Lindos DS10

Software Audio Test Set

User Manual

First Edition

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1. Introduction to the DS10

The DS10 is a Windows software package, which turns any PC with a sound card into an audio test system that combines the functions of an oscillator (signal generator), level meter, spectrum analyser, noise meter, peak programme meter and distortion meter, along with frequency and phase measurement.

Its main feature though is 'Sequence Testing', introduced by Lindos twenty years ago in the LA100 test set, and universally used by broadcasters and studios for routine quality assurance ever since. It takes just two clicks to run a typical test sequence that can quantify any audio system in terms of levels, frequency response, phase, distortion and noise in as little as twenty seconds. If Lin4WinXP (another software package common to all Lindos test systems) is also running, an impressive results sheet with graphs and pass/fail tolerance testing appears immediately on the screen ready to save, print, or even publish to the Web on our Test Sheet Database for all to see! Sequence tests carry (FSK) coded information, which ensures that they are self-synchronising and fully automatic, even between two DigiSonics.

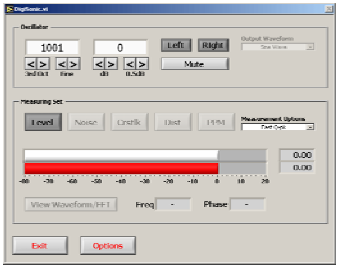
The DigiSonic can generate and receive LA100 and MiniSonic sequences and digital processing has enabled us to include swept distortion and swept phase tests as well.

All measurements are completely real-time and great attention has been paid to ensure that the software behaves more like hardware than software.

DO NOT INSERT THE DONGLE UNTIL AFTER INSTALLING THE DIGISONIC SOFTWARE!

2. Installing the DigiSonic

The DigiSonic is available for download from ~~www.lindos.co.uk~~ and is also supplied on a CD found on the inside front cover of this manual. When the dongle is not found the software runs in demonstration mode, which provides a basic oscillator and level meter.

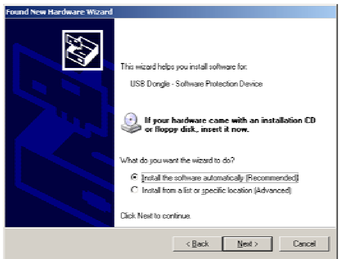


1. Insert the DigiSonic CD into your computer
2. Open the 'DigiSonic Setup' directory and double click on setup.exe.
3. Follow the on-screen prompts and install the software.

If you encounter any problems installing the software restart the computer, ensure that no other applications are running and try again.

The software will work with Windows 2000 and XP. A 1.5GHz processor is necessary for all of the DigiSonic's features to work correctly.

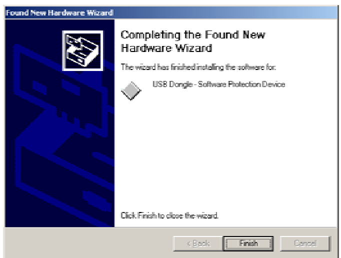
To run the DigiSonic, click on the Windows Start button, follow the programs menu arrow and click on the DigiSonic icon. It is good idea to create a shortcut on the desktop.



When the DigiSonic starts, it will run in demonstration mode and most of the buttons will be greyed out. Insert the dongle and wait for the 'Find new hardware wizard' window to appear.

Select 'No, not at this time' and click next.

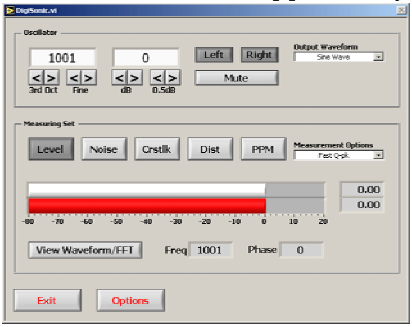
Select 'Install the software automatically' and click next



Click 'Finish' to complete the dongle set-up procedure and the DigiSonic buttons will ungrey.

2.1 Calibrating the Sound Card

The DigiSonic will function with any sound card that is supported by Windows, for testing in the digital domain only a sound card that support AES3/EBU or SPDIF is all that is required. More commonly soundcards incorporate A/D and D/A conversion and it is important to realise that the quality of testing in the analogue domain can only be as good as the quality of the convertors.



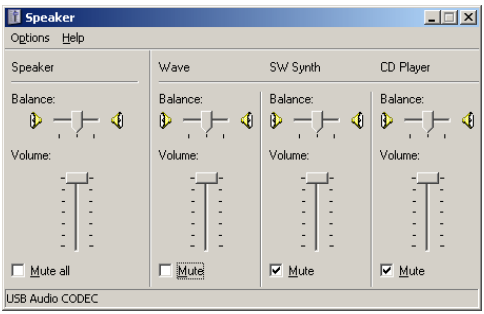
The software starts-up outputting 1kHz at 0dB on whichever sound card is selected within Windows from the 'Sound and Audio Devices' category in the 'Control Panel'.

Calibrated analogue input and output levels will only be obtained if the convertors output and input levels are accurately adjusted. With some professional convertors, the levels will be 0dBu (775mVrms) to within a few tenths of a dB, which may be good enough for many purposes. With consumer cards there is no standard level, but it has been found that some do (by chance) come close to the Lindos 'UniSon' standard of –6dBu (387mVrms). For best accuracy a calibrated sine-wave source will be needed – otherwise use calibration method 2 below.

If a calibrated oscillator or measuring set is not available then we can hire you a MiniSonic (see ~~www.lindos.co.uk/prices.html~~) or alternatively send your sound card to us and we'll measure the errors for you. Contact Lindos for more information.

2.2 Calibration Method 1

Connect the calibrated sine-wave source to the input of the sound card. The DigiSonic provides the ability to enter software correction factors for the sound card, but first you need to use the input gain control to calibrate input and closely as possible.



the  Double-click on speaker icon (found on the right-hand side of the Windows toolbar) to open the card's volume control settings.

Select 'Properties' from the 'Options' menu on the volume control window. If the sound card has an input level control (the YellowTEC PUC and some other cards do not have) then select the 'Recording' button (otherwise greyed out) and click OK to open 'Recording Control Window'. Adjust the input level slider until the measured level reads around 0dB. At this stage the two channels may read differently. The two digital level readings represent the measuring errors.

Enter the negation of these errors (i.e. positive become negative and vice-versa) for both channels into the Left In (dB) and Right In (dB) boxes. Click OK and check that both channels read zero.

Disconnect the sine-wave source and connect the sound card's output to its input. Double-click on the speaker icon again to open another 'Volume Control Window'. Mute any output sliders other than 'Wave' to eliminate their noise contributions. Set the card's output 'Wave' slider to full and adjust the volume/speaker slider until the measuring set reads as close to zero as possible. Again read off the errors and enter the negated readings into the Left Out (dB) and Right Out (dB) boxes. Click OK and check that the levels read zero.

2.3 Calibration Method 2

In the absence of both a calibrated signal source and a calibrated level meter (PPM), the best that can be done is to calibrate each input channel in turn to an arbitrary level (maximum output setting is perhaps the best choice). Plug the left input into the right output and read the (left) input level. Enter the negation of this into the L channel calibration window as above. Then plug the right input channel into the left output and read the right channel level. Enter the negation of this into the (right) input calibration box. Then finally connect left-to-left and right-to-right. Read the right input level, and enter the negation of this into the (right) output calibration box. Note that if you use this method without calibrating the left channels output first, setting 0dBu may not produce 0.775Vrms out of the card. This requires particular consideration when measuring the available headroom of a system, and when calibrating other equipment using a DigiSonic.

2.4 Installing Lin4WinXP

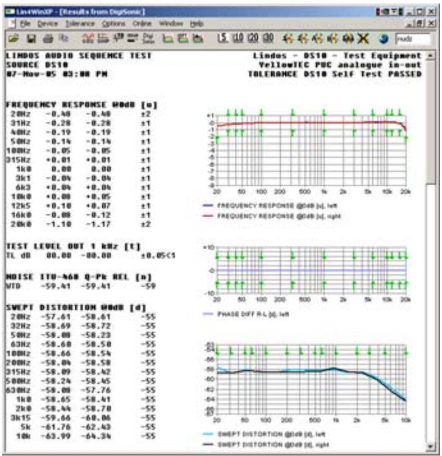
Lin4WinXP can control a DigiSonic locally (on the same PC) or remotely over a network. As well providing access to the Sequence testing facilities built into the DigiSonic, all of the manual measurement features on the DigiSonic can be controlled from Lin4Win's control panel.

1. Open the 'Lin4Win Setup' directory found on the same CD as the DigiSonic and double click on Setup.exe.
2. Follow the onscreen instructions and install the software. Create a shortcut on your desktop or toolbar.

If you encounter any problems installing the software restart the computer, ensure that no other applications are running and try again.

The software will work with Windows 95,98,NT, 2000 and XP.

Double click on the Lin4WinXP icon and immediately the results and control panel windows will appear. Click the DigiSonic icon on the toolbar at the top of the main results window to establish a local TCP/IP connection with the DigiSonic. For remote connection see section 4.1 Remote Connection



2.5 Running a Sequence

A DigiSonic Sequence consists of an unlimited sequence of segments; each segment is building block that generates a particular series of tones, a sweep, or silence, to test a particular parameter (frequency response, distortion or noise etc). It begins by sending its identifying character, (in ASCII code) as a frequency-shift-keyed (FSK 110-baud) signal, which acts as a synchronising trigger for a series of timed measurements. The responding DigiSonic does not have to be the unit that is generating the sequence.

Enter a list of segments typically 'nurhz' into the sequence text box on the control panel (or on the far right of the toolbar at the top of the main window) and press return (or click the Run button). This immediately puts the DigiSonic into Sequence mode and runs and receives the sequence.

Each segment lasts typically 5 seconds and when the Sequence is complete the results are displayed in the results window. Double click on a graph to view it in detail, or on the text to edit the header information.

The function of **Seq** is to put the DigiSonic into Sequence mode ready to receive a Sequence from another device. Pressing **Seq** before running a sequence also offers the advantage of providing a list of segments and sequences in the function drop-down menu on the control panel.

What follows is a more detailed description of each of the DigiSonic's features. You may wish to skip to section '*4.0* Using the DigiSonic with Lin4WinXP', *'5 Sequence Testing'* or *'5.5 Segment Details (V1.0)'.*

3. Stand-Alone Use (Without Lin4WinXP)

The maximum sample rate that can be used by the DigiSonic is currently limited to 44.1kHz (48kHz 24bit coming soon!). The maximum frequency that the DigiSonic allows the user to select is half the sample rate.

3.1 Oscillator Use

3.1.1 Setting the Output Waveform

Sine Wave - Selects a sinusoidal output waveform.

Square Wave - Selects square wave output.

Triangular - Selects triangular wave output.

White Noise - Generates white noise at the (rms) output level selected. Note that the output frequency controls become greyed out.

Pink Noise - This is not currently implemented, but will be available in a future update. Contact Lindos if you would like this feature.

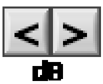
3.1.2 **Setting the Output Frequency**

Lindos standard frequencies are generated (for compatibility with other Lindos test sets).

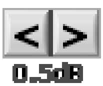
Adjusts the output frequency in 3^{rd} octave steps.

Adjusts the output frequency in 12^{th} octave steps.

3.1.3 Setting the Output Level



Adjusts the output level in 1dB steps.



Adjusts the output level in 0.5dB steps.

3.1.4 Channel Selection and Mute

The DigiSonic starts up driving both channels at 0dB 1kHz. Click **Left** to toggle the left channel off and on.

Click **Right** to toggle the right channel off and on.

Click **Mute** to toggle between muted output and the output state set by the **Left** and **Right** channel buttons.

3.2 Level Measurement

Click **Level** to measure the level on the incoming signal, the default option switches in a rectifier with fast Quasi-Peak (rms-reading) characteristics.

Level Option 1 – Fast Q-pk

The term Quasi-Peak refers to any rectifier (detector) that responds essentially to the peak of the signal, but takes a length of time to respond (integration time).

Level Option 2 – Super-fast Q-pk

This selects level measurement with a very fast response; reducing the attack and decay times by a factor of ten.

Level Option 3 – RMS

This takes the root mean square of the digital samples received over the last 10ms period.

3.3 Noise Measurement

Click **Noise** and the oscillator output is automatically muted (though it can be de-muted) and measurements are made according to the option selected. The default is Option 1.

Noise Option 1 - ITU-468 weighted Q-pk

Noise Option 2 - Unweighted Q-pk

Noise Option 3 - Unweighted RMS

Noise Option 4 - A-weighted Q-pk

Noise Option 5 - A-weighted RMS

Noise is often expressed as 'Signal to Noise Ratio' or 'Dynamic Range'. Dynamic range is often taken to mean the difference between max signal level and noise level, but it is also used to refer to the difference between max signal level (full scale digital) and ~~muted~~ noise level (suppressing the dither in a digital system).

What matters is how far the noise is below a typical signal, and it is a good idea to establish a nominal level, call it 'alignment level' (AL) and measure noise relative to this. All broadcast measurements are made relative to 'alignment level', which usually corresponds to 0dBu (0.775Vrms). Maximum permitted level, or full-scale digital (digital FS), will be above this by an amount referred to as 'headroom', usually 18dB (EBU recommendation) on digital recordings, or 24dB on some original material. Headroom is typically 12dB or less on broadcast paths, where the EBU recommends +9dB max level when measured on a PPM (which does not read the true peaks).

The merit of this system is that a noise measurement made relative to AL will always be a reasonable guide to the true intrusiveness of the noise onto typical material, regardless of the headroom. Where less headroom is available for transmission, compression or limiting will usually be used to reduce brief peaks, but this has little effect on overall perceived loudness. A recording may start life with 24dB of headroom, and end up with only 9dB of headroom at the listener's radio or television. Its 'dynamic range' has indeed been reduced, but it will not sound any noisier, assuming that it is the peaks that have been limited. Instead it will have lost 'sparkle' and impact. Specifying noise and headroom separately reflects these different qualities properly.

3.4 Distortion Measurement

Click Dist to measure the level of all harmonics and noise in the frequency range 20Hz to 20kHz. Various weighting filter options are provided, but Lindos recommends using ITU-R 468 weighting, which takes proper account of the high order harmonics caused by crossover distortion or digital quantising errors, and also gives due emphasis to the high frequency modulation noise that can cause a characteristic 'roughness' on tape or compact cassette. Such 'Distortion-Residue' readings will generally be higher than if they were measured using the THD (total harmonic distortion rms) method, and on harsh crossover-distortion they may be 15 or 20 dB higher. Lindos suggests reserving the term 'Distortion Residue' to mean 1kHz distortion measured ITU-R 468 weighted and Quasi-peak as an alternative to the traditional THD method.

Distortion Option 1 – Distortion Residue (ITU-468 weighted Q-pk

Distortion Option 2 - THD+N rms (bandwidth set by sample rate to 20kHz approx).

3.5 Crosstalk Measurement

Click Crstlk. Operation is exactly as for noise, with the same 468-Weighting and Quasi-Peak response, but one channel of the oscillator output is always muted while the other is driven at the set level and frequency, which can be adjusted while reading crosstalk. Crosstalk can be measured at any frequency or level, simply by changing the oscillator settings.

Crosstalk Option 1 – ITU-R 468 Weighted
Crosstalk Option 2 – Unweighted Quasi-Peak
Crosstalk Option 3 – Unweighted rms

3.6 Frequency and Phase Measurements

Frequency and phase measurements are displayed at the bottom of the DigiSonic's panel. Note that the left channel is considered the reference, so that a +ve reading indicates that the right channel leads the left and a –ve reading means that the right channel lags behind the left. This was chosen because the left channel is used for FSK signalling on sequences and so is the preferred channel to use as a directly connected reference when measuring absolute phase in-out on a piece of equipment. There are circumstances though in which the phase delay in a single channel is of interest, for example when attempting to 'time align' separate drivers in a wide range loudspeaker. In this case, the left channel input and output of the DigiSonic should be connected directly and treated as the reference signal (by plugging the XLRs together or by making up a special lead incorporating the link. The right channel input is then connected to the output of the device under test (crossover network etc) so that what is displayed becomes the input-to-output phase shift.

3.7 PPM (Peak Programme Meter) Use

Click PPM. Both channels are displayed with accurate IEC (type Ilb ie UK/BBC – British Broadcasting Corporation) dynamics, so this mode is ideal for checking the legality of programme levels for broadcast or recording.

PPM Option 2 – Fast attack PPM

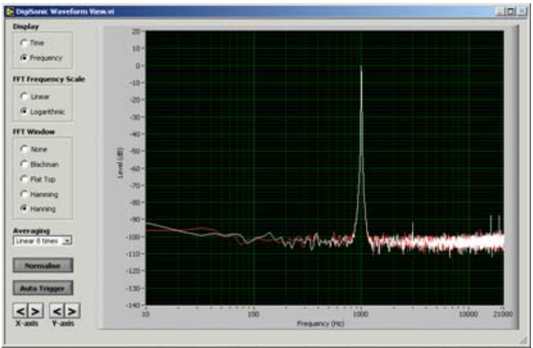
Provides an alternative PPM with very fast attack (integration time of under 1ms) on both channels. Decay time is as standard.

PPM Option 3 – Dual Speed (not currently implemented)

Selects dual-attack PPM operating on the left channel only. The main bar retains standard PPM characteristics while the 'shadowbar' has the very fast attack of option 2. This will be found useful for detecting whether brief peaks have been compressed. On uncompressed live percussive sounds a difference of 5 to 15dB will be observed between the two readings.

3.8 Spectrum Analysis

Click the **View Waveform/FFT** button to open the spectrum analyser and oscilloscope view. By default this window opens in frequency (spectrum analyser) mode displaying both channels overlaid on top of each other – the red trace represents the right channel.



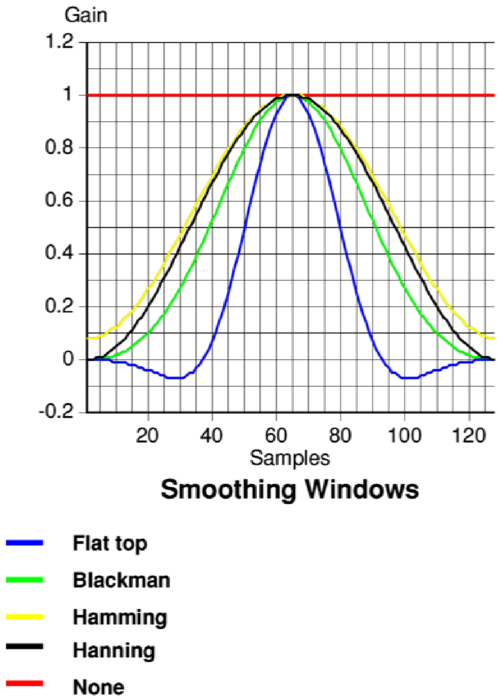
The spectrum analyser displays the frequency content of the input signal in real-time, but the interpretation of spectral plots requires careful consideration. This arises out of a fundamental difference between sine waves and other signals such as noise. The amplitude of a sine wave is easily defined by its rms value. The rms value of a band of noise however depends entirely on the bandwidth used for measurement, leading to the confusing fact that a spectrum analysis of white noise using infinitely small bandwidth for high resolution, gives no reading at all (or a very large number of very small readings!).

As implemented here, using FFT (Fast Fourier Transform), the power content is divided into 2048 frequency 'bins' which are of equal bandwidth (22Hz) across the spectrum. This gives rise to a display on white noise, which is a straight horizontal line at the level of power seen in the analysis bandwidth (the wider the bandwidth the higher the line). There is no such thing as spectral content at a given frequency for noise or any waveform that is not just pure sine waves – the bandwidth must be defined. There is such a thing as noise-power density at a given frequency, which is measured in Watts/Hz, this relates to typical noise figure specifications like 12 nV/√Hz as might be seen in an Op-Amp specification.

Clicking normalise shifts the display vertically such that the highest peak reads 0dB, which can help resolve the above problem when looking, for example, for the relative levels of harmonics in a distorted sine-wave.

3.8.1 FFT Windowing Options

The spectrum analyser splits up the incoming audio into chunk, each of length 2^12 (4096) samples. However, it is unlikely that this number of samples will correspond to an integer number of waveforms, and therefore the first and last samples in the 4096-sample chunk are unlikely to both be zero. This creates an edge at the start and finish of the chunk which, when analysed, results in wideband noise (spectral leakage). Windowing provides a solution to this problem by weighting the samples such that those at the edges are reduced to zero. The amount of edge attenuation applied determines the amount by which the spectral leakage is reduced and thus increases the available dynamic range for measurements. However, it also reduces the power within the signal and consequently a trade-off arises.



Several of the more common smoothing windows have been provided and their weighting functions are shown above.

3.8.2 FFT Averaging Options

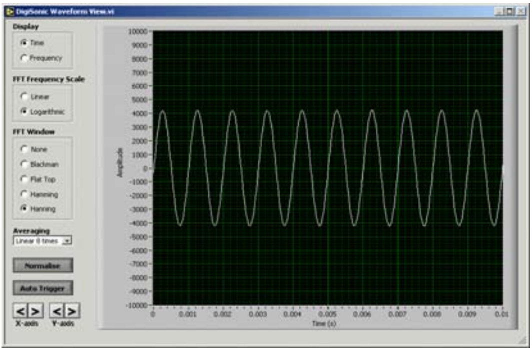
Averaging the results for each frequency band (bin) over a number of FFT analysis iterations smoothes out the variations in power within each bin. Because the frequency content of noise changes rapidly with time, an FFT with no averaging will display a large amount of hash at the noise floor. Averaging the bins helps remove this hash and produces a much clearer plot.

The default option provides eight times linear averaging. Though this produces the clearest display it is also the most processor intensive. Currently only linear averaging options are provided.

3.9 Oscilloscope View

Click **Time** to switch to the oscilloscope view, this changes mode of measurement from the frequency domain to the time domain. Again, both traces are overlaid on top of each other and the red trace is for the right channel.

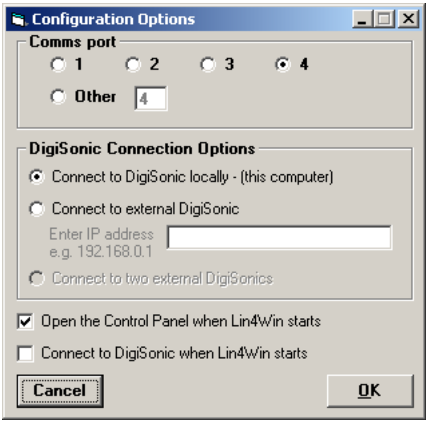
Auto Trigger is by default switched on. This triggers the display on the left channel's waveform and stabilizes the trace by also starting the view at a zero crossing.



X-axis allows the scale on the x-axis to be adjusted.
Y-axis allows the scale on the y-axis to be adjusted.

4. Using the DigiSonic with Lin4WinXP

Consult the enclosed Lin4WinXP manual for detailed information on Lin4Win's features. The new options and buttons specific to operation with the DigiSonic are explained below:



4.1 Remote Connection

Connect to DigiSonic locally sets the remote host to the local port of the machine running this software.

Connect to external DigiSonic allows the entry of an IP address in text field. To establish the IP address of a computer on a local network, start a windows command prompt, type ipconfig and press return.

Connection to two external DigiSonics is not currently implemented, but this option will allow Lin4Win to control a separate DigiSonic oscillator and DigiSonic measuring set at different IP addresses. Contact Lindos for more information.

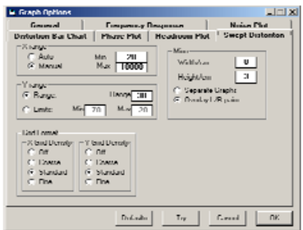
Connect to DigiSonic when Lin4Win starts enables Lin4Win to start-up in TCP/IP connect mode. If the user only has a DigiSonic, it would be sensible to check this option to avoid having to click the DigiSonic toolbar button every time Lin4Win is loaded.

4.2 The DigiSonic Toolbar Button

The DigiSonic button on the toolbar on the main window toggles the communication type between serial and TCP/IP. It is perfectly permissible to have a DigiSonic running while also communicating with an LA100 or MiniSonic; this button will toggle between the two.

4.3 Swept Distortion Graph Options

New options are provided to customisation of the swept distortion plot. These options are the same as those described for other graphs in section 5.3.3 of the Lin4Win manual.



5. Sequence Testing

5.1 Downloading and Interpreting Results

Though this is automatic whenever a sequence is run, or received, it is of course possible to download the stored results from any DigiSonic unit at any time by simply clicking the **GET** button on the Lin4Win control panel. Results can be downloaded from the unit again and again, unless they are overwritten by running a new sequence or cleared by clicking reset or restarting the software

A typical results page begins with a title, date and time (inserted by the PC), the latter being useful as a reference code when a number of similar results are being compared or put into order. A header on the right hand side can be entered by the user, and will usually contain the equipment make and type along with details of the settings or test conditions.

Then come the listings of results on the left, with corresponding graphical representations on the right. These graphs can each be viewed individually in a separate window, on a larger scale, and with numerical cursor readout using the **Graph Viewer**. They can also have their appearance altered in terms of height, width, scaling, normalising and colours using Graph Options. The bar chart representation of Distortion Residue is particularly interesting as it contains a lot of information designed to allow at-a-glance assessment of performance. The first bar represents the Noise level, followed by bars for relative Distortion Residue at up to six levels. The first of these (at –20dB) is often the most revealing. In the absence of significant distortion it will indicate a figure 20dB worse than the noise bar (because it is showing noise relative to the –20dB level and not alignment level) but crossover distortion (in amplifiers), quantising errors (in digital convertors), and modulation noise (on tape) will all show as further worsening. Manufacturers are keen to produce very low figures for distortion measured at MOL (maximum output level) or FS (full scale), but in practice these are of little relevance compared to performance at –20dB AL (typically -46dB on CD) where low relative levels are harder to achieve and defects are more likely to be heard.

5.2 Inter-working with the MS10

The DigiSonic has been designed to mimic the MiniSonic as closely as possible. DigiSonic segments are subset of MiniSonic segments with the exception of swept distortion (segment d), which is not supported by the MiniSonic.

5.3 Inter-working with the LA100

The DigiSonic and the MS10 normally uses segments that are different from those recognised by the LA100, because their design made a faster and better set possible. To distinguish them from LA100 segments, DigiSonic and MiniSonic segments are characterised by lower-case characters (the LA100 uses mostly upper-case characters).

The LA102 is only able to measure on one channel at a time and has to switch channels after completing left channels sequence measurements and run the sequence again for the right channel. Currently the DigiSonic only generates segment U.

When the DigiSonic receives an LA100 sequence, because it can make measurements on both channels simultaneously and because the LA101 can output on both channel simultaneously all of the sequence results apart from right channel crosstalk results will be logged on the first sequence pass. To run a LA100 sequence just once (and thus halving the time taken to complete), press the L/R key on the LA101 when it is in sequence mode before running the sequence until L (as opposed to L+R) is displayed in the top right hand corner of the graphical display. The sequence is still output on both channels.

5.4 Notes on Sequences and Segments

Segment letters can be entered in any order, either in the control panel or at the top of the main (results) window (press return to send) – the actual sending order used by the DigiSonic is fixed.

Segment q actually tests distortion at +11.5, +14.5, and +17.5dB rather than the nominal +12, +15 and +18dB quoted for simplicity. This provides a margin of safety to avoid 'just clipping' when equipment is not precisely aligned.

5.5 Segment Details (V1.0)

DigiSonic Segments:

SEGd	Swept Distortion	ITU-R 468 weighted distortion plot 20Hz to 10kHz 10s
SEG/n	Sequence Separator	Where n is an integer e.g. nu/2r generates nunur
SEG<n	Repeat Sequence	e.g. u< runs sweeps indefinitely refreshing after each.

MiniSonic Segments Supported:

SEG n	Master-seg	Test level
SEG u	Sweep	20Hz – 20kHz @ 0dB 5 secs
SEG r	Distortion Residue	Distortion Residue 1kHz @ -20, 0, +8dB 5s
SEG q	Dist Res to +18	Distortion Residue 1kHz @ 0, +12, +18dB 5s
SEG h	Headroom Plot	1kHz 0dB to +18dB in 1dB steps, 5s
SEG l	Long Noise Plot	Noise Weighted L-channel 20s
SEG z	Phase (FSK only)	Phase Plot (use with u)

LA100 Segments Supported:		***(not currently generated by the DS10)**
SEGT*	Test Level 1kHz	1kHz @ 0dB (includes 'MS10' start text) 1s
SEGV*	Test Level 400Hz	400Hz @ 0dB (includes 'MS10' start text) 1s
SEGU	Sweep	20Hz – 20kHz @ 0dB 5s
SEGR*	Tape Sweep	20Hz – 20kHz @ -10dB 5s
SEGD*	Distortion (THD)	100Hz, 1kHz, 6.3kHz @ +8dB 6s
SEGG*	Distortion 50us	40, 100, 315, 1k, 6k3, 10kHz with 50us de-emphasis 18s
SEGI*	Distortion (THD)	100, 1kHz @ +8 and –10dB 8 secs
SEGA*	Crosstalk	40, 100, 315, 1k, 6k3, 10kHz with 50us de-emphasis 6s
SEGN*	Noise	mute for 8s
SEGW*	Wow & Flutter	3.150kHz for 12s
SEGY*	Phase	40, 100, 1k, 6k3, 10k, 15k (mean) 50us de-emphasis is not provided
SEQZ*	Phase	40, 100, 1k, 6k3, 10k, 15k (mean)
SEQC*	Crosstalk	40,100,315,1k,6.3k,10kHz 6s

Special segments are available on request.

6. Appendix A - Remote Commands

Full remote control is possible using remote commands sent as ASCII characters over TCP/IP.

The DigiSonic uses two ports for TCP/IP communication. This is to maintain the independence to the two sides of the DigiSonic and facilitate remote control of independent DigiSonic oscillators and measuring sets.

Oscillator - 2055
Measuring Set - 2056

General Purpose Measuring Set Commands

ID?	Read Identity (dongle serial number followed by 'DigiSonic - DS10 V1.0')
TR	Total Reset
RS	Partial Reset (leaves sequence data stored)
D8	Start continuous TCP/IP data stream output (see website for further information)
D0	Stop streamed output

Oscillator Commands

FC*n*	Set Output Frequency by binary code (100 = 1kHz) (0 = mute) 1/12th Octaves 4Hz to 40kHz
LC*n*	Set Output Level by binary code (0 = mute 200 = 0dB) 0.5dB steps -95 to +18dB NB this also sets the an internal value, used to restore level after muting.
LL*n*	Set Output Level L chan only (codes as above) (restore value not affected)
LR*n*	Set Output Level R chan only (codes as above) (restore value not affected)

Measurement Commands

IC*n*	Set input Channel (1=left 2=right 0=both) can be shortened to IC for both
FN*nm*	Set measurement Function and Option: n and m are ASCII 0 to 7. m is optional. n : 0=Phase,1= Level, 2=Noise,3=Crosstalk,4=Distortion,5=PPM, 6=Seq , 7=Freq Option 0 is normal, eg FN1 (or FN10) is level, FN12 is super-fast level.
F?	Read input Frequency in Hz (BCD ie binary coded decimal)
L?	Read input Level in dB (ASCII decimal - CR terminated) (true value inc range)
P?	Read Phase in degrees (ASCII decimal - CR terminated)
SQ	Run a stereo test sequence. Send SQ to send the currently set Sequence
S?*n*	Read Sweep graph for register r (1 = left chan 2 = right)
SGx	Set Segment, x is the ASCII segment character, SG0 resets all segments To set up a sequence send SG0 followed by SGn, SGu, SGr for example
SR?	Read Sequence results (Lindos format)

7. Appendix B – Known Problems and Bugs

It is advisable to turn off any screen savers on your PC. Otherwise then the screen saver executes it will crash the DigiSonic's oscillator.

Removing the soundcard hardware at any point while the DigiSonic is running will cause an exception error message to appear. Unfortunately because of the way the software works there is no provision for catching this error and dealing with it sensibly. Press Crtl+Alt Delete to bring up the task manager and click end task to close the DigiSonic.

A 1.5GHz processor is necessary for all of the DigiSonic's features to work correctly, though many of the less processor intensive functions will operate on slower machines. If the processor cannot cope with demands being placed on it, glitches will be heard on the audio. Using other applications while the DigiSonic is running will also often cause glitches to be heard on the audio.