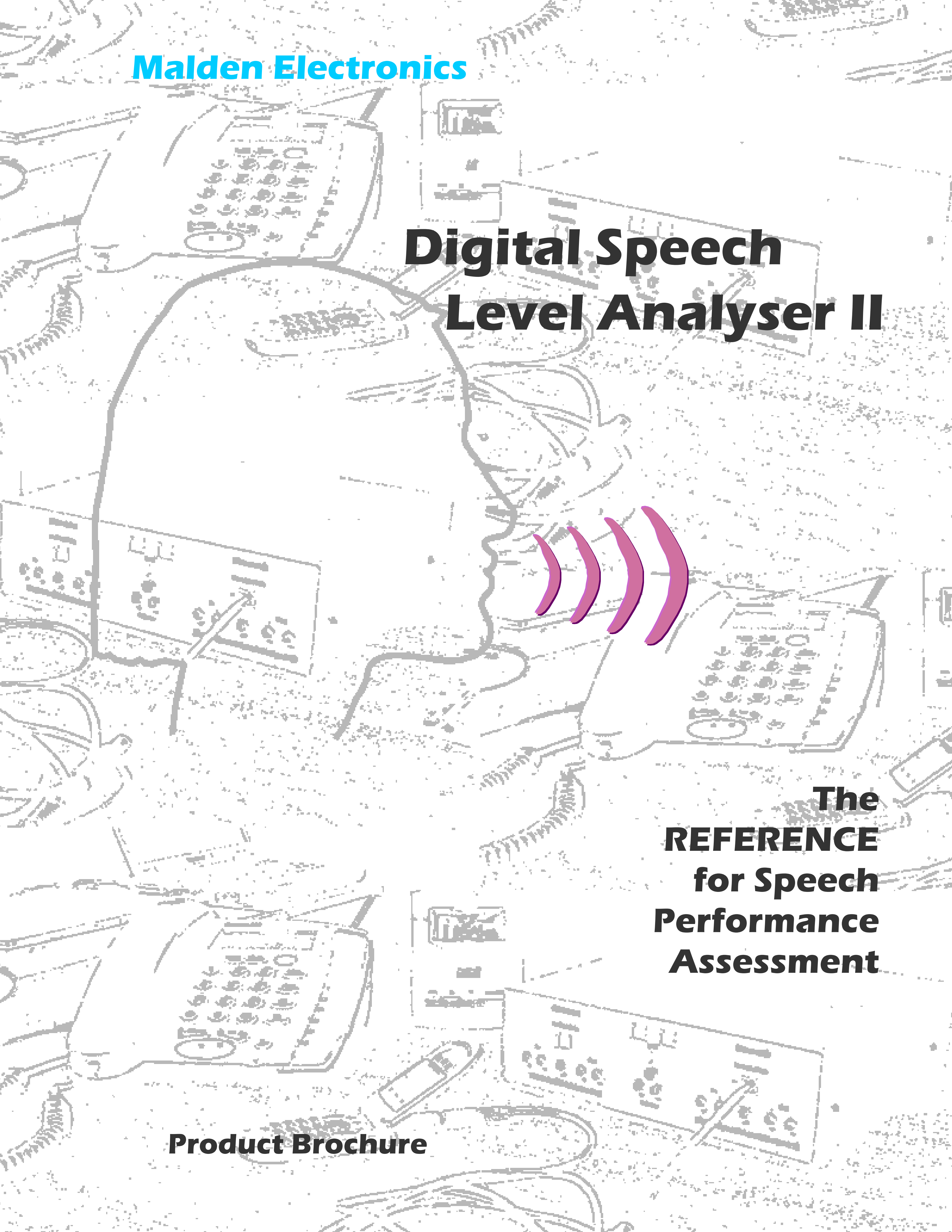


**Malden Electronics**

# Digital Speech Level Analyser II

## The REFERENCE for Speech Performance Assessment

**Product Brochure**

## Product Overview

The **Digital Speech Level Analyser** (DSLA) is a professional measurement system for objective speech quality prediction and speech level measurement in telecommunications equipment and networks.

**Exacting Design**

DSLA is distinguished from other speech test products by an exacting design standard which places it in a class of its own. All DSLA circuitry is purpose designed in-house by Malden Electronics. This achieves performance levels not possible with products based on a personal computer sound card and interface adaptors. Analogue interfaces, signal converters, noise performance, linearity and dynamic range are all specified to superior standards, making DSLA the choice of professionals for speech performance assessment.

**Based on Experience**

Originally launched in 1997 to meet the needs of the 'voice over data' revolution, the DSLA has been continuously developed and enhanced, evolving today into DSLA II, a state of the art measurement system for speech performance assessment. The specialisation of Malden Electronics in this field has enabled it to respond rapidly to customers' requirements, resulting in a system of unparalleled precision, capability and flexibility. It has become the system of preference in applications ranging from core technology development to network operations.

**Generic Solution**

DSLA is used in the research, development, acoustic and test laboratories of telecommunication equipment manufacturers, supporting development and testing of VoIP, GSM, UMTS, DECT, TETRA/TETRAPOL and VoDSL terminals, media gateways, echo cancellers, integrated access devices, PBX equipment and telephone switches. DSLA is used in product evaluation and system selection laboratories as well as in the installation, maintenance and Quality of Service (QoS) departments of network operators and service providers.

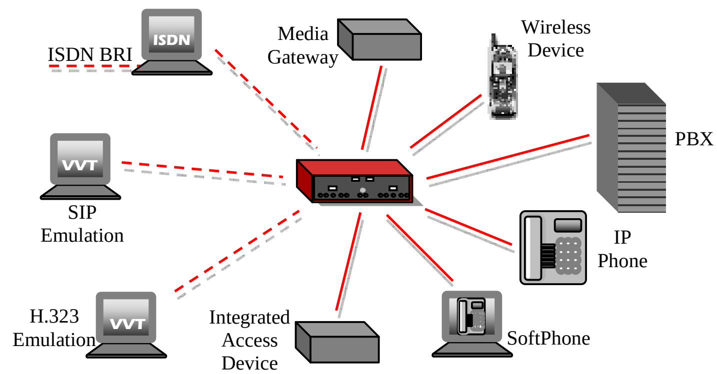
**International Standards**

DSLA measures the mean active speech level (MASL) according to ITU-T P.56 and performs ITU-T P.862 (PESQ) objective speech quality scoring plus improved MOS prediction according to ITU-T P.862.1. The MASL technology allows the DSLA to generate speech signals at any defined level, and to measure mean active speech level, noise level, peak signal level and activity factor. In addition to speech level measurements and speech quality scores, DSLA offers key speech performance indicators in numerical and graphical form, essential for the analysis of distortions in the degraded voice signal caused by noise, codec distortion, voice activity detector (VAD) performance or network impairments such as delay, packet loss, jitter buffer behaviour and echo.

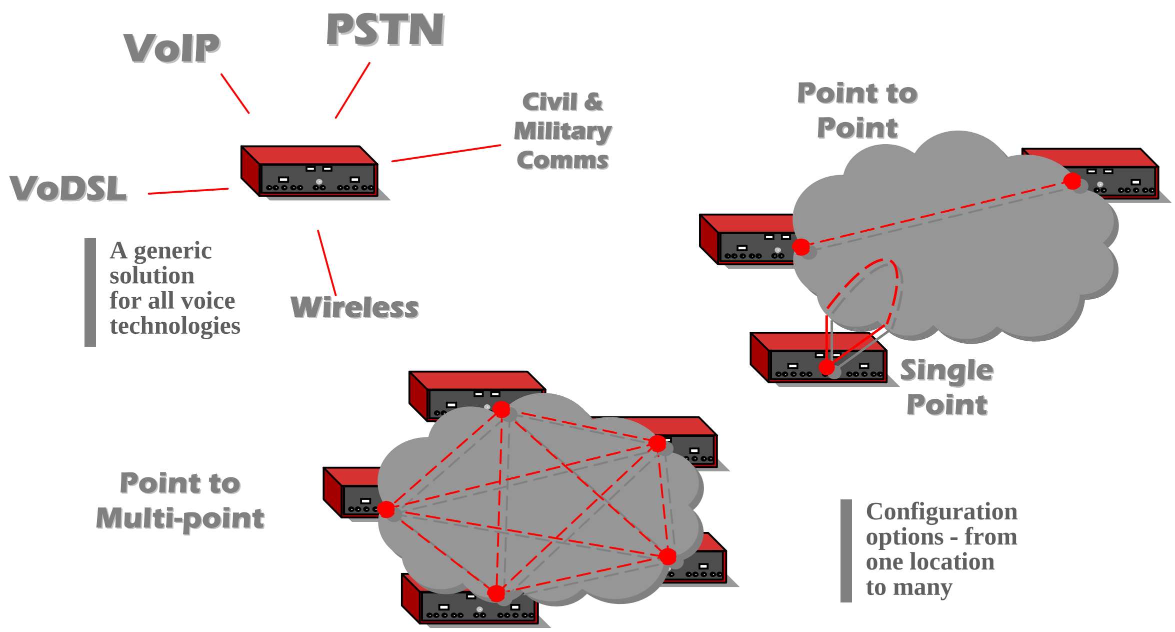
**Many Configurations**

Telephone, Handset and 4-wire Balanced analogue interfaces are incorporated into the two-channel DSLA terminal. These can be used alone or in conjunction with a variety of optional digital interfaces including ISDN BRI and SIP/H.323 test agents, providing a measurement capability between many types of physical and virtual network end points.

**Analogue & Digital End-Points**



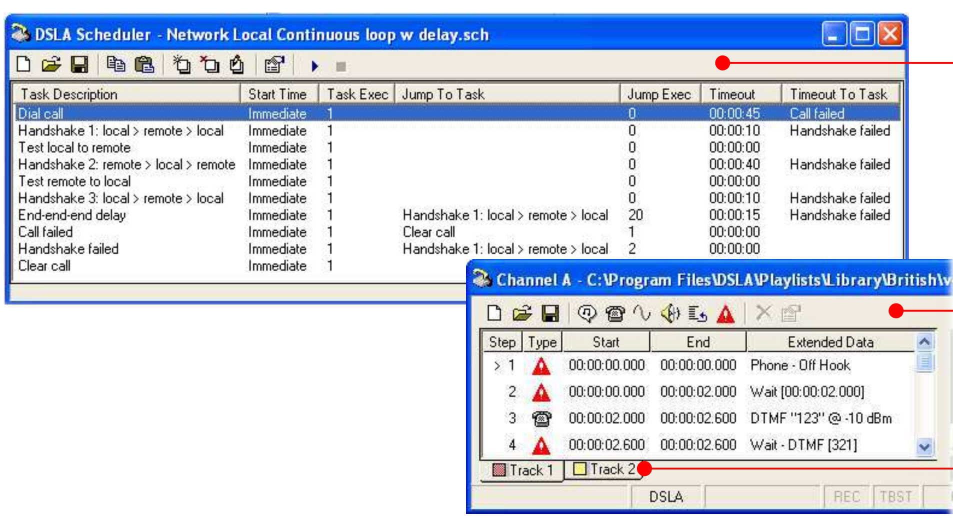
## DSLA: Superior speech performance assessment for modern voice networks




A generic solution for all voice technologies

Configuration options - from one location to many

## Two levels of scripting provide extreme flexibility




| Task Description | Start Time | Task Exec | Jump To Task | Jump Exec | Timeout | Timeout To Task |
|---|---|---|---|---|---|---|
| Dial call | Immediate | 1 | | 0 | 00:00:45 | Call failed |
| Handshake 1: local > remote > local | Immediate | 1 | | 0 | 00:00:10 | Handshake failed |
| Test local to remote | Immediate | 1 | | 0 | 00:00:00 | |
| Handshake 2: remote > local > remote | Immediate | 1 | | 0 | 00:00:40 | Handshake failed |
| Test remote to local | Immediate | 1 | | 0 | 00:00:00 | |
| Handshake 3: local > remote > local | Immediate | 1 | | 0 | 00:00:10 | Handshake failed |
| End-end delay | Immediate | 1 | Handshake 1: local > remote > local | 20 | 00:00:15 | Handshake failed |
| Call failed | Immediate | 1 | Clear call | 1 | 00:00:00 | |
| Handshake failed | Immediate | 1 | Handshake 1: local > remote > local | 2 | 00:00:00 | |
| Clear call | Immediate | 1 | | | | |



| Step | Type | Start | End | Extended Data |
|---|---|---|---|---|
| 1 | | 00:00:00.000 | 00:00:00.000 | Phone - Off Hook |
| 2 | | 00:00:00.000 | 00:00:02.000 | Wait [00:00:02.000] |
| 3 | | 00:00:02.000 | 00:00:02.600 | DTMF "123" @ -10 dBm |
| 4 | | 00:00:02.600 | 00:00:02.600 | Wait - DTMF [321] |



High-level commands provide powerful control of short-term or long-term testing

Low-level commands provide flexibility and precision

Dual tracks enable two scripts to run in parallel – for example: mix speech and noise

## The choice of professionals

- **True end-to-end assessment accurately predicts users' perceptions of quality**
    - **A professional grade measuring instrument and system**
        - **Outstanding Price/Performance ratio**
            - **Used internationally by major equipment developers and network operators**

### DSLA is the REFERENCE for Speech Performance Assessment

# Applications

**Subjective Assessment**

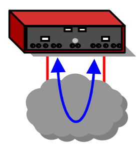
All voice transmission techniques are prone to losses and distortions. New-generation networks are affected by many impairments, including coding distortion, noise, crosstalk in the network elements, errors, packet loss, speech latency and jitter. The traditional way to measure the end-user's perception of speech quality is by the *subjective* listening methods described in ITU-T Recommendation P.800, which is the basis of the Mean Opinion Score (MOS). MOS assessment is both complex and expensive to perform.

**Objective Assessment**

DSLA uses *objective* methods for speech quality assessment, such as the PESQ, PAMS and PSQM speech quality metrics. These techniques aim to predict the results of subjective evaluation. The PESQ Score, in particular, is mapped to the MOS scale to provide an accurate prediction of subjective MOS over a wide range of coding and network conditions. The objective measuring method is now commonly used for laboratory tests of network elements and also for field and maintenance applications in the heterogeneous network infrastructure. DSLA enables development and network engineers to set up and to maintain the quality of voice transmission to achieve the necessary balance between quality of service (QoS) performance and investment and operating costs.

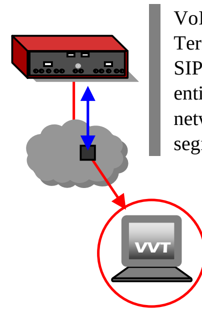
DSLA supports a wide variety of configurations, from the laboratory to international networks. Some examples are shown here:

**Local**



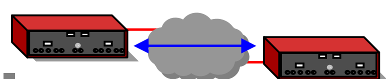
Replaces two physical end-points for end to end test

**Physical to Virtual End Point**



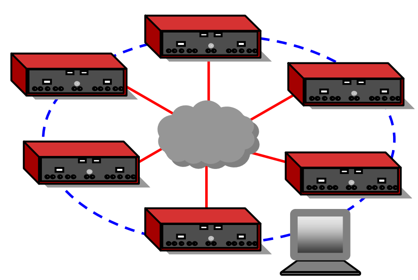

VoIP Virtual Terminals emulate SIP or H.323 entities, enabling network segmentation

**Local / Remote**



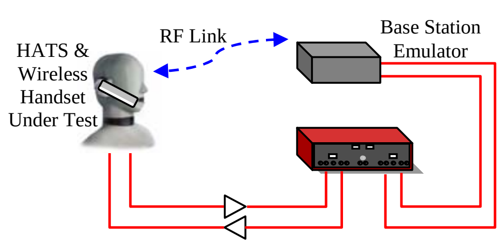
Two or more DSLA's test across the network, between locations

**Multi-DSLA**



Multi-DSLA configuration with Central Control

**With Acoustic Test Tools**



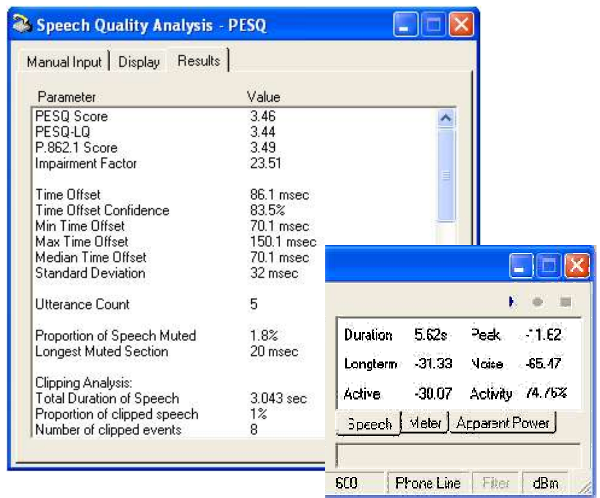

Use with Head & Torso Simulator for Acoustic Test of Wireless Handset

## Results

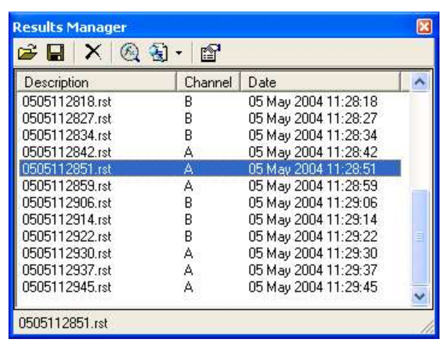
DSLA provides a comprehensive set of tools for the generation, management, display, remote viewing, logging, summary and detailed analysis of results. Users have widely varying priorities for results. Developers typically require a critical and detailed view of many aspects of speech performance - this is provided by the DSLA graphical results presentation and further enhanced by the Performance Examiner option. Network operators typically require a statistical analysis, with emphasis on patterns of performance over different periods.

DSLA allows the user to configure results presentation to meet specific requirements, selecting from a range of options for logging, setting limits, handling exceptions and generating alarm messages.

**Concise displays of data with per field logging selection**




**Powerful results management simplifies viewing**

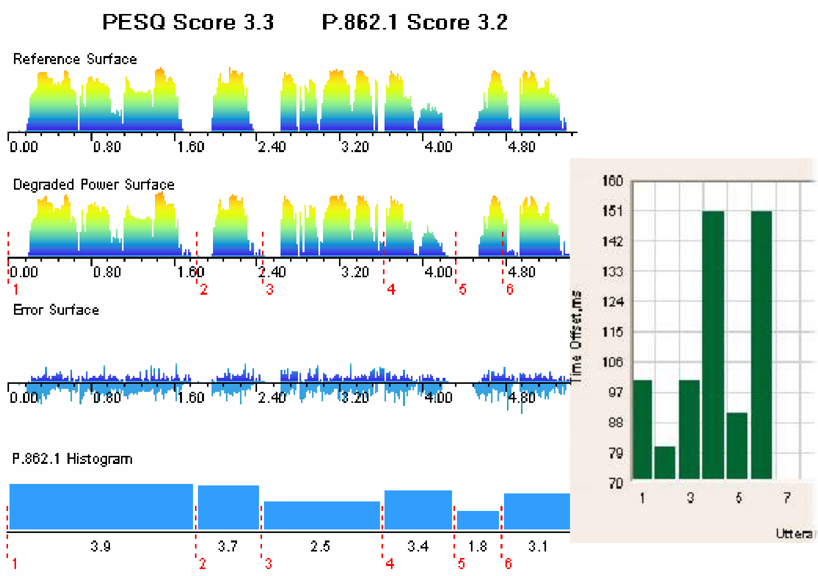



**Selectable logging format; Spreadsheet-compatible file**

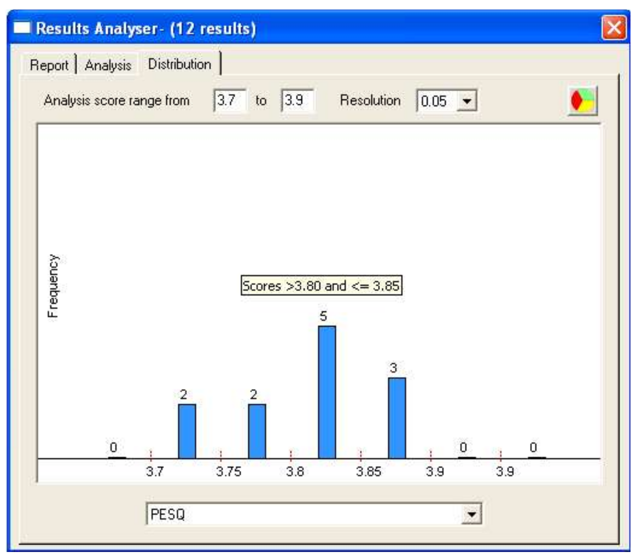
| DESCRIPTION | DUR(s) | ACTLVL | PEAK | NSE | ACT% | PESQ | P862.1 | OFFSET | MIN | MAX | STDDEV | DATE/TIME |
|---|---|---|---|---|---|---|---|---|---|---|---|---|
| Handset A-Handset B | | | | | | | | | | | | |
| Chan B 1311153659.rst | 5.91 | -34.29 | -17.15 | -66.32 | 73.34 | 4.08 | 4.23 | 66 | 65.9 | 66 | 0.1 | 13/11/2004 15:36 |
| Chan B 1311153708.rst | 6.46 | -33.88 | -17.74 | -66.1 | 72.24 | 4.04 | 4.2 | 65.9 | 65.9 | 65.9 | 0 | 13/11/2004 15:37 |
| Chan B 1311153716.rst | 5.62 | -34.24 | -17.96 | -65.08 | 73.35 | 4.02 | 4.18 | 66 | 65.9 | 66 | 0.1 | 13/11/2004 15:37 |
| Chan A 1311153725.rst | 5.91 | -35.36 | -18.48 | -67.77 | 73.46 | 4.02 | 4.18 | 65.1 | 65 | 65.1 | 0.1 | 13/11/2004 15:37 |
| Chan A 1311153734.rst | 6.46 | -35.18 | -19.01 | -66.32 | 72.02 | 4.03 | 4.19 | 65 | 65 | 65 | 0 | 13/11/2004 15:37 |
| Chan A 1311153742.rst | 5.62 | -35.57 | -19.22 | -68.32 | 73.72 | 4 | 4.16 | 65.1 | 65 | 65.1 | 0.1 | 13/11/2004 15:37 |
| Chan B 1311153751.rst | 5.75 | -34.13 | -11.66 | -67.02 | 70.27 | 4.06 | 4.21 | 66 | 66 | 66 | 0 | 13/11/2004 15:37 |
| Chan B 1311153800.rst | 6.39 | -33.8 | -13.63 | -67.51 | 69.85 | 4.1 | 4.25 | 66 | 66 | 66 | 0 | 13/11/2004 15:38 |
| Chan B 1311153808.rst | 5.5 | -34.39 | -14.66 | -67.26 | 72.16 | 4.06 | 4.21 | 66 | 66 | 66 | 0 | 13/11/2004 15:38 |
| Chan A 1311153817.rst | 5.75 | -35.2 | -12.74 | -68.04 | 70.16 | 4.02 | 4.18 | 65.1 | 65 | 65.1 | 0.1 | 13/11/2004 15:38 |
| Chan A 1311153826.rst | 6.39 | -35.02 | -14.97 | -66.1 | 69.61 | 4.08 | 4.23 | 65 | 65 | 65 | 0 | 13/11/2004 15:38 |
| Chan A 1311153834.rst | 5.5 | -35.77 | -15.76 | -67.51 | 72.55 | 4.01 | 4.17 | 65 | 65 | 65 | 0 | 13/11/2004 15:38 |

## Results, cont.

**Graphs provide insight into the detail of voice performance**




**Manual and automatic summary and analysis of multiple test data**




**Detailed analysis of DTMF sequences, including frequency and level of each tone (DTMF Analysis Option)**

DTMF Analysis Results - 13 Nov 2004 11:42:00

Expected Tx Digits: 10
Detected Rx Digits: 10

| Tx | Rx | On ms | Off ms | Twist | LF(Hz) | Shift% | LF(dBm) | HF(Hz) | Shift% | HF(dBm) |
|---|---|---|---|---|---|---|---|---|---|---|
| 1 | 1 | 195.9 | 100.1 | 3.9 | 697 | 0.0 | -21.6 | 1209 | 0.0 | -17.7 |
| 2 | 2 | 200.1 | 100.1 | 5.3 | 697 | 0.0 | -21.7 | 1336 | 0.0 | -16.4 |
| 3 | 3 | 199.8 | 100.0 | 6.7 | 697 | 0.0 | -22.0 | 1477 | 0.0 | -15.3 |
| 4 | 4 | 200.0 | 100.0 | 3.4 | 771 | 0.1 | -21.2 | 1209 | 0.0 | -17.8 |
| 5 | 5 | 199.9 | 100.4 | 5.0 | 771 | 0.1 | -21.4 | 1336 | 0.0 | -16.4 |
| 6 | 6 | 199.5 | 100.3 | 5.9 | 770 | 0.0 | -21.2 | 1477 | 0.0 | -15.3 |
| 7 | 7 | 199.9 | 100.1 | 3.1 | 852 | 0.0 | -20.7 | 1209 | 0.0 | -17.6 |
| 8 | 8 | 199.8 | 100.5 | 4.1 | 853 | 0.1 | -20.7 | 1336 | 0.0 | -16.6 |
| 9 | 9 | 199.8 | 100.0 | 5.4 | 853 | 0.1 | -20.7 | 1477 | 0.0 | -15.3 |
| 0 | 0 | 99.6 | 0.0 | 3.5 | 941 | 0.0 | -20.2 | 1336 | 0.0 | -16.7 |

# DSLA System Functions

**Single DSLA Terminal**

- Measure between:
  - Analogue phone line / Analogue phone line
  - Analogue phone line / Handset interface
  - Analogue phone line / 4-wire balanced interface
  - Handset interface / 4-wire balanced interface
  - Any analogue interface / SIP/H.323 test agent (option)
  - SIP/H.323 test agent / SIP/H.323 test agent (option)
  - Any analogue interface / ISDN BRI interface (option)
  - ISDN BRI interface / SIP/H.323 test agent (option)
  - ISDN BRI interface / ISDN BRI interface (option)

**Two or more DSLA Terminals**

- Local or remote control
- Measure between:
  - Any analogue interface at one unit / Any at another unit

**Automated Call Control**

- On hook, off hook, dial, ring detect for POTS services
- Call control for Cisco Call Manager / 79xx series IP phone environment (other types to follow)

**Speech quality measurements**

- Speech quality scoring using PESQ and optionally PAMS and PSQM & PSQM+ (ITU-T P.861, withdrawn)
- PESQ LQ and ITU-T P.862.1 mapping to MOS
- Utterance-by-utterance PESQ and PAMS score histograms
- Estimation and graphical presentation of transfer function.
- Graphical presentation of PESQ and PAMS analysis
- 2D and 3D graphical presentation of reference, degraded and error surface signals with zoom and playback functions
- Visualisation of speech impairment effects such as noise, codec distortion, temporal clipping, packet loss, synchronisation loss, jitter buffer offsets.
- Waveform display of reference and degraded signals
- PSQM scoring with time alignment and weighting factor (option)

**With Performance Examiner Option:**

- Frame by frame PESQ score and display
- Frame by frame delay variation
- Frame-by-frame time offsets
- Frame by frame signal level and gain
- Bark and linear signal spectrum
- Transfer function estimation

**DTMF**

- Generate standard DTMF strings with all 16 characters
- Detect specified DTMF string

**With DTMF Analysis Option:**

- Generate DTMF strings with control over parameters
- Measure and report characteristics of received DTMF string up to 20 digits

**Signal Handling**

- Full support for generation and analysis of narrowband and wideband (8kHz) telephony signals
- Monitoring output with independent selection of input and output signals for each channel

**Speech Level Measurements**

- Speech level measurement to ITU-T P.56, method B
- Mean active speech level, peak level, noise level, RMS level, measurement duration, activity factor
- Graphical presentation of speech level distribution
- True level measurements for DTMF and tone bursts
- Measurement of apparent power and power meter for the RMS power of external signals with 1.5 sec integration
- Measurement and simulation of doubletalk

**Echo Measurements**

- True measurement of mean active speech level of echo signal in analogue domain
- Measurement of echo at 2-wire or 4-wire interface, even in presence of sidetone
- Measurement of echo canceller performance with/without double talk
- Simulation of echo delay and level for echo canceller performance analysis

**Measurement of VAD Performance**

- Graphical presentation of front-end clipping in the PESQ or PAMS degraded and error surfaces
- Frame-by-frame VAD performance graphs and data: front-end clipping, back-end clipping, hangover (Performance Examiner option)

**Packet Loss**

- Graphical presentation of packet losses in the degraded and error surfaces
- Zoom function with time scale for detailed signal analysis

**Delay**

- Different methods for one-way delay measurements, using speech as the stimulus
- True round trip delay measurement using end-end-end method (for measurement between two DSLA terminals)

**Delay Variation and Jitter buffer performance**

- Utterance-by-utterance time offset with histograms.
- Minimum, maximum, mean, median, standard deviation of time offset

**Results Management**

- Excel-compatible log file with field selection
- Report all or report by exception
- Manual and automatic results summary with conditional actions
- Automatic e-mail generation on exception, based on single measurement or minimum, maximum, mean or std. dev. of multiple measurements
- Graphical presentation with rotating 3D surface explorer
- Waveform diagrams with zoom, overlay, re-size and re-colour
- Spectrograms with zoom, overlay, re-size and re-colour
- Save graphs as bitmap images
- Export data from most graphs
- Composite graph/data files can be viewed in DSLA or with downloadable Speech Performance Viewer (SPV) application

# Technical Specification

**Test Signals**

- Artificial Speech Test Stimulus (ASTS) British or American English; 8k and 16k sample rate
- Any user supplied speech material in wav or PCM format, generated with user definable mean active speech level with setting range -99dBm to +10dBm
- Sine wave 20Hz to 15kHz, setting range -99dBm to +10dBm any duration
- Swept sine wave 20Hz to 15kHz, setting range -99dBm to +10dBm any duration
- White noise setting range -99dBm to +10dBm any duration
- DTMF setting range -99dBm to +10dBm any duration
- DTMF user definable twist, frequency offset and break duration (option)
- Conversational speech with/without double-talk
- Filter signals to MIRS, HATS or user definable filter (option)
- Two independent tracks on each channel to create complex mixed signals

**Test Signal Presentation**

- Two balanced and floating inputs (600 ± 0.1ohm or 1Mohm) and outputs (600 ± 0.1ohm) on 4mm sockets
- Two telephone ports 600ohm or complex impedance on RJ11. Output level limited to +6dBm
- Two balanced and floating inputs (10kohm) and outputs (25ohm) on handset ports on RJ22. Output level attenuated by 28dB on specification above. Network isolated.
- Monitor output with switching/mixing of input and output of each channel

**Optional Features**

- Equaliser - applies any desired filter characteristic to 8 or 16k sample rate sound files. A typical HATS, plus IRS and mIRS preset filters are included. Equalisation is applied manually or automatically.
- DTMF Analysis - analyses a DTMF string up to 20 digits, bridging 10msec gaps. Measurements of on/off times, level, frequency and % shift for each tone, plus twist and an option to compare the detected digit to the expected digit. Advanced DTMF generator sets the interdigit silence, frequency offset and twist away from the standard to test network equipment DTMF capability.
- Performance Examiner – displays detailed speech performance data in graphical form, including frame-by-frame score and time offset, clipping, level and noise analysis. Includes a scalable graphics window with selectable colours and transparency, facilitating overlays for comparison. Recommended for users requiring the most detailed understanding of speech transmission performance.
- VoIP Virtual Terminal - a test agent which runs on a networked PC, emulating a SIP/H.323 phone under control of the DSLA system. VoIP VT originates or terminates a call and receives or generates speech signals. The degraded file is returned to the DSLA user interface for analysis.
- ISDN BRI Interface - presents an ISDN BRI interface as a virtual port of DSLA terminal. PCMCIA or PCI card.

**Measurements**

- Speech level measurement ITU-T P.56 Method B
- DTMF detection ITU-T Q.24, Bellcore TR-TSY-000181, EIA/TIA-464A
- Noise in speech to within 20dB of mean active speech level
- Peak and true RMS Levels
- Units of measurement dBm, dBV, dBr, mV
- Tone burst measurement mode
- Measurement of doubletalk (percentage of measurement period where speech is present on both channels)
- Measurement of Mean Apparent Power
- Linearity 0.1dB for levels -60 to +10dBm
- Linearity 0.1dB for frequencies 20Hz to 14.8 kHz
- Noise floor -85dBm or better
- Range of measured levels -75dBm to +19dBm
- Minimum measurable mean active speech level -65dBm
- Dynamic range of 4-wire inputs 104dB

**Calibration**

- Full calibration report supplied
- Calibration cycle 3 years

**Minimum PC System Requirements**

- Windows 2000 or XP
- Pentium-class processor, 200MHz
- 64Mb RAM
- Screen resolution 1024 x 768 recommended
- DirectX 8.1 or higher recommended

**Dimensions (mm)**

- Approx 72 high x 218 wide x 200 deep

**Weight**

- Approx 3kg excluding packaging and accessories

**Power Supply**

- 100 – 240Vac or 9 – 18Vdc, 12W

**Operating Temperature Range**

- - 2 to + 40$^{o}$C

**CE Mark**

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