loss, stability loss, echo control characteristics and speech quality in sending direction in presence of background noise.

For this kind of test, an ideal network connection is assumed. It is established with a radiocommunication tester like for example R&S® CMW500. The near-end connection to the mobile terminal is of acoustic nature and established with artificial ear and artificial mouth, preferably mounted in a head-and-torso simulator.

Acoustic minimum performance testing is the basis of acoustic conformance testing according to GCF and PTCRB.

A test solution from Rohde & Schwarz for this purpose comprises the R&S® UPV audio analyzer with options R&S® UPV-K9 and R&S® UPV-K91 and the R&S® CMW500 wideband radiocommunication tester.

1.1.2 Speech quality test with base station simulator

This test category applies pre-defined non-ideal conditions to the connection between base station simulator and mobile phone terminal under test. The purpose is to assess the speech quality of the device under test with different codec modes and rates, under RF fading conditions as well as under IP impairment conditions. Tests with impairments are mainly performed in downlink (receiving) direction. The resulting speech quality is depending on RF receiver performance, bit error correction and concealment of not recoverable transmission errors. As the acoustic performance is not of interest, the near-end connection between mobile phone terminal under test and audio analyzer is established electrically using an adapter cable which simulates a headset connected to the mobile phone terminal.

3GPP TR 26.954 (3GPP) describes speech quality tests with base station simulator and electric connection under various impairment conditions. Furthermore, speech quality tests with base station simulator are part of operator acceptance test lists for mobile phone terminals.

A test solution from Rohde & Schwarz for this purpose comprises the R&S® UPV audio analyzer with options R&S® UPV-K1, R&S® UPV-K4 and R&S® UPV-K63, the R&S® CMW500 wideband radiocommunication tester and software CMWRun.

Application note 1GA62 (Lechner, 2014) provides a tool for level adjustment and test automation with PESQ® and POLQA® without control of the R&S® CMW500 wideband radiocommunication tester.

1.1.3 Drive test and walk test

The purpose of drive test and walk test is the characterization of the coverage of mobile networks. A “reference” mobile terminal is used to establish a voice connection. It is moved through the network in order to determine the speech quality of the mobile connection throughout a part of the coverage area of a network. In many cases the evaluation of the degraded speech signals is performed offline, and the near-end connection at the reference mobile is implemented by file recording and / or file playback.

Drive tests are often performed by mobile network operators for internal quality control.

Products from Rohde & Schwarz for this purpose are R&S® ROMES4 and SwissQual QualiPoc.

1.1.4 Speech quality test in live network

This kind of test is closest to the real user experience. The speech connection is established between two mobile phone terminals or from a terminal under test to a reference terminal via a real live network. This test can reveal interoperability problems between the terminal in question and the network, but it also includes the acoustic quality, the quality of the speech codec and the influence of RF performance on the speech quality. The connection between test equipment and each mobile can either be established acoustically like for 1.1.1, but also electric connection according to 1.1.2 is possible.

This application note provides a program for performing speech quality tests on mobile phones in a live network.

1.2 POLQA® measurements

1.2.1 Objective perceptual speech test methods

POLQA® and PESQ® are objective perceptual speech test methods.

A **speech** test method rates the quality of a speech signal as opposed to a conversational rating where impairments like round-trip delay are judged for their impact on a conversation.

A **perceptual** test method delivers results which assess degradations the same way as human beings would do. Non-perceptual test methods usually determine physical entities like voltage level.

A possible implementation of a perceptual test is a subjective listening test, e.g. according to ITU-T P.800. In a subjective test, speech signals are presented to subjects under standardized listening conditions (level, playback equipment) with a well-defined judgement task or question. Such a task could be to categorize the quality of a presented speech signal according to the question “how good does the voice sample sound?” into one of five steps between 5 = “Excellent” and 1 = “Bad”. The answers from many subjects are averaged to obtain a so-called “MOS” (Mean Opinion Score).

An **objective** perceptual speech test method implements a complex algorithm which predicts results expected from subjective tests by applying psychoacoustic and perceptive models to the degraded test signals. The algorithm is verified by comparing the results to MOS scores obtained in subjective listening tests. A mapping function may be applied for better matching between the objective and subjective MOS scores.

PESQ® and POLQA® are so-called full-reference test methods. They analyze the original (“Reference”) speech signal and the degraded signal and compare attributes extracted from both for obtaining an assessment of the degradation.

POLQA® is standardized in ITU-T P.863 (ITU-T, 09/2014). An application guide is available in ITU-T P.863.1 (ITU-T, 09/2014-1).

1.2.2 Bandwidth and listening situation

In telephony different bandwidths are defined:

Narrowband (300 Hz ~ 3400 Hz)

Wideband (100 Hz ~ 7 kHz)

Super-wideband (50 Hz ~ 14 kHz)

Fullband (20 Hz ~ 20 kHz).

PESQ® provides a narrowband and a wideband mode. In both cases listening tests for the verification of the algorithm were performed monaural using a so-called IRS type receiver. The same applies to the POLQA® narrowband mode.

In POLQA® super-wideband mode the subjective tests are performed with high-quality binaural headphones. POLQA® does not provide a wideband mode.

ITU-T P.863 mode	Signal bandwidth	Listening situation	Scale	Output
Super-wideband	Superwideband wideband narrowband	Flat headphone diffuse field equalized, diotic presentation	MOS ACR superwideband scale	MOS-LQOsw
Narrowband	Narrowband	IRS receive handset, monotic presentation	Narrowband scale comparable to [b-ITU-T P.862] in conjunction with [b-ITU-T P.862.1]	MOS-LQOn

Table 1-1: Modes of POLQA® and their application (from ITU-T P.863)

POLQA® super-wideband mode can be applied to narrowband, wideband and super-wideband connections. The MOS scores will also reflect the degradation caused by the respective bandwidth limitation.

POLQA® narrowband mode is provided for compatibility with assessment of narrowband telephony and backward compatibility with P.862.1 (PESQ with mapping of PESQ scores to a MOS scale). In all other application cases the super-wideband mode of POLQA® should be used.

In particular for assessment of transmission paths including acoustic components, only POLQA® in the super-wideband mode is validated. Therefore this application note will concentrate in the following on POLQA® in the super-wideband mode.

1.2.3 Reference signals

Clean speech “reference” signals should preferably have the following properties:

- at least three seconds of active speech
- at least one second of silence between active speech periods
- no more than 6 seconds of active speech
- total length of test sample including silence should be no more than 12 seconds
- active speech level of -26 dBov
- 16-bit linear PCM encoded
- noise floor < -80 dBov (A)
- speech pauses should not be digital zero but contain noise signal < -80dBov(A)

The level in the speech reference files is specified in terms of “dBov” (“dB overload”) which is the rms level referenced to a full-scale square wave signal. A full-scale sinewave signal has a rms level of -3 dBov.

Usually one reference signal file contains a pair of sentences separated by a short silence.

Reference signals for POLQA® super-wideband should be sampled at a rate of 48 kHz and bandwidth-limited to 50 Hz ~ 14 kHz.

The reference signals should have a leading and a trailing silence at least as long as the expected end-to-end delay in the transmission path under test. This allows temporal alignment of the degraded signal with the reference signal without losing portions of the speech.

Codecs and other signal processing algorithms as well as the POLQA® and PESQ® algorithms may react differently to different voices. Therefore it is recommended to conduct tests with female and male voices and average the results.

Speech samples “English female 1” and “English male 2” of ITU-T recommendation ITU-T P.501, Annex C fulfill the requirements for POLQA® reference files. They should additionally be filtered to the SWB frequency range for the super-wideband mode of POLQA®. These filtered files are included with the installer of this application note.

1.2.4 POLQA® on the R&S® UPV audio analyzer

The POLQA® algorithm according to ITU-T P.863 is provided with option UPV-K63. A description of all parameters and results is given in the current manual of the R&S® UPV audio analyzer and in the on-screen context help. A detailed whitepaper is available in application note 1MA202. Application note 1GA62 provides an application program for level adjustment and test automation to be used with speech quality tests utilizing a base station simulator like R&S® CMW500.

1.2.5 Conditioning of the speech signals

The POLQA® algorithm in super-wideband mode as defined in ITU-T P.863 rates the absolute listening level. For obtaining a correct rating, a “gain” factor has to be specified, calculated from several gain factors like artificial ear sensitivity and a factor depending on the analyzer input range. If there is not sufficient information for calculating this gain factor, the level of the degraded signal should be normalized for the POLQA® algorithm. From UPV firmware 3.3.1, the analyzer function “POLQA” offers an “Auto Gain” parameter which can be activated for this purpose.

To achieve a correct spectrum of the speech signal in the sound pressure at the so-called “Mouth Reference Point” (MRP), an equalization of the electric input signal is required. The application program provided with this application note comprises functionality for determining the frequency response and absolute sensitivity of the artificial mouth, and for equalizing the test signals with the inverse frequency response, such that a speech signal with correct spectrum and an active speech level of -4.7 dBPa at the mouth reference point (MRP) can be generated.

The microphone in the artificial ear delivers an electric signal which is proportional to the sound pressure at its membrane (“Eardrum Reference Point”, DRP). For POLQA® super-wideband evaluation, the signal recorded at the artificial ear of a head-and-torso simulator has to be diffuse-field equalized. To obtain a flat frequency response related to the sound pressure in a diffuse sound field, appropriate corrections have to be applied which can be automatically loaded as an analyzer filter by the application program. A filter for diffuse field correction of the type 3.3 artificial ear of the head-and-torso simulator is included with the installer of the application program.

If an electric connection to one or both mobile phone terminals is established to replace the acoustic coupling, the application program allows to specify the test signal level relative to full-scale of the input and output of the transmission path.

The application program provided with this application note loads the (equalized) test signal into the generator of the R&S® UPV audio analyzer and sets generator level and analyzer range according to the calibration values of acoustic or electric connections, respectively.

2 Using the application program

2.1 Preparations

2.1.1 Prerequisites

Using the application program requires option R&S® UPV-K1 (universal sequence controller). This option comprises a Visual Basic .NET 2008 IDE, which is, however, not required here.

The POLQA® evaluation requires option R&S® UPV-K63.

2.1.2 Installation of the application program

Copy the installer file “1GA68_xxx.msi” to an appropriate folder on the D: partition of the hard disk of the R&S® UPV audio analyzer, e.g. “D:\R&S Software\Application Programs”. Start the installer and follow the instructions on the screen.

The installer creates shortcuts “Live Network Test” on the desktop and in folder “R&S UPV Applications” of the programs menu.

2.1.3 Configuring a quick launch button for starting the application

This step is optional and simplifies the start of the application program out of the UPV firmware window.

The R&S® UPV audio analyzer offers 8 quick launch buttons which can be utilized either for loading pre-defined setups or for starting pre-defined macros. The quick launch buttons are on the toolbar which is either displayed permanently on top of the UPV window (can be set in “Utilities → Config Panel”) or appears on the right side when the mouse is moved to the right edge of the UPV window.

The assignment of the buttons can be defined in “Utilities → Quick Launch Config ...”:

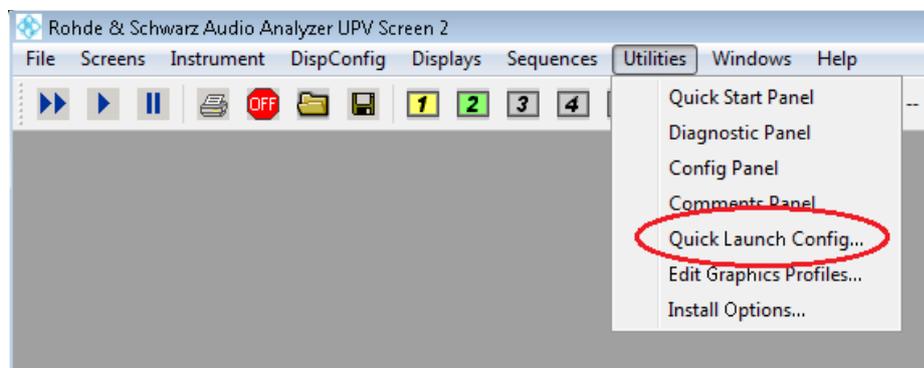


Fig. 2-1: Starting the Quick Launch Config Panel

This menu opens the following configuration panel:

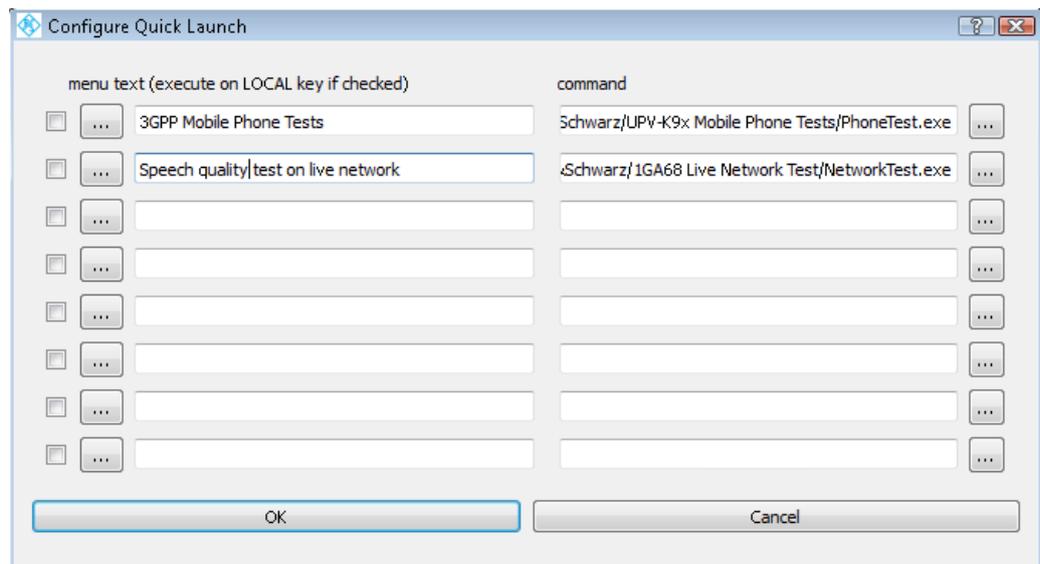


Fig. 2-2: Quick Launch Config Panel

Use the “...” buttons on the right side to browse to the executable “C:\Program Files\Rohde&Schwarz\1GA68 Live Network Test\NetworkTest.exe”. The “...” buttons on the left side can be utilized to specify icons on the buttons. Text entered in the text box on the left side will be displayed as tool tip text for the respective button when the mouse cursor hovered over the area of the button.

2.1.4 Test setup

The characteristic of this application note is the speech connection between two mobile phone terminals over a live public mobile network. Each terminal may be connected with the R&S® UPV audio analyzer either by acoustic coupling or by electric connection.

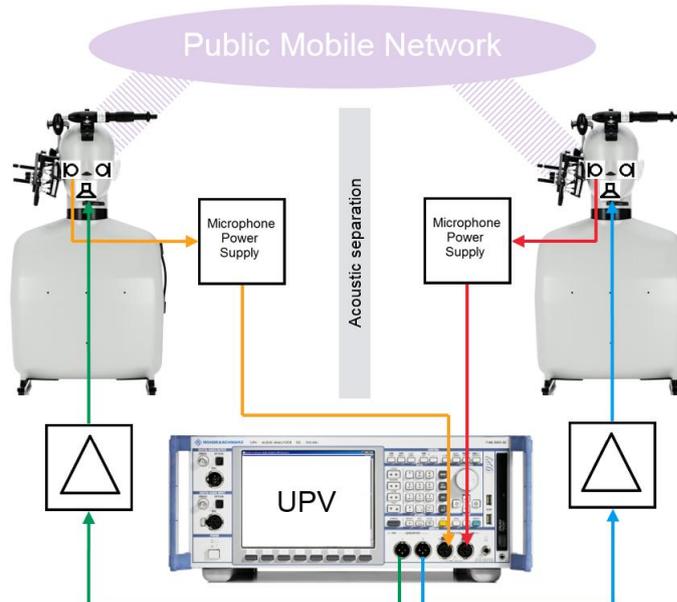


Fig. 2-3: Physical test setup for live network test with acoustic coupling to both mobile phones

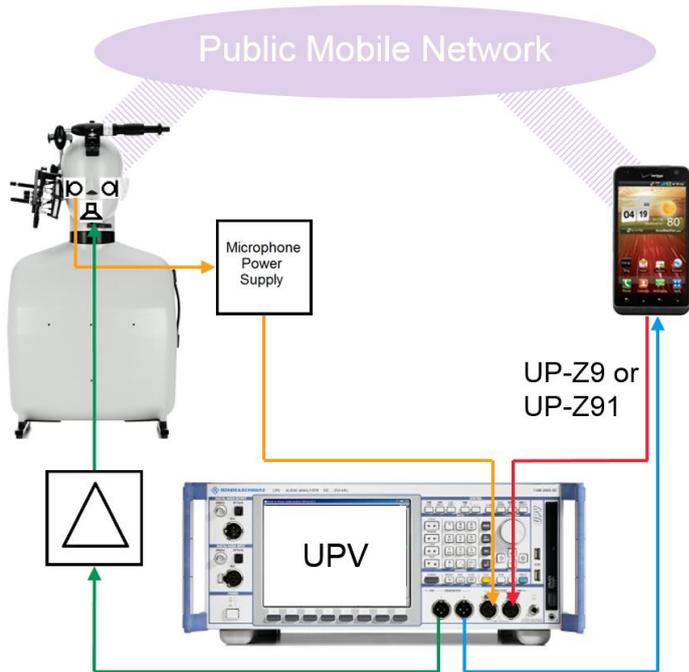


Fig. 2-4: Physical test setup for live network test with acoustic coupling to first mobile phone and electric coupling to reference mobile phone

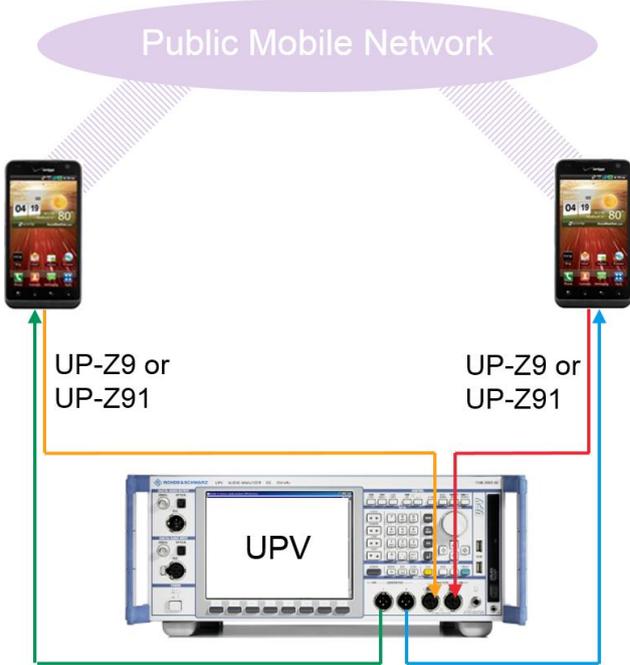


Fig. 2-5: Physical test setup for live network test with electric coupling to both mobile phones

2.2 User interface

The main window of the application program provides eight tabs.

The first tab is dedicated to acoustic calibration functions.

The next two tabs are labeled according to the two transmission types “1 → 2” and “2 → 1”. Each tab provides controls for configuration of the speech quality tests in the respective direction.

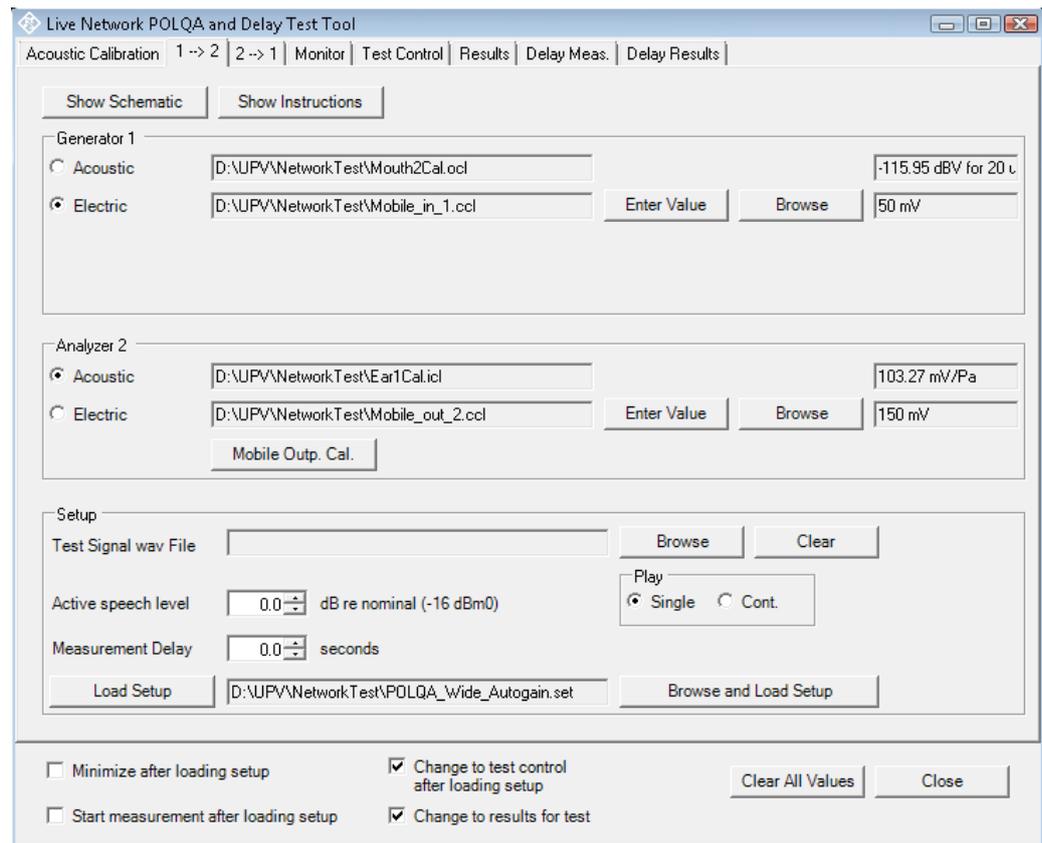


Fig. 2-6: “1 → 2” tab of the user interface

In addition there is one tab each for monitor, test control and results, delay measurement and delay results.

Two help buttons on each of the second and third tabs give instructions in schematic and in text how to set up the connections. Connections between each of the two mobile phone terminals and the UPV can be established by acoustic coupling or electric by cable.

The next section of the user interface provides a row for each required calibration value. The value, if available, is shown on the respective right end. In the center a textbox shows the name of the file which stores the current calibration value. Stored values can be loaded using the “Browse” buttons.

Another section follows which allows to specify a wave file to be used as test signal file, and the full-scale value of the wave file relative to the determined full-scale value of the signal path input. An additional measurement delay can be set for long transmission delays in the network. Without additional delay setting, the maximum

tolerable end-to-end delay is as long as the trailing silence of the reference speech files, which is 1 second with the files provided with this application note. A value > 0 in the “Measurement Delay” setting delays the start of the acquisition by the given value against the start of the playback in the generator “Play” function. This additional delay value is later on added to the average, maximum and minimum delay determined by the POLQA® evaluation.

The last section in each tab handles the setup to be used for the test. The text box in the center shows the file name. With the “Load Setup” button, the setup is loaded to the UPV, the generator reference value, signal file name and voltage value are entered and the analyzer range is set appropriately. To load a setup from a file different from the one shown in the text box, use the “Browse and Load Setup” button.

The checkboxes below the tabs allow to automatically minimize the window of the application program after the setup has been loaded, and/or to start a (single) measurement immediately afterwards. Two more checkboxes allow to configure automatic change to the test control tab after a setup has been loaded and automatic change to the results tab as soon as a test sequence starts.

The user interface of the last five tabs is described with monitor, test automation and reporting functionality further below.

The program stores all settings and calibration values in a special file from where they are loaded when the program starts. The button “Clear all Values” can be used to clear the settings and calibration values in all tabs.

2.3 Acoustic calibration

2.3.1 Calibration of the measurement microphone

A calibrated measurement microphone is required for measuring the sound pressure at the mouth reference point (MRP) during calibration of the artificial mouth.

Use a ¼” measurement microphone for the calibration of an artificial mouth according to ITU-P.58 being integral part of a head-and-torso simulator (HATS).

The following steps describe the calibration of the measurement microphone.

- 1. Insert the measurement microphone fully into the adapter of the sound level calibrator and switch on the calibrator. A ¼” adapter is required.



Fig. 2-7: Insertion of a ¼” measurement microphone into the calibrator with adapter

- Select the acoustic calibration level (1 Pa or 10 Pa) and calibration frequency according to the sound pressure and frequency generated by the acoustic calibrator in use.
- If the frequency of the sound generated in the acoustic calibrator is 1 kHz or 250 Hz, checkbox “Selective” should be enabled and the appropriate radio button for the respective frequency should be checked. If the calibrator generates another frequency, checkbox “Selective” must be disabled.
- Click the “Mic Calibration” button in the “Reference Microphone Calibration” group box.
- The application program measures the microphone output voltage and calculates the microphone sensitivity according to the calibration level selected in the “Level” group box.
- Specify a name for the file storing the calibration value of the reference microphone and click “Ok”.

2.3.2 Calibration of the artificial mouth

- Connect the reference microphone via power supply or conditioning amplifier to the UPV analyzer input with which it was calibrated.
- Select the correct analyzer channel with the radio buttons next to “Reference mic. Connected to”. The appropriate microphone calibration value must be loaded for the channel in use.
- Fix the reference microphone in front of the artificial mouth with the center of the membrane at the Mouth Reference Point (MRP).



Fig. 2-8: Position of the reference microphone for calibration of the HATS artificial mouth

- Select the calibration frequency range. Minimum for wideband telephony is 100 Hz to 8000 Hz. Observe the frequency range of the artificial mouth. Older types of artificial mouth cannot be equalized for the super-wideband frequency range.

- Checkbox “Selective” should be enabled in order to improve immunity against acoustic background noise in the test box.
- Click the “Mouth Calibration” button.
- The application program measures the sensitivity of the loudspeaker of the artificial mouth and the frequency response at -4.7 dBPa. The inverse frequency response is applied as equalization and the equalized frequency response is measured again. The equalization is updated with the inverse deviation from this measurement, and the equalized frequency response is measured once more as final check.
- Specify a name for the file storing the calibration value of the artificial mouth and click “Ok”.

2.3.3 Calibration of the artificial ear

The following instructions apply to the calibration of the type 3.3 artificial ear of the head-and-torso simulator.

- Connect the microphone of the artificial ear to R&S®UPV analyzer (input 1 or 2, respectively) via a microphone power supply or conditioning amplifier.
- Remove the pinna from the artificial ear according to the manufacturer’s instructions

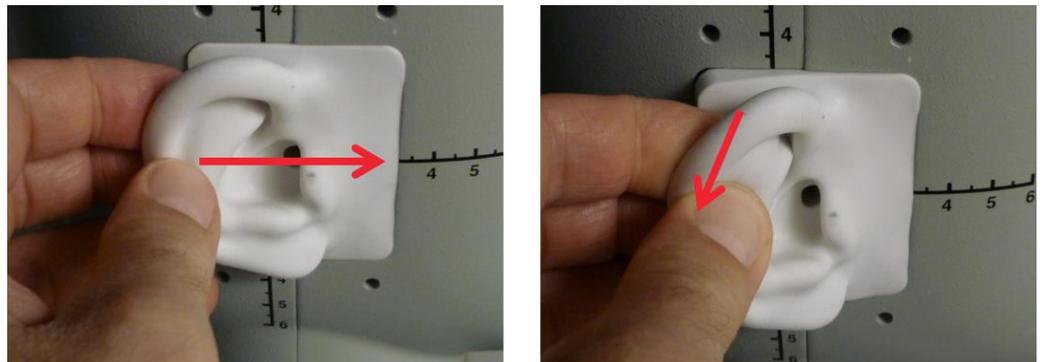


Fig. 2-9: Removal of the pinna for calibration of the artificial ear type 3.3

- Connect the sound level calibrator tightly to the artificial ear using the appropriate adapter and switch the calibrator on.

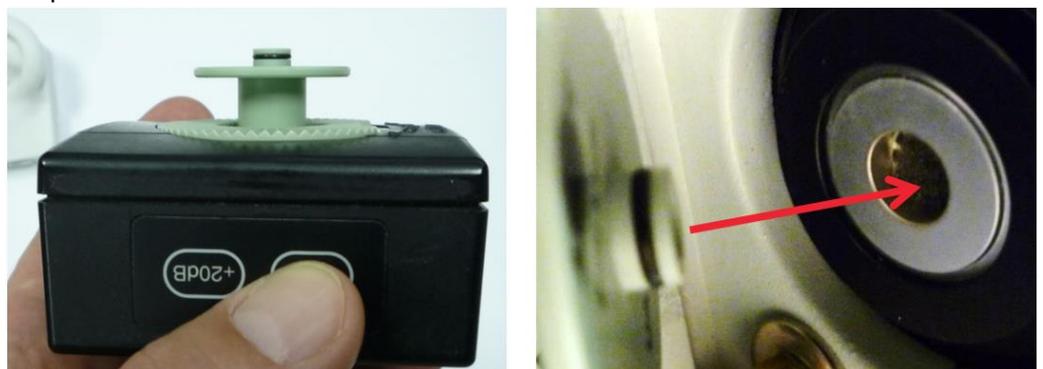


Fig. 2-10: Insertion of the calibrator with adapter into the ear simulator

- Click the “Ear Calibration” button.
- The output voltage of the microphone in the ear is measured and the sensitivity displayed with reference to 1 Pa. If the voltage measured is below 3 mV or fluctuating by more than 0.2 dB, an error message is displayed. Possible error sources are, for example, a switched-off microphone power supply or a disabled calibrator. In this case, the program requests that the test be repeated. After switching on the microphone power supply, wait approx. 20 s before restarting the calibration.
- Specify a name for the file on the hard disk in which the measured reference value is stored.

2.4 Using the monitor function

The “Monitor” tab can be used to listen to the speech signal in the current connection, using the built-in loudspeaker of the UPV or a headset connected to the headset jack on the front panel.

Before the monitor is switched on, an appropriate setup with appropriate signal file has to be loaded, using one of the second or third tab.

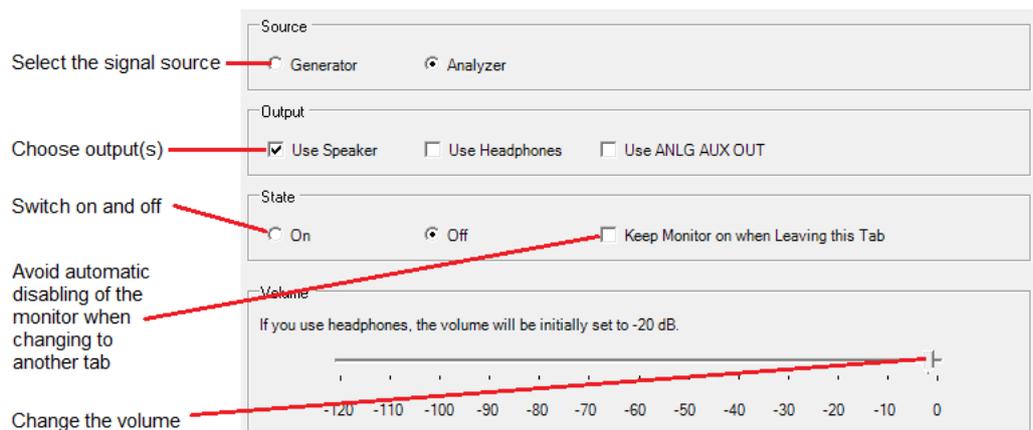


Fig. 2-11: Controls in the Monitor tab

When the headphone output is enabled and the volume setting is higher than -20 dB, the volume is automatically reduced to -20 dB to protect the listener’s ears. The volume can subsequently be re-adjusted also to higher values. When another tab in the tab control is selected, the monitor output is disabled unless the “Keep Monitor on when Leaving this Tab” checkbox is checked. In the latter case the speaker remains active even if a new setup is loaded or automated tests are started. The controls in the “Monitor” tab are disabled while automated tests are running.

2.5 Speech quality test control

2.5.1 User interface of the “Test Control” tab

The “Test Control” tab provides controls for the configuration of automatic tests.

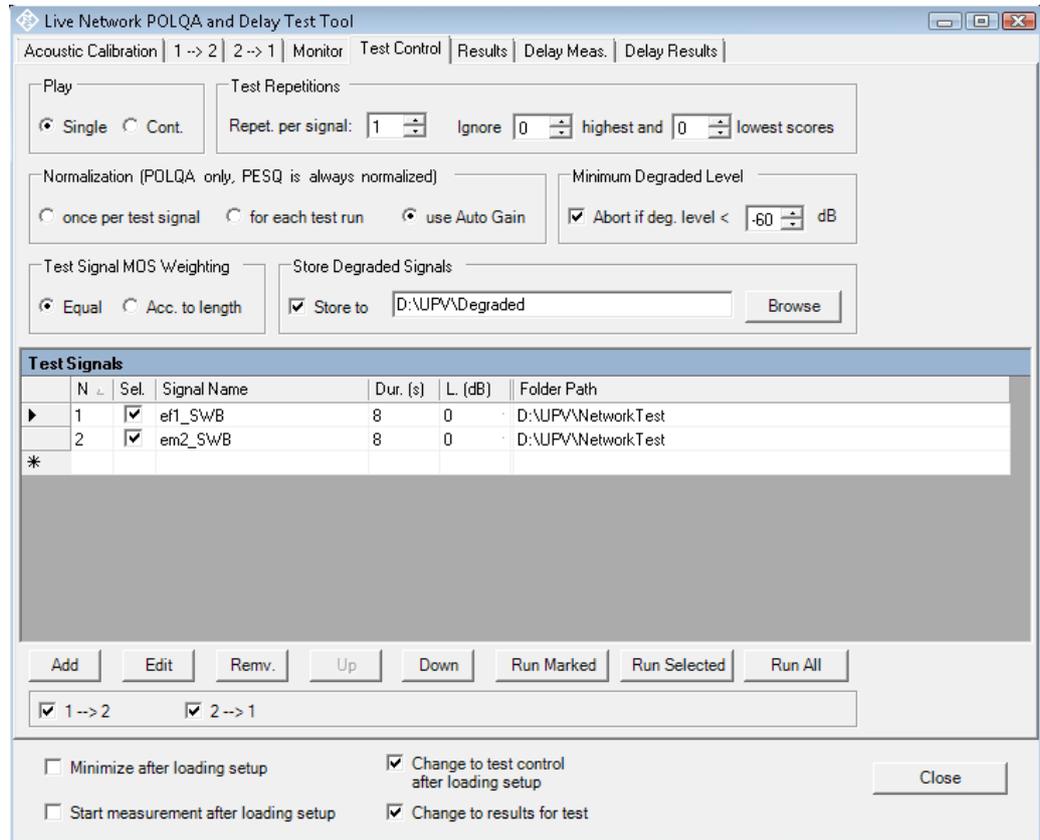


Fig. 2-12: Test Control tab

2.5.2 Compiling a list of test signals

The data grid in the center of the tab shows the compilation of test signals with which the tests are to be run.

Test Signals						
	N	Sel.	Signal Name	Dur. (s)	L. (dB)	Folder Path
▶	1	<input checked="" type="checkbox"/>	ef02_s001_SWB	8	0	D:\NTU-T Speech\SWB
	2	<input checked="" type="checkbox"/>	ef02_s002_SWB	8	0	D:\NTU-T Speech\SWB
	3	<input checked="" type="checkbox"/>	ef03_s003_SWB	8	0	D:\NTU-T Speech\SWB
	4	<input checked="" type="checkbox"/>	ef03_s004_SWB	8	0	D:\NTU-T Speech\SWB
	5	<input checked="" type="checkbox"/>	em01_s003_SWB	8	0	D:\NTU-T Speech\SWB
	6	<input checked="" type="checkbox"/>	em01_s004_SWB	8	0	D:\NTU-T Speech\SWB
	7	<input checked="" type="checkbox"/>	em02_s001_SWB	8	0	D:\NTU-T Speech\SWB
	8	<input checked="" type="checkbox"/>	em02_s002_SWB	8	0	D:\NTU-T Speech\SWB
*						

Fig. 2-13: List of test signals

The context menu of the test signal list provides the following entries:



Fig. 2-14: Context menu of test signal list

The list can be edited using the buttons below the left half of the grid or with the respective entries in the context menu. The “Add” and “Edit” buttons open an entry window where the source file and the relative level can be set:

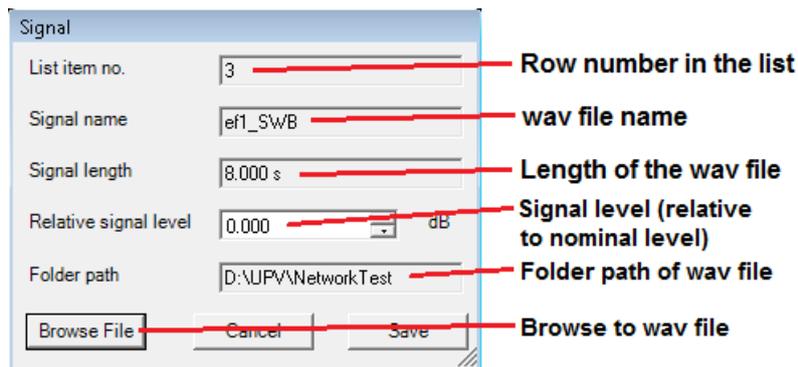


Fig. 2-15: Entry window for editing test signals or adding test signals to the list

To add a signal to the list, click the “Add” button in the Test Control tab, click the “Browse File” button in the signal window and browse to the desired speech signal file. The relative signal level is preset with the value from the tab two or three of the main window. It can be adjusted individually for each signal.

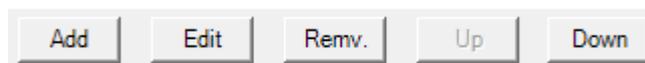


Fig. 2-16: Buttons for modifying the signal list

With the buttons “Edit”, “Remv.”, “Up” and “Down”, entries in the signal list and their order in the list can be modified.

A subset of the signal files in the list can be selected for use with the checkboxes in the “Sel.” Column.

Test signal lists can be saved, loaded and appended using the respective entries in the context menu of the test signal list.

2.5.3 Test and averaging options

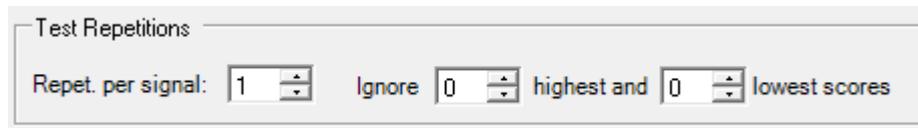


Fig. 2-17: Test repetitions group box in the test control tab

Each signal can be applied multiple times as specified in the “Test Repetitions” group box. Erroneously extreme results can be excluded from averaging. However, all results will appear in the detailed results list and taken into account for the minimum and maximum result values in the overview.

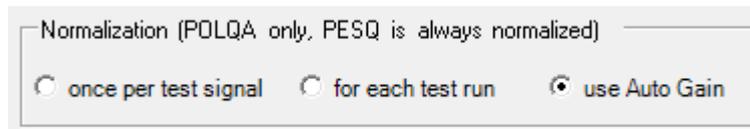


Fig. 2-18: POLQA® normalization group box in the test control tab

As POLQA® as standardized in ITU-T P.863 does not apply automatic level normalization of the degraded signal, additional normalization should be applied unless a meaningful “Gain” value calculated from all involved external gains is available and can be entered in the Analyzer function panel. The calculation of such a gain factor is not subject of this document. The first two normalization options calculate a gain factor such that the degraded level reported from the POLQA® evaluation is -26 dB. Normalization can be done once per test signal keeping the “Gain” value determined in the first test run for the subsequent tests with the same signal. Another option is to repeat the normalization for each test run which may increase test time. From UPV firmware 3.3.1, analyzer function “POLQA” offers an “Auto Gain” feature which does the level normalization for each test run in the analyzer. This way, a repetition of the test with changed gain value is not necessary.

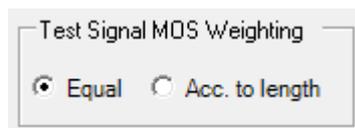


Fig. 2-19: Weighting group box in the test control tab

The results from all test runs on the same signal are averaged with the possible exclusions specified in the “Test Repetitions” field. For each sequence run, a total average is calculated from the averaged per-test-signal results. This averaging over all applied test signals can be done with equal weight or weighted according to the respective length of the test signal files. These options can be chosen in the “Test Signal MOS Weighting” group box.

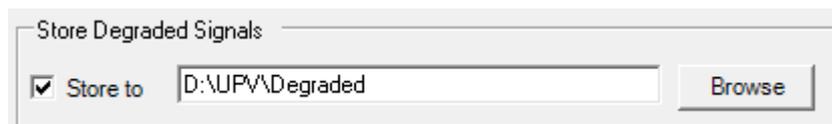


Fig. 2-20: Folder for degraded signals in the test control tab

Degraded test signals can be collected in a specified folder for subsequent analysis of degradation. The folder is specified in the “Store Degraded Signals” group box. The wav file names are automatically generated from the test file name and a number

extension. Each degraded file name will appear in the detailed results list to unambiguously assign the degraded signal file to each result.



Fig. 2-21: Minimum degraded level groupbox in the test control tab

The “Minimum Degraded Level” group box allows to set a threshold for the degraded level. If the associated checkbox is activated, a running test sequence is aborted when this limit is violated. This allows to automatically react to a lost speech connection.

2.5.4 Starting a test sequence



Fig. 2-22: Direction group box in the test control tab

Before starting a test sequence, the desired test direction(s) should be selected in the group box below the start buttons. Thus it is possible to run tests in both directions in one sequence.

If no direction is selected before the start of a test sequence, it is assumed that all tests are to be run with a setup that has been loaded to the UPV firmware either manually or using the functionality of tab two or three in this program. A confirmation for this is prompted before the sequence starts.

For each selected direction, the setup specified in the respective tab of the user interface is automatically loaded. However, test signals and levels are set according to the test signal list in the “Test Control” tab.

2.6 Speech quality test results

The last tab in the user interface is dedicated to viewing, reporting, exporting and managing results.

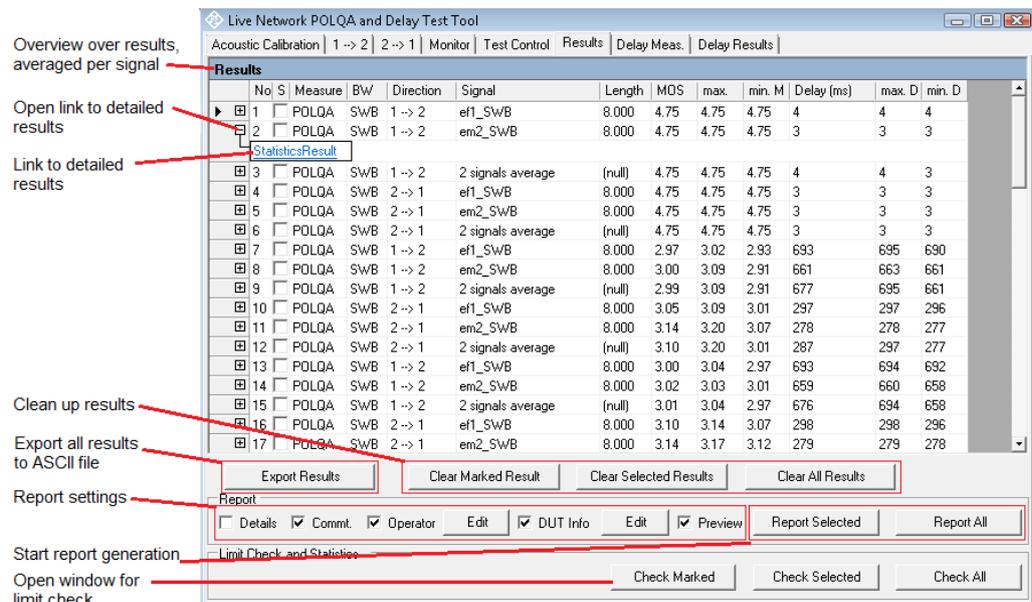


Fig. 2-23: Results tab

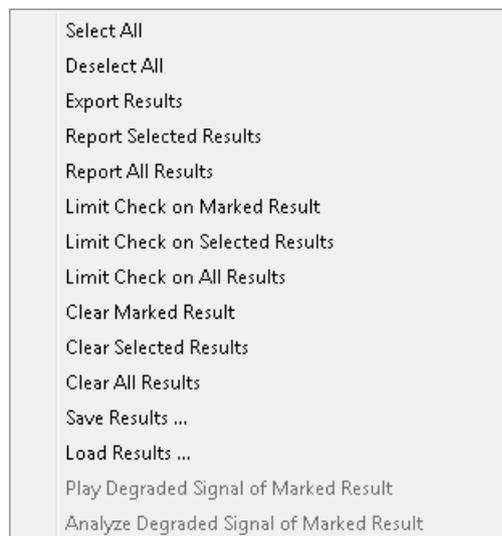


Fig. 2-24: Context menu of the results datagrid

The context menu can be used for reporting and deleting results, and for saving and loading result lists.

2.6.1 Viewing results in the data grid

Initially the data grid shows the averages for each test signal per sequence run.

Results													
No	S	Measure	BW	Direction	Signal	Length	MOS	max.	min.	M	Delay (ms)	max. D	min. D
13	✓	POLQA	SWB	1->2	ef1_SWB	8.000	3.00	3.04	2.97	693	694	692	
14	✓	POLQA	SWB	1->2	em2_SWB	8.000	3.02	3.03	3.01	659	660	658	
15	✓	POLQA	SWB	1->2	2 signals average	(null)	3.01	3.04	2.97	676	694	658	
16	✓	POLQA	SWB	2->1	ef1_SWB	8.000	3.10	3.14	3.07	298	298	296	
17	✓	POLQA	SWB	2->1	em2_SWB	8.000	3.14	3.17	3.12	279	279	278	
18	✓	POLQA	SWB	2->1	2 signals average	(null)	3.12	3.17	3.07	288	298	278	
19	✓	POLQA	SWB	1->2	ef1_SWB	8.000	3.54	3.56	3.53	725	726	724	
20	✓	POLQA	SWB	1->2	em2_SWB	8.000	3.55	3.65	3.50	695	695	694	
21	✓	POLQA	SWB	1->2	2 signals average	(null)	3.55	3.65	3.50	710	726	694	
22	✓	POLQA	SWB	2->1	ef1_SWB	8.000	3.47	3.51	3.41	299	299	298	
23	✓	POLQA	SWB	2->1	em2_SWB	8.000	3.62	3.65	3.59	280	280	279	
24	✓	POLQA	SWB	2->1	2 signals average	(null)	3.55	3.65	3.41	289	299	279	
25	✓	POLQA	SWB	1->2	ef1_SWB	8.000	3.56	3.62	3.47	728	729	728	
26	✓	POLQA	SWB	1->2	em2_SWB	8.000	3.80	3.84	3.72	699	699	698	
27	✓	POLQA	SWB	1->2	2 signals average	(null)	3.68	3.84	3.47	714	729	698	
28	✓	POLQA	SWB	1->2	ef1_SWB	8.000	3.46	3.51	3.43	735	738	733	
29	✓	POLQA	SWB	1->2	em2_SWB	8.000	3.45	3.51	3.38	735	735	734	
30	✓	POLQA	SWB	1->2	2 signals average	(null)	3.46	3.51	3.38	735	738	733	

Fig. 2-25: Overview over averaged results in the results data grid

Clicking on one of the small “+” signs in the row headers opens a new row with a link to another table showing the detailed results from which the respective row was averaged:

10	✓	POLQA	SWB	2->1	ef1_SWB	8.000	3.05	3.09	3.01	297	297	296
StatisticsResult												

Fig. 2-26: Link to detailed results in the results data grid

Clicking on the link displays the detailed data:

Results														
◀ Statistics: No.: 10 Sel.: ✓ Measurement: POLQA BW: SWB Direction: 2->1 Signal: ef1_SWB Length (s): 8.000 M ▶														
No	Measure	BW	Direction	Signal	MOS	Gain (Length	Ref. L	Deg. L	Att. (d	Delay (S/N R	S/N D	Act.
1	POLQA	SWB	2->1	ef1_SWB_eq2	3.09	0.00	8.000	-27.25	-42.12	13.74	296	36.54	39.49	0.73
2	POLQA	SWB	2->1	ef1_SWB_eq2	3.01	0.00	8.000	-27.25	-42.24	13.82	296	36.54	39.80	0.73
3	POLQA	SWB	2->1	ef1_SWB_eq2	3.04	0.00	8.000	-27.25	-42.30	13.92	297	36.54	40.67	0.73
*														

Fig. 2-27: Detailed results table in the results data grid

The back arrow “←” in the caption is the link for closing the detailed table and returning to the statistic overview table.

In the details view, the entry “Play Degraded Signal of Marked Result” is enabled in the context menu of the results data grid if the degraded signal is available in a wav file. When this functionality is chosen, a window for controlling the playback of the degraded signal opens:

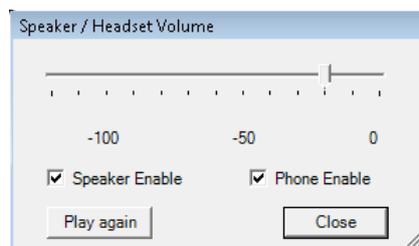


Fig. 2-28: Control window for playback of degraded signals

The window allows to choose the output and set the volume. The file is played once. Then the playback can be restarted using the “Play again” button.

Also from the results data grid it is possible to analyze a stored degraded signal with “Analyze Degraded Signal of Marked Result” and display the spectrum and waveform in a separate window. This enables investigation of causes for degradation, such as dropouts, hum noise and short noise artefacts.

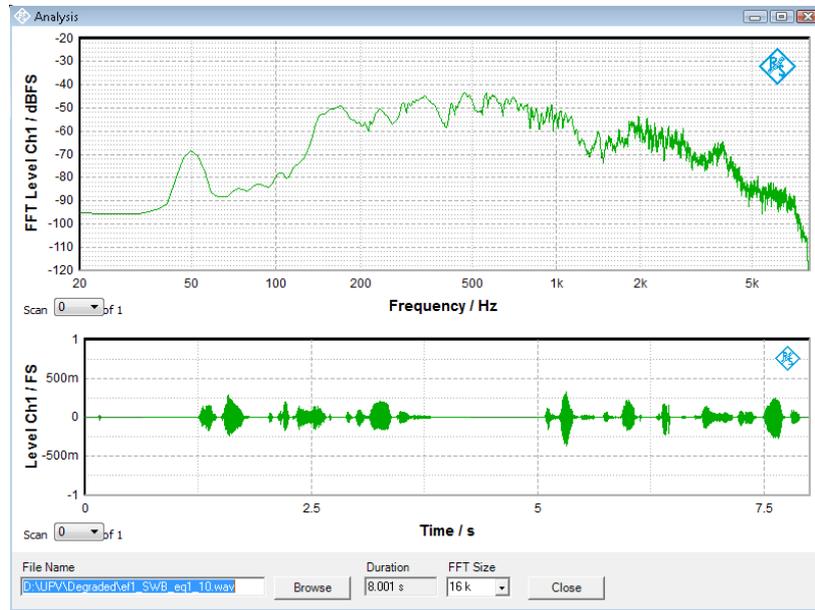


Fig. 2-29: Spectral and time-domain analysis of degraded signal

With the “Browse” button it is possible to load and analyze any wav file. The frequency resolution can be changed with the “FFT Size” combobox. This window must be closed before other functions of the application program can be operated.

2.6.2 Statistical evaluation and limit check of result data

In the overview over averaged results in the results data grid it is possible to open a window for statistical evaluation and limit check. Use context menu item “Limit Check on Marked result”, “Limit Check on Selected Results” or “Limit Check on All Results”.

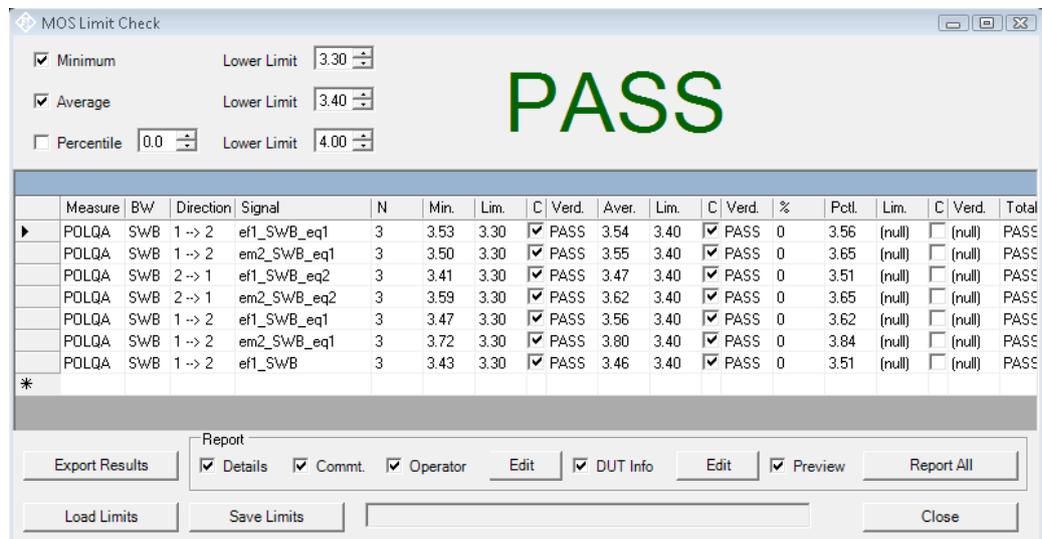


Fig. 2-30: Statistical evaluation and limit check of result data

The data grid shows minimum, average and percentile value of the MOS for each chosen test signal from the averaged results overview. Percentile means the MOS value which is greater than the specified percentage of individual results. For example 90 percent of all results are below the 90th percentile of all results.

Every change in the limit data triggers a re-calculation of the limit check with update of all verdicts. There is a verdict for each individual check, a verdict per data row as summary for each chosen test signal and a total verdict for the whole checked data.

The limit and percentile specifications can be stored to a file and retrieved from there at a later time.

The window for statistical evaluation and limit check offers similar export and reporting functionality as the test results overview(see below).

2.6.3 Entering auxiliary data

In the “Report” group box, data to be included into the printed report can be administered:

**Fig. 2-31: Report group box in the results tab**

The “Details” checkbox decides whether tables with the detailed results are included into the printed report. In this case the statistic results will be separated row by row, each with the associated detailed results table below.

If the “Commt.” Checkbox is activated, a comment will be included into the report. An entry window will pop up when the report is started.

The “Operator” checkbox specifies whether an operator’s name is to be included into the printed report. For entering a name, click the “Edit” button next to the checkbox.

The “DUT” checkbox allows to add information about the device under test. For entering name, type, manufacturer and serial number, click the “Edit” button next to it.

2.6.4 Previewing a report

For previewing a report in the Windows print preview, activate the “Preview” checkbox before clicking “Rep. Sel.” Or “Rep. All”. The latter button starts preview of all results in the data grid. The “Rep. Sel.” Button starts exclusive preview of results checked in the “S” column of the table. Finally the Windows print preview opens with a preview of the report as configured:

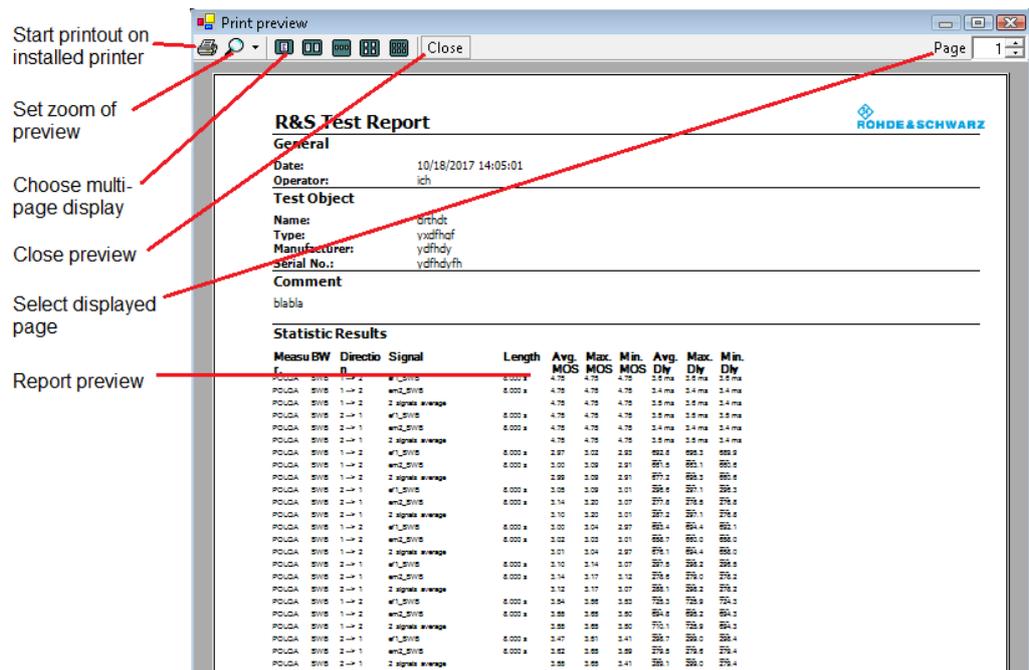


Fig. 2-32: Report preview

2.6.5 Printing a report

A report can be printed on any printer installed in the Windows operating system, including XPS document writer or pdf distiller.

The printing can either be started from the preview window by clicking the printer icon top left or by clicking one of the “Re. Sel.” or “Rep. All” buttons without the “Preview” checkbox activated.

The printer can only be selected if the latter way of starting the printout is chosen.

2.6.6 Export of result data to a spreadsheet file

The “Export results” button saves all available results to a tab-separated txt file or a comma-separated csv file which can easily be imported to a spreadsheet.

	A	B	C	D	E	F	G	H	I	J	K	L	M	N
1	Statistics													
2	No.	Measur	Bandwidth	Direction	Signal	Length	Avg. MOS	Max. MOS	Min. MOS	Avg. delay	Max. delay	Min. delay	Identifier	
3	1	POLQA	SWB	Mobile Loc	02_s001	8.500208	3.908971	3.953615	3.857836	445.5949	464.6147	424.2723	01/14/2013 09:40:45	
4	2	POLQA	SWB	Mobile Loc	02_s002	8.5	3.945699	4.0004	3.861595	429.8408	450.0769	410.1642	01/14/2013 09:43:56	
5	3	POLQA	SWB	Mobile Loc	02_s003	8.500021	3.851822	3.923364	3.752919	483.9594	507.8291	462.425	01/14/2013 09:44:59	
6	4	POLQA	SWB	Mobile Loc	02_s004	8.500031	3.728467	3.766167	3.683336	473.7631	445.7938	401.2763	01/14/2013 09:48:45	

Fig. 2-33: Test results imported to a spreadsheet (excerpt)

2.7 Delay measurements

The “Delay Measurements” tab provides controls for measuring the end-to-end delay of a speech connection.

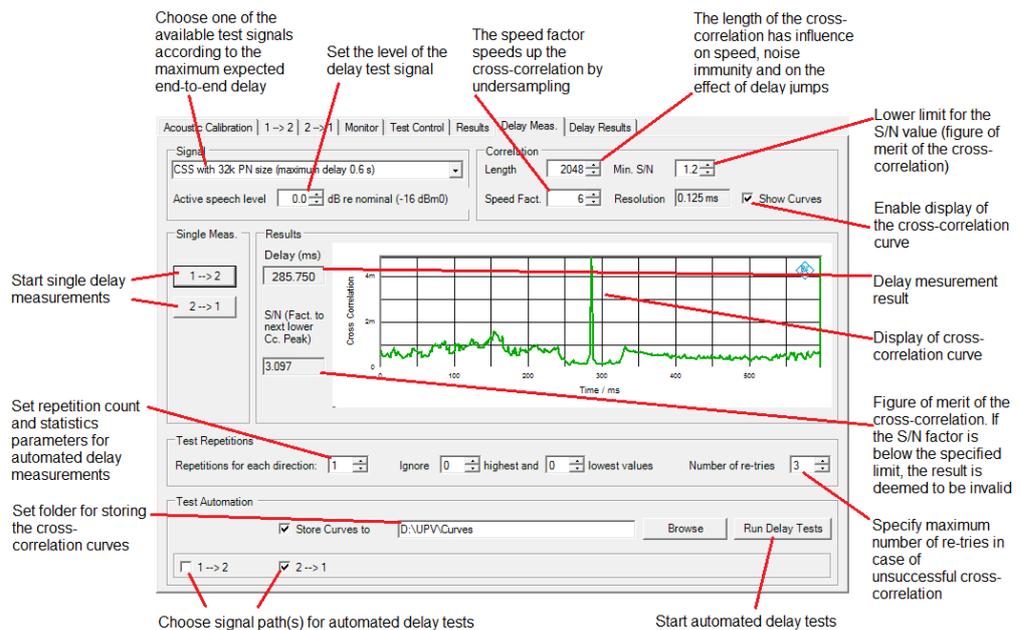


Fig. 2-34: Controls of the delay measurement tab

Delay Measurements can be performed on a pseudo-random noise signal by cross correlation. The test signal further comprises a leading voiced part to activate the speech coder. The length of the pseudorandom part determines the maximum measureable end-to-end delay. With a 32k samples pseudorandom noise, a maximum delay of 0.6 seconds can be measured. With the 64k samples pseudorandom noise, the maximum delay which can be measured is 1.2 seconds. The delay measurement works with an end-to-end delay > 10 ms. For use with codecs which do not transmit the composite source signal with sufficient fidelity, there is also the option to use a real-speech signal from ITU-T P.501.

To speed up the cross correlation calculation, the number of samples can be reduced by undersampling the test signal before cross correlation ("speed factor"). This of course is a tradeoff against time resolution and noise immunity. For each available test signal a dedicated factor is stored by the program.

During delay tests with impairments on the speech connection (fading, packet jitter and packet loss on a VoLTE connection) delay jumps may occur which render a unique delay result difficult to obtain. By reducing the length of the cross correlation operation in the "Length" field, delay measurements on heavily impaired speech connections may be improved.

The "Min S/N" field allows to specify a minimum ratio of the maximum cross correlation peak to the next lower peak. This allows to suppress invalid results in automated tests.

The display of the cross correlation curve is activated with the "Show Curves" checkbox.

The buttons in the "Single Measurement" groupbox start delay measurements for each of the two signal paths and display the result in the "Results" groupbox. The S/N factor displayed is the amplitude ratio between the highest cross correlation peak and the second highest cross correlation peak. If this ratio is less than the min. S/N value set in

the “Correlation” group box, the result is deemed to be invalid. In this case you can reduce the speed factor to improve the noise immunity of the delay measurement.

The controls in the “Test Repetitions” group box allow to set the parameters for statistics in automated testing. “Number of re-tries” specifies the number of attempts in case the S/N values is repeatedly below the specified limit.

In the “Test Automation” groupbox it is possible to start multiple delay measurements for each of the selected signal paths and do a statistical evaluation as specified in the “Test Repetitions” groupbox. It is possible to store the cross correlation result as curve in an ASCII file to a specified folder.

2.8 Delay results

Reporting and handling of delay results is similar to that of the speech quality test results. Please refer to sections 2.8.1. to 2.8.6.

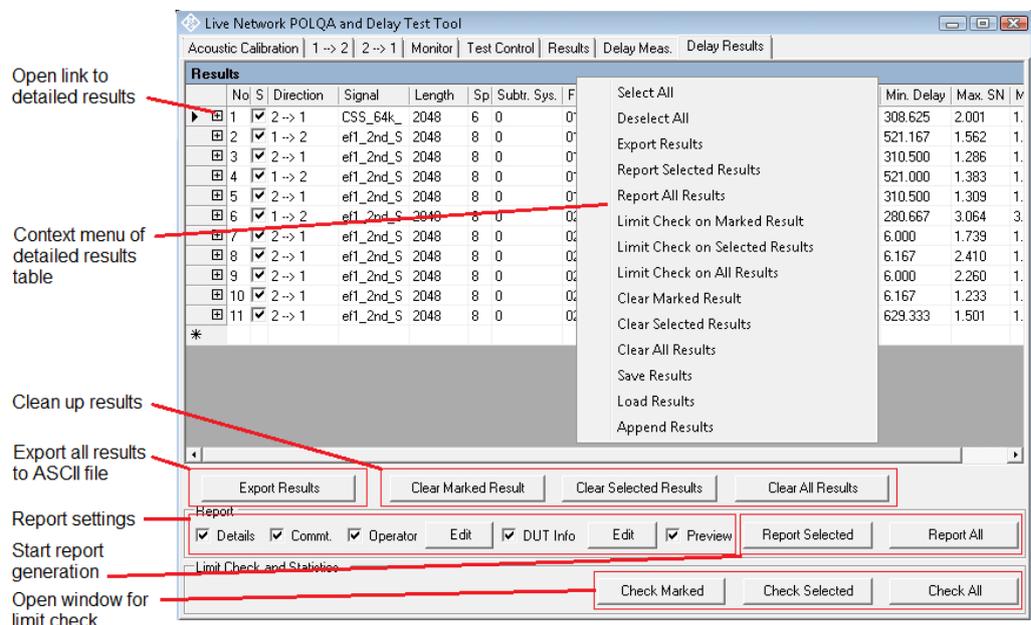


Fig. 2-35: Controls of the delay results tab

3 Literature

- [1] **3GPP** TR 26.954 14.0.0 [Online] // www.3gpp.org.
- [2] **ITU-T** Application guide for Recommendation ITU-T P.863 // P.863.1. - Geneva : ITU, 09/2014-1.
- [3] **ITU-T** Perceptual objective listening quality assessment // P.863. - Geneva : ITU, 09/2014.
- [4] **Lechner Thomas** 1GA62: Test Automation Tool for POLQA® and PESQ® Speech Quality Tests [Online] // Rohde & Schwarz corporate web site. - Rohde & Schwarz GmbH & Co. KG, September 11, 2014. - 3e. - October 16, 2017. - https://www.rohde-schwarz.com/applications/test-automation-tool-for-polqa-and-pesq-speech-quality-tests-application-note_56280-38466.html.

4 Ordering Information

Designation	Type	Order No.
Audio analyzer	R&S®UPV	1146.2003.02
Universal sequence controller	R&S®UPV-K1	1401.7009.02
Listening quality analysis POLQA	R&S®UPV-K63	1402.1156.02

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