Digital Speech Level Analyser

User Guide

Revision 4.4

Malden Electronics Ltd

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End User License Agreement

In the End-User Agreement below, the following terms are employed with these definitions:

- The End-User is the person or entity that has purchased the Software and is using it in the course of its business.
- An Affiliate is a legal entity directly or indirectly controlling, controlled by or under common control with an End-User. Control of an entity shall exist through the direct or indirect:

  a) control of 50% or more of the nominal value of the issued share capital of the entity or of 50% or more of the entity's shares entitling the holder to vote for the election of directors or persons performing similar functions,

  or

  b) right by any other means to elect or appoint directors of the entity (or persons performing similar functions) who have a majority vote.

- The Software is the application program and related programs including the speech quality metric(s) and firmware that have been supplied to the End-User and have been installed from the original media along with any product enhancements supplied by Malden Electronics Limited and come into being upon a single computer platform that is connected to a single Digital Speech Level Analyser (DSLA).

By installing and using the Software the End-User agrees to the following:

1. The Software and firmware shall only be used in conjunction with the DSLA.
2. One instance of the Software may be caused to come into being on one computer platform.
3. The Artificial Speech Test Stimulus (ASTS) may only be used for the purpose of making a measurement or a sequence of measurements, as part of which at least one such copy of the ASTS shall be processed through the Software. No more than five simultaneous copies of the ASTS may be being generated at any one time.
4. The End-User shall not copy the Software in whole or in part, other than is essential for the proper operation of the Software or for normal security back-up purposes.
5. The End-User shall not modify, translate, reverse-engineer or decompile the Software except to the extent permitted by law.
6. The End-User shall maintain the Software in confidence and ensure that it is protected from unauthorised copying or disclosure by measures that are no less stringent than those it uses to protect its own valuable information and that are, in any case, no less than reasonable in the circumstances.
7. The End-User shall prohibit the use of the Software by anyone other than the End-User, its employees and agents.
8. The acknowledgement of the rights in the Software shall not be removed from the Software or any installation of it
9. The End-User shall not transfer or assign the End-User Agreement except to an Affiliate of the End-User.
10. The validity construction and performance of this Agreement shall be governed by and interpreted in accordance with the laws of England.

Introduction

What is the DSLA?

The Digital Speech Level Analyser is a true RMS voltmeter which measures mean active speech level in accordance with ITU-T Recommendation P.56 Method B. This technique provides accurate and repeatable measurements of the energy or amplitude of complex discontinuous signals found in telecommunications and signal processing applications.

The DSLA generates any speech signal to stimulate networks and network elements, in particular an Artificial Speech Test Stimulus (ASTS) in different languages.

Options include:

- An Artificial Speech Test Stimulus (ASTS).
- The Perceptual Analysis/Measurement System (PAMS) providing a prediction of Listening Effort and Listening Quality mean opinion scores.
- Perceptual Evaluation of Speech Quality (PESQ) – ITU-T Rec. P.862 and mappings to LQO P.862.1 and PESQ-WB P.862.2
- Perceptual Speech Quality Measurement (PSQM) – (was ITU-T Rec. P.861)
- Sound file equalisation.
- DTMF analysis.

Getting Started

1. Unpack the DSLA and its power supply.
2. Check that the units are undamaged.
3. Connect the power supply jack to the jack socket on rear panel of the DSLA. The power supply unit provided will accept any AC mains supply in the range 90 to 260 volts, 50 or 60 Hz. A red LED on the front panel will light when power is applied to the unit.
    - If the mains power connector on the end of the power cable is not appropriate for connection to the type of socket in use, it may be changed. The mains power lead has three cores; Brown is live, Blue is neutral and Green/Yellow is earth. Ensure that the correct connections are made to the mains power connector. The Brown wire should connect to live supply or the terminal marked L. The Blue wire should connect to neutral or the terminal marked N. The Green/Yellow wire should connect to earth or terminal marked E or with the earth symbol. Connect the mains power lead to the power supply unit.
4. Install the software supplied as described in: Installing the Software.
5. Connect the DSLA to your PC. (See: Connecting a DSLA to your PC).
6. A monitor output is provided on 4mm banana connectors and a 3.5mm headphone socket. The monitor output is capable of driving headphones or amplified speakers.

Installing the Software

System Requirements

Before you install the product, you should make sure that your computer meets the minimum system requirements.

- Windows 2000/XP.
- Pentium-class PC (200 MHz or higher recommended)
- 64 megabytes of RAM
- Screen resolution of 1024 x 768 @ 16 colour quality recommended
- DirectX 8.1 or higher recommended

To install the software

1. Insert the CD in the CD-ROM drive. The installation process should start automatically, if however it does not, follow these steps:
   - From the Windows Start Bar, click Run.
   - Type <DRIVE>:\SETUP.EXE where <DRIVE> is the CD-ROM drive you are using and click OK.
2. The setup program prompts you through the installation process. Follow the instructions on the screen.

Notes

- You should exit all applications before running the setup program for software installation.
- The Quickstart Schedules and Playlists are structured for operation from the C: drive. If you have installed the software on another drive, it is helpful initially to have a copy of the Playlists folder in C:\Program Files\DSLA.

Using DSLA

Connecting a DSLA to your PC

To control the DSLA, it must be connected to your PC via a standard RS232 serial port, or to your LAN using the Ethernet TCP/IP connection. The default connection setting is to use a serial port connection.

If you wish, you may leave the DSLA connected via both serial and LAN connections, although only one control method is in use at any one time.

When using a LAN connection, the DSLA will be assigned an IP address via DHCP if a DHCP server is present. If your LAN does not use a DHCP server, you must manually assign the DSLA an IP address.

The current DSLA connection status is displayed in the lower left corner of the DSLA application's status bar.

To connect a DSLA to your PC

1. Do one or both of the following:
   - Connect the DSLA to your PC's serial port using the supplied 9 way serial lead.
   - Connect the DSLA to your PC's Ethernet connection using the supplied lead, or to your LAN using a standard Ethernet lead (not supplied).
2. Power up the DSLA.
3. Start the DSLA application.
4. On the **Tools** menu, click **Options**.
5. Do one of the following:
   - To use a serial connection select the **Use Serial Port** option.
   - To use a LAN connection, click the **Browse** button and select the DSLA you wish to connect to.
6. Click **OK**.

To manually assign an IP Address to the DSLA

1. Establish a serial connection to the DSLA.
2. On the **Tools** menu, click **Options**.
3. In the IP Address box, enter the **IP address** you want to assign to the DSLA.
4. Click **OK**.
5. Click **Yes** to the 'DSLA IP Configuration' dialog box that appears.

Signal Connections

Each channel has two identical sets of signal connections. The telephone jacks enable the DSLA to behave as a telephone; it can go off-hook, dial DTMF strings, output speech and measure the incoming signal. The 4mm connections provide a 4-wire connection point, i.e. an input and output. Finally, the Handset Port is an RJ22 type connector and enables simple, reliable connection to any microphone/earpiece device, including digital telephone (IP phone) handset, cellular phone handsfree port, PC microphone/line out jacks.

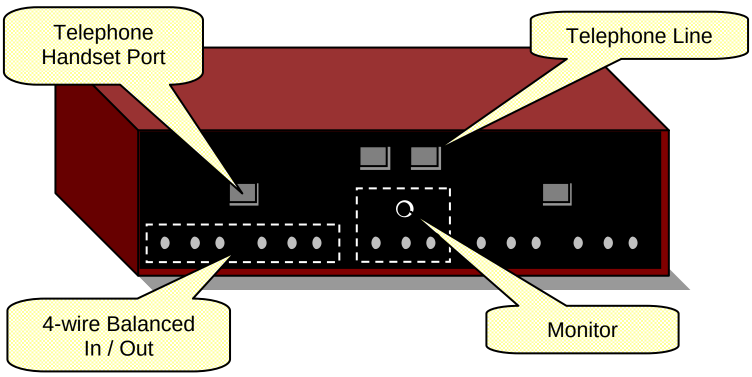
The measurement input is selected in the Channel Window status bar. The output is always present on both the telephone jack and the 4mm connector of each channel and is switched to the handset port when selected. It is possible for the DSLA to measure the signal on the telephone line on the 4mm inputs in high impedance mode. Characterisation of the line is easily performed in this way.

The 4mm connectors have accurate 600 ohm terminations. The telephone jacks are 600 ohm terminated but the signal developed across the phone line will be dependent upon the impedance presented by the phone line and line card at the switch. There is a complex impedance termination for the telephone jacks. The impedance is 270 ohms in series with 150nF paralleled with 750 0hms. The output impedance of the handset port is 25 ohms.

DSLA was designed to generate speech signals. The peak to mean ratio of speech is typically 15-18dB so the DSLA will generate speech at 0dBm mean active speech level, with some headroom. The balanced outputs on each channel provide a clean signal at these (and lower) levels.

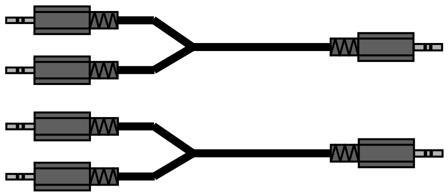
The telephone connections apply the signal directly to the network which clips around +8dBm (or less). The mean active speech level on the network is about –26dBm, depending upon the country and specific network types. The DSLA can generate a –20dBm mean active speech level signal, which will appear on the network at an appropriate level depending upon the network loss plan.

DSLA with Handset Front Panel Signal Connections



Examples of signal connectors

Analogue Connection	DSLA Port	Connectors
Telephone Network	Telephone Line	RJ11
Telephone Handset Port	Telephone Handset	RJ-22
PC Sound Card Line In / Line Out	4-wire Balanced	Type C
PC Sound Card Mic In / Line Out	Handset	Type A + Type B
Cellular Telephone	Handset	Type B + suitable handsfree/2.5mm adaptor available from cellular accessory retailers
Acoustic Test System (Head & Torso Simulator; Artificial Ear & Mouth)	4-wire Balanced	Depends on specific equipment in use

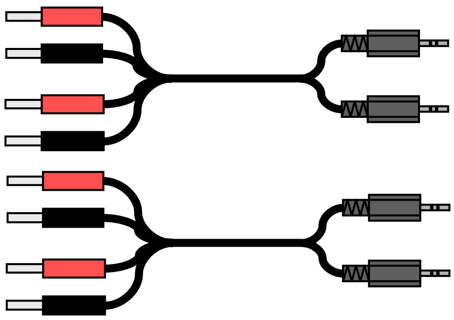


Type A

2 x 3.5mm male jack to
1 x 3.5mm male jack

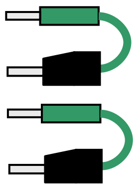
Type B

RJ-22 plug to
3.5mm female jack



Type C

4 x 4mm plug to
2 x 3.5mm male jack



Type D

4mm plug to
4mm plug jumper

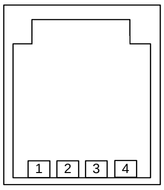
Handset Port

A connection at microphone level to the handset port of any telephone (POTS, ISDN, IP, etc) or to the handsfree port of a cellphone can be achieved directly from the DSLA balanced output. However, the Playlist will require a signal of perhaps –50dBm mean active speech level to avoid clipping the microphone input stage of the equipment under test. This is not sufficiently above the noise floor of the DSLA and will mean that some impairment exists in the source signal.

The impedance of the DSLA balanced output is 600 ohms. Typically, the microphone will have a lower impedance and therefore the DSLA will be mismatched to the microphone input stage of the telephone or cellphone.

Finally, an electret microphone has a bias voltage which is applied to the DSLA output if the DSLA is connected to a microphone amplifier. Problems can arise from power supply leakage currents and inappropriate earthing.

The DSLA Handset Port is an RJ22 connector. The connections are presented as shown in the diagram below



Pin	Description
1	Output –
2	Input -
3	Input +
4	Output +

Using the Handset Port

Remove the coiled lead from the telephone handset and connect it to the Handset port. A cell phone or other device will require an appropriate adapter to connect to the hands free port or the soundcard etc.

The call must be established, possibly using the telephone’s keypad. Set the DSLA monitor to input on each channel so that call progress tones and messages can be heard on the monitor loudspeaker or headphones. Alternatively, the call can be established automatically from the POTS line on the other channel of DSLA if that is to be the termination or it may be possible to establish the call using an appropriate executable program called from a Playlist to control the network under test.

Once the call has been established, Playlists that have been developed for telephone circuit tests can be used directly with the handset port without any change to speech levels. In some equipment, the handset port is the only way to establish a four-wire connection that can be used for the measurement of echo and other parameters.

The Quickstart schedules and playlists provide a number of useful examples of measurements of speech levels and speech quality between the Handset ports and other network terminations.

The user should recognise that the microphone input stage of the telephone base unit or cellphone should have an input impedance greater than 1Kohms and the gain to the network junction should be no more than 30dB.

Using Quick Start Examples

The Quick Start examples are easily started by selecting an item from the ‘Help|Quick Start Examples’ menu. The Quick Start examples are designed to give DSLA users an insight in using DSLA. For most users, the Quick Start examples should only need a little editing to achieve first time results. However due to the varied nature of the DSLA test environments, you may need to make some changes to best suit your needs.

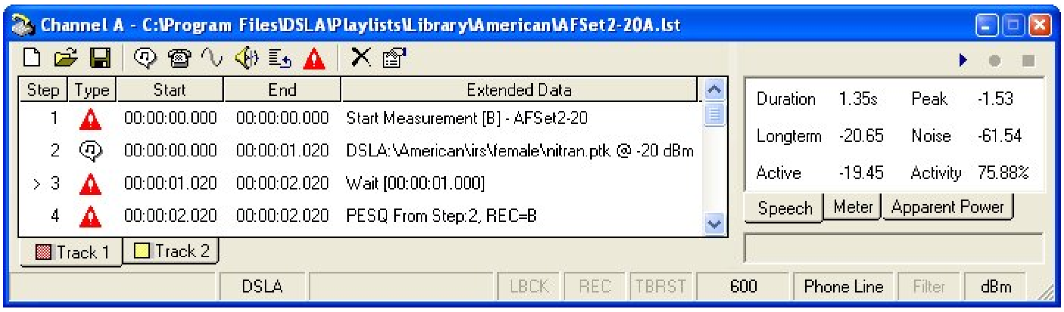
Schedules and playlists are provided to establish and verify a connection, measure speech quality at different levels and to measure speech quality at a fixed level with a range of speech material for male and female talkers.

Before You Begin

- If this is your first time using the DSLA, you might like to familiarise yourself with some of the DSLA’s basic concepts, such as Playlists and using the Scheduler.
- Many of the Quick Start examples use the ASTS files, so you must ensure that you have copied your ASTS files to the DSLA before use. See ‘Copying Sound Files to DSLA’ for more information.
- Sound Files on your hard disk in the Phonytalk folder (Local Files) can be used within a Playlist, as can files that have been downloaded to DSLA memory.
- Make sure that you have correctly connected the DSLA hardware to your test environment.
- If you are using a Quick Start example that uses a PSTN dial-up connection, then you may also need to edit the dialling string within the relevant ‘Call Setup’ Playlist to dial your required number. Remember to save the Playlist afterwards before using the schedule.

Channel Window

Menu: View|Channel (A/B)



Using the Channel window(s) you can:

- Start and stop measurements.
- View Speech Level, Voltmeter and Apparent Power measurements.
- Open, create, edit and save Playlist scripts.
- Configure DSLA channel settings such as input impedance etc.

Using the Channel window

If you need to do more with the DSLA than just simple speech level measurements, you will need to either open or create a Playlist script.

A Playlist consists of a series of events that define what the DSLA is to output (or *play*), to control measurement progress, and control various other DSLA functions. A Playlist has two tracks, on which events can be added. The Playlist events are interpreted by the DSLA when a channel is started.

To start a measurement

1. Open the **Channel** window(s) you wish to start a measurement on.
2. On the Channel window toolbar, click **Start** ▸ to start the measurement on that channel.

To stop a measurement

1. On the Channel window toolbar, click **Stop** ■ to stop the measurement on that channel.

To create a Playlist

1. On the View menu, select either **Channel A** or **Channel B**.
2. Click the **Track** tab to select the required track you want to add events to.
3. Do one of the following:
   - To add an event to a Playlist, click the required **Event** icon. (See: Playlist Events)
   - To delete an event from a Playlist, click the event to be deleted, and then click **Delete**
   - To edit an event, double-click the event you wish to edit, or click the event and then click **Properties**

Notes

- To add an event to the end of a track, click the blank space following the last event, so that the event pointer ['>' symbol] points to an empty step.
- To add an event in the middle of a Playlist, click the step where you wish to insert an event, and then click the required event icon on the Channel toolbar.

To insert an existing Playlist

You may wish to insert smaller, predefined Playlist sequences into the current Playlist to save you recreating the same sequences over again. An existing Playlist can be inserted anywhere within the current Playlist, or appended to the end of the current Playlist.

1. Click the **Playlist step** where you wish to insert an existing Playlist.
2. Right-click in the Playlist window and select **Insert Playlist** from the context menu.
3. In the 'Look in' box, click the drive or folder that contains the Playlist.
4. In the folder list, double-click folders until you open the folder that contains the Playlist you want.
5. Double-click the Playlist you want to open.

Note

- When inserting a Playlist within a Playlist which contains Repeat or PAMS events, take care to make sure the events, 'Start From Step' property is still valid.

To save a Playlist

1. Click **Save Playlist** on the Channel window that contains the Playlist you want to save.
2. Select the format to save the Playlist file as using the **File Type** drop-down list box.
3. In the File name box, type a name for the Playlist.
4. Click **Save**.

Viewing the Results

The measurement results are displayed in the right hand side of the channel window. There are three selectable result displays:

Speech	*Duration*: Shows the measurement period so far in seconds. *Longterm*: Shows the average level of the input signal in the selected measurement unit over the entire measurement period. This is a true RMS reading. *Active*: This is the mean active speech level in the selected measurement unit. The active level result is repeatable for 5 seconds or more of speech. *Activity*: This is the activity factor expressed as a percentage of the total time where the mean active speech level is exceeded, i.e. speech is present. *Peak*: This is the highest instantaneous level in the selected measurement unit.
Voltmeter	Displays level in current measurement units.
Apparent Power	Displays apparent power in current measurement units.

Additional Results

A delay figure, expressed in milliseconds, is displayed in the channel status bar. (Only when a Delay Event is used within a Playlist). Delay of signal is accurately expressed within 3ms for a sine wave or within 5ms for speech, provided the speech signal on both channels is active when the measurement is stopped.

The Dual Activity result is displayed at the bottom of the DSLA window when measurements are made on both channels simultaneously.

Setting channel properties

The current channel settings are displayed in the right-hand side of the status bar. Clicking on the setting will toggle through the available options.

The available settings are:

Input Impedance:	600, High, Complex
Input Selection:	Balanced, Phone Line, Handset
Filter:	Off, On (200Hz-5kHz bandpass filter)
Measurement Units:	dBm, dBv, dBr and mV

Playlist Synchronisation

There are a maximum of four tracks of Playlist sequences (two per channel). However, the initiation / execution of any Playlist item is performed sequentially (one at a time) and different Playlist commands take varying amount of time to execute. For example, the Start Measurement command takes around 2ms to execute and will therefore introduce this amount of delay between measurements on both channels.

Where it is necessary for Playlists to be in synchronisation (especially when starting a measurement) it is recommended that one Playlist issue a ‘Sync On’ command and the other Playlist(s) wait until the former is ready by containing a’ Wait For Sync’ command. It may be necessary to experiment to determine which Playlist should issue 'Sync On'.

Playlist Events

Speech

Plays back a sound file at any defined mean active speech level.

Icon 

Parameters

File	File to playback. Can be either a DSLA memory resident file, or a local file (Ethernet option only)
Output Level	Specifies the output level of the file in the units at the time the event is created.

DTMF

Generates the DTMF tones 0-9 ABCD #*.

Icon 

Parameters

DTMF Sequence	Specifies the DTMF sequence to generate.
Dial Outside Line	Prefixes the DTMF sequence with the specified number you use to access an outside line.
Duration	Specifies the duration of each DTMF digit.
Output Level	Specifies the output level of the DTMF digits in the current units.

Remarks

- Use the ‘,’ (comma) character in the DTMF sequence to insert a 2 second pause.
- DTMF Analysis Option users can specify advanced DTMF parameters, such as inter-digit duration by clicking the **Advanced** button and selecting the **Enable Advanced Options** check box.

Noise

Generates ‘White’ noise.

Icon 

Parameters

Duration	Specifies the duration of the noise signal.
Output Level	Specifies the output level of the DTMF digits in the current units.

Sinewave

Generates either a fixed frequency Sinewave from 10–15kHz, or a swept Sinewave at any defined level.

Icon

Parameters

Use Fixed Frequency	Specifies the fixed frequency in Hz.
From / To Frequency	Specifies the swept frequency in Hz.
Duration	Specifies the duration of the Sinewave.
Output Level	Specifies the output level of the Sinewave in the current units.

Repeat

Repeats a group of Playlist events.

Icon

Parameters

From Step	Specifies the event step from which you wish to start the repeat group from.
Repeat	Specifies the number of times to repeat.

Example

The following example starts a measurement on Channel A, and then repeats two speech phrases 10 times.

Step	Type	Description
1		Start Measurement [A] - Repeat Example
2		Speech 1
3		Speech 2
4		Step = 2, Repeat = 10

Continue Measurement

Re-starts a mean active speech level measurements without resetting the previously accumulated results.

Icon

Delay - Active Level

Calculates network delay using the Active Level method.

Icon

Remarks

- This event must be used after a 'Start Measurement [BOTH] event.
- Recommended for tone signals and compressed speech.
- The signal delay is accurately expressed within 3ms for a sine wave or within 5ms for speech, provided the speech signal on both channels is active when the measurement is stopped.

Example:

Channel A Track 1

Step	Type	Description
1		Start Measurement [BOTH]
2		Wait [00:00:01:00]
3		Delay - Active Level
4		Stop Measurement [BOTH]

Channel A Track 2

Step	Type	Description
1		Speech file 1
2		Speech file 2

The signal sent on Channel A should be received by Channel B. The Wait duration should be set to ensure that the delayed speech received by Channel B is active when the measurement stops.

Delay - End-End-End

Calculates network end-end-end delay.

Icon 

Parameters

Start Measurement from Step	Specifies the event step from which the measurement begins.

Remarks

- This delay measurement differs from round-trip delay in that ITU-T G.107 specifies RTD within network junctions. This is a true measurement from one end point to another.

Example:

The following example sets Channel A to start a measurement and play the deluttfem.wav speech phrase. Channel A then waits for the deluttm.wav speech phrase played from Channel B and finally calculates the End-End-End delay.

Channel B is set to wait until the deluttfem.wav speech phrase played from Channel A is detected and then plays the deluttm.wav speech phrase.

Channel A

Step	Type	Description
1		Start Measurement [A] – E3 Example
2		deluttfem.wav @ -20dBm
3		Wait - Speech [deluttm.wav]
4		Delay [End-End-End] – from step [2]

Channel B

Step	Type	Description
1		Wait – Speech [deluttfem.wav]
2		deluttm.wav @ -20dBm

Delay - Time Align

Calculates delay between the reference and delayed signals using the Time Alignment method in the DSLA.

Icon

Parameters

Ref Channel	Specifies the reference channel (A / B).
Ref Input, Output	Specifies whether the reference data is present at the input or output of the reference channel.
Dly Channel	Specifies the delayed signal input channel.

Remarks

- This event makes allowance for the delay inherent in the DSLA and so may differ in result from the average time offset displayed in the Speech Quality Analysis results window.

Example

The following example starts a measurement on Channel A and initiates a delay measurement on the signal received by Channel A using the Time Align method.

Channel A Track 1

Step	Type	Description
1		Start Measurement [A] – Delay [TA] Example
3		Speech file 1
4		Speech file 2
5		Stop Measurement [A]

Channel A Track 2

Step	Type	Description
1		Delay –TA [Ref Chan :A ,OUT Delay Chan :A]

DTMF Analysis (available only if DTMF Analysis option installed)

Analyses received DTMF tones.

Icon

Parameters

Expected Sequence	Specifies the expected DTMF digits.
Record from Channel	Specifies the receive channel for the DTMF tones.

Execute Application

Executes an external application.

Icon 

Parameters

File Path	Full file path to the application to be executed.
Description	Description to identify this event.
Wait for Process	If enabled, the DSLA application will wait until the external application terminates.

Remarks

- This event must be used in conjunction with a 'Start Measurement' event.
- This event can be placed anywhere within a Playlist, but the specified application will only be executed after all other events on both channels have stopped.

Gain

Allows the input level to be amplified or attenuated after the A/D converter.

Icon 

Parameters

Gain	Specifies the amount of gain/attenuation to be applied to the input signal in dB.

Remarks

- 40dB maximum gain is available.
- You must reset the gain to 0dB after using this event.

Phone Off-Hook

Sets the DSLA phone interface to off-hook state.

Icon 

Remarks

It is important to wait a second for the line card to respond and the DSLA phone circuit to be powered after going off-hook.

Phone On-Hook

Sets the DSLA phone interface to on-hook state.

Icon 

Silence

Inserts a silence period between 'Speech' events when using a 'Speech Quality Analysis' event.

Icon 

Parameter

Duration	Specifies the silence duration.

Remarks

- This event is not the same as using a 'Wait' event.

Start Measurement

Starts mean active speech level measurements on the selected channel(s).

Icon 

Parameters

Caption [Optional]	Description used to identify this result set.
Starts	Specifies on which channel(s) to start a measurement.

Remarks

- You must use this event if you want your results logged to a results file, or included in the Results Manager for later analysis.

Stop Measurement

Stops mean active speech level measurements on the selected channel(s).

Icon 

Parameters

Stops	Specifies on which channel(s) to stop a measurement

Speech Quality Analysis

Executes a speech quality analysis test.

Icon

Parameters

Start Measurement from Step	Specifies the event step from where the reference measurement data starts.
Record from Channel	Specifies the receive channel for the degraded data source.

Remarks

- This event must be used in conjunction with a ‘Start Measurement’ and at least 4 seconds of ‘Speech’ events.
- A measurement must have been started on the channel that the ‘Record from Channel’ parameter has been set to.

Example

The following example starts a measurement on Channel A, plays three speech files and executes a speech quality analysis test using the Channel A input.

Step	Type	Description
1		Start Measurement [A] - PAMS Example
2		Speech file 1
3		Speech file 2
4		Speech file …
5		Wait [00:00:01:00]
6		PAMS from Step 2, REC = A

Step 5 is important to allow sufficient time for the speech signal to traverse the network..

Sync

Continues Playlist execution after a ‘Wait for Sync’ event has been executed.

Icon

Parameters

Sync Status	On / Off

Remarks

- The Sync status must be set ‘Off’ before another ‘Wait for Sync’ event is used.

Wait

Waits for the specified duration.

Icon 

Parameters

Duration	Specifies the duration to wait before Playlist execution continues.

Wait - Active Level

Waits until the specified Active Level has been reached for the given minimum period.

Icon 

Parameter

Active Level	Specifies the mean active speech level to detect
Minimum Period	Specifies the minimum period to detect the specified mean active speech level before Playlist execution continues.

Wait - Channel

Waits until the specified channel and track are not waiting.

Icon 

Parameters

Channel	A, B
Track	1, 2

Wait - DTMF

Waits until the specified DTMF sequence has been detected.

Icon 

Parameters

DTMF Sequence	Specifies the DTMF sequence to detect before Playlist execution continues.

Wait - Ring Detect

Waits until the specified number of ring signals have been detected.

Icon 

Parameters

Ring	Specifies the number of times the ring signal should be detected before Playlist execution continues.

Note that the ringing voltage must exceed 18Vrms.

Wait - Speech

Waits until the specified speech file has been detected.

Icon 

Parameters

Speech File	Specifies the speech file to detect before Playlist execution continues.

Wait - Sync

Waits until a ‘Sync = On’ event is executed.

Icon 

Wait - Tone

Waits until the specified tone has been detected.

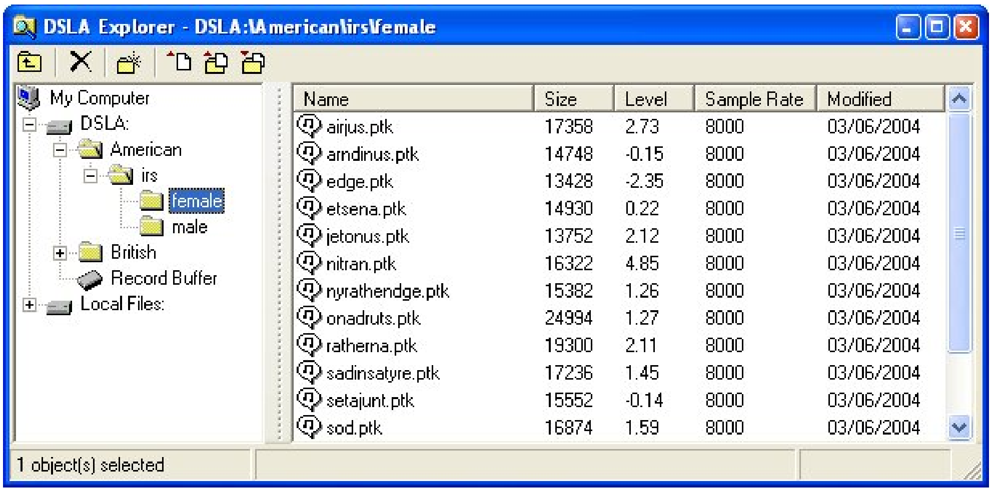
Icon 

Parameters

Frequency	Specifies the frequency of the tone to detect in Hz.
Tolerance	Specifies the tolerance of the Frequency parameter in Hz.
Tone Duration	Specifies the minimum duration of the tone to be detected in mS.
Wait Until Tone Stops	By default, Playlist execution continues after the tone has been detected. If this option is enabled, Playlist execution will continue once the detected tone stops.

DSLA Explorer

Menu: Tools|DSLA Explorer



Using the DSLA Explorer, you can:

- Create and delete folders to organise sound files within DSLA memory.
- Delete old or unused sound files from DSLA memory.
- Copy new sound files from your PC to DSLA.
- Save sound files from DSLA to your PC in a variety of file formats.
- Save the contents of the DSLA record buffers.
- Preview sound files within DSLA memory.

Using DSLA Explorer

To create a new folder

1. Open the **DSLA Explorer** window (Tools|DSLA Explorer).
2. In the folder view, select where you want to create a new folder.
3. On the DSLA Explorer toolbar, click **Create New Folder**
   The new folder appears with a temporary name.
4. Type a name for the new folder, and then press **ENTER**.

To rename a file or folder

1. Open the **DSLA Explorer** window (Tools|DSLA Explorer).
2. Click the **file** or **folder** you want to rename.
3. Press **F2**.
4. Type the new name, and press **ENTER**.

To delete a file or folder

1. Open the **DSLA Explorer** window (Tools|DSLA Explorer).
2. Click the **file** or **folder** you want to delete.
3. On the DSLA Explorer toolbar, click **Delete** .

To refresh DSLA Explorer

1. Press F5 to refresh the DSLA Explorer window if you have added files to the Phonytalk folder.

Copying sound files to DSLA

Before you can use the DSLA to playback sound files, such as WAV files or ASTS files, they must first be copied to DSLA memory. You can either copy individual files, or you can choose to copy a folder containing your files to DSLA.

Copying a folder can be useful when you want to use your sound files as the source for speech quality testing, as your sound files must exist within the same DSLA folder hierarchy as they appear on the PC in order that the speech quality metric can access them.

Ethernet equipped DSLA's can also replay sound files directly from the PC hard disk.

To copy sound files to DSLA

1. Open the **DSLA Explorer** window (Tools|DSLA Explorer).
2. Select or create the folder where you wish to copy the sound file(s).
3. Do one of the following:
   - To copy individual sound files to the DSLA, click **Send Sound File** Using the Open dialog box, select the file(s) you wish to send to the DSLA, and click **Open**.
   - To copy a folder and all its sub-folders to the DSLA, click Send **Folder Structure** Using the Browse for Folder dialog box, select the folder you wish to copy, and click OK.

Saving sound files to your PC

You can save sound files stored in DSLA memory to your PC (except for ASTS files). Sound files can be saved in the following formats:

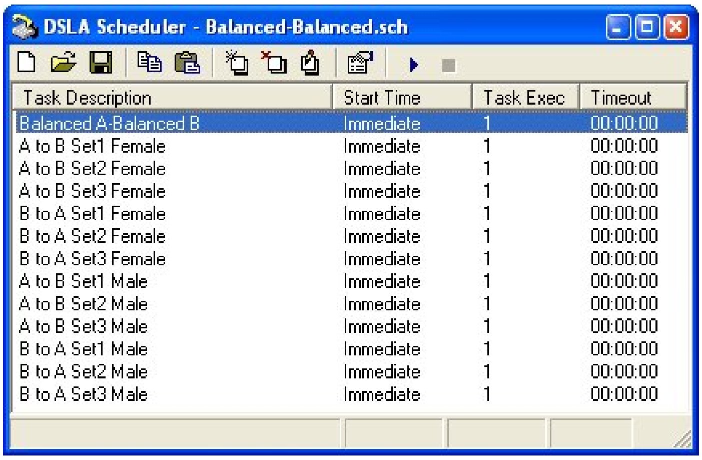
- Windows Wave PCM, 8/16k, 16 bit, mono
- A/mu-Law Wave 8k, 16 bit, mono
- A/mu-Law Raw 8k, 16 bit, mono
- PCM Raw 8k, 16 bit, mono

To save sound files to your PC

1. Open the **DSLA Explorer** window (Tools|DSLA Explorer).
2. Click the sound file you want to save to disk.
3. On the DSLA Explorer toolbar, click **Save Sound File**
4. Using the Save dialog box, select the folder where you wish to save the file.
5. Select the format to save the sound file as using the **Save as type** drop-down list box.
6. Click **Save**.

DSLA Scheduler

Menu: Tools|Scheduler



Using the DSLA Scheduler, you can:

- Automate execution of DSLA Playlist scripts.
- Execute a test set repeatedly, or a specified number of times.

The DSLA Scheduler is ideal for use where a number of different tests and/or setup routines are required. A Schedule consists of a sequence of tasks which are used to define individual test setups.

Using DSLA Scheduler

To create a schedule

1. Open the **Scheduler** window (Tools|Scheduler).
2. Do one of the following:
   - To add a new task, click **Add Task**
     Configure the task as required using the 'Task Properties' dialog box.
   - To insert a task, right-click on the task where you wish to make an insertion and select 'Insert' from the context menu.
   - To edit a task, double click the task you wish to edit.
   - To delete a task, select the task you wish to delete and click **Delete Task**
3. Repeat Step 2 until you have added all the required steps for this schedule.
4. Click **Scheduler Properties**
   Configure the Scheduler using the Scheduler Properties dialog box.

To open a schedule

1. Open the **Scheduler** window (Tools|Scheduler).
2. Click **Open Schedule**
3. Navigate to the folder that contains the schedule file.
4. Double-click the Schedule you want to open.

Note

- Schedule files can also by opened by selecting the appropriate Schedule file name displayed in the Help|Quick Start menus.

To save a schedule

1. Click **Save Schedule**
2. In the File name box, type a name for the Schedule.
3. Click **Save**.

To delete a schedule task

1. Click the **task** to be deleted.
2. Click **Delete Task**

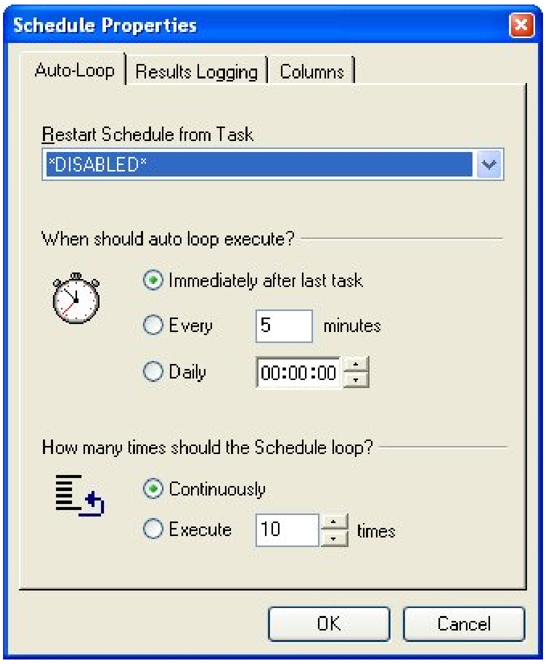
To start a schedule

1. Open the **Scheduler** window (Tools|Scheduler).
2. Do one of the following:
   - To open an existing schedule file, click **Open Schedule**
   - Create a new Schedule.
3. Click the **task** from which you wish playback to start.
4. Click **Start Schedule**

To stop a schedule

1. Click **Stop Schedule**

Scheduler Properties



The Scheduler Properties dialog box is used to configure the current schedule's operation.

Auto-Loop

This section specifies what happens when the last task in task list has finished executing. The Scheduler can be set to either end after the last task, or, restart from any task within the task list. If the Scheduler is set to re-start, you can specify the period at which the re-start occurs, and how many times the re-start should occur before stopping.

Restart Schedule from Task	Specifies the task at which the Scheduler will restart. If this property is set to *DISABLED* the Scheduler will stop after the last task has finished executing.
Immediately after last task	Specifies that the restart starts immediately after the last task has finished executing.
Every X minutes	Value specifying the number of minutes the Scheduler should wait before the restart occurs.
Daily	Value (HH:MM:SS) specifying the time at which the restart should occur.
Continuously	Specifies that the Scheduler should continue restarting until 'Stop Schedule' ■ is clicked.
Execute X times	Value specifying the number of times the Scheduler should restart.

Results Logging

This section allows you to specify where ASCII test results are written. You can choose either the default results log file as specified in Tools|Options|Results Manager|Results Log, or specify another file path.

Use my default results log file	Uses the results log file specified in Tools|Options|Results Manager|Speech Measurements.
Use this file path to log my results	Uses the file path specified in the file path text box. If this file does not exist it is created.

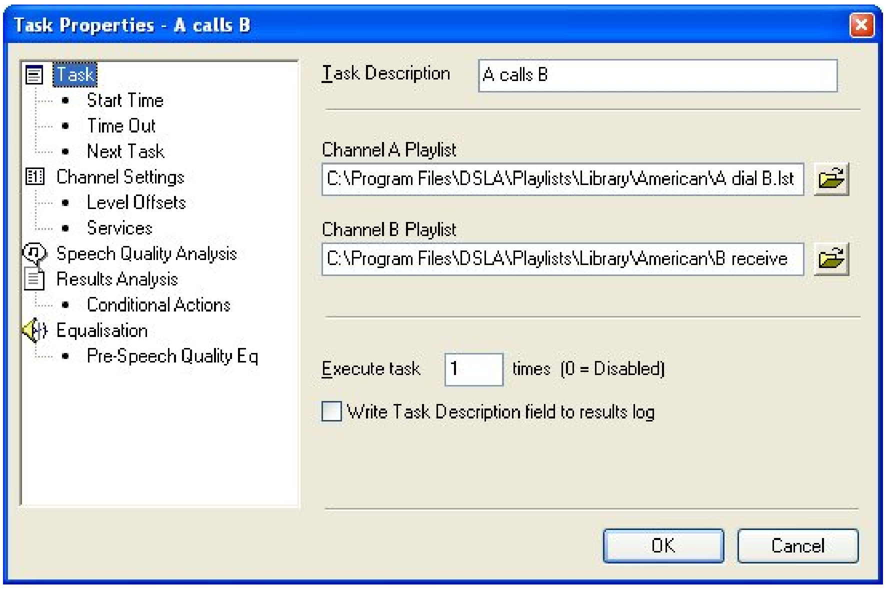
Columns

This section allows you to select which columns are displayed in the Scheduler task list.

To change the columns displayed in the Scheduler task list

1. To add a column, click the check box next to the column name.
2. To remove a column, clear the check box next to the column name.
3. To restore the default view, click 'Restore'.

Scheduler Task Properties



The Task Properties dialog box is used to configure how a task executes. It is divided into several option pages as described below:

Task options

This section is used to configure the general task settings.

Task Description	Description used to identify this task in the Scheduler task list.
Channel A/B Playlist	Specifies the Playlists to load into Channel A/B. Use the Browse buttons, or enter the full file path to an existing Playlist.
Execute	Value specifying how many times this task will execute. A setting of 0 will disable the task.
Write Task Description field to results log	Appends the Task Description field to the results log. Useful for identifying particular tasks in the results log file.

Task|Start Time

This section specifies when the task should begin executing.

Run this task immediately	Specifies that this task executes immediately after the last task has finished executing.
Every X minutes	Value specifying the number of minutes the Scheduler should wait before this task executes.
Daily	Value (HH:MM:SS) specifying the time at which this task will start.

Task|Time Out

Using the default settings, the Scheduler will wait indefinitely for each task to complete. However, some test scenarios may benefit from allowing the Scheduler to execute another task if the current task may not complete execution – i.e. when waiting for an exchange to answer a call.

Timeout	Sets the timeout period for this task.
Next Task	Specifies the task that executes after the timeout period has expired.

Task|Next Task

Using the default settings, the Scheduler will step through tasks sequentially. Some test scenarios may require more control over the task sequencing options, such as creating inter-schedule loops, or out of sequence jumps.

Next Task	Specifies the next task to execute.
Execute	Value specifying the number of times the 'Next Task' setting is executed. Once this value is exceeded, the default behaviour applies.

Channel Properties

This section configures the channel setup properties. The settings specified here override the current channel settings. The next task to be added to the Schedule will inherit the previously specified settings, so there is not normally any need to keep specifying these settings.

Input Impedance	600, High,Complex
Input Selection	Balanced, Phone Line, Handset
Filter	On, Off (200Hz-5kHz bandpass filter)
Tone Burst	Tone Burst mode
Sample Rate	Use Current Setting, 8000, 16000

Speech Quality Analysis

This section allows you to override the default speech quality metric processing options as specified in Tools|Options|Speech Quality Analysis.

Results Analysis

This section allows you process the last group of results since the last analysis was executed. You may also choose to save the analysis results to the specified ASCII formatted file.

Results Analysis|Conditional Actions

If you have chosen the Results Analysis option, you can specify a conditional expression, which when evaluated is true, will execute the following selected actions

Retry from Task	Select this action to specify which task will execute next.
Send E-Mail To	Select this action to have an alert E-mail sent to the specified address.

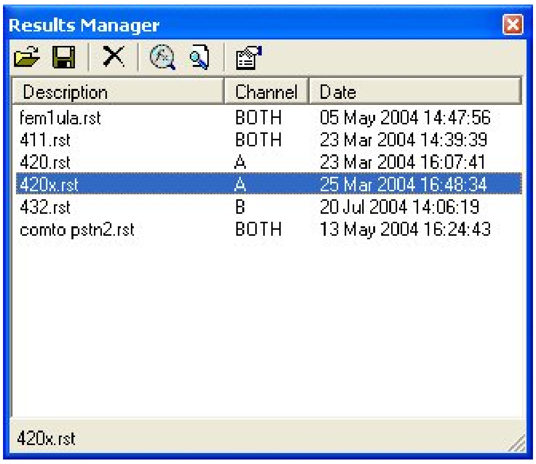
Equalisation

This section allows you to specify a filter preset that will be applied to any sound files played by DSLA. If your Playlist uses DSLA memory resident sound files, the required pre-equalized sound files must exist in DSLA memory. If your Playlist is using local files, the selected filter will be applied to the source sound files before it is streamed to DSLA.

See **DSLA Equaliser** for more information on using equalised sound files.

Results Manager

Menu: Tools|Results Manager



Using Results Manager, you can:

- Open, save, delete and view result sets.

Using Results Manager

A result set is added to the Results Manager history window after a measurement has been generated using:

- a Playlist Start Measurement event.
- the Manual Input section on the PAMS/PESQ window.
- the Speech Quality Batch Processor.

Selecting a result set displayed in the Results Manager history window will display the results data in all the relevant open display windows.

To open a results set (*.rst file)

1. Open the **Results Manager** window (Tools|Results Manager).
2. On the Results Manager toolbar, click **Open**
3. Navigate to the folder where the file is located.
4. Select the results file(s) that you want to open, and then click **Open**.

To view results set properties

1. Open the **Results Manager** window (Tools|Results Manager).
2. On the Results Manager toolbar, click **Properties**

To delete a results set

1. Open the **Results Manager** window (Tools|Results Manager).
2. Click the **results set** you wish to delete.
3. On the Results Manager toolbar, click **Delete**.

Note

- Deleting a result set, only removes the results set from the Results Manager history window, it does not delete result sets saved to disk.

To view the results log file

1. Open the **Results Manager** window (Tools|Results Manager).
2. On the Results Manager toolbar, click **View Results Log**

Saving Results

After a measurement has been generated, the results data can be saved automatically to a tab delimited ASCII format file and/or a results (*.rst) file. You are also able to save individual result sets manually, in either format, via the Results Manager window.

The measurement parameters that are written to the ASCII file are user definable, but graphical data is not saved. A results (*.rst) file contains all the numerical and graphical data and can be opened and viewed later using Results Manager or the Speech Performance Viewer available from our website.

Result limits (Tools|Options|Results Manager|Speech Measurements|Limits) can be defined for various measurement parameters so that only those results that exceed the specified limits are saved.

Note:

- All unsaved results are lost when the DSLA application is closed.

To automatically save results to an ASCII format file

1. On the Tools menu, click **Options**.
2. Select the **Results Manager|Speech Measurements** options page.
3. Check the **Write results to an ASCII log file** option box.
4. If you only want to save results that exceed your defined measurement limits, check the **Only log results that exceed limits** option box.
5. In the **File Path** textbox, type the full path to the results log file.
6. In the **Results Log Fields** list box, check the measurement parameters you want written to the log file.
7. Click **OK**.

To automatically save results to a results (*.rst) file:

1. On the Tools menu, click **Options**.
2. Select the **Results Manager** options page.
3. Check the **Auto-save result files** option box.
4. If you only want to save results that exceed your defined measurement limits, check the **Only save results that exceed limits** option box.
5. Click **OK.**

To manually save results to a log (ASCII) or results (*.rst) file

1. Open the **Results Manager** window (Tools|Results Manager).
2. Click the **results set** you wish to save.
   - To select multiple result sets at the same time, use Ctrl+Click.
3. On the Results Manager toolbar, click **Save Results**
4. Navigate to the folder where you want to save the file.
5. Select the format to save the results file as using the **File Type** drop-down list box.
6. Type a name for the result set in the **File Name** text box.
7. Click **Save**.

Specifying Result Limits

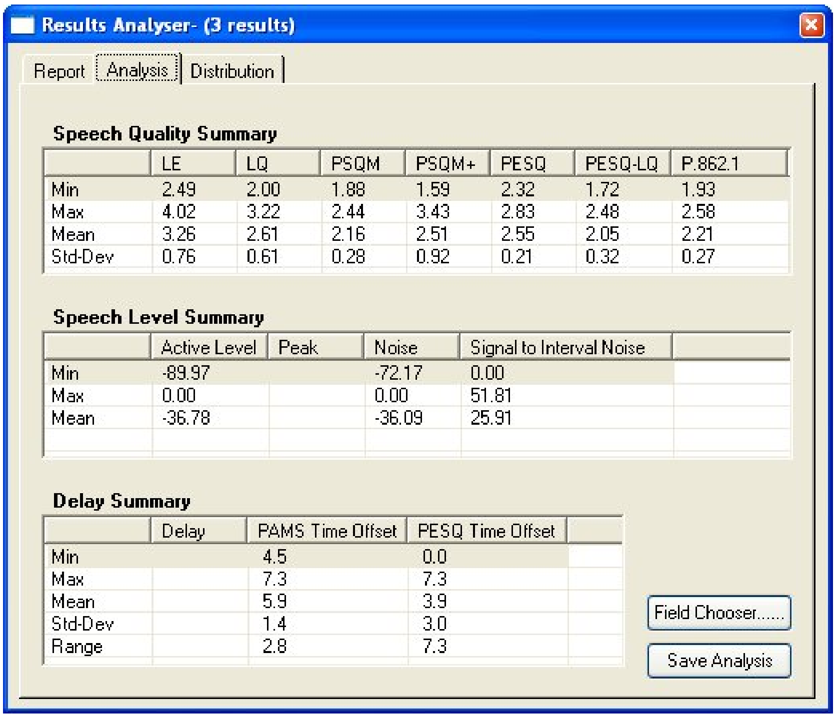
By default, Results Manager will save all results set data when configured to log or auto-save results. However, if you are only interested in results data that you consider to have failed a test condition, then this can lead to large amounts of unwanted data. By specifying result limits, Results Manager will only log or save results sets that fall outside those limits.

To specify result limits

1. On the Tools menu, click **Options**.
2. Select the **Results Manager|Speech Measurements|Limits** options page.
3. Type a value in the **Limits** section for any measurement parameter that you want to specify a limit for.
4. Click **OK**.

Results Analyser

Menu: Tools|Results Manager



Using the Results Analyser you can:

- Easily view result statistics in both numerical and graphical formats.
- Save result analysis data to an ASCII format file.

Analysing your results using Results Manager

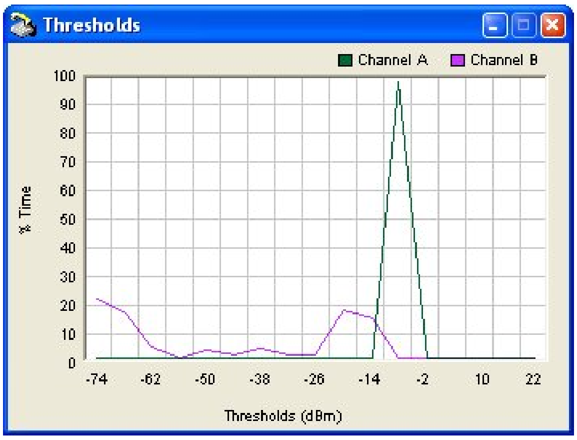
1. Open the **Results Manager** window (Tools|Results Manager).
2. Select the group of results you want analyse.
3. Click **Results Analyser**

Analysing your results using DSLA Scheduler

1. Open or create a new Schedule.
2. Select a task and click **Edit Task**
3. Select the **Results Analysis** options page.
4. Check the **Analyse Results** option box.

Thresholds

Menu: View|Thresholds



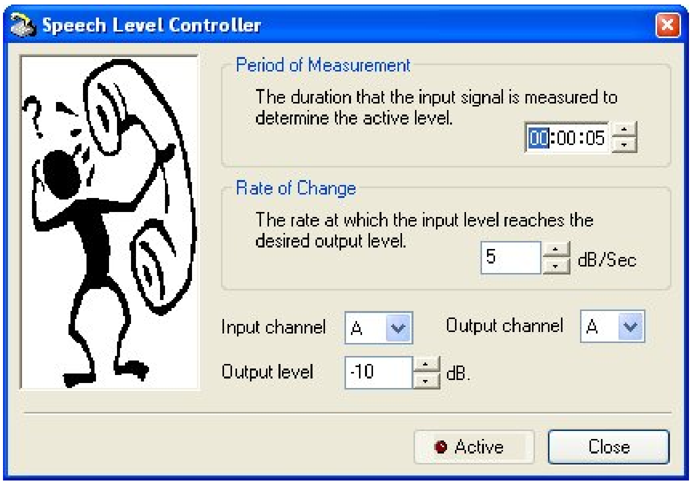
The ITU. Rec. P.56 speech measurement algorithm implemented in the DSLA involves comparing and accumulating the input signal to an array of 17 fixed thresholds. These thresholds are referred to as the Activity Count Array.

The result of these comparisons is displayed as histograms for each channel and allows you to see how the signal energy has been distributed during the measurement period.

Typically there will be two peaks in the signal distribution, one will be the residual or noise, and the other will be the speech activity.

Speech Level Controller

Menu: Tools|Speech Level Controller



Using the Speech Level Controller, you can:

- Configure the DSLA to output a constant mean active speech level for a given variable input level. The input material is output on the desired channel at the level specified.

Using the Speech Level Controller

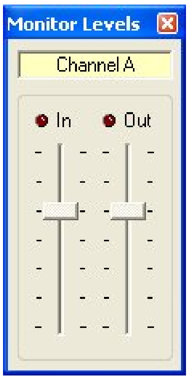
1. Connect the input signal to the desired input connection.
2. Connect the receiving equipment to the desired output connection.
3. Set the parameters on the Speech Level Controller dialog box to the required settings.
4. Click **Active**.

Note

- You cannot run a Playlist on the same channel as the 'Output Channel' selection, whilst speech level control is active.

Monitor Levels

Menu: Tools|Monitor Levels



Using the Monitor Levels window, you can:

- Set the output gain and channel source for the monitor output on the front panel of the DSLA hardware.

To change the monitor levels

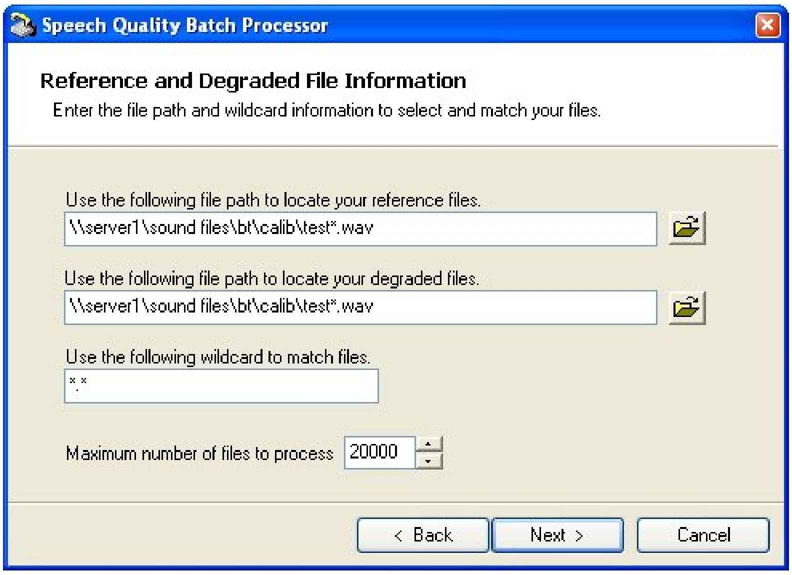
1. Click the channel indication label to toggle between Channel A and B.
2. Click the In/Out LED option button(s) to enable/disable monitoring for the In/Out section on each channel.
3. Adjust the slider(s) to set the monitor level.

Note

- It's usually best to monitor the inputs on Channel A and B only, as monitoring the outputs as well can lead to confusion.

Speech Quality Batch Processor

Menu: Tools|Speech Quality Batch Processor



Using the Speech Quality Batch Processor, you can:

- Analyse speech quality of a large number of sound files without user intervention.
- Generate an ASCII formatted file containing the scores which can then be imported into a spreadsheet as a tab-delimited text file for further analysis.

Using the Speech Quality Batch Processor

1. Open the **Speech Quality Batch Processor** window (Tools|Speech Quality Batch Processor).
2. Do one of the following:
   - To create a new profile, type a **profile name** in the Profile Name list box.
   - To use a previous profile, select a profile name from the drop down list.
3. Click **Next.**
4. Specify the **reference** and **degraded** file paths to process into the reference and degraded file path text boxes. (See: Specifying files for processing)
5. Type a wildcard string into the **Wildcard** text box to match your reference and degraded files. (See: Setting the wildcard string)
6. Click **Next.**

7. Verify the matched files displayed in the **Matched File Preview**:
   - Click Back if you need to revise your file matching specifications.
   - Use Ctrl+Click to select any files in the Matched File Preview display that you do not want to process, and then press Delete.
   - Click Next when the Matched File Preview correctly displays the files you want to process.
8. Type the file path to the **results log** file.
9. Click **Next.**
10. To view or change the options used by the Batch Processor to process your files, click **Options.**
11. Click **Next** to begin processing your files.

Specifying files for processing

The reference and degraded file paths must include either a filename, or a wildcard as part of the file path.

Examples of file paths

c:\myfolder\myfile.wav	Use the file, ‘myfile.wav’ in the folder, ‘c:\myfolder’.
c:\myfolder*	Use all the files within the folder, ‘c:\myfolder’.
c:\myfolder\???.wav	Use any file that has a three character filename followed by a .wav extension.

Setting the wildcard string

Because of the various methods used for naming files, a wildcard string is used to find a match between the reference and degraded file names. Therefore, the wildcard string used is dependent on the logic behind your file naming convention.

The general rules to follow when entering wildcard strings are as follows:

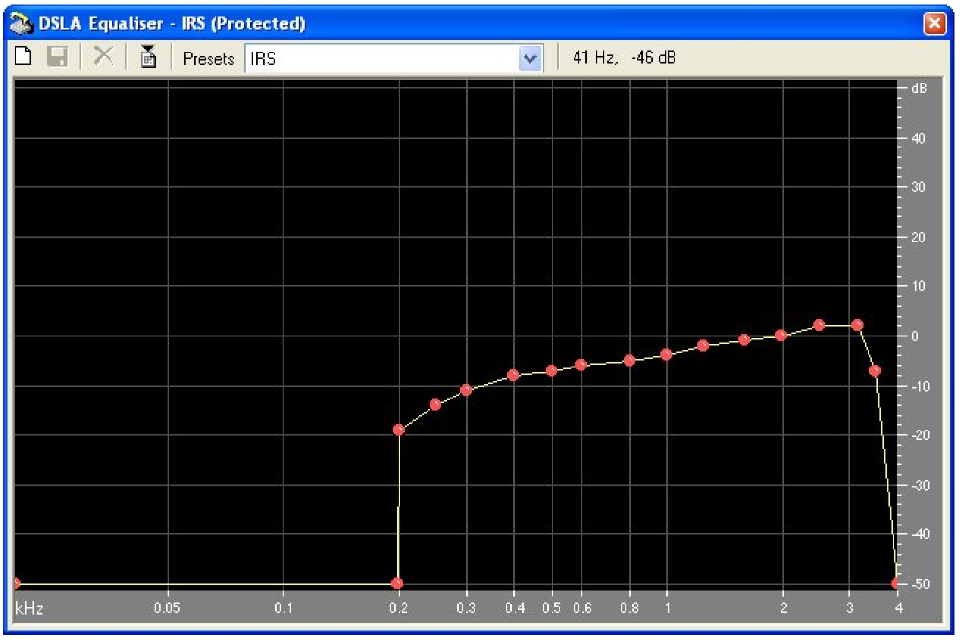
- Use the ‘?’ character where a matching character in both filenames must be present.
- Use the ‘*’ character to ignore miss-matching filename characters.
- Use [text] to search for the string text within the degraded filename. The search text is not matched against the reference file.
- Use [N] where N is a number of characters to skip within the degraded filename.

Examples of file matching wildcards

Reference	Degraded	Wildcard	Match
Cb.wav	AN0_Cb.pcm	[_]??*	True
Cb.wav	AN0_Cb.pcm	[_]??.wav	False
Cb.wav	AN0_Ft.pcm	[_]??*	False
F00p01.17r	F00p01.12d	?00p??*	True
F00p01.17r	F00p02.12d	?00p??*	False
M00p01.17r	M00p01.13d	?00p??*	True
M00p01.17r	F00p01.12d	?00p??*	False
abfile.pcm	rec_abfile_deg	[4]??*	True
abfile.pcm	rec_abfile_deg	[rec_]??*	True
abfile.pcm	rec_cdfile_deg	[rec_]??*	False

DSLA Equaliser

Menu: Tools|Equaliser



Using the DSLA Equaliser you can:

- Create filter characteristics that can then be applied to your sound files.

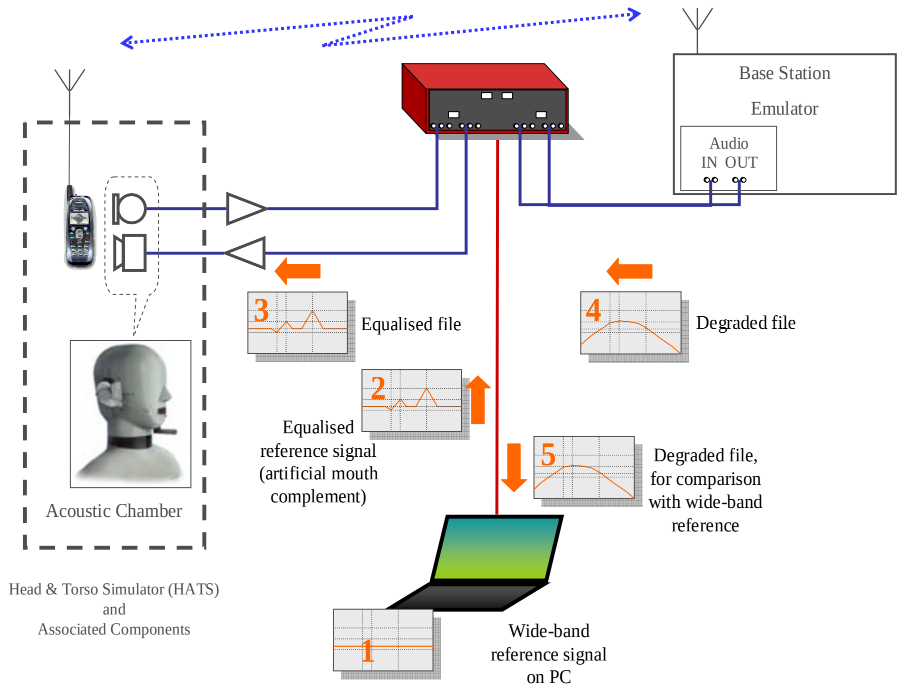
Introduction

Parts of the telephone network expect speech signals to lie within particular bandwidths or to have some spectral characteristic. For example, applying a wideband speech signal with significant energy above 4kHz directly into a network may cause impairments because the network equipment expects a signal rolled off above 3.8kHz.

Using the DSLA Equaliser, you can apply any desired filter characteristic to 8 or 16k sample rate sound files. You can create your own filter characteristics for your particular application, or use one of the following supplied filter characteristics:

- A typical Brüel & Kjær Head and Torso Simulator (HATS) artificial mouth characteristic.
- Intermediate Reference System Send (IRS).
- Modified Intermediate Reference System Send (mIRS)(ITU-T Rec. P.830).
- Low and High pass filters.

Filter presets created with the DSLA Equaliser can be used in conjunction with the Scheduler to enable reference speech to be filtered for playback via DSLA, but the original reference speech is still used for Speech Quality Analysis. This feature is useful for the HATS application where wideband speech is equalised for the artificial mouth characteristic but you need to compare the degraded speech with the original wideband reference material.



Using the DSLA Equaliser

To create or modify a filter preset

1. Open the **DSLA Equaliser** window (Tools|Equaliser).
2. Do one of the following:
    - To create a new filter preset, click **New Filter**
    - To modify an existing filter preset, select the desired filter from the **Presets** drop-down listbox.
3. Double-click to insert a node at the current frequency and gain point.
4. Right-click an existing node to specify the exact frequency and gain, or to remove the node.
5. Click and drag an existing node to move the node to a new frequency/gain point.

Note

- To modify a protected filter preset, you must first right-click within the drawing area and de-select the **Protected** menu option.

To save a filter preset

1. Click Save
2. Type a name for your profile in the **Save Filter Profile** dialog box.
3. Click the **Protected** check box if you want to protect your profile from begin accidentally changed in the editor.
4. Click **Save.**

To delete a filter preset

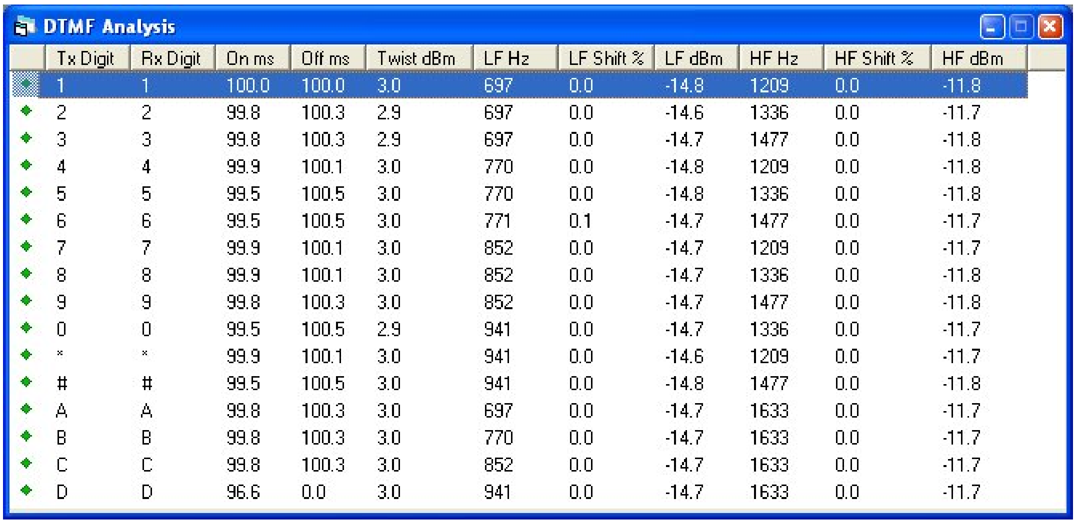
1. Select the **filter preset** you want to delete from the drop-down listbox.
2. If the preset has been protected, right-click within the drawing area and de-select the **Protected** menu option.
3. Click **Delete Filter**.

Applying a filter preset to your sound files

1. Open the **DSLA Equaliser** window (Tools|Equaliser).
2. Click **Apply Filter**
3. Follow the steps in the **Equalisation Wizard**.

DTMF Analysis

Menu: View|DTMF Analysis



DTMF Analysis

Tx Digit	Rx Digit	On ms	Off ms	Twist dBm	LF Hz	LF Shift %	LF dBm	HF Hz	HF Shift %	HF dBm
1	1	100.0	100.0	3.0	697	0.0	-14.8	1209	0.0	-11.8
2	2	99.8	100.3	2.9	697	0.0	-14.6	1336	0.0	-11.7
3	3	99.8	100.3	2.9	697	0.0	-14.7	1477	0.0	-11.8
4	4	99.9	100.1	3.0	770	0.0	-14.8	1209	0.0	-11.8
5	5	99.5	100.5	3.0	770	0.0	-14.8	1336	0.0	-11.8
6	6	99.5	100.5	3.0	771	0.1	-14.7	1477	0.0	-11.7
7	7	99.9	100.1	3.0	852	0.0	-14.7	1209	0.0	-11.7
8	8	99.9	100.1	3.0	852	0.0	-14.7	1336	0.0	-11.8
9	9	99.8	100.3	3.0	852	0.0	-14.7	1477	0.0	-11.8
0	0	99.5	100.5	2.9	941	0.0	-14.7	1336	0.0	-11.7
*	*	99.9	100.1	3.0	941	0.0	-14.6	1209	0.0	-11.7
#	#	99.5	100.5	3.0	941	0.0	-14.8	1477	0.0	-11.8
A	A	99.8	100.3	3.0	697	0.0	-14.7	1633	0.0	-11.7
B	B	99.8	100.3	3.0	770	0.0	-14.7	1633	0.0	-11.7
C	C	99.8	100.3	3.0	852	0.0	-14.7	1633	0.0	-11.7
D	D	96.6	0.0	3.0	941	0.0	-14.7	1633	0.0	-11.7

The DTMF Analysis window reports:

- Received digit status:
  Green (OK), Blue (OK, but on specification limits), Red (Failed)
- Decoded digits
- On/Off durations
- Twist
- Received high and low frequencies
- Received levels.

Analysing DTMF Signals

Measurements can be made on DTMF signals whether generated by DSLA or received from the network. The DSLA can generate any of the 16 DTMF signals with any duration, inter-digit silence, twist or frequency shift.

An analysis is perform on the DTMF digits received by DSLA by including the DTMF Analysis control event in your Playlist. The following Playlist example will output all the available DTMF digits and then analyse the received digits on Channel B.

Step	Type	Description
1		Start Measurement [B] – DTMF Analysis Example
2		DTMF “1234567890#*ABCD” @ -10 dBm
3		DTMF Analysis [REC=B]

Writing DTMF Analysis results to an ASCII log file

1. On the **Tools** menu, click **Options**.
2. Select the **Results Manager|DTMF Measurements** options page.
3. Check the **Write Results to an ASCII log file** option box.
4. Click OK.

Using DTMF result limits

1. On the **Tools** menu, click **Options**.
2. Select the **Results Manager|DTMF Measurements** options page.
3. Type the limits you would like applied into the **Limits** text boxes.
4. Click OK.

Operational departures from standard (Bellcore TR-TSY-000181, EIA/TIA-464A, ITU-T Q.24) values:-

Frequency Deviation - All digits detected at ±1.5%, all digits rejected at ±3.5%

Minimum Tone Duration - All digits detected with 40ms pulse width, all digits rejected with 23ms pulse width

Minimum Interdigit Interval - 40ms gap causes two digits to be detected for all digits. 10ms gap is bridged and a single digit is detected for all digits.

Minimum Cycle Time - 93ms

Accept Levels - Detects all digits in range of +1 to -37dBm. Rejects all digits ≤ -55 dBm.

Twist (ratio of high group power to low) - Detects all digits for twist of +4 to -8dB

Noise - All digits detected for noise up to 15dB below tone level

Echo - Detects all digits with echo more than 10dB below tone level with echo up to 20ms delay

Miscellaneous

Upgrading the DSLA's firmware

The DSLA operating system can be upgraded by sending an .EPR file to the DSLA via either a serial connection or an Ethernet connection (firmware version 4.31 or better). The original firmware revision .EPR file that shipped with your DSLA can be found in the DSLA installation folder.

1. Establish a connection to the DSLA.
2. On the **Tools** menu, click **Options**
3. Select the **DSLA|Firmware Update** options page.
4. Click **Install**.
5. Using the Open dialog box, navigate to the folder where the .EPR file is located.
6. Select the .EPR file that you want to use and then click **Open**.

Specifying the default user interface language

By default, the DSLA user interface is displayed using the current operating system regional settings. However, you can override this setting by specifying one of the following supported languages:

- Chinese (Simplified)
- Chinese (Traditional)
- English
- French

1. On the **Tools** menu, click **Options**.
2. Select the **General|International Settings** options page.
3. Select the required user interface language from the drop-down listbox.
4. Click **OK**.
5. Restart the DSLA application.

Logging and suppressing error messages

Operational errors generated whilst using DSLA can be suppressed and logged to the file <InstallDir>\DSLA\Results\Error.log. This option is particularly useful when using the Scheduler and unattended operation is required.

1. On the **Tools** menu, click **Options**.
2. Select the **General** options page.
3. Check the Log and suppress errors checkbox.
4. Click **OK**.

Using headerless (raw) sound files

Headerless (RAW) sound files, can only be used as a source for speech quality analysis - either via the 'Manual Input' section of the for Speech Quality Analysis display windows, or when using the Speech Quality Batch Processor.

Before using headerless files, you must make sure that you have correctly specified the file format that you are using. Not doing so may give incorrect result measurements. Currently supported sound file formats are: 8/16k sample rates, mono, 16-bit, Intel / Motorola format.

1. On the **Tools** menu, click **Options**.
2. Select the **Speech Quality Metrics|Sound File Data Format** options page.
3. Set the format parameters to match your file type.
4. Click **OK**.

Specifying default folder paths

You can specify the default folder paths used for speech sets, Playlist scripts, result sets and downloaded files.

1. On the **Tools** menu, click **Options.**
2. Select the **General** options page.
3. In the **Folder Location** box, click the folder you wish to modify.
4. Click **Modify**.
5. In the 'Browse for Folder' dialog box, select the folder you want to use as the default for the folder location you selected.
6. Click **OK**.

Recording

Recorded files are viewed using the DSLA Explorer. They appear in the explorer at the DSLA root level. Playback and manipulation of recorded files is performed in the same manner as speech files.

To start recording

1. Open the **Channel window**(s) you wish to use.
2. On the channel status bar, click **REC** to enable the record button on the toolbar.
3. Click the record button on the channel toolbar.

To stop recording

1. On the Channel window toolbar, click **Stop** ■

Items that appear in the status bar

The status bar, which is the horizontal area at the bottom of the DSLA main window, provides information about the current state of the DSLA connection and any other contextual information.

Item	Description
Connection Status	Displays the connection status to the DSLA.
Dual Activity	Displays the Dual Activity value.
Auto On-Hook	Normally, whenever a channel stops – either by clicking **Stop** ■ or when Playlist execution ends, the line sockets on the DSLA hardware are set on-hook. However, you can disable this function should this not be desirable. When 'AUTO ON-HOOK' is disabled, this option appears dimmed. You can also force the line status to on-hook by right-clicking 'AUTO ON-HOOK' and selecting 'Go On-Hook' from the pop-up menu.
Sample Rate	Sets the DSLA sample rate – either 8k or 16k. When 16k is selected, the Channel A/B bandpass filter is forced to the off state.
AB	You can set both channels to start and stop together by enabling the 'AB' option. When 'AB' is disabled, this option appears dimmed.

Command line options

The following command line options can be specified to control the DSLA application upon startup:

Usage: medsla[ip].exe [/p refpath,degpath] [/s filepath] [s:/run filepath]

Parameter	Description
/p refpath,degpath	Use the /p option to execute the default speech quality assessment options on the specified reference and degraded files.
/s[:run][:xxxx] filepath	Use the **/s** option to load the specified schedule file. The following flags can used in conjunction with the /s option: **:run** Specify this flag to automatically start the Scheduler. When the Scheduler has finished executing, the DSLA application will close. **:xxxx** Where **xxxx** is a dial string. If specified, the first occurrence of a DTMF event in a Playlist will be substituted with this string.
/fmt	Use the /fmt option to erase all files and folders from DSLA memory.

Result log field definitions

Heading	Meaning	Description
DUR(s)	Duration	Test duration shown in seconds.
LNGTRM	Long Term	Average level of the input signal. This is a true RMS reading. Expressed in the current units.
ACTLVL	Active Level	Mean active speech level. Expressed in the current units.
PEAK	Peak	Highest instantaneous level. Expressed in the current units.
NSE	Noise	Noise level. Expressed in the current units.
ACT%	Activity Factor	Percentage of the total time where the mean active speech level is exceeded, i.e. speech is present.
DLY	Delay	Delay in milliseconds of degraded signal with respect to the reference signal.
DAF	Dual Activity Factor	Dual Activity Factor score.
LE	Listening Effort	Listening effort score.
LQ	Listening Quality	Listening quality score.
OFFSET	Time Offset	Average of the utterance time offsets of the degraded signal with respect to the reference signal in milliseconds.
CONF%	Confidence	Confidence in measurement of time offset. Expressed as a percentage.
MIN	Minimum	Minimum utterance time offset in milliseconds.
MAX	Maximum	Maximum utterance time offset in milliseconds.
STDDEV	Standard Deviation	The Standard Deviation of the utterance time offsets.
UTTCNT	Utterance Count	Utterance count of the degraded signal – maybe more than the reference signal due to silence intervals in speech or time shifts in active speech.
QLE	Listening Effort Q	
QLQ	Listening Quality Q	
MUTED%	Proportion of Speech Muted	

LNGMUTE	Longest Muted Section	
PESQ	PESQ Score	PESQ MOS prediction
PESQLQ	PESQ Listening Quality Score	PESQ MOS prediction transformed to LQ scale
P.862.1 Score	P.8621.Listening Quality Score	PESQ MOS prediction mapped to LQO scale
P.862.2 Score	P.862.2 Listening Quality Score	PESQ MOS prediction mapped to PESQ-WB scale
FILEID	Metric history file path	The file path of the reference and degraded files.
DATE/TIME	Time when test completed	

Notes

- All time values are expressed in milliseconds unless otherwise stated.
- Some field values may be expressed as N/A. This means that this field was not applicable for that particular result.

Speech Quality Analysis

Modern communications networks include elements, for example, lossy coding, error-prone channels and voice activity detection that cannot reliably be assessed by such conventional engineering metrics as signal-to-noise ratio.

One way to measure customers' perception of the quality of these systems is to conduct a subjective test involving panels of human subjects. However, these tests are expensive and unsuitable for such applications as real-time monitoring. Objective speech quality metrics are designed to predict the scores that would be obtained from subjective tests.

The DSLA supports three speech quality metrics:

- Perceptual Analysis Measurement System (PAMS)
- Perceptual Evaluation of Speech Quality (PESQ ITU-T Rec. P.862)
- Perceptual Speech Quality Measure (PSQM was ITU-T Rec. P861)

PAMS Overview

To measure speech quality, PAMS uses a sensory model to compare the original, unprocessed signal with the degraded version at the output of the communications system. This process is shown in Figure 1 and determines Mean Opinion Score predictions, on a scale of 1-5, for the listening quality and listening effort. The score calculated by PAMS will be typically within one half a MOS of that determined by a well controlled subjective test in a laboratory.

Reference should be made to ITU-T Recommendation P.800 Methods for subjective determination of transmission quality for further information regarding the methods and procedures for conducting subjective tests.

Listening tests are often conducted through telephone handsets and the bandwidth of the test material is adjusted appropriately. It may be necessary to filter the reference speech material to perhaps the Intermediate Reference System send characteristic (ITU-T Recommendation P.48) or some other characteristic to ensure that the reference and degraded signals are to be correctly compared. The absence of such filtering may produce a significantly lower score than might be expected.

The PAMS algorithm processes the reference and degraded speech to produce an auditory sensation surface for each. These surfaces can be compared in a perceptually relevant manner to derive a number of error parameters. Applying appropriate weighting to the error parameters and combining the results enables a MOS prediction to be made.

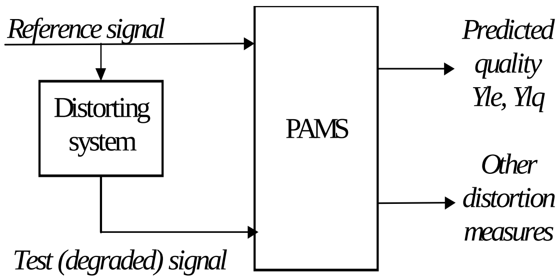
PAMS incorporates many new developments that distinguish it from earlier codec assessment models such as those given in ITU-T P.861. These innovations allow PAMS to be used with confidence to assess end-to-end speech quality as well as the effect of individual elements such as codecs.

Release 3.0 of PAMS further extends the ability of the model to identify variable delay – characteristic of packet-based transmission systems – and introduces an equalisation process designed to enhance the robustness of PAMS with measurements made over connections including analogue filtering. This release also includes re-calibrated listening quality and listening effort mappings, as the subjective test database has been extended to include a wider range of distortion types.

These developments provide PAMS release 3.0 with unparalleled accuracy at predicting quality across many different voice transmission technologies. PAMS has been used successfully to assess conventional fixed and mobile networks as well as numerous packet-based transmission systems.

The DSLA and user interface have been designed to provide a simple access to this powerful algorithm, either directly from the analogue connection or from speech files recorded elsewhere.

Figure 1: Usage of PAMS



Operations Performed by PAMS

The processing carried out by PAMS is illustrated in Figure 2. The model includes the following stages.

Time Alignment

PAMS is a listening model and has no knowledge of the delay of the system. In order to compare the reference and degraded signals, however, they need to be lined up with each other. Time alignment does this by dividing the signal into sections known as utterances, and finding the delay (time offset) of each section. This enables it to cancel any bulk delay and most delay changes that might be caused by, for example, packet-based transmission. The DSLA has a delay of 4msec. This is accounted for in the delay measurements reported by the DSLA. The PAMS time offset measurements do not take account of the DSLA delay. This means that a time offset reported by PAMS on a file collected via the DSLA will be 4msec greater than that reported by the DSLA.

Level Alignment

Likewise, PAMS has no knowledge of level changes in the system. It therefore assumes a constant listening level of approximately 79dB SPL [P.830], applying a gain to both the reference and degraded signals to bring them to this level.

Equalisation

Analogue connections often introduce some degree of filtering. For example, the send path of a telephone handset usually filters speech with a characteristic similar to the standard 'modified IRS send' characteristic [P.830]. It is generally accepted that this has less effect on quality than coding distortions. PAMS identifies any filtering that has taken place in the network and cancels its effect.

Auditory Transform

In order to compare the reference and degraded signals in a meaningful way – taking account of how a listener would have heard them – they are passed through an auditory transform that mimics certain key properties of human hearing. This gives a representation in time and frequency of the perceived loudness of the signal, known as the sensation surface. The difference between the sensation surfaces for the reference and degraded files is known as the error surface; this shows any audible differences introduced by the system under test.

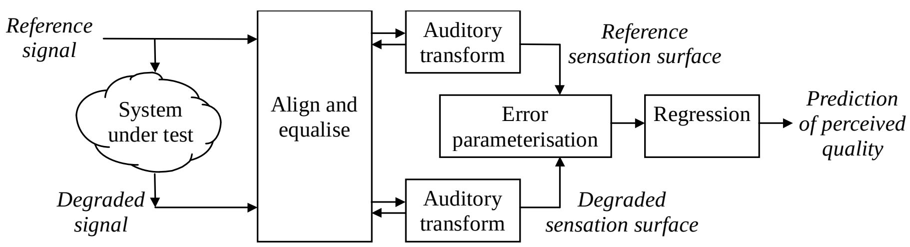
Error parameterisation

The error surface is analysed in several different ways, for example calculating separately the average positive distortion (sounds added to the degraded signal) and the average negative distortion (e.g. parts of speech that have been attenuated). This analysis gives a number of error parameters that summarise the amount of each type of audible error.

Regression

Finally, the error parameters are mapped onto predictions of perceived listening quality and listening effort. These mappings are calculated and verified using a very large database of subjective tests to ensure that PAMS is able to predict quality for a wide range of distortion types.

Figure 2: Processing performed in PAMS



Listening Effort and Listening Quality

PAMS returns quality scores on two different opinion scales, Ylq (listening quality) and Yle (listening effort). These scales are standard in the industry and are defined in [P.800]. They are reproduced in the table below, along with the prompt that is given to subjects. Both Ylq and Yle scores lie between 1 and 5 and are usually quoted to two decimal places.

The scores give a measure of customers' perception of quality. A PAMS score of 5 means that no distortion is measured. As the amount of distortion increases the quality falls. Because they relate to different aspects of subjectivity, Yle and Ylq scores are normally different if there is perceived distortion; Yle is usually higher than Ylq.

	Listening Quality	Listening Effort
5	Excellent	Complete relaxation possible; no effort required
4	Good	Attention necessary; no appreciable effort required
3	Fair	Moderate effort required
2	Poor	Considerable effort required
1	Bad	No meaning understood with any feasible effort

PAMS Metric Controls

Modified IRS send filter

From release 3.0, PAMS includes the ability to autodetect whether the distorting system includes an MIRS send filter. This should be the default behaviour and it is recommended that users do not change this.

The MIRS send filter represents the transfer function from mouth to junction in a fixed network telephone handset [P.830]. This provides a boost of about 6dB to high frequencies and strongly attenuates below 300Hz. Applying this characteristic before speech enters the network controls the dynamic range and can improve intelligibility. However, current mobile telephone handsets do not tend to follow the MIRS response and so this filter may be inappropriate in measured or simulated mobile conditions.

The MIRS send filter on its own tends to have little subjective impact, so it is important that its effect is eliminated before the PAMS comparison. To accomplish this, the filter is autodetected and applied if necessary, to the reference signal only, to equalise it to the degraded signal. If the filter is detected incorrectly, for example due to noise in the degraded file, this behaviour may be overridden by specifying that the filter should, or should not, be included.

Transfer function estimation and equalisation

The IRS send response is one example of a filter that may be encountered in a network. Filtering may occur at any analogue interface, and generally has limited subjective effect. However, PAMS tends to over-estimate the effect of filtering unless the reference signal is equalised to the degraded signal. To address this, release 3.0 of PAMS includes a default option to perform identification of the filtering in the network and equalise the reference as required. The method used is robust even in the presence of high levels of non-linear distortion such as channel errors.

Receive acoustic filters

One component of the auditory transformation used in PAMS is a filter which models the acoustic path from the junction, through a telephone receiver, and into the inner ear. By default, this filter is applied to both the reference and degraded signals. It is appropriate to omit this filter in two applications.

Head and torso simulator (HATS) ear measurements will already have applied a similar characteristic as part of the acoustic path. In this case, the receive acoustic filters should be included for the reference but omitted for the degraded file.

Wideband telephony, operating at 16k sample rate, normally uses listening equipment that filters the signals much less strongly than a conventional narrowband receiver. To model this, the receive acoustic filters should be omitted for both the reference and the degraded file.

Transfer function estimation/equalisation should normally be left on in both of these applications.

PAMS Results Returned

PAMS calculates many parameters describing differences between the reference and degraded files. These parameters are described in this section.

Listening Quality Ylq

PAMS returns a Ylq score designed to relate to the listening quality subjective opinion scale defined in P.800. The PAMS score is continuous between 1 and 5. A new mapping was introduced in release 3.0 and it is recommended that this is used for most measurement applications.

The result of the old release 2.0 Ylq mapping is also calculated and may be available.

In addition, two further quantities are derived, converting the release 3.0 Ylq scores to the equivalent Q domain on a dB scale. The equivalent Q values may be used with Q domain models such as the E-model [G.107].

Note

- The release 2.0 Ylq mapping will only be identical to those of PAMS release 2.0 if transfer function estimation/equalisation is disabled, which is not recommended.

Listening Effort Yle

PAMS also provides a Yle score on the less-used listening effort scale. This is also a new mapping in release 3.0. Additionally, the release 2.0 Yle is calculated if transfer function estimation/equalisation is disabled, which is not recommended. Listening quality is the dominant opinion scale in subjective testing; PAMS provides listening effort scores as this may be of value in testing systems for which intelligibility is more of a concern or where overall quality is generally low.

Time Offset

In order to deal with variable time offset, PAMS sub-divides the signal into a number of utterances. Each utterance is time aligned separately. The average of the time offsets is displayed along with the range and standard deviation. A graphical display of the utterance by utterance variation of time offset is shown.

Utterance Count

The number of utterances and segments that PAMS has determined in the degraded file. This may be more or less than expected if there has been some temporal clipping, delay changes or extensive silent intervals.

Muting

To assist with diagnosing systems in which muting (temporal clipping) is a distortion, two measures of muting – taking account of perceptual processes such as masking – are returned. Muting is the replacement of speech with silence or comfort noise.

- Proportion of the speech that has been muted
- Length in seconds of the longest muted part of speech

Reference MIRS Send Filter Auto-Detection

Applied if the bandwidth of the degraded signal is significantly less than that of the reference signal.

PESQ Overview

PESQ provides an objective measure that predicts the results of subjective listening tests on telephony systems. To measure speech quality, PESQ uses a sensory model to compare the original, unprocessed signal with the degraded version at the output of the communications system. This process is shown in Figure 1 and is explained in more detail in the next section.

The result of comparing the reference and degraded signals is a quality score. This score is analogous to the subjective “Mean Opinion Score” (MOS) measured using panel tests according to ITU-T P.800. The PESQ scores are calibrated using a large database of subjective tests.

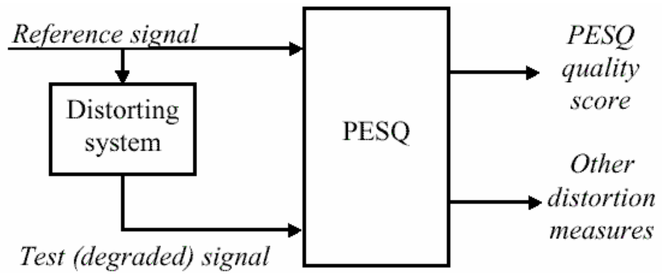
PESQ incorporates many new developments that distinguish it from earlier models for assessing codecs, for example, PSQM and MNB (ITU-T P.861). These innovations allow PESQ to be used with confidence to assess end-to-end speech quality as well as the effect of such individual elements as codecs.

This is a fully conformant implementation of PESQ as defined in ITU-T P.862 with the mappings to P.862.1 LQO, P.862.2 PESQ-WB and Ie. Additionally, it provides extensions to allow PESQ to be used with wideband telephony or head and torso simulator (HATS) measurements.

The ITU-T selection process that resulted in the standardisation of PESQ involved a wide range of conditions, with demanding correlation requirements set to ensure that it has good performance in assessing conventional fixed and mobile networks and packet-based transmission systems.

The DSLA and user interface have been designed to provide a simple access to this powerful algorithm, either directly from the analogue connection or from speech files recorded elsewhere.

Figure 1: Usage of PESQ

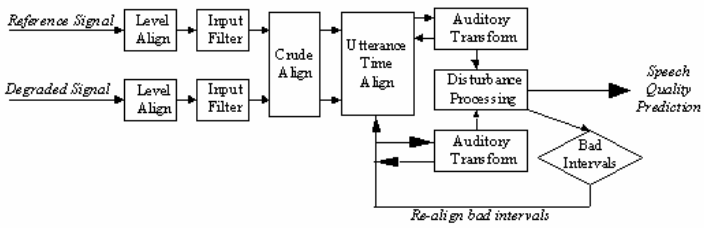


PESQ takes into account the following sources of signal degradation: coding distortions, errors, packet loss, delay and variable delay, and filtering in analogue network components.

PESQ does not take into account the subjective effect of level changes in the network, echo, and the effect of round-trip delay on conversation.

Operations Performed by PESQ

The processing carried out by PESQ is illustrated below.



The model includes the following stages:

Level alignment

In order to compare the signals, the reference speech signal and the degraded signal should be at the same, constant power level. This is necessary because the reference signal does not have to be to be at a defined level and because the gain of the system under test is unknown before testing.

PESQ assumes that the subjective listening level is a constant 79dB SPL at the ear reference point [ITU-T P.830, section 8.1.2]. A gain is applied to both the reference and degraded signals to bring them to this level.

Input filtering

Analogue connections often introduce some degree of filtering. For example, the send path of a telephone handset usually filters speech with a characteristic similar to the standard ‘modified IRS send’ characteristic [ITU-T P.830]. It is generally accepted that this has less effect on quality than coding distortions do. PESQ compensates for any filtering that has taken place in the network.

Time alignment

The system may include a delay, which may be variable. In order to compare the reference and degraded signals, they need to be lined up with each other. PESQ is a listening model and has no knowledge of the delay of the system. PESQ applies voice activity detection to the signals to identify those parts of the signal that are speech, ignoring noise.

Time alignment is then done in three stages:

- First, PESQ aligns the overall speech signals (utterances). An utterance is a continuous speech burst identified by the voice activity detector. It may contain pauses that are no longer than a pre-determined threshold (200ms). This process detects delay over major sections of the degraded signal compared to the reference signal.
- Second, PESQ aligns overlapping sections of the speech (frames). This process detects delay that is variable over the length of an utterance, as this can be significant in packet-based networks.

- There is also a third stage after the next operation (the auditory transform), in which “bad intervals” (sections of the speech with very large disturbance) are realigned. This step improves the model’s accuracy with a small number of files where delay changes are not correctly identified by the initial time alignment process.

The DSLA has a delay of 4msec. This is accounted for in the delay measurements reported by the DSLA. The PESQ time offset measurements do not take account of the DSLA delay. This means that a time offset reported by PESQ on a file collected via the DSLA will be 4msec greater than that reported by the DSLA.

Auditory transform

In order to compare the reference and degraded signals, taking account of how a listener would have heard them, each is passed through an auditory transform that mimics certain key properties of human hearing. This gives a representation in time and frequency of the perceived loudness of the signal, known as the sensation surface.

Disturbance processing

The difference between the sensation surfaces for the reference and degraded files is known as the error surface; this shows any audible differences introduced by the system under test. The error surface is analysed by a process that takes account of the effect that small distortions in a signal are inaudible in the presence of loud signals (masking).

From the positive and negative errors, two disturbance parameters are calculated. They are calculated as non-linear averages over specific areas of the error surface. These disturbance parameters are:

- the absolute (symmetric) disturbance – a measure of absolute audible error
- the additive (asymmetric) disturbance – a measure of audible errors that are significantly louder than the reference

This analysis gives two error parameters that summarise the amount of each type of audible error. Finally, the error parameters are converted to a quality score, which is a linear combination of the average symmetric disturbance value and the average asymmetric disturbance value.

PESQ Quality Scores

PESQ returns a quality score on the listening quality scale, which is standard in the industry and is defined in [ITU-T P.800]. This is reproduced in the table below, along with the prompt that is given to subjects. Listening quality scores lie between 1 and 5, but the PESQ score does not exceed 4.5, as this is usually the maximum obtained in a subjective test.

Score	Quality of the Speech
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

The score gives a measure of customers' perception of quality. The highest PESQ score, 4.5, means that no distortion is measured. As the amount of distortion increases the quality falls.

Several alternative mappings of the PESQ raw score have been developed to attain a better correlation with subjective test.

PESQ-LQ

PESQ-LQ has many advantages over the standard PESQ score provided by P.862. PESQ-LQ is closer to subjective MOS. PESQ-LQ is on a 1–5 scale, the same range as MOS. (Note that PESQ-LQ is limited to a maximum of 4.5 as MOS is seldom greater than this.)

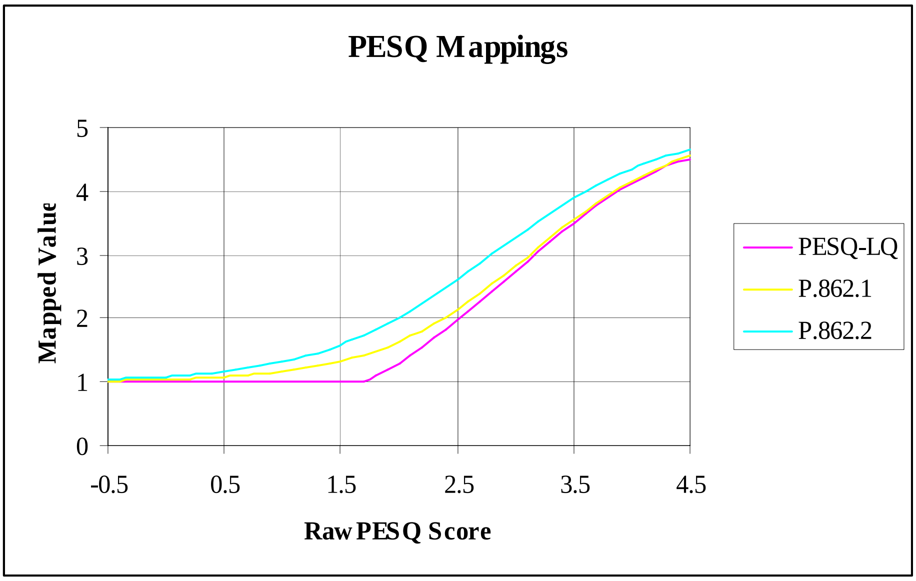
On average, over a large number of subjective tests, PESQ-LQ relates closely to subjective MOS. This is shown by **Error! Reference source not found.**, which plots subjective MOS compared with PESQ-LQ for 11 subjective tests. The solid line in this graph shows the ideal one-to-one relationship.

P.862.1 Score

The aim of separate recommendation ITU-T P.862.1 is to provide a single mapping from raw P.862 score to the Listening Quality Objective Mean Opinion Score (MOS-LQO).

P.862.2 Score

The aim of separate recommendation ITU-T P.862.2 is to provide a single mapping from raw P.862 score to the Listening Quality Objective Mean Opinion Score (PESQ-WB).



Summary

- The P.862.1 mapped score is a function that maps more closely to subjective MOS according to the ACR LQ method defined in ITU-T P.800/P.830.
- Subjective MOS does vary from test to test, but PESQ-LQ is a good predictor of MOS taking account of a large number of subjective tests.
- Results obtained with PESQ will vary from PAMS, PSQM, etc. PESQ is the most accurate.

PESQ Metric Controls

The models that PESQ can implement are:

- Standard P.862 (narrowband handset on reference and degraded signals) raw score
- HATS ear recording on degraded signal, unprocessed (wideband) reference signal
- Wideband model (headphone listening) and P.862.2 mapped score
- PESQ LQ mapped score
- P.862.1 mapped score

The default for narrowband measurements is the P.862 raw score and P.862.1 mapped score.

PESQ Input Filters

Depending on the choice of model, PESQ determines internally which input filter to apply.

In the standard narrowband PESQ measurements, a frequency-domain input filter is applied to both the reference and degraded files before time alignment and psychoacoustic processing. The filter used, which is similar to the modified IRS receive filter specified in P.830, is shown in Figure 2. This approximates the filter characteristic of a standard handset.

For wideband measurements, a more appropriate filter is used with a flat response above 100Hz and a gentle roll-off below this point, modelling the attenuation of the headphones and ear at low frequencies. The filter uses a simple IIR implementation. The response of the 16kHz implementation is shown in Figure 3. The 8kHz implementation has the same gain (within 0.1 dB) in the 1Hz–4kHz range.

For HATS measurements using a telephone handset, the standard narrowband input filter is applied to the reference (to model a standard handset), but the wideband filter is applied to the degraded file as the HATS recording will automatically include the handset path.

Figure 2: PESQ narrowband input filter characteristic

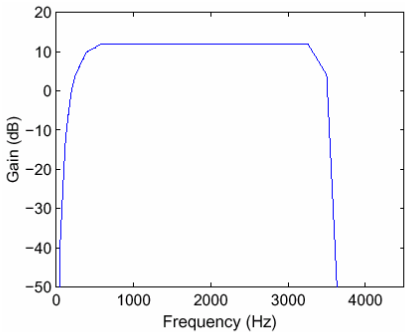


Figure 3: PESQ wideband input filter characteristic

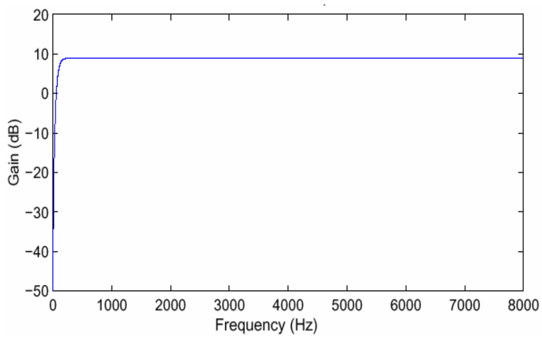


Table of PESQ Filter Options

Option	Default	Model
Reference Acoustic Filter	On	Standard P.862 Narrowband model
Degraded Acoustic Filter	On	
Reference Acoustic Filter	On	Wideband reference signal HATS ear recording
Degraded Acoustic Filter	Off	
Reference Acoustic Filter	Off	Wideband model for reference and degraded signals
Degraded Acoustic Filter	Off	

PESQ Results Returned

PESQ Score

PESQ returns a quality score on the listening quality scale, which is standard in the industry and is defined in [ITU-T P.800]. The range of the score is from 4.5 to 1.

PESQ-LQ Score

PESQ-LQ Score is a function that maps more closely to subjective MOS

P.862.1 Score

P.862.1 score is a function that maps more closely to subjective MOS in the narrowband condition.

P.862.2 Score

P.862.2 Score is a function that maps more closely to subjective MOS when the wideband model is used.

Time Offset

In order to deal with variable delay, PESQ sub-divides the signal into a number of utterances. Each utterance is time-aligned separately. The average of the time offsets is displayed along with the range and standard deviation. A graphical display of the utterance by utterance variation of time offset is shown.

Frame-by-frame delay

Frame-by-frame delay is the delay measure used in calculating the PESQ quality score. Utterances are broken up into frames of 32 ms duration. Frames use a window function that gives greater weight to the central 16ms of each frame, and there is an overlap between successive frames of 50%. Effectively, therefore, each frame is 16ms long; this can be thought of as “sampling” the values every 16ms. PESQ calculates the delay in each frame, based on the nearest utterance.

The frame-by-frame delay is the best way of tracking how delay varies during the signal. This processing ensures that insertions into the degraded signal are ignored, and deletions from the signal are partially ignored.

Utterance Count

The number of utterances that PAMS has determined in the degraded file. This may be more or less than expected if there has been some temporal clipping or extensive silent intervals.

Perceptual Speech Quality Measurement

The implementation of PSQM is based upon ITU-T Rec. P.861. The algorithm and functionality are described in P.861 and not repeated here.

The silent interval weighting factor (Wsil) can be set to 0, 0.2, 0.4, 0.6, 0.8, or 1.0. The usual setting is 0.2 as described in P.861 9.5.4 Note 2. The setting of the silent interval weighting factor (Wsil) significantly affects the PSQM value.

The mapping of the PSQM value to Mean Opinion Score or MOS is described in P.861. 0 is equivalent to excellent and 6.5 is bad on the Listening Quality Scale defined in ITU-T Rec. P.800. For simplicity, a linear mapping has been employed.

PSQM Value 0 = MOS LQ 5
PSQM Value 6.5 = MOS LQ 1

So, MOS LQ = 5 – (4 * PSQM / 6.5)

PSQM	0	1	2	3	4	5	6	6.5
MOS	5.00	4.38	3.77	3.15	2.54	1.92	1.31	1.00

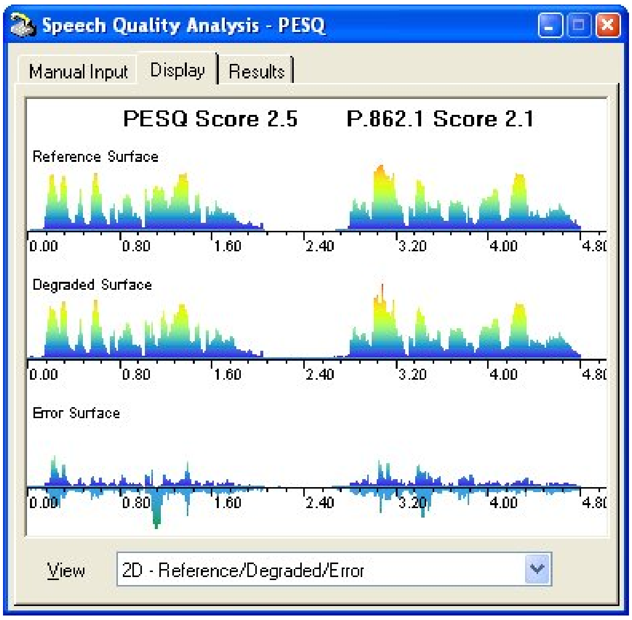
Other mappings may be more appropriate. Other PSQM implementations on other platforms may well use different mappings leading to different MOS. The use of different Wsil will lead to differences in PSQM values.

Note that the PSQM values that are returned by DSLA for packet based networks may be better than other implementations of PSQM on other platforms. This is because the DSLA uses superior techniques for time and level alignment. The PSQM values should be consistent across platforms for non-time varying networks.

The PSQM and mapped MOS results are presented for the Wsil value selected when the PAMS process is called. The results are written to the results.log file if selected in Options dialog box (Tools|Options|Results). The MOS value may be similar to the LQ value reported in the PAMS window. There may be differences due to the types of distortion that have been applied to the degraded signal. The limitations of PSQM, designed as a codec evaluation tool, when used in network tests have been described elsewhere.

PAMS – Speech Quality Analysis Display

Menu: View|PAMS



Using the PAMS display window you can:

- View 2D and 3D views of the PAMS sensation surfaces.
- Zoom, playback and re-analyse surface views.
- Display utterance markers.
- Save surface views as a Bitmap file for importing into a word processor etc.
- View graphical plots for transfer function, signal spectra and utterance time offsets.
- Execute the PAMS process on your own sound files.

About the PAMS Display

Sensation surface and error surface views

The sensation surfaces show the perceived loudness, on the phon scale, of the signals in time and frequency. The frequency scale is the Bark scale, and the time interval between successive samples is 4ms. The sensation surfaces are very useful, clearly showing the signals' content.

The difference between the sensation surfaces of the reference and degraded speech is the error surface. This means that errors that have added to the signal (for example, noise) have positive values and appear in the top error surface view, while parts of the signal that have been attenuated or muted have negative values and can be seen clearly below the error surface. The magnitude of the error is related to how audible and annoying it will be.

Examination of the error surface can provide useful insights into the degradation that has occurred. Noise will show as a positive error in the silences between utterances. Degradation due to a voice activity detector will probably appear as negative excursions in the error surface at the beginning of an utterance. High frequency loss can be seen in the three dimensional view.

The 2D view of the time aligned reference, degraded and error surfaces show the following properties clearly:-

- Front-end clipping causes large but short negative errors at the start of speech bursts.
- Muting can be seen as prolonged negative errors during speech, where the degraded sensation surface falls close to zero.
- Addition of background noise shows up as positive errors, and is most apparent in silent periods.
- Coding distortion generally causes low-level errors throughout speech bursts, although this is very codec-dependent.
- Bit or frame errors tend to cause localised distortion, which may be positive or negative. This effect is dependent on the codec and any error concealment algorithm used.

Transfer function and signal spectra views

Even if equalisation is disabled, the system's transfer function and the spectrum of the reference and degraded signals are computed. All of these quantities are calculated in dB for each of the 19 frequency bands.

If the IRS send filter is auto-detected, it is applied to the reference signal before the transfer function is calculated and so will not be part of the transfer function estimate.

Utterance time offset view

The distribution of utterance time offsets clearly shows how the speech transmission performance is being affected by variations in packet arrival time or jitter buffer resizing.

Using the PAMS – Speech Quality Analysis Display

Selecting a view

1. Open the PAMS – Speech Quality Analysis Display window (View|PAMS).
2. Click the **Display** tab.
3. Select the required view from the **View** list box.

Analysing an area of the display

1. Select the area you wish to analyse by dragging over the display with the left mouse button held down to highlight the required area.
2. Right-click the display and then click **Selection|Analyse**.

Note

- The analysed area will start from the beginning of the nearest utterance start point to the end of the nearest utterance end point.

Listening to an area of the display

1. Select the area you wish to listen to by dragging over the display with the left mouse button held down to highlight the required area.
2. Right-click the display and select **Selection|Play**.

Zooming the view

1. Select the area you wish to zoom by dragging over the display with the left mouse button held down to highlight the required area.
2. Right-click the display and select **Selection|Zoom**.
3. Restore a zoomed area by right-clicking the display and selecting **Restore View**.

Changing the degraded sensation surface display options

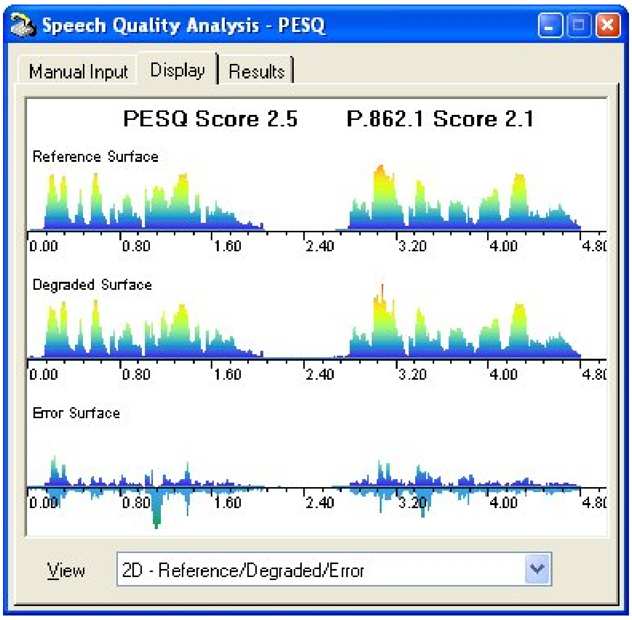
1. Right-click on the display, and then click, **Degraded Display Options**.
2. Select an option from the menu.

Saving the view as a Bitmap

1. Right-click on the display and click, **Save Image As**.
2. To save the image in a different folder, click a different drive in the Save in box, or double-click a different folder in the folder list.
3. In the File name box, type a name for the document.
4. Click **Save**.

PESQ – Speech Quality Analysis Display

Menu: View|PESQ



Using the PESQ display window you can:

- View 2D and 3D views of the PESQ sensation surfaces.
- Zoom, playback and re-analyse surface views.
- Display utterance markers.
- Save surface views as a Bitmap file for importing into a word processor etc.
- View graphical plots for transfer function and utterance time offsets.
- Execute the PESQ process on your own sound files.

About the PESQ Display

Sensation surface and error surface views

The sensation surfaces show the perceived loudness, on the Sone scale, of the signals in time and frequency. The frequency scale is the Bark scale, and the time interval between successive samples is 16ms. The sensation surfaces are very useful, clearly showing the signals' content.

An alternative view of the power distribution in different frequency bands provides a more useful representation of the signal.

The difference between the sensation surfaces of the reference and degraded speech is the error surface. This means that errors that have added to the signal (for example, noise) have positive values and appear in the top error surface view, while parts of the signal that have been attenuated or muted have negative values and can be seen clearly below the error surface. The magnitude of the error is related to how audible and annoying it will be.

Examination of the error surface can provide useful insights into the degradation that has occurred. Noise will show as a positive error in the silences between utterances. Degradation due to a voice activity detector will probably appear as negative excursions in the error surface at the beginning of an utterance. High frequency loss can be seen in the three dimensional view.

The 2D view of the time aligned reference, degraded and error surfaces show the following properties clearly:-

- Front-end clipping causes large but short negative errors at the start of speech bursts.
- Muting can be seen as prolonged negative errors during speech, where the degraded sensation surface falls close to zero.
- Addition of background noise shows up as positive errors, and is most apparent in silent periods.
- Coding distortion generally causes low-level errors throughout speech bursts, although this is very codec-dependent.
- Bit or frame errors tend to cause localised distortion, which may be positive or negative. This effect is dependent on the codec and any error concealment algorithm used.

Transfer function

Even if equalisation is disabled, the system's transfer function and the spectrum of the reference and degraded signals are computed. All of these quantities are calculated in dB for each of the 42 or 49 frequency bands.

Utterance time offset view

The distribution of utterance time offsets clearly shows how the speech transmission performance is being affected by variations in packet arrival time or jitter buffer resizing.

Using the PESQ – Speech Quality Analysis Display

Selecting a view

1. Open the PESQ – Speech Quality Analysis Display window (View|PESQ).
2. Click the **Display** tab.
3. Select the required view from the **View** list box.

Analysing an area of the display

1. Select the area you wish to analyse by dragging over the display with the left mouse button held down to highlight the required area.
2. Right-click the display and then click **Selection|Analyse**.

Note

- The analysed area will start from the beginning of the nearest utterance start point to the end of the nearest utterance end point.

Listening to an area of the display

1. Select the area you wish to listen to by dragging over the display with the left mouse button held down to highlight the required area.
2. Right-click the display and select **Selection|Play**.

Zooming the view

1. Select the area you wish to zoom by dragging over the display with the left mouse button held down to highlight the required area.
2. Right-click the display and select **Selection|Zoom**.
3. Restore a zoomed area by right-clicking the display and selecting **Restore View**.

Changing the degraded sensation surface display options

1. Right-click on the display, and then click, **Degraded Display Options**.
2. Select an option from the menu.

Saving the view as a Bitmap

1. Right-click on the display and click, **Save Image As**.
2. To save the image in a different folder, click a different drive in the Save in box, or double-click a different folder in the folder list.
3. In the File name box, type a name for the document.
4. Click **Save**.

Using DSLA to Analyse Speech Quality

When using Speech Quality Analysis with the DSLA, a degraded signal is recorded by the DSLA using a Playlist script. The degraded file is then compared with a reference file(s) stored on your PC's hard disk. The reference file(s) can be either the ASTS speech set, or your own WAV formatted files.

Example Playlist implementation using Speech Quality Analysis

1. Create a Playlist on Channel A which contains the following events:

Step	Type	Description
1		Start Measurement [A] - PAMS Example
2		Speech element 1
3		Speech element 2
4		Speech element …
5		Wait [00:00:01:00]
6		PAMS from Step 2, REC = A

2. Set the **Wait** event **Duration** property to 1 second - this event allows any network delay to be taken into account.
3. Set the **Speech Quality Analysis** event **Start measuring from Step** property to the event index of the first speech element.
4. Set the **Speech Quality Analysis** event **Record from Channel** property to the channel which is to receive the input – or degraded signal.
5. On the **Channel A** window toolbar, click **Start**

Notes

- Your reference files must exist within the default Phonytalk folder as defined in Tools|Options|General and must also be copied to DSLA memory and placed within the same folder hierarchy.

  Examples: (default folder path set to: 'c:\programs\dsla\phonytalk')

Local file path	**DSLA file path**
c:\programs\dsla\phonytalk\file1.wav	dsla:\file1.wav
c:\programs\dsla\phonytalk\sub-folder\file2.wav	dsla:\sub-folder\file2.wav

- Do not use more than 15 seconds of speech or the record buffer will lose the earlier material.
- There must be at least 4 seconds of speech for either PAMS or PESQ to produce a prediction.
- Add a short wait period before the Speech Quality Analysis statement to allow for network delay.
- An Ethernet connected DSLA will replay and recover speech of any duration, limited only by the hard disk space.
- There is a fifty utterance limit for Speech Quality Analysis.

Analysing Speech Quality Using Your Own Sound Files

Speech Quality Analysis can be used to process any WAV or headerless (RAW), mono channel, 8/16k, sound files.

1. Open a Speech Quality Analysis window.
2. Select the **Manual Input** tab.
3. Enter the file paths for the reference and degraded files, or select a previously used file from the drop-down lists.
4. Click **OK**.

Notes

- The DSLA must be connected to the PC for the process to be enabled.
- If using headerless (RAW) files, be sure to define the format you are using. (Tools|Options|Speech Quality Analysis|Sound File Data Format).
- There must be at least 4 seconds of speech for Speech Quality Analysis to produce a prediction.
- You can preview a file listed in the file path boxes by clicking Preview.

16k Sample Rate Operation

16k sample rate sound files can only be used with Ethernet DSLA using an IP connection. There is sufficient memory in DSLA for about 35 seconds of 16k sample rate speech files.This allows all of the 16k ASTS files from either the IRS or SP folder to exist in DSLA memory, but not both together. There is sufficient memory in DSLA II for about 130secs of 16k sample rate speech files.

When using a Playlist that includes a speech quality analysis test using more than 8 seconds of speech, DSLA will stream the degraded file back to the PC, therefore sufficient Ethernet bandwidth must be available between DSLA and the PC.

The 16k sample rate ASTS is installed in the <InstallPath>\Phonytalk\16k folder. Create an American or British folder in the root of DSLA and download the appropriate ASTS files. A Playlist calling these files will automatically use 16k versions of the files in this folder or in the DSLA if the sample rate is set to 16k on the status bar or by the schedule. This means that your existing Quick Start schedules and Playlists can be used without making any changes to them.

When you select 16k in the DSLA status bar then only those files in DSLA that are 16k appear in the DSLA Explorer window. You can add 16k sample rate versions of your own speech files to the 16k folder in the same structure as existed in the Phonytalk folder for the 8k versions and, if there is sufficient space in memory, download them to DSLA.

Both 16k and 8k files can exist in the same folder in DSLA, even though they are in the Phonytalk /16k and Phonytalk folders respectively. Switching DSLA between 8k and 16k gives access to the different sample rate files.

Note that selecting 16k in the DSLA status bar forces the channel bandpass filter to the off state. The channel bandpass filter is useful for speech level measurements but the attenuation of high and low frequencies in the degraded file shows a significant mean opinion score impairment at 16k sample rate.

You can create 16k schedule tasks or 8k schedule tasks or leave the task undefined so that it is controlled by the DSLA status bar. Each schedule task has a 16k/8k property in the Channel Properties tab. Schedules created by earlier versions of the DSLA User Interface (<3.70) will assume the 16k/8k status in the DSLA status bar when they are executed. If such a schedule is saved it will not have a 16k/8k property unless it has been set by the user. If you wish to create new schedules that will work with either 16k or 8k speech files, depending upon the DSLA status bar status, do not set the schedule task 16k/8k property.

Alternatively, you can specify file paths to 8k files or to 16k files in Playlists and define the 16k or 8k sample rate in the Channel Properties tab in each schedule task. Sample rates cannot be mixed in a Playlist.

16k sample rate Intermediate Reference System send characteristic ASTS files are supplied (IRS folder) along with flat response files (SP folder). You can use the IRS files with any telephone network.

The SP files should only be used with artificial mouth or where DSLA is directly connected to a wideband telephony or communications system. If the degraded signal is recorded through a HATS artificial ear, then the degraded acoustic filter should be unchecked in the Speech Quality Metric Control, either manually in the Speech Quality Metrics properties window or via the schedule task Metric Control tab. This control will be applied to both PAMS and PESQ if you are using both and selects the appropriate model in the speech quality metric for HATS ear recording.

The speech quality metric settings should be changed where the signal is to be received by a wideband communication device employing headphones for listening (PC to PC speech communications, for example). Both the reference and the degraded acoustic filters should be unchecked in the Speech Quality Metric Control, either manually in the Speech Quality Metrics properties window or via the schedule task Metric Control tab. This control will be applied to both PAMS and PESQ if you are using both and selects the appropriate model in the speech quality metric for wideband telephony. The P.862.2 score will be reported instead of P.862.1.

Speech path	**Source Speech Material**	**PESQ/PAMS Degraded acoustic filter**
h/free mic jack>GSM>base	IRS	ON
Base>GSM>h/free e/p jack	IRS	ON
Mouth>GSM>base	SP, Equalised for Mouth	ON
Base>GSM>Microphone	IRS*	ON
HATS mouth>GSM>base	SP, Equalised for Mouth	ON
Base>GSM>HATS ear	IRS*	OFF

* Unless IRS filtering is applied in the audio module of the base station emulator, in which case use SP.

Typical Subjective Test Scores for Various Codecs

Coder Type	Application	Standard	Typical MOS(LQ)	Bit Rate kb/s
LPC vocoder	Secure Telephony	FS-1015	2.36	2.4
PSI-CELP	Cellular	PDC HR	3.51	3.45
IMBE	Satellite	IN-M	3.19	4.15
CELP	Secure Telephony	FS-1016	3.09	4.8
MPC-MLQ & ACELP	Compression	G.723.1	4	6.3
MPC-MLQ & ACELP	Compression	G.723.1	3.51	5.3
VSELP	Cellular	GSM HR	3.51	5.6
VSELP	Cellular	PDC	3.51	6.7
CS-ACELP	Cellular	IS641	4.04	7.4
VSELP	Cellular	IS54	4.54	7.95
CS-ACELP	Compression	G.729	4.04	8
CELP	Cellular	IS96	3.5	8.5
Multipulse LPC	Satellite	AMTS	3.24	9.6
RPE-LTP	Cellular	GSM	3.56	13
LD-CELP	Compression	G.728	4.05	16
ADPCM	Compression	G.726	4.05	32
ADPCM	Compression	G.727	4.05	32
Companded PCM	Compression	G.711	4.17	64

Calibration

The DSLA has a three year calibration cycle. Calibration can only be performed by Malden Electronics Ltd. Contact us to arrange the return of the unit. It will be tested and re-calibrated if necessary. A new calibration certificate will be returned with the unit.

It is easy to verify that the DSLA is still in calibration. It is a true RMS voltmeter, so you can compare its measurements of a sinusoidal waveform with those of another measuring instrument that is in calibration. Similarly, the sine wave output levels and frequencies can be verified with a known good instrument. The speech level determination is performed in software and is based upon the signals observed at the A/D converter. If the sine wave measurements are correct then the speech level measurements must be as well.

Frequently Asked Questions

Q. What type of cable is required to connect my PC to the DSLA?

A. You need to use a 'straight through' (extension) serial cable.

Q. What is the minimum pin connections required for the serial cable?

A.

PC (9 way)	PC (25 way)	DSLA
2	3	2
3	2	3
5	7	5

Q. What is the meaning of the Offset Confidence figure shown in the PAMS 'Results' display, or displayed in the Channel status bar when using the 'Delay – Time Align' event?

A. The confidence figure is a percentage value, showing the reliability of the delay or offset figure that has been calculated. It is proportional to the number of points that have been matched successfully.

Q. What does the colour scaling indicate in the PAMS window?

A. It allows you to differentiate the differences between signal levels at different times – particularly in the 3D views of the auditory sensation surfaces. Signal energy is indicated from blue – low signal energy, to red – high signal energy.

Q. What is the difference between the SP and IRS speech sets?

A. The SP speech set contains utterances that are flat to 4kHz and are ideally suited for use where a wide bandwidth signal is expected. i.e. Into a standard mouth acoustically coupled to a microphone. The IRS speech set contains utterances that are bandwidth limited using the Intermediate Reference System Send characteristic (ITU-T Recommendation P.79) and are ideally suited for direct injection into the telephone network.

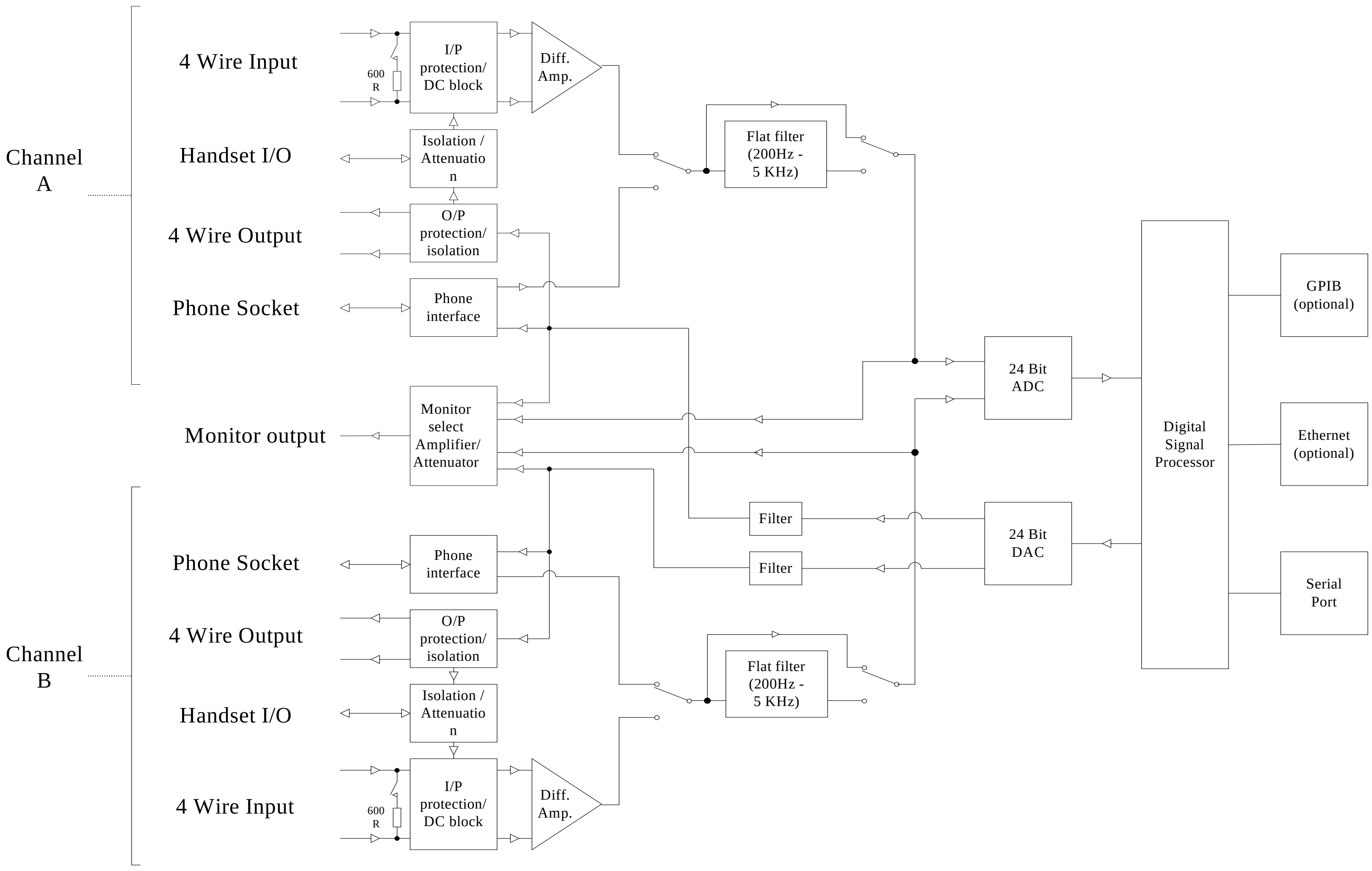
Q. Can I automatically suppress error messages that may appear whilst running an automated test sequence?

A. Yes. Open the Options dialog box (Tools|Options) and select the 'Suppress and Log Error Messages' option found in the 'General' options page.

Q. Where can I get the latest version of the DSLA software?

A. For the latest DSLA software updates and information, visit our website at: http://www.malden.co.uk

DSLA Block Diagram



Glossary of Terms and Abbreviations

BABT	The British Approvals Board for Telecommunications
ITU	The International Telecommunications Union
IEEE	Institution of Electrical and Electronic Engineers
P.48	ITU Recommendation P.48. Specification for an Intermediate Reference System (IRS).
P.56	ITU Recommendation P.56 for Objective Measurement of Active Speech Levels Method B.
P.800	ITU Recommendation P.800 for Subjective determination of Transmission Quality.
P.830	ITU Recommendation P.830 for Subjective performance assessment of telephone-band and wideband digital codecs
P.861	ITU Recommendation P.861 for Objective quality measurement of telephone-band codecs (300-3400 Hz)
P.862	ITU-T Recommendation P.862 for Perceptual Evaluation of Speech Quality (PESQ), an objective method for end-to-end speech quality assessment of networks and speech codecs
dB	Decibel
RMS	Root-Mean-Square.

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Technical Support

Before contacting Technical Support, you should check this manual. If you still cannot solve the problem, you can obtain product support in several ways:

Telephone: 020 8786 9145

Email: Ask questions and receive answers from Technical Support via the Internet. Just send email addressed to the account listed below. You will receive a reply via email.

support@malden.co.uk