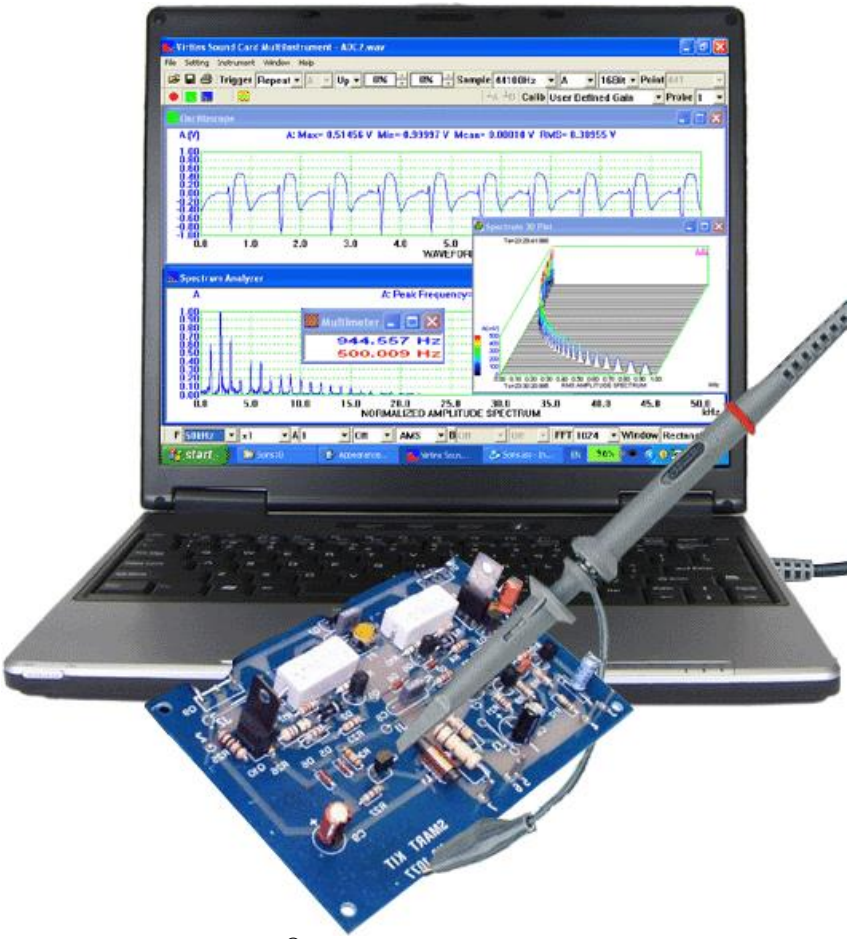
VIRTINS® Multi-Instrument 3.9.15 Manual



This is the manual for VIRTINS® Multi-Instrument. It contains manuals for:

- VIRTINS® Oscilloscope
- VIRTINS® Spectrum Analyzer
- VIRTINS® Signal Generator
- VIRTINS® Multimeter
- VIRTINS® Data Logger
- VIRTINS® Spectrum 3D Plot
- VIRTINS® Device Test Plan
- VIRTINS® LCR Meter
- VIRTINS® Derived Data Point (DDP) Viewer
- VIRTINS® Derived Data Curve (DDC)

If you have only purchased a subset of the full functions, then only the relevant portions of this document are applicable.

Note: VIRTINS TECHNOLOGY reserves the right to make modifications to this manual at any time without notice. This manual may contain typographical errors.

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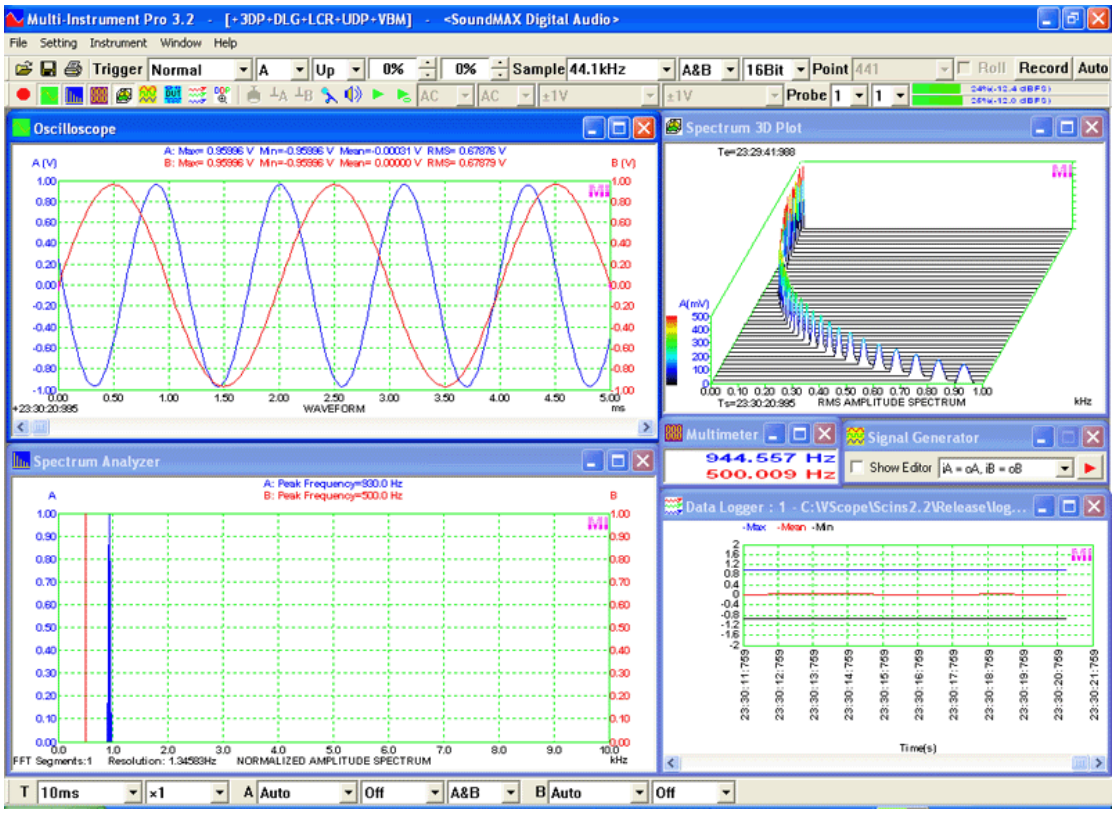
1 Introduction

1.1 Overview



VIRTINS® Multi-Instrument is a powerful multi-function virtual instrument software. It is a professional tool for time, frequency, time-frequency and modulation domain analyses. It supports a variety of hardware ranging from sound cards which are available in almost all computers to proprietary ADC and DAC hardware such as NI DAQmx cards, VT DSO, VT RTA, VT IEPE, VT CAMP, RTX6001 and so on. It consists of the following instruments and functions.

(1) Dual-channel Oscilloscope

It provides five types of views:

- Real time waveform of Channel A and Channel B
- Real time waveform of Channel A + Channel B
- Real time waveform of Channel A - Channel B
- Real time waveform of Channel A × Channel B
- Real time Lissajous Pattern for Channel A and Channel B

Statistical data such as Maximum, Minimum, Mean, Root Mean Square (RMS), Skewness and Kurtosis values are also calculated and displayed. Each frame of data is time stamped with an accuracy of 1 ms and the time stamp is shown at the lower left corner of the Oscilloscope view.

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The acquired signal can be inter-frame averaged (synchronously averaged), time aligned, demodulated (AM, FM, PM), DC removed, half-wave or full-wave rectified, and digitally filtered before any other analyses. The types of digital filters supported are: Low Pass, High Pass, Band Pass, Band Stop and Arbitrary. The class of the filter can be FFT, FIR or IIR. Reverberation, speech intelligibility, waveform discontinuity, step response, echo, damping ratio analyses are supported.

The displayed waveform can be played (output) or cyclically played (output) via the default computer sound card, if the sampling rate and bit resolution of the waveform is compatible with it.

The Oscilloscope also provides a Record Mode which can be used to record data to the hard disk continuously until the recording process is stopped manually or 2 gigabytes of data have been recorded, whichever is earlier. The display of the Oscilloscope can also work in Roll Mode, with which the screen will be updated in real time even if the sweep time is long.

The Oscilloscope can perform waveform conversion among acceleration, velocity and displacement when acceleration, velocity or displacement sensors are used.

The Oscilloscope can display both analog and digital signals in one window and thus can be used as a MSO (Mixed Signal Oscilloscope) when used with a hardware MSO.

The Oscilloscope supports digital persistence display mode, Equivalent Time Sampling (ETS) mode, SINC interpolation between samples.

(2) Dual-channel Spectrum Analyzer

It provides seven types of views:

- Real time Amplitude Spectrum / Power Spectrum Density / Impedance Spectrum
- Real time Phase Spectrum
- Real time Auto Correlation Function (Linear or Circular)
- Real time Cross Correlation Function (Linear or Circular, Original or Generalized)
- Real time Coherence / Non-Coherence Function
- Real time Transfer Function (Bode Plot, Frequency Response, or Gain and Phase Plot) / Impedance Analyzer
- Real time Impulse Response

with adjustable FFT size ranging from 128 to 4194304, adjustable window overlap percentage (0%~99.9%), and 69 selectable window functions such as Rectangle, Triangle, Hann, Hamming, Blackman, Kaiser and so on. It supports display in logarithmic scale for both X axis and Y axis (dBr, dBV, dBu, dBFS, dB), octave analysis (1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96), frequency compensation, frequency weighting (flat, A weighting, B weighting, C weighting, ITU-R 468 weighting), moving average smoothing (linear or 1/1~1/96 octave), DC removal, peak hold, linear average, exponential average, cross power

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spectrum vector average ("cross correlation average") and parameter measurements (THD, THD+N, SINAD, SNR, Noise Level, IMD-SMPTE/DIN, IMD-CCIF2, IMD-CCIF3, DIM, Crosstalk, Bandwidth, Harmonics, Peaks, Energy in user defined frequency bands, Wow & Flutter, Sound Loudness & Sharpness, Total Non-Coherent Distortion + Noise， GedLee Metric, Total Distortion + Noise, Noise Rating & Noise Criterion, Sound Intensity), etc.

(3) Dual-channel Signal Generator (Sweep/Arbitrary/Function/Burst Generator)

It provides the following types of waveforms/tones:

- Sine
- Rectangle (with adjustable duty cycle)
- Triangle
- Saw Tooth
- White Noise
- Pink Noise
- Multi-Tones
- Arbitrary waveform via user-configurable waveform library (WFLibrary)
- Maximum Length Sequence (MLS), with adjustable length (127~16777215)
- Dual Tone Multi-Frequency (DTMF)
- Unit Impulse
- Unit Step
- Musical Scale
- Wave File (*.WAV)

at either a fixed frequency, or a frequency that sweeps linearly or logarithmically within a specified frequency range and time duration.

The output signal can be looped back partially (i.e. only one channel while the other channel can be used for field input) or fully (i.e. both channels), via the software itself, to the input of the Oscilloscope for display and analysis in real time. Synchronized operation between the Signal Generator and the Oscilloscope with a timing accuracy in the same order of the sampling frequency is supported. Amplitude sweeping, burst signal generation, output signal fade in/out and modulation (AM, FM, PM) are also supported. In addition to the streaming mode, the software supports DDS (Direct Digital Synthesis) mode when the hardware used supports DDS.

(4) Dual-channel Multimeter

It provides the following types of digital displays:

- RMS
- dBV
- dBu
- dB
- dB(A)
- dB(Z)
- dB(C)
- Frequency Counter

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- RPM (Revolutions Per Minute)
- Counter
- Duty Cycle
- F/V (Frequency Voltage Conversion)
- Cycle RMS
- Cycle Mean
- Pulse Width
- Vibrometer (RMS, Peak/PP, Crest Factor values for acceleration, velocity and displacement)

The above items from Frequency Counter to Pulse Width involve a pulse counting process, and the software allows you to configure the counter trigger level and trigger hysteresis in order to shape the analog signal to rectangular pulses before counting. It also allows you to specify the frequency dividing ratio of the counter. Modulation domain analysis is performed during this counting process to measure jitter statistics.

(5) Dual-channel Data Logger

It provides long time data logging function for 254 Derived Data Points (DDP) and 16 User Defined Data Points (UDDP), including RMS value, Peak Frequency, Sound Pressure Level, RPM, THD, etc. Up to 8 data logger windows can be opened and each window can trace up to 8 variables. The logged data files can be reloaded into the data logger for review.

Three logging methods are supported: Fastest (i.e. take one record whenever new data are available), Time Interval (i.e. take one record whenever new data are available and the specified time duration has elapsed since the last update), Update Threshold (i.e. take one record whenever new data are available and the change compared with the last update exceeds the specified update threshold).

(6) Dual channel Spectrum 3D Plot

It is used to trace the spectrum variation with time. Two types of plots are provided:

- Waterfall, with an adjustable tilt angle of T axis, an adjustable height of Y axis, and selectable color palettes.
- Spectrogram, with selectable color palettes and a smoothing option

The Spectrum 3D Plot can be generated by multiple frames of data (Inter-frame Mode) or one single frame of data (Intra-frame Mode). The former supports Short Time Fourier Transform (STFT) only while the latter supports both STFT and Cumulative Spectral Decay (CSD).

(7) Device Test Plan

Device Test Plan provides a mechanism for you to configure and conduct your own device test steps. It takes the advantage of the sound card's (or other ADC/DAC hardware's) capability of simultaneous input & output, to generate a

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stimulus to the Device Under Test (DUT) and acquire the response from that device at the same time. Different stimuli can be generated and the response can be analyzed in different ways. The DUT can be marked as Pass or Fail after a sequence of test steps and a test report can be generated. Device Test Plan supports 30 instructions with corresponding parameters. Test results (e.g. Gain vs Frequency, Phase vs Frequency, etc.) can be plotted in up to 8 X-Y plots and reported in one textual log window in real time. Device Test Plan supports connection with external systems through serial communication.

Device Test Plan can also be used to perform other functions such as data file batch processing, batch signal event capturing and storing, etc.

(8) LCR Meter

It is used to measure the value of an inductor, a capacitor, a resistor, or the impedance of a network of them. Two types of external connections are supported:

- Serial connection for high impedance measurement
- Parallel connection for low impedance measurement

(9) Derived Data Point (DDP) Viewer

It is used to display the value of a DDP (Derived Data Point) in a dedicated window with a bigger font size. It is possible to specify the DDP's high-high, high, low, low-low limits for alarming. Different alarm sounds can be configured for different types of alarms. Alarm acknowledgement is supported. Up to 32 DDP viewers can be opened.

DDP viewer can also be used to define and display a UDDP (User Defined Data Points). Various math functions can be used in the mathematical expression defining the UDDP.

A DDP array viewer is also provided. It can be used to display a report consisting of multiple DDPs or DDP arrays. The supported reports include:

- Harmonic Frequencies, RMS, Phases
- Octave Bands, RMS
- Peak Frequencies, RMS, Phases
- Frequency Bands, RMS
- Reverberation / Speech Intelligibility (1/1 Octave)
- Reverberation / Speech Intelligibility (1/3 Octave)

(10) Derived Data Curve (DDC)

It is used to derive a dedicated data curve from one or multiple frames of sampled data and display it in a dedicated graph window. Up to 8 DDC windows can be opened. The supported DDCs include:

- Energy Time Curve (Log-Squared)
- Energy Time Curve (Envelop)
- Energy Time Curve (dBSPL)

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- Impulse Response Schroeder Integration Curve
- Step Response Curve (via Impulse Response Integration)
- Frequency Time Curve (Demodulated)
- X-Y Plot
- Shock Response Spectrum (Absolute Acceleration, Relative Velocity, Relative Displacement, Pseudo Velocity, Equivalent Static Acceleration)
- Frequency Time Curve (Timed)
- RPM Time Curve
- Spectrum Analysis on Selection

The above instruments (1)~(6) are basic instruments and can run simultaneously, while (7) and (8) ride on the top of the basic instruments and require the cooperation of some or all the basic instruments to achieve the specified functions.

A/D and D/A

For a basic configuration, the sound card is used as the signal input and output device and no additional A/D and D/A hardware is required. The sampling frequency depends on the capability of the sound card. The software allows you to select a sampling frequency of up to 768 kHz and a sampling bit resolution of 8 bits, 16 bits or 24 bits, as long as they are supported by the sound card used. Both the sound card MME driver (Windows default) and the sound card ASIO driver (often used in Pro Audio, *ASIO is a trademark and software of Steinberg Media Technologies GmbH*) are supported by the software.

The software is able to interface to other ADC and DAC hardware based on the standard data acquisition software interface specification (vtDAQ® and vtDAO®) developed by Virtins Technology. In this case, the sampling capability is determined by the respective hardware. The ADC and DAC hardware can be selected independently in the software. For example, you can run a DSO (Digital Storage Oscilloscope) hardware for ADC and a sound card for DAC simultaneously.

Triggering

The software supports both software trigger and hardware trigger.

For sound card based data acquisition, it is possible to specify a software trigger condition for collecting a frame of data. A negative or positive trigger delay can be specified so that collecting data can be started before or after the trigger event. The software features a specially designed data acquisition approach which is able to monitor the input signal continuously without missing any trigger event, before a frame of data is collected into the PC memory after the trigger event is found. This makes the software suitable for transient signal recording. Both level triggering and differential triggering are supported.

Calibration

For sound card based data acquisition, the software supports the calibration of input and output channels so that absolute values in engineering units can be used for display, analysis or export. It is able to take into account the change of the sound card

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internal gain settings (e.g. Mic gain, Mic Boost, Line In gain) automatically so that the calibration will not change with these gain settings. It also allows accounting for the external attenuation ratio if an external attenuation circuit is used.

Graph Operation

Zooming and Scrolling is supported in all graphs, enabling you to investigate the fine details of the data. This is very important when a large amount of data are displayed on one screen.

A cursor reader is provided in each graph to show the x and y readings of the actual measurement point. For Spectrum 3D Plot, the cursor reader supports the readout of x, y, t values of the actual measurement point.

Two markers are provided in each graph to get the x and y readings of the actual measurement points nearest (in horizontal direction) to the points of the mouse clicks. The differences in the readings of the two markers are also displayed.

Five chart types are supported: Line, Scatter, Column, Bar, and Step. Line width and colors of the graph are configurable. Options are provided to either display all data points (slow) or only display one data points per vertical raster line (fast).

The data in any graph can be copied into the clipboard as text and later pasted into other software such as Microsoft Excel for further analysis. The image of the graph can be copied into the clipboard as Bitmap image and later pasted into other software such as Microsoft Word.

Reference Curves

Up to five reference curves can be set for each channel in each graph. The reference curve can be configured by either copying the current curve, or loading a properly formatted text file or a previously saved reference file from the hard disk. The reference curves can be defined as high-high, high, low, low-low limits for the real-time curves, and the alarm statuses can be obtained through the respective DDPs.

File Import and Export

The collected frame of data can be saved as a wave file (*.wav) or exported as a text file (*.txt). All analysis results can be exported as text files (*.txt). All graphs can be exported as bitmap files (*.bmp) or printed out directly. A long wave file can be imported frame by frame either manually or automatically.

Wave files with PCM format or properly formatted text files can be imported for analysis. The signal generated by the Signal Generator can be saved as a wave file (*.wav) or a text file (*.txt) for a given duration of up to 1000 seconds as long as the spare space on the hard disk is sufficient. The saved files can be in turn imported for display and analysis or used by other software.

Data Merging and Extraction

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Combining data from individual channels of different wave files and extracting part of data from a wave file are supported.

Save/Load, Lock/Unlock Panel Setting

You can save the current instrument panel setting either as default or as a customized panel setting for later use. You can configure up to 20 most frequently used panel settings in the Hot Panel Setting Toolbar so that these settings can be loaded by just a single mouse click. You can also lock the panel setting so that only authorized users can unlock it and then modify the setting.

Controls/Options Enabling and Disabling

Graphical User Interface items such as menu items, buttons, combo boxes, edit boxes, radio buttons, checkboxes are enabled/disabled based on context, so as to void any mis-understanding and mis-operation.

Displayed Precision for Numerical Values

The precision (decimal places) of the numerical values displayed on the screen are automatically adjusted based on the precision of their sources.

Multilingual User Interface

The software supports Multilingual User Interface under Windows 2000, XP, 2003, Vista, 7, 8, 8.1, 10, 11 and above. The currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

Function Allocation in Different Licnese Levels

A complete Multi-Instrument software package consists of basic and add-on modules with all features in each of them. The basic modules include Oscilloscope, Spectrum Analyzer, Signal Generator, Multimeter, Derived Data Point Viewer, Derived Data Curve, and General Functions. The add-on modules include Spectrum 3D Plot, Data Logger, LCR Meter, Device Test Plan, Vibrometer, and Dedicated Hardware Support.

There are six license levels to access the basic modules: Sound Card Oscilloscope, Sound Card Spectrum Analyzer, Sound Card Signal Generator, Multi-Instrument Lite, Multi-Instrument Standard, and Multi-Instrument Pro. The add-on modules need to be purchased separately. They can only run with Multi-Instrument Lite, Standard, or Pro, except that Vibrometer can only run with Multi-Instrument Standard or Pro. The following table shows the function allocation among different license levels. Please note that a license of Multi-Instrument Full Package contains Multi-Instrument Pro and all add-on modules.

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*Legend: √ - Function available * - Function available in Multi-Instrument Full Package only*

		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument Standard	Multi-Instrument Pro
General Functions							
DAC / ADC Hardware	Sound Card MME	√	√	√	√	√	√
	Sound Card ASIO						√
	Other Hardware				√	√	√
	vtDAQ, vtDAO software development kit	License automatically activated with the presence of the corresponding hardware, e.g. a USB hardkey or a VT DSO.					
File Operation	Load WAV File	√	√	√	√	√	√
	Load TXT File					√	√
	Load WAV File Frame by Frame (fore Long WAV File)					√	√
	Combine WAV Files	√	√	√	√	√	√
	Extract Data and save them into a new WAV File	√	√	√	√	√	√
	Save/Load Panel Setting	√	√	√	√	√	√
Data Export	Copy Text to Clipboard	√	√	√	√	√	√
	Copy BMP to Clipboard	√	√	√	√	√	√
	Print Preview	√	√	√	√	√	√
	Print	√	√	√	√	√	√
	Export as TXT File	√	√	√	√	√	√
	Export as BMP File	√	√	√	√	√	√
Trigger Settings	Trigger Mode	√	√		√	√	√
	Trigger Source	√	√		√	√	√
	Trigger Edge	√	√		√	√	√
	Trigger Level	√	√		√	√	√
	Trigger Delay	√	√		√	√	√
	High Frequency Rejection	√	√		√	√	√
	Noise Rejection	√	√		√	√	√
Sampling Settings	Sampling Rate	√	√	√	√	√	√
	Sampling Channels	√	√	√	√	√	√
	Sampling Bit Resolution	√	√	√	√	√	√
	Record Length	√	√		√	√	√
Calibration	Input	√	√		√	√	√
	Output			√	√	√	√
	Probe	√	√		√	√	√
	Sound Pressure Level	√	√		√	√	√
	F/V Conversion					√	√
	Latency for Sync. Output/Input						√
	Sensor Sensitivity	√	√		√	√	√
	Load Factor for Power Calculation	√	√		√	√	√
Graph Operation	Zoom	√	√	√	√	√	√
	Scroll	√	√	√	√	√	√
	Cursor Reader	√	√	√	√	√	√
	Marker	√	√	√	√	√	√
	Chart Type	√	√	√	√	√	√

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		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument Standard	Multi-Instrument Pro
	Line Width	√	√	√	√	√	√
	Color	√	√	√	√	√	√
	Fast/Slow Display Mode	√	√	√	√	√	√
	Refresh Delay	√	√	√	√	√	√
	Font Size	√	√	√	√	√	√
	Roll Mode					√	√
	Reference Curves & Limits					√	√
Others	Gain Adjustment	√	√	√	√	√	√
	Input Peak Indicator	√	√	√	√	√	√
	Sound Card Selection	√	√	√	√	√	√
	Sampling Parameter Auto Setting	√	√	√	√	√	√
	Multilingual GUIs	√	√	√	√	√	√
	Show/Hide Toolbar	√	√	√	√	√	√
	Lock/Unlock Panel Setting	√	√	√	√	√	√
	Hot Panel Setting Toolbar	√	√	√	√	√	√
	ActiveX Automation Server	√	√	√	√	√	√
	AutoRanging	√	√	√	√	√	√
	AutoScaling	√	√		√	√	√
	Input Channel Operation	√	√		√	√	√
Oscilloscope							
Type	Individual Waveform	√	√	√ (offline)	√	√	√
	Waveform Addition	√	√	√ (offline)	√	√	√
	Waveform Subtraction	√	√	√ (offline)	√	√	√
	Waveform Multiplication	√	√	√ (offline)	√	√	√
	Lissajous Pattern	√	√	√ (offline)	√	√	√
Inter-Frame Processing	Linear Average					√	√
	Exponential Average					√	√
Intra-Frame	Time Delay Removal					√	√
Demodulation (Intra-Frame)	AM					√	√
	FM					√	√
	PM					√	√

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		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument Standard	Multi-Instrument Pro
Digital Filtering (Intra-Frame Processing)	Remove DC					√	√
	Rectification					√	√
	FFT Low Pass					√	√
	FFT High Pass					√	√
	FFT Band Pass					√	√
	FFT Band Stop					√	√
	FFT Frequency Response					√	√
	FIR Low Pass					√	√
	FIR High Pass					√	√
	FIR Band Pass					√	√
	FIR Band Stop					√	√
	FIR Frequency Response					√	√
	IIR Coefficients					√	√
Parameter Measurement	Reverberation / Speech Intelligibility						√
	Discontinuity						√
	Step Response						√
	Echo						*
	Damping Ratio						*
Others	Max, Min, Mean, RMS, Skewness, Kurtosis	√	√	√ (offline)	√	√	√
	Record Mode					√	√
	Persistence Display Mode	√	√		√	√	√
	Equivalent Time Sampling Mode	√	√		√	√	√
	Analog & Digital Signal Mixed Display				√	√	√
	SINC Interpolation	√	√	√	√	√	√
Spectrum Analyzer							
Type	Amplitude Spectrum / Power Spectrum Density / Impedance Spectrum		√		√	√	√
	Phase Spectrum		√		√	√	√
	Auto-correlation (Linear/Circular)		√		√	√	√
	Cross-correlation (Linear/Circular) (Original /Generalized)		√		√	√	√
	Coherence/Non-Coherence						√
	Transfer Function / Impedance Analyzer						√
	Impulse Response						√
Intra-Frame	Frequency Compensation		√		√	√	√
	Frequency Weighting		√		√	√	√
	Remove DC		√		√	√	√

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		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument Standard	Multi-Instrument Pro
	Smoothing via Moving Average (Linear/Octave)		√		√	√	√
Inter-Frame Processing	Peak Hold		√		√	√	√
	Linear Average		√		√	√	√
	Exponential Average		√		√	√	√
	Cross Correlation Average (Cross Power Spectrum Vector Average)		√		√	√	√
Parameter Measurement	THD,THD+N,SNR, SINAD,Noise Level, ENOB		√		√	√	√
	IMD/DIM		√		√	√	√
	Bandwidth		√		√	√	√
	Crosstalk		√		√	√	√
	Harmonics & Phase		√		√	√	√
	Energy in User Defined Frequency Band		√		√	√	√
	Peak Detection, SFDR, TD+N		√		√	√	√
	Wow & Flutter						*
	Sound Loudness						√
	Sound Loudness Level						√
	Sound Sharpness						√
	Sound Articulation Index						√
	Noise Rating & Criterion						√
	Total Non-Coherent Distortion + Noise						√
	GedLee Metric						√
	Sound Intensity						*
FFT	FFT Size 128~32768		√		√	√	√
	FFT Size 65536~4194304						√
	Intra-Frame Average		√		√	√	√
	Window function		√		√	√	√
	Window Overlap		√		√	√	√
Others	Octave Analysis (1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96)		√		√	√	√
	Linear / Log Scale for X and Y		√		√	√	√
	Peak Marker / Label		√		√	√	√
Signal Generator							
Waveform	Sine			√	√	√	√
	Rectangle			√	√	√	√
	Triangle			√	√	√	√
	Saw Tooth			√	√	√	√
	White Noise			√	√	√	√

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		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument Standard	Multi-Instrument Pro
	Pink Noise			√	√	√	√
	MultiTones			√	√	√	√
	Arbitrary Waveform			√	√	√	√
	MLS			√	√	√	√
	DTMF			√	√	√	√
	Musical Scale			√	√	√	√
	Wave File					√	√
	Play Waveform in Oscilloscope	√	√	√	√	√	√
	Cyclic Play Waveform in Oscilloscope	√	√	√	√	√	√
Sweep	Frequency Sweep (Linear/Log)			√	√	√	√
	Amplitude Sweep (Linear/Log)			√	√	√	√
	Forward + Reverse Sweep			√	√	√	√
Burst (Mask)	Normal Phase			√	√	√	√
	Locked Phase			√	√	√	√
	Window-Shaped Burst			√	√	√	√
	On/Off Amplitude Ratio			√	√	√	√
Fade	Fade In			√	√	√	√
	Fade Out			√	√	√	√
Modulation	AM			√	√	√	√
	FM			√	√	√	√
	PM			√	√	√	√
Others	Software Loopback (all channels)			√	√	√	√
	Software Loopback (1 channel)				√	√	√
	Sync. with Oscilloscope						√
	Save as WAV file			√	√	√	√
	Save as TXT file			√	√	√	√
	DDS				√	√	√
	DC Offset				√	√	√
Multimeter							
Type	RMS					√	√
	dBV					√	√
	dBu					√	√
	dB					√	√
	dB(A)					√	√
	dB(Z)					√	√
	dB(C)					√	√
	Frequency Counter				√	√	√
	RPM					√	√
	Counter					√	√
	Duty Cycle					√	√
	Frequency/Voltage					√	√
	Cycle RMS					√	√
	Cycle Mean					√	√
	Pulse Width					√	√

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		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument Standard	Multi-Instrument Pro
	Jitter Statistics						√
Settings	Counter Trigger Hysteresis				√	√	√
	Counter Trigger Level				√	√	√
	Frequency Divider				√	√	√
DDP (Derived Data Point) Viewer							
Function	DDP & UDDP display						√
	HH, H, L, LL Alarm						√
	Set Display Precision						√
	Define UDDP						√
	Alarm Sound						√
	Alarm Acknowledge						√
	Inter-frame Linear / Exponential Average						√
DDP Array Viewer	Harmonic Frequencies, RMS, Phases Report						√
	Octave Bands, RMS Report						√
	Peak Frequencies, RMS, Phases Report						√
	Frequency Bands, RMS Report						√
	Reverberation / Speech Intelligibility Report (1/1 Octave)						√
	Reverberation / Speech Intelligibility (1/3 Octave)						√
Derived Data Curve (DDC)							
Function	Energy Time Curve (Log-Squared)						√
	Energy Time Curve (Envelop)						√
	Energy Time Curve (dBSPL)						√
	Impulse Response Schroeder Integration Curve						√
	Step Response Curve (via Impulse Response Integration)						√
	Frequency Time Curve (Demodulated)						√
	X-Y Plot						√

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		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument Standard	Multi-Instrument Pro
	Shock Response Spectrum						√
	Frequency Time Curve (Timed)						√
	RPM Time Curve						√

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Legend: Blank - Function available if purchased Shaded Blank - Function NOT available in that license level

		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument	Multi-Instrument Pro
Spectrum 3D Plot							
Type	Waterfall Plot (Inter-frame, STFT)						
	Waterfall Plot (Intra-frame, STFT)						
	Waterfall Plot (Intra-frame, CSD)						
	Spectrogram (Inter-frame, STFT)						
	Spectrogram (Intra-frame, STFT)						
	Spectrogram (Intra-frame, CSD)						
Settings	Spectrogram Color Palette						
	Waterfall Color Palette						
	Waterfall Tilt Angle						
	Waterfall / Spectrogram Height						
	Linear / Log Scale for X and Y						
	Number of Spectral Profiles (10~200)						
Others	3D Cursor Reader						
	Octave Analysis (1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96)						
	Spectrogram Smoothing						
Data Logger							
Real Time Logging							
Load Historical Log File							
Three logging methods (Fastest, Time Interval, Update Threshold)							
270 derived data points available for logging							
Up to 8 × 8 = 64 variables can be logged simultaneously							
LCR Meter							
High Impedance Measurement							
Low Impedance Measurement							
Up to 8 X-Y Plots (Linear/Log)							
Device Test Plan							
30 Instructions							
Create/Edit/Lock/Execute/Load/Save a Device Test Plan							
Up to 8 X-Y Plots (Linear/Log)							
Device Test Plan Log							
Automatic Mutli-Step Generation							
User Log In / Out							
Volatile & Non-volatile							

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	Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument	Multi-Instrument Pro
Variables						
Vibrometer						
RMS, Peak/PP, Crest Factor for acceleration, velocity, displacement (in Multimeter)						
Waveform conversion among acceleration, velocity and displacement (in Oscilloscope)						
SI / English units						
Dedicated Hardware Support						
RTX6001 Remote /Local Control						

The Dedicated Hardware Support add-on module is intended for a seamless integration with some third-party hardware devices. The following third-party hardware devices are under this category:

- RTX6001 Audio Analyzer
  Note that without this add-on module, RTX6001 will work as a sound card only and is still supported by Multi-Instrument's generic sound card drivers. With this add-on module, the input and output data will be scaled properly by the software without calibration, and RTX6001 is fully controllable from PC.

1.2 System Requirement

Microsoft Windows XP or greater. Both 32-bit and 64-bit Windows are supported. Screen resolution is recommended to be greater than 1024 × 600 pixels.

For a sound card based data acquisition system, an 8, 16 24 or 32 bit Windows compatible sound card is required.

1.3 Quick Start Guide

1.3.1 Start Multi-Instrument

To start Multi-Instrument, on the Windows desktop, select [Start]>[All Programs]>[Multi-Instrument]>[VIRTINS Multi-Instrument].

If the software license has not been activated and the trial period has not expired, the software will work in trial mode. Under this mode, the software will have the full functionality of the full package of the software except that a text "trial" will be shown in the center of every graph window.

If the software license has not been activated and the trial period has expired, the software will refuse to run.

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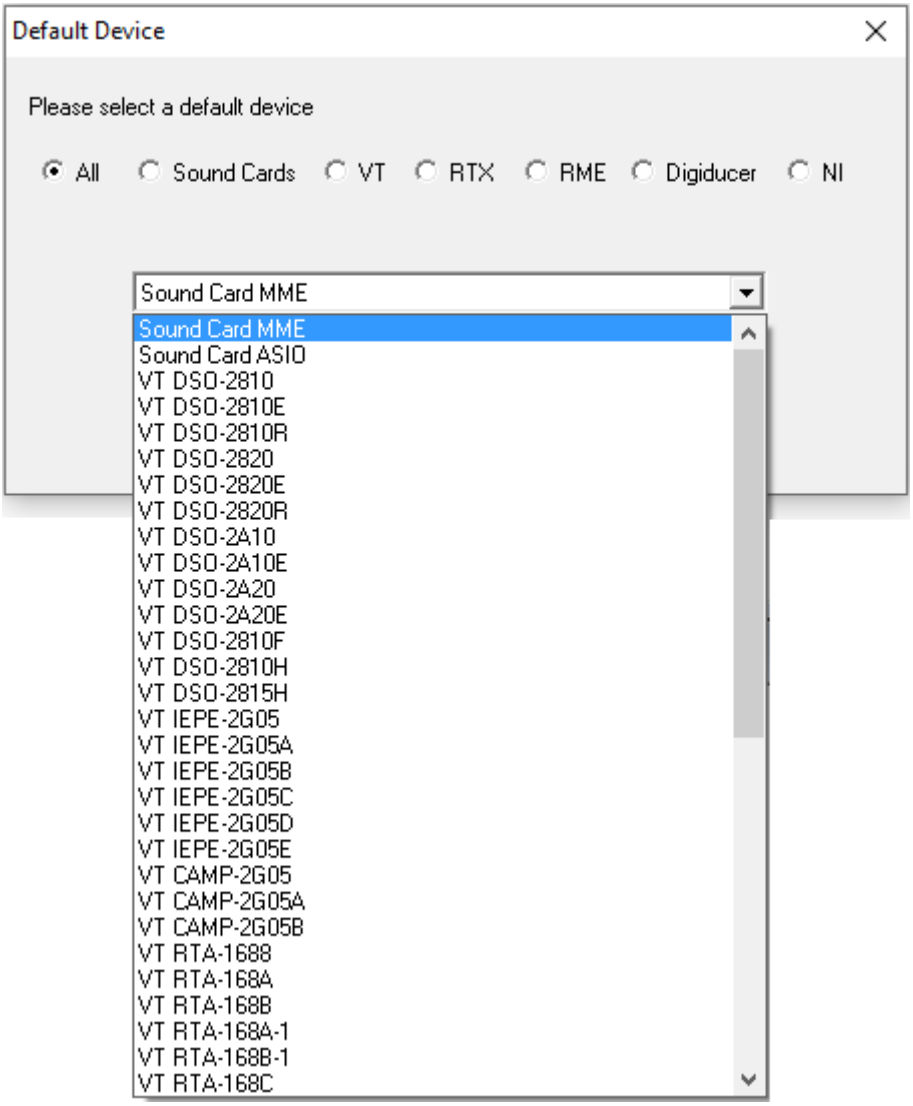
If the software license has been activated, by a softkey (activation code), a USB hardkey, or a VT device, the software will work in licensed mode. Under this mode, the software will have the functionality corresponding to the license level activated.

1.3.1.1 Default Device Selection

If the software is started for the very first time after installation, it will prompt the user to select a default device. The default device can also be changed later via [Setting]>[ADC Device], [Setting]>[DAC Device], and [Setting]>[Configure Hot Panel Setting Toolbar], or simply [Setting]>[Restore to Factory Default]. **However, if [Restore to Factory Default] command is executed, all calibration data entered manually via [Setting]>[Calibration] after software installation will be reset to the default values of the selected product. To avoid the loss of the manually entered calibration data, you can save them as a calibration file first.**



1.3.1.2 Default Skin Selection

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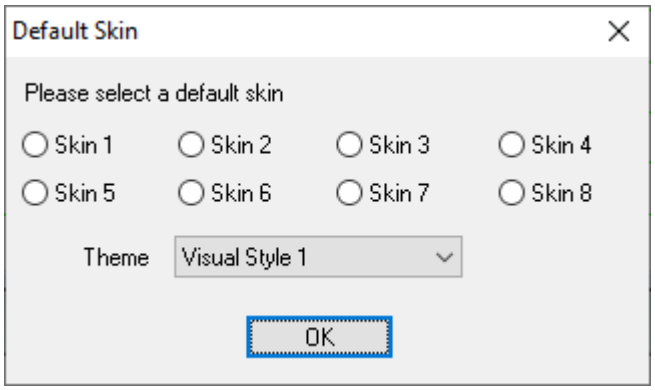
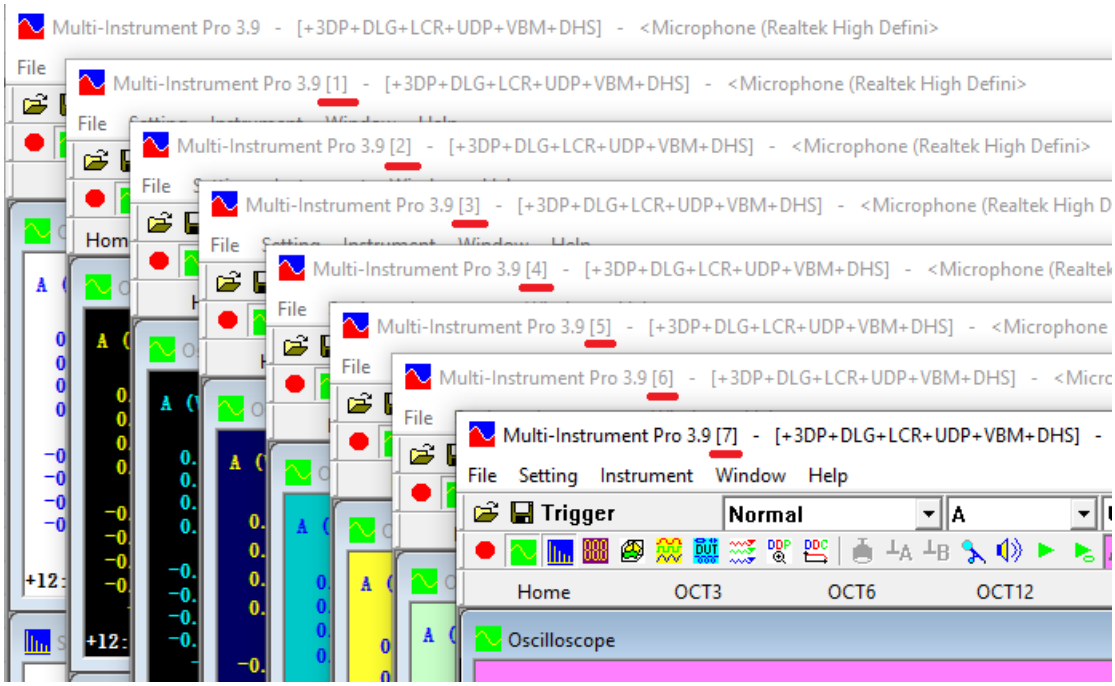
If the software is started for the very first time after installation, it will prompt the user to select a default color scheme (Skin) and theme (visual style). The default skin can also be changed later via [Setting]>[Display].



1.3.1.3 Run Multiple Instances

It is possible to launch multiple instances of the software on the same computer. Each instance has its own default configuration. It is possible to use different ADC/DAC devices or different channels of the same device in different instances. This effectively increases the total number of channels supported. The location of each instance on the screen is memorized by the software.

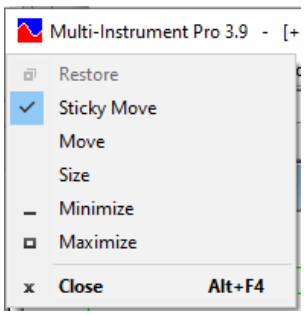
Each instance except the first one has a unique number in the brackets [] in the title bar of the main window to differentiate itself from others (see figure below).



If multiple instances point to the same hardware device (either the same or different channels), the device driver must support multi-client applications, otherwise, clash will occur. Sound card MME drivers and some sound card ASIO drivers support multi-client applications, but VT DSO drivers do not.

The Oscilloscopes and Signal Generators among multiple instances can be synchronized respectively to start and stop.

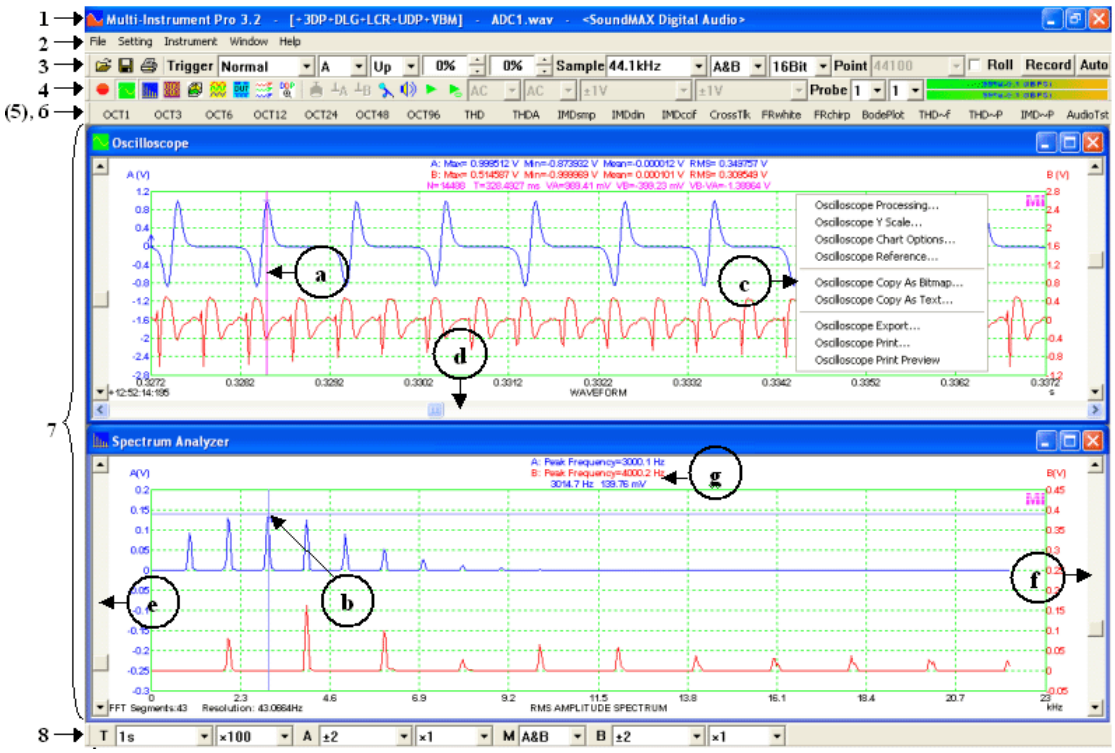
1.3.1.3.1 Sticky Move



"Sticky Move" option can be accessed by clicking the software icon at the upper left corner of the main window. It is ticked by default. With this option selected, all screen components belonging to the same software instance will move together if you drag the title bar of the main window. Otherwise, only the Oscilloscope, Spectrum Analyzer, Spectrum 3D Plot, and Multimeter will move together with the main window.

1.3.2 Screen Layout and Components

The main window of the software is divided into the following eight areas: (1)~(8).



(1) Title bar

It is located at the top of the screen. It contains software title, software version number, add-on modules activated, file name opened (if any), ADC device being used, Panel Setting File name loaded (if any, when the Panel Setting is locked), and instance no.

Depending on the license level activated, the software title can be:

- Sound Card Oscilloscope
- Sound Card Spectrum Analyzer
- Sound Card Signal Generator
- Multi-Instrument Lite
- Multi-Instrument Std (i.e. Standard)
- Multi-Instrument Pro

Depending on the license level activated, the add-on modules activated can be:

- 3DP (i.e. Spectrum 3D Plot)
- DLG (i.e. Data Logger)
- LCR (i.e. LCR Meter)
- UDP (i.e. Device Test Plan)
- VBM (i.e. Vibrometer)
- DHS (i.e. Dedicated Hardware Support)

(2) Menu bar

It is located at just below the title bar. Different instruments may have different menu items in each submenu.

(3) Sampling Parameter Toolbar

It is located just below the menu bar. It contains three parts:

- File Input & Output

- Trigger Parameters
  It contains (from left to right):
  Trigger Mode, Trigger Source, Trigger Edge, Trigger Level, and Trigger Delay, Trigger Frequency Rejection

- Sampling Parameters (for ADC)
  It contains (from left to right):
  Sampling Frequency, Sampling Channels, Sampling Bit Resolution, Record Length per sweep, Roll Mode Checkbox, Record Button, and Sampling Parameter Auto Setting Button.

This toolbar is common for all instruments. Note that the sampling parameters for DAC are located in the Signal Generator panel.

(4) Instrument & Miscellaneous Toolbar

It is located just below the Sampling Parameter Toolbar, consisting of the following two parts:

- Instrument Toolbar (left hand side)
  It contains buttons for (from left to right):
  Oscilloscope Run/Stop, Oscilloscope, Spectrum Analyzer, Multimeter, Spectrum 3D Plot, Signal Generator, Device Test Plan, Data Logger, DDP Viewer, Derived Data Curve (DDC).

- Miscellaneous Toolbar (right hand side)
  It contains (from left to right):
  Invert Input Signal button, Channel A Zeroing button, Channel B Zeroing button, Windows Recording Control, Windows Volume Control, Waveform Play button, Waveform Cyclic Play Button, ADC Channel A Coupling Type, ADC Channel B Coupling Type, ADC Channel A Range, ADC Channel B Range, Probe A Current Switch Position, Probe B Current Switch Position, and Input Peak Level Indicators for Channel A and Channel B.

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This toolbar is common for all instruments.

(5) Long Wave File Navigation Toolbar (Hidden by default)

It is located just below the Instrument and Miscellaneous Toolbar, and is visible only when a WAV file is opened with [File]>[Open Frame by Frame] command. It contains the following controls (from left to right):

Frame Up button, Frame Down button, Frame Auto Scroll button, Current Frame Position Slider, Current Frame Position/Length of the File, Frame Overlap Percentage.

(6) Hot Panel Setting Toolbar

It is a toolbar containing 20 configurable buttons. You can assign each button with a frequently used panel setting file via [Setting]>[Configure Hot Panel Setting Toolbar], so that that panel setting file can be loaded when the button is pressed. You can show or hide this toolbar via [Setting]>[Show Hot Panel Setting Toolbar].

By default, when "Sound Card" is selected as the default device, the 20 buttons are pre-configured as follows:

1. Home: Default setting.

2. OCT3: 1/3 Octave Analysis (Avg. 10)
   Pink noise will be generated by pressing the start button of the Signal Generator. The pink noise will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. A flat curve in the Spectrum Analyzer would indicate a flat magnitude frequency response of the DUT.

3. Polarity: Polarity Test with Crest Factor Check
   0.5 ms half-inverted sawtooth pulse will be generated every 100 ms by pressing the start button of the Signal Generator. The signal will be injected into the DUT (Device Under Test) such as a speaker, and the response of the DUT will be captured and analyzed by the Oscilloscope. The polarity of the DUT can be seen from the direction of the pulse displayed in the oscilloscope. In addition, a DDP viewer is configured to display the measured polarity using its background color.

4. THD: THD,THD+N,SNR,SINAD,Noise Level, ENOB (Avg. 10)
   A 1 kHz sine wave will be generated by pressing the start button of the Signal Generator. The sine wave will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. All the above parameters of the DUT will be measured and displayed. This panel setting uses Kaiser 6 window function in the Spectrum Analyzer to suppress spectral leakage.

5. THDa: THD,THD+N,SNR,SINAD,Noise Level, ENOB (A-Weighted) (Avg. 10)

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Similar to THD, but with A-weighting profile applied to the spectrum.

6. THDnsl: THD,THD+N,SNR,SINAD,Noise Level, ENOB (Avg. 10)-No Spectral Leakage
   A 1 kHz (to be precise, a frequency very close to 1 kHz in order to avoid spectral leakage) sine wave will be generated by pressing the start button of the Signal Generator. The sine wave will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. All the above parameters of the DUT will be measured and displayed. This panel setting should be used only if the same sound card is used for signal input and output. In case different sound cards are used for signal input and output, you should change the Window Function in the Spectrum Analyzer from Rectangle to Kaiser 6.

7. THDres: 1kHz Sine wave vs Its Duplicate with Fundamental attenuated by 179 dB via FIR
   A 1 kHz sine wave will be generated by pressing the start button of the Signal Generator. The sine wave will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. [Setting]>[ADC Device]>"Channel Operation" is set to "A=iA, B=iA" to duplicate the same single-channel response from the DUT into both Ch. A and Ch. B. A band-stop linear-phase FIR filter is applied to Ch. B only to attenuate the 1kHz fundamental by 179 dB, leaving only the residual harmonics and noises in Ch. B for waveform comparison with the original signal in Ch. A. The FIR time delay in Ch. B is removed so that the two waveforms in both channels are time aligned. In case the amplitude of the waveform in Ch. B is too small, right click the Oscilloscope and select "AutoScaling". DDP Array viewer is used to show the harmonic frequencies, amplitudes, phase information. Please note that the phase information will only be available if [FFT Size]<=[Record Length]< 2×[FFT Size] and there is no inter-frame average. Peak Markers are used to indicate the harmonic peaks. They will show the correct peak heights even when spectral leakage occurs.

8. IMDsmp: IMD SMPTE (60 Hz + 7 kHz, 4:1) (Avg. 10)
   A 60 Hz and a 7 kHz sine waves mixed at an amplitude ratio of 4:1 will be generated by pressing the start button of the Signal Generator. The mixed signals will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. The SMPTE IMD value of the DUT will be measured and displayed.

9. IMDdin: IMD DIN (250 Hz + 8 kHz, 4:1) (Avg. 10)
   A 250 Hz and an 8 kHz sine waves mixed at an amplitude ratio of 4:1 will be generated by pressing the start button of the Signal Generator. The mixed signal will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. The DIN IMD value of the DUT will be measured and displayed.

10. IMDccif: IMD CCIF2 (19 kHz + 20 kHz, 1:1) (Avg. 10)
    A 19 kHz and a 20 kHz sine waves mixed at an amplitude ratio of 1:1 will be generated by pressing the start button of the Signal Generator. The mixed signal

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will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. The CCIF2 IMD value of the DUT will be measured and displayed.

11. CrossTlk: Crosstalk A->B, THD, THD+N, SNR, SINAD, ENOB (Avg. 10)
    A 1 kHz sine wave will be generated by pressing the start button of the Signal Generator. The generated signal will be injected into Channel A of the DUT (Device Under Test) and the input of Channel B of the DUT will be grounded. The response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. All the above parameters of the DUT will be measured and displayed.

12. FRwhite: Magnitude Frequency Response (White Noise, Avg. 30)
    White noise will be generated by pressing the start button of the Signal Generator. The white noise will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. The curve in the Spectrum Analyzer indicates the magnitude frequency response of the DUT.

13. FRpwn: Magnitude Frequency Response (Periodic White Noise)
    Periodic white noise will be generated by pressing the start button of the Signal Generator. The periodic white noise will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. The curve in the Spectrum Analyzer indicates the magnitude frequency response of the DUT. One advantage of using periodic white noise is that its spectrum is ideally flat even without averaging.

14. FRswph: Magnitude Frequency Response (Frequency Sweep, Peak Hold forever)
    A frequency linearly swept sine wave (chirp) will be generated by pressing the start button of the Signal Generator. The chirp signal will be injected into the DUT (Device Under Test), and the response of the DUT will be captured by the Oscilloscope and Spectrum Analyzer. The curve in the Spectrum Analyzer indicates the magnitude frequency response of the DUT. Glitches may appear if the sweep time is reduced or FFT size is increased.

15. FRswlog: Magnitude Frequency Response (Frequency Log Sweep+1/96 Octave Analysis)
    A frequency logarithmically swept sine wave (chirp) will be generated by pressing the start button of the Signal Generator. The chirp signal will be injected into the DUT (Device Under Test), and the response of the DUT will be captured by the Oscilloscope and Spectrum Analyzer. The curve in the Spectrum Analyzer indicates the magnitude frequency response of the DUT. The trigger level should be set properly such that the oscilloscope will capture the entire swept sine signal. Logarithmic sweep and octave analysis is used to provide sufficient SNR at lower frequency range. If linear sweep is used, narrow-band FFT analysis should be used instead.

16. BodePlot: Bode Plot (Transfer Function, Frequency Response) (White Noise, Avg. 50)

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White noise will be generated by pressing the start button of the Signal Generator. The generated white noise will be split into two: one injected into the DUT (Device Under Test), and the other injected directly into Channel B of the Oscilloscope. The response of the DUT will be injected into Channel A of the Oscilloscope. The curve in Channel A of the Spectrum Analyzer will show the gain vs frequency plot of the DUT, and the curve in Channel B of the Spectrum Analyzer will show the phase shift vs frequency plot of the DUT.

17. THD~f: THD, THD+N, SNR, Magnitude Response vs Frequency Plot
    Please refer to Section 8.8.3.

18. THD~P: THD, THD+N vs Magnitude, Power Plot
    Please refer to Section 8.8.4.

19. IMD~P: SMPTE IMD vs Magnitude, Power Plot
    Please refer to Section 8.8.5.

20. AudioTst: Automated Audio Parameter Test
    Please refer to Section 8.8.7.

Note: In the above tests, it is assumed that the performance of the test set-up itself (e.g. the sound card that is used as the measurement instrument) is better than that of the DUT (preferably one order better), so that the measurement errors introduced by the test set-up itself can be ignored. The test set-up's specifications can be obtained from the hardware manufacturers. A test set-up with a low noise level, high bit resolution, wide bandwidth, flat frequency response, and low distortion should be used. The parameters of the test set-up can also be obtained by directly looping back its output to its input, and then performing the above tests. Some of the imperfection of the test set-up can be compensated by software, such as the non-flat magnitude frequency response. But some cannot, such as distortion and noises.

Please refer to the following documents for some sound card loop back test examples:

- *EMU Tracker Pre Test Report using Multi-Instrument*
  Download link:
  https://www.virtins.com/doc/D1004/EMU_Tracker_Pre_Report_D1004.pdf

- *M-Audio Mobile TrackerPre Test Report using Multi-Instrument*

  Download link:
  https://www.virtins.com/doc/D1005/M-Audio_MobilePre_Report_D1005.pdf

- *VT XLR-to-USB Pre Test Report using Multi-Instrument*
  Download link:
  https://www.virtins.com/doc/D1006/VT_XLR-to-USB_Pre_Report_D1006.pdf

- *EMU 0204 Test Report using Multi-Instrument*
  Download link:
  https://www.virtins.com/doc/D1007/EMU_0204_Report.pdf

- *Focusrite Scarlett Solo Test Report using Multi-Instrument*

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Download link:
https://www.virtins.com/doc/Focusrite-Scarlett-Solo-Test-Report-using-Multi-Instrument.pdf

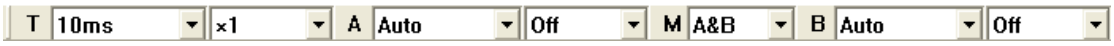
(7) Display area

It is positioned in the middle of the screen. It is used to display various instrument views and panels. Multiple views and panels can be displayed in this area simultaneously. This area is intentionally designed as big as possible in order to accommodate multiple views/panels and display the fine details of the data. Compared with conventional instrument, one distinct advantage of virtual instrument is that it utilizes the computer screen for display, which is much bigger than the screen of a conventional oscilloscope. However, this advantage has not yet been fully taken in many other virtual instrument software applications in the market, as they simply mimic the conventional instrument panel on the computer screen and thus only a small portion of the computer screen is used for the display of the measurement data. In contrast, Multi-Instrument gives priority to the display of the measurement data on the screen and at the same time maintains a user friendly human machine interface (HMI) by using common Windows gadgets.

(8) View Parameter Toolbar

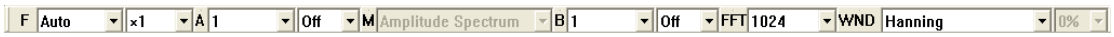
It is located at the bottom of the screen. It is used to adjust those display-related parameters. Each view (e.g. Oscilloscope view, Spectrum Analyzer view, Multimeter view, Spectrum 3D Plot view) has its own View Parameter Toolbar and only the one for the foremost view will be shown, in order to save screen space. Clicking anywhere within an instrument view will bring forward its respective View Parameter Toolbar and make it visible.

- View Parameter Toolbar for Oscilloscope



It contains (from left to right):
T Range, T Multiplier, A Range, A Multiplier, Oscilloscope Type, B Range, B Multiplier.

- View Parameter Toolbar for Spectrum Analyzer



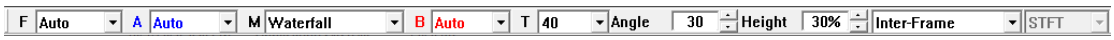
It contains (from left to right):
Horizontal Axis Range, Horizontal Axis Multiplier, A Range, A Multiplier, Spectrum Analyzer Type, B Range, B Multiplier, FFT Size, Window Function, Window Overlap Percentage.

- View Parameter Toolbar for Multimeter

It contains (from left to right):

Display/Hide A, Counter A Trigger Level, Counter A Trigger Hysteresis, A Divider, Multimeter Type, Display/Hide B, Counter B Trigger Level, Counter B Trigger Hysteresis, B Divider.

- View Parameter Toolbar for Spectrum 3D Plot



It contains (from left to right):
F Range, A Range, Spectrum 3D Plot Type, B Range, T Range, Tilt Angle of T, Height Percentage of Y, Processing Mode, Analyzing Mode.

(a) Cursor Reader (left mouse click)
Please refer to the section "Cursor Reader and Markers" in the next chapter for details.

(b) Marker (double left mouse click, with or without CTRL or SHIFT)
Please refer to the section "Cursor Reader and Markers" in the next chapter for details.

(c) Context Menu (right mouse click)
Please refer to the section "Context Menu" in the following chapters for details.

(d) Horizontal Scrollbar (put the mouse cursor just below the horizontal axis till it becomes a magnifying glass, then left click to zoom in, right click to zoom out, and move the scroll box to scroll horizontally. Zoom in/out can also performed using the mouse wheel.)

Please refer to the sections about the multiplier for horizontal axis in the following chapters for details.

(e) Channel A Scrollbar (put the mouse cursor on the left side of A axis till it becomes a magnifying glass, then left click to zoom in, right click to zoom out, and move the scroll box to scroll vertically for Channel A. Zoom in/out can also performed using SHIFT + mouse wheel.)

Please refer to the section "Channel A Multiplier" in the following chapters for details.

(f) Channel B Scrollbar (put the mouse cursor on the right side of A axis till it becomes a magnifying glass, then left click to zoom in, right click to zoom out, and move the scroll box to scroll vertically for Channel B. Zoom in/out can also performed using CTRL + mouse wheel.)

Please refer to the section "Channel B Multiplier" in the following chapters for details.

(g) Text display area of a graph window
It displays the DDP (Derived Data Point) values of that graph window. When the cursor reader / marker is shown, it will also display the cursor / marker readings.

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1.3.3 Change ADC/DAC device

You can change the ADC device being used via [Setting]>[ADC Device]>[Device Model].

You can change the DAC device being used via [Setting]>[DAC Device]>[Device Model].

1.3.4 Basic operations

To get familiar with the basic operation of the software quickly, it is recommended to use your computer's sound card with a built-in or external microphone. You can go to [Setting]>[ADC Device], and choose "Sound Card MME" as the Device Model, and choose a sound card (for Windows before Vista) or an input source of a sound card (for Windows after Vista) in the Device No. field. You can adjust the microphone gain of your sound card via the Recording Control under Windows Control Panel. You can also access the Recording Control via the Recording Control button in the Instrument & Miscellaneous Toolbar.

Now, if you start the oscilloscope and speak before the microphone, you should see the waveform and spectra of your voice. To get real time display of your voice, it is recommended to set the sweep time of the oscilloscope to be 5 ms ~100 ms. To change the sweep time, click anywhere within the oscilloscope window and change the sweep time at the bottom left corner. You can change the trigger parameters and sampling parameters in the sampling parameter toolbar and observe their effects. The trigger level and trigger delay can also be changed by dragging the respective markers along A axis and the horizontal axis at the top in the Oscilloscope window.

If the oscilloscope sweep time is too long (e.g. greater than 1s), the screen update will become slow, then you can tick the "Roll" checkbox at the upper right corner of the screen to get real time but partial update of the screen.

You can record your voice continuously by stopping the oscilloscope, uncheck the "Roll" checkbox if it is checked, and then press the "Record" button at the upper right corner of the screen. Press the "Record" button again to stop recording, and the recorded WAV file will be opened automatically. To get out of the Record Mode, click anywhere within the oscilloscope window and change the sweep time at the bottom left corner of the screen from "Record" to something else.

To get familiar with the operation of the Signal Generator, it is recommended to use your computer's sound card with a speaker. You can go to [Setting]>[DAC Device], and choose "Sound Card MME" as the Device Model, and choose a sound card (for Windows before Vista) or an output destination of a sound card (for Windows after Vista) in the Device No. field. You can adjust the volume of the sound card via the Volume Control under Windows Control Panel. You can also access the Volume Control via the Volume Control button in the Instrument & Miscellaneous Toolbar.

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Clicking the Signal Generator button in the Instrument and Miscellaneous Toolbar will open the Signal Generator panel. The basic operation of the Signal Generator is straight forward. Please refer to Chapter Signal Generator for details.

Whenever you want to revert back to the default panel setting, stop the oscilloscope and go to [File]>[New], or simply press “Home” in the Hot Panel Setting Toolbar.

If you want to see some more basic tests with the software, please refer to: *Basic Oscilloscope, Spectrum Analyzer, Multimeter and Signal Generator Functional Tests using Multi-Instrument and Pocket Multi-Instrument with Sound Cards*

Download link:
https://www.virtins.com/doc/D1001/Basic_Oscilloscope_Spectrum_Analyzer_Multimeter_and_Signal_Generator_Functional_Tests_using_Multi-Instrument_and_Pocket_Multi-Instrument_with_Sound_Cards_D1001.pdf

1.4 Input & Output Connection for Sound Card Based Systems

For sound card based systems, signals to be tested should be connected to either the "MIC" jack or the "Line in" jack, and the generated signals will be output via the "Speaker" jack or the "Line out" jack.

Typically, the "MIC" jack of a sound card has an input impedance in the range of 600 Ω ~ 50 kΩ (card dependent). Its ADC full-scale input voltage (1/2 Vpp) is in the range of 1 mV ~ 500 mV (card dependent), and is adjustable through the Recording Control under Windows Control Panel or hardware gain control (if any) provided by the sound card. Most built-in sound cards have only one mic input and the same mic signal is split and fed into the two input channels.

The "Line In" jack of a sound card has an input impedance typically around 1 kΩ ~ 50 kΩ (card dependent). The ADC full-scale input voltage (1/2 Vpp) of this connection is in the range of 500 mV ~ 2 V (card dependent), and is adjustable through the Recording Control under Windows Control Panel or hardware gain control (if any) provided by the sound card. Generally, the "Line In" should be used as the primary input connection as it offers better Signal-to-Noise Ratio (SNR) and bandwidth.

The "Line Out" connection of a sound card has an output impedance typically in the range of 20 Ω ~ 500 Ω (card dependent) and can output signal up to 2 V typically (card dependent). It has better SNR than the "Speaker" connection.

The "Speaker" connection of a sound card has a typical output impedance of 4Ω ~ 8 Ω (card dependent) and output power of 2 W (card dependent). The headphone connection of a sound card has a typical output impedance of 4Ω ~ 100 Ω and output power of 100 mW.

For input connection, the simplest way is to directly connect the signal under test to the sound card "Line In" or "MIC" jack (see the figure below). However, this kind of connection requires the tester to be extremely careful to ensure the input signal is within the allowable range before connecting. Otherwise the sound card or even the

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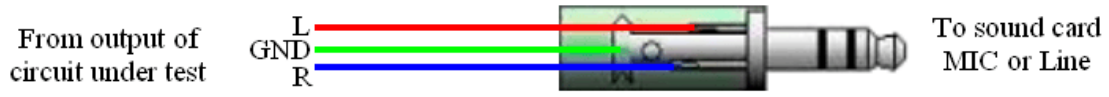
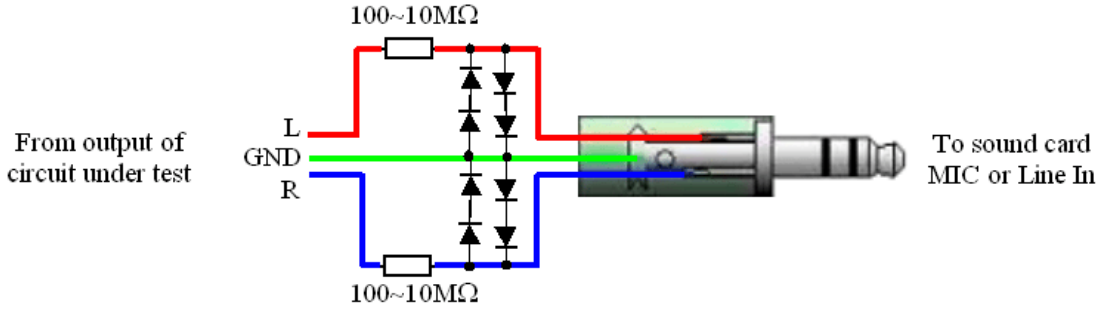
PC may be damaged. The maximum allowable input voltage is about 3V (card dependent).



Simplest input connection

In order to prevent the sound card from excessive input voltage, the following limiter circuit can be added. The two Silicon diodes will clamp the input voltage at about 2 × 0.65 = 1.3 (V). If the sound card ADC full scale is affected, one more Silicon diode can be added in series to clamp the input voltage at about 3 × 0.65 = 1.95 (V) instead. The protection is limited to ± 50 V maximum (also depending on the resister's value and maximum allowable current, and the diode's maximum allowable current). If the amplitude of the signal to be measured exceeds the allowable range of the sound card, it must be attenuated before connecting. The simplest way to attenuate the input signal is to add a resistor in series to the sound card's input. This resistor and the sound card's input impedance form a voltage divider, the higher the resistor's value, the higher the attenuation ratio, and the higher the input impedance of this measurement circuit.



Input connection with input protection

In order to get good measurement result, the output impedance of the circuit under test must be lower than the input impedance of the sound card's measurement circuit. Otherwise the signal cannot be properly transmitted from the circuit to the sound card. If necessary, you can add a pre-amplifier before the sound card's input. The pre-amplifier is responsible for impedance conversion, signal amplification or attenuation, and input protection.

The following figure shows the output connection for the Signal Generator. The resistor is used to prevent accidental short circuit of the output. It can be omitted if you are careful enough. As the output impedance of a sound card is very low, there should not be any impedance matching problem when connected to external circuits.

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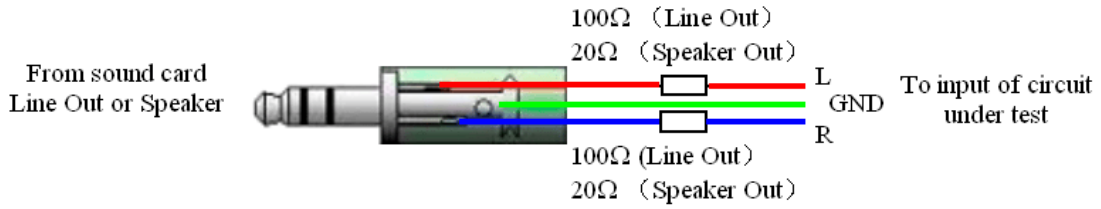

Output connection with short circuit protection

It should be noted that for many sound cards (typically the internal sound card of a desktop PC or a laptop PC with built-in AC power supply adapter), the ground line of input and output is connected to the mains earth. This is not a problem if the circuit under test is floating (i.e. isolated from the mains earth). Otherwise, you must make sure that the ground line is connected to a point on the circuit that is also at earth potential.

The above mentioned connection circuits and pre-amplifiers are NOT supplied with the software unless otherwise stated explicitly during purchasing.

For non-sound-card based systems, please refer to the respective hardware manual for the input & output connection.

1.5 Specifications

1.5.1 Oscilloscope

1 ADC Hardware Related Specifications

Sound Card based systems:

1) Sweep Time: 100 μs~500 s (computer memory dependent).

2) Bandwidth: 1 Hz - 200 kHz (sound card dependent).

3) Maximum Allowable Input Voltage (if connected directly): about 3 V (sound card dependent).

4) Selectable sampling frequency (up to 768kHz), sampling bit resolution (8, 16, 24 or 32 bits) and sampling channels (one or two) (sound card dependent). 32-bit float format is supported as well.

5) Four trigger modes: Auto, Normal, Single and Slow.

6) Support triggering by rising edge, falling edge, or both at a specified trigger level or for a specified amount of change, in the selected input channel. Support trigger frequency rejection.

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7) Support pre-triggering and post triggering from 0 to 100% of the specified record length.

8) Continuously monitor the input signal such that no trigger event is missed before data collection

9) Support calibration of the input channels with the input gain setting automatically being taken into account. Re-calibration is not necessary even if the gain setting changes.

Non Sound Card based systems:

Please refer to the respective hardware manual.

2 Can be used as a transient signal recorder to capture and store raw data continuously into a sequence of WAV files in the hard disk.

3 Five view types: Real time waveform of Channel A and Channel B, Real time waveform of Channel A + Channel B, Real time waveform of Channel A - Channel B, Real time waveform of Channel A × Channel B, Real time Lissajous Pattern for Channel A and Channel B.

4 Independent X axis and Y axis zooming and scrolling.

5 WAV files and properly formatted TXT files can be imported for display and analysis.

6 Collected measurement data can be saved as WAV files or exported as TXT files.

7 Data curve can be print-previewed, printed out directly or saved as BMP files.

8 Support adding notes to the measurement data.

9 Fast display refresh rate: about 50 frames per second (tested with a sound card under Windows XP SP2 on IBM ThinkPad R51 Laptop PC with Intel Pentium M processor 1.60 GHz, with sweep time=10 ms and FFT size=1024 and both the Oscilloscope and the Spectrum Analyzer running under "Auto" trigger mode). Thus data are displayed and analyzed in "true" real time.

10 The colors of display, font size, screen refresh rate are configurable.

11 Number of points (samples) to be collected per sweep can be fine tuned at one point's resolution.

12 Display the Maximum, Minimum, Mean, RMS, Skewness and Kurtosis values of the data per sweep. Therefore it can be used as a voltmeter.

13 Support one cursor reader and two markers which stick to the measurement data.

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14 Five chart types: Line, Scatter, Column, Bar, and Step. Line width is adjustable. Support SINC interpolation between samples.

15 Support combining data from individual channels of different wave files and extracting part of data from a wave file.

16 Support normal and inverted display of a waveform.

17 Up to five reference curves can be set for each channel. The reference curve can be configured by either copying the current curve, or loading a properly formatted text file or a previously saved reference file from the hard disk. Reference curves can be assigned as High-High, High, Low, Low-Low limits.

18 Support digital filtering (intra-frame processing) such as low pass, high pass, band pass, band stop and arbitrary. The filter class can be FFT, FIR or IIR. Support inter-frame processing including linear and exponential average. Support demodulation (AM, FM, PM), DC removal and rectification. Support removal of time delay between two channels. Support time domain parameter measurements including reverberation and speech intelligibility, waveform discontinuity, step response, echo, and damping ratio.

19 Support loading long WAV file frame by frame, either manually or automatically.

20 Each frame of data is time stamped with an accuracy of 1 ms.

21 The data in the graph can be copied into the clipboard as text and later pasted into other software such as Microsoft Excel for further analysis. The image of the graph can be copied into the clipboard as Bitmap image and later pasted into other software such as Microsoft Word.

22 The waveform displayed in the Oscilloscope can be played or cyclically played via the default computer sound card, if the sampling rate and bit resolution of the waveform is compatible with that sound card. Otherwise, it is possible to adjust the replay sampling rate so that the signal is pitch shifted into the audio frequency range.

23 The screen display can work in “Roll” mode.

24 Support waveform conversion among acceleration, velocity and displacement. Support both SI and English unit systems.

25 Support engineering unit conversion.

26 Support auto setting of sampling parameters such as sampling frequency, sweep time and full-scale ADC range. Support Auto Ranging and Auto Scaling.

27 Support digital persistence (phosphorescent, rainbow) mode and equivalent time sampling mode.

28 Support mixed signal (analog and digital) display.

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29 Supports Multilingual User Interface under Windows XP, Vista, 7, 8, 8.1, 10, 11 and above. Currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

1.5.2 Spectrum Analyzer

1 Seven view types: Real time Amplitude Spectrum, Real time Phase Spectrum, Real time Auto Correlation Function, Real time Cross Correlation Function, Real time Coherence/Non-Coherence Function, Real time Transfer Function / Impedance Analyzer, Real time Impulse Response.

2 Independent X axis and Y axis zooming and scrolling.

3 In Amplitude Spectrum, Y axis supports relative modes in linear and dBr scale, and absolute mode in RMS voltage, dBV, dBu, dB, dBFS scale. X axis supports linear, logarithmic and octave scale (1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96). Support amplitude / power spectrum density display. Y axis can be converted to impedance display.

4 Analysis results can be exported as TXT files.

5 Data curves can be print-previewed, printed out directly or saved as BMP files.

6 Fast display refresh rate: about 50 frames per second (tested with a sound card under Windows XP SP2 on IBM ThinkPad R51 Laptop PC with Intel Pentium M processor 1.60 GHz, with sweep time=10 ms and FFT size=1024 and both the Oscilloscope and the Spectrum Analyzer running under "Auto" mode). Thus data are displayed and analyzed in "true" real time.

7 The colors of display, font size, screen refresh rate are configurable.

8 FFT size can be adjusted from 128 to 4194304 points.

9 Allow record length to be different from FFT size. If the FFT size is greater than the record length, then zero(s) will be added at the end of the actual measurement data during FFT computation. If the FFT size is less than the record length, then the measurement data will be split into different segments with the size of each segment equal to the FFT size. Segment overlap percentage can be selected in the range of 0%~99.9%. The final result will be obtained by averaging the FFT results from all segments.

10 Support 69 window functions: Rectangle, Triangle (or Fejer), Hann, Hamming, Blackman, Exact Blackman, Blackman Harris, Blackman Nuttall, Flat Top, Exponential, Gaussian, Welch (or Riesz), Cosine, Riemann (or Lanczos), Parzen, Tukey, Bohman, Poisson, Hann-Poisson, Cauchy, Bartlett-Hann, Kaiser, etc.

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11 Display peak frequency with a sub-FFT-bin-size accuracy in Amplitude Spectrum display, second peak time delay and corresponding coefficient in Auto Correlation Function display, peak time delay and corresponding coefficient in Cross Correlation Function display, peak frequency and corresponding coefficient in Coherence Function display, peak frequency and corresponding gain and phase in Transfer Function display, peak time and corresponding value in Impulse Response display.

12 Allow the measurement of Total Harmonic Distortion (THD), THD+Noise (THD+N), Signal in Noise and Distortion (SINAD), Signal-to-Noise Ratio (SNR), Noise Level (NL), Total Non-Coherent Distortion+Noise (TNCD), GedLee Metric, and Sound Intensity in a specified frequency range.

13 Allow the measurement of IMD-SMPTE/DIN, IMD-CCIF2, IMD-CCIF3, DIM, Crosstalk, Bandwidth (-3dB), Harmonics, Energy in user defined frequency bands, Peaks, Total Distortion+Noise (TD+N), Wow & Flutter, sound loudness, loudness level, sharpness & articulation index, noise rating & noise criterion.

14 In THD measurement and harmonics analysis, the fundamental frequency can be detected automatically based on the peak frequency or another variable such as the RPM reading from a tachometer (e.g. for the purpose of order tracking). It can also be specified manually.

15 Support one cursor reader and two markers which stick to the measurement data. Support peak markers.

16 Intra-frame processing includes: Remove DC Component, Frequency Compensation, Frequency Weighting (flat, A weighting, B weighting, C weighting, ITU-R 468 weighting), and Smooth via Moving Average (linear or octave). Frequency compensation is achieved via loading a user configurable text-based Frequency Compensation File (*.fcf).

17 Four inter-frame processing methods: None, Peak Hold, Linear Average, Exponential Average, Cross Power Spectrum Vector Average (“Cross Correlation Average”). The number of frames (2~200, forever) for peak hold or linear averaging can be specified. The process can be reset during runtime if “forever” is chosen.

18 Five chart types: Line, Scatter, Column, Bar, and Step. Line width is adjustable.

19 Up to five reference curves can be set for each channel. The reference curve can be configured by either copying the current curve, or loading a properly formatted text file or a previously saved reference file from the hard disk. Reference curves can be assigned as High-High, High, Low, Low-Low limits.

20 The data in the graph can be copied into the clipboard as text and later pasted into other software such as Microsoft Excel for further analysis. The image of the graph can be copied into the clipboard as Bitmap image and later pasted into other software such as Microsoft Word.

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21 Supports Multilingual User Interface under Windows XP, Vista, 7, 8, 8.1, 10, 11 and above. Currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

1.5.3 Signal Generator (Sweep/Arbitrary/Function/Noise/Burst Generator)

1 DAC Hardware Related Specifications

Sound Card Based Systems:

1) Bandwidth: 1 Hz - 200 kHz (sound card dependent).

2) Maximum Output Voltage: about 2 V (sound card dependent).

3) Selectable sampling frequency (up to 768 kHz), sampling bit resolution (8, 16, 24 or 32 bits) and sampling channels (one or two) (sound card dependent).

4) Support calibration of the output channels.

Non Sound Card Based Systems:

Please refer to the respective hardware manual.

2 Support predefined waveforms: Sine, Rectangle, Triangle and Saw Tooth. The duty cycle for a rectangle wave is adjustable.

3 Support non-repetitive pink noise and white noise generation. It is possible to generate identical or different noises across channels.

4 Support multi-tones generation. A Multi-Tones is a combination of predefined waveforms with different amplitudes, frequencies, and phases. Pink noise and white noise with specified amplitude can also be added into the multi-tones. Maximum 200 tones can be combined in each channel. In addition to the manual configuration method, automatic configuration method is provided to generate multitones that are aligned to FFT bands or fractional octave bands within a specified frequency range. When the multitones are aligned to FFT bands, it is possible to specify the signal's frequency response: white noise (flat), pink noise (1/f) or arbitrary (a frequency response file *.frf).

5 Support arbitrary waveform generation through user defined waveform library. A waveform library is a TXT file containing the coordinates of each point in one cycle of the waveform. There is no limit as to how many points can be used to define a waveform. Two types of libraries are supported: Waveform Library and Bit Perfect Library.

6 Support Maximum Length Sequence (MLS) generation. The length can be selected from 127 to 16777215.

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7 Support Dual Tone Multi-Frequency (DTMF) generation.

8 Support musical scale tone generation.

9 The predefined waveform, multi-tones and user-defined waveform can be generated at a frequency that sweeps linearly or logarithmically within a specified frequency range and time duration. Option is provided to allow a reverse sweep after a forward sweep.

10 The predefined waveform, white noise, pink noise, multi-tones, user-defined waveform and MLS can be generated at an amplitude that sweeps linearly or logarithmically within a specified amplitude range and time duration. Option is provided to allow a reverse sweep after a forward sweep.

11 Allow specifying the phase difference between the two channels if the output signals have the same frequency.

12 The amplitude of the output signal is adjustable. Support Auto Ranging.

13 The generated signal can be saved as a WAV file or a TXT file for a duration of up to 1000 s.

14 The output signal can be looped back partially (i.e. only one channel while the other channel can be used for field input) or fully (i.e. both channels), via the software itself, to the input of the oscilloscope for display and analysis in real time. Synchronized operation between the Signal Generator and the Oscilloscope with a timing accuracy in the same order of the sampling frequency is also supported. It is possible to specify when to start the Oscilloscope after the Signal Generator is started.

15 It is possible to add a mask with specified periodic on/off timing to the signal to be output so that a burst-type signal can be generated. You can choose whether to phase-lock each burst. The mask can be specified as a window function in order to shape the envelop of the output signal. It is also possible to specify the off/on amplitude ratio.

16 Support fade in/out and modulation (AM, FM, PM) of the output signal.

17 Support DDS mode for those devices that supports DDS mode.

18 Support DC offset adjustment for those devices that support it.

19 Supports Multilingual User Interface under Windows XP, Vista, 7, 8, 8.1, 10, 11 and above. Currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

1.5.4 Multimeter

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1 Display the RMS voltage value of the current frame of data in Vrms, dBV or dBu.

2 Display the sound pressure level of the current frame of data in dB, dB(A), dB(Z) or dB(C).

3 Display frequency (via the Frequency Counter), RPM (Revolutions Per Minute), total counts (via the Counter), duty cycle, F/V voltage (via the Frequency Voltage Converter), Cycle RMS, Cycle Mean, Pulse Width for the current frame of data. You are allowed to configure the counter trigger level and the counter trigger hysteresis in order to rectify the analog signal to rectangular pulses before these analyses. Jitter statistics including peak(%), max. positive (%), max. negative (%), normalized standard deviation (%) are also measured.

4 A frequency divider can be configured for each channel for the Frequency Counter, RPM meter, Counter, F/V converter.

5 Display the RMS, Peak/PP, Crest Factor values for acceleration, velocity, displacement if acceleration, velocity or displacement sensors are used.

6 Supports Multilingual User Interface under Windows XP, Vista, 7, 8, 8.1, 10, 11 and above. Currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

1.5.5 Data Logger

1 Provide long time data logging function for 254 Derived Data Points (DDP) and 16 User Defined Data Point (UDDP), including RMS value, Peak Frequency, Sound Pressure Level, RPM, THD, etc.

2 Up to eight data logger windows can be opened and each window can trace up to 8 variables. You can configure which derived variables to be logged.

3 Color of each trace can be configured.

4 The range of Y axis can be configured. Y axis can be displayed in linear or logarithmic scale.

5 X axis is always a time axis with an accuracy of 1ms. The span of X axis can be configured. The screen automatically scrolls as new data are continuously fed into the right of the window.

6 Data are logged in text format. Each log file contains a maximum 32767 (configurable) lines of data. The file name reflects the time stamp of that file. Log files can be reloaded into the data logger window for display.

7 Three logging methods: Fastest (i.e. whenever new data are available), Time Interval (i.e. whenever new data are available and the specified time duration has elapsed since the last update), Update Threshold (i.e. whenever new data are

available and the change compared with the last update exceeds the specified update threshold).

8 The data in the graph can be copied into the clipboard as text and later pasted into other software such as Microsoft Excel for further analysis. The image of the graph can be copied into the clipboard as Bitmap image and later pasted into other software such as Microsoft Word.

9 Recorded data can be exported as TXT files.

10 Data curves can be printed out directly or saved as BMP files.

11 Supports Multilingual User Interface under Windows XP, Vista, 7, 8, 8.1, 10, 11 and above. Currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

1.5.6 Spectrum 3D Plot

1 Support Waterfall Plot and Spectrogram. It can be generated by multiple frames of data (Inter-frame Mode) or one single frame of data (Intra-frame mode). The former supports Short Time Fourier Transform (STFT) only while the latter supports both STFT and Cumulative Spectral Decay (CSD).

2 For Waterfall Plot, the tilt angle of T axis and the height percentage of Y axis are adjustable. Six color palettes are available: No Color, Rainbow, Bluish, Yellowish, Grayscale, Inverted Grayscale.

3 For Spectrogram, five color palettes are available: Rainbow, Bluish, Yellowish, Grayscale, Inverted Grayscale. A smoothing option is provided.

4 X axis can be displayed in linear, logarithmic, or 1/1 ~ 1/96 octave scale.

5 Y axis is displayed in absolute mode in RMS Voltage, dBV, dBu, dB or dBFS.

6 Number of spectral profiles can be set from 10 to 200.

7 Spectral profiles are time stamped with an accuracy of 1 ms.

8 A 3D cursor reader can be used to display the X, Y, T readings of an actual measurement point. The X-Y profile at that point is highlighted and also displayed in a separate X-Y plot.

9 The data in the graph can be copied into the clipboard as text and later pasted into other software such as Microsoft Excel for further analysis. The image of the graph can be copied into the clipboard as Bitmap image and later pasted into other software such as Microsoft Word.

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10 Analysis results can be exported as TXT files. The 3D data can be sliced by either time or frequency.

11 Data curves can be print-previewed, printed out directly or saved as BMP files.

12 Independent X axis and Y axis zooming.

13 Supports Multilingual User Interface under Windows XP, Vista, 7, 8, 8.1, 10, 11 and above. Currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

1.5.7 Device Test Plan

1 Provides a mechanism for you to configure and conduct your own device test steps. It takes the advantage of the sound card's (or other ADC/DAC hardware's) capability of simultaneous input & output, to generate a stimulus to the Device Under Test (DUT) and acquire the response from that device at the same time. Different stimuli can be generated and the response can be analyzed in different ways.

2 Supports 30 instructions with corresponding parameters. Support using variables in scripts.

3 Parameters to be tested can be selected from 254 DDPs and 16 UDDPs, including RMS value, Peak Frequency, Sound Pressure Level, RPM, Gain, THD, etc.

4 Test results (e.g. Gain vs Frequency, Phase vs Frequency, etc.) can be plotted in up to 8 X-Y plots and reported in one textual log window called Device Test Plan Log in real time. The data in X-Y Plots can be exported as TXT files or copied into the clipboard as text and later pasted into other software such as Microsoft Excel for further analysis. The graph of the X-Y plot can be copied into the clipboard as Bitmap image and later pasted into other software such as Microsoft Word. The graphs can also be printed out directly or exported as BMP files. The data in the Device Test Plan Log can be saved as a TXT file.

5 Support batch file processing and batch signal event capturing and storing.

6 A device test plan can be created, edited, modified, saved, locked, reloaded, and executed.

7 There are two types of device test plans: locked and unlocked. A locked device test plan cannot be modified within the software after it has been created.

8 Support Pass/Fail check. Support connection with external systems through serial communication.

9 Supports Multilingual User Interface under Windows XP, Vista, 7, 8, 8.1, 10, 11 and above. Currently supported languages are English, French, German, Italian,

Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

1.5.8 LCR Meter

1 Can measure the value of an inductor, a capacitor, a resistor, or the impedance of a network of them in a wide range and display the result in a big font.

2 Two types of external connections are supported: serial connection for high impedance measurement, and parallel connection for low impedance measurement. Serial connection uses the sound card's input impedance as reference, while parallel connection uses an external resistor of a relatively small value as reference.

3 Support the calibration of the sound card input impedance which may vary with frequency.

4 The value of the external reference resistor (if any) can be entered.

5 The measurement range is displayed and updated when the relevant settings change.

6 The LCR Meter is a special Device Test Plan with built-in LCR measurement algorithm. The default LCR test plan uses 1 kHz sine wave as the test tone. However, you can configure your own LCR test plans (e.g. with different test tones) if necessary.

7 Same as Device Test Plan, the test results can be plotted in up to 8 X-Y plots. The variables for X and Y axes can be configured.

8 Supports Multilingual User Interface under Windows XP, Vista, 7, 8, 8.1, 10, 11 and above. Currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean.

1.5.9 Derived Data Point (DDP) Viewer

1 Display the value of a Derived Data Point (DDP) or User Defined Data Point (UDDP) in a standalone window. Maximum 32 windows can be opened.

2 Allow High-High, High, Low, Low-Low alarming. Alarm sounds can be configured. Alarm acknowledgement is supported.

3 Displayed precision can be specified.

4 Allow the user to define a UDDP using a mathematical expression. Various math functions are supported.

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5 Support inter-frame processing including linear and exponential averaging.

6 Support DDP array viewer which can be used to generate the following reports:

- Harmonic Frequencies, RMS, Phases
- Octave Bands, RMS
- Peak Frequencies, RMS, Phases
- Frequency Bands, RMS
- Reverberation / Speech Intelligibility (1/1 Octave)
- Reverberation / Speech Intelligibility (1/3 Octave)

1.5.10 Derived Data Curve (DDC)

1 Derive a dedicated data curve from a frame of sampled data and display it in a standalone graph window. Maximum 8 DDC windows can be opened.

2 The DDC window is similar to the X-Y Plot in Device Test Plan except that its context menu contains a [DDC Processing] menu item which is used to select a predefined DDC and configure its relevant parameters. The supported DDCs are:

- Energy Time Curve (Log-Squared)
- Energy Time Curve (Envelop)
- Energy Time Curve (dBSPL)
- Impulse Response Schroeder Integration Curve
- Step Response Curve (via Impulse Response Integration)
- Frequency Time Curve (Demodulated)
- X-Y Plot
- Shock Response Spectrum (Absolute Acceleration, Relative Velocity, Relative Displacement, Pseudo Velocity, Equivalent Static Acceleration)
- Frequency Time Curve (Timed)
- RPM Time Curve
- Spectrum Analysis on Selection

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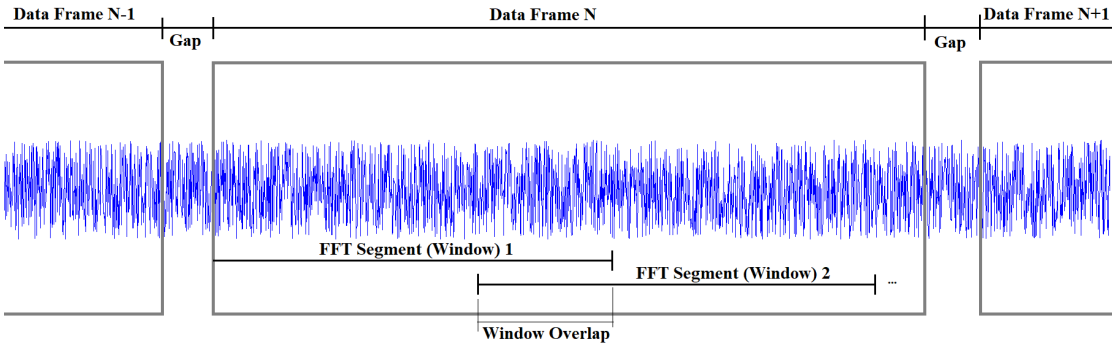
1.6 Signal Flow

1.6.1 Intra-Frame Processing and Inter-Frame Processing

Physical quantities can be converted to electronic signals by sensors. The analog electronic signals can be quantized by ADC devices and then processed and analyzed by virtual instrument software. Generally, in Multi-Instrument, after quantization, the signals flow from time domain (Oscilloscope) to frequency domain (Spectrum Analyzer) and then time-frequency domain (Spectrum 3D Plot). The data are not processed sample by sample but frame by frame instead. The number of samples (Record Length) contained in a data frame equals to [Sampling Frequency]×[Sampling Duration]. Processing performed within a data frame is called Intra-frame Processing while that performed among multiple data frames is called Inter-frame Processing. For example, when FFT size is less than Record Length, a data frame will be divided into multiple partially-overlapped FFT segments and the resulting spectrum will be based on their averaged results. This is intra-frame averaging. If averaging is performed further among different data frames, then it is inter-frame averaging.

A data frame can be acquired upon being triggered or by free run. During real time processing, there is usually a gap between adjacent data frames unless the sampling rate is sufficiently low compared to the data processing rate under free run mode (e.g. Auto Trigger Mode for sound cards). In case multiple frames of data must be sampled and processed without any gaps in between, record data continuously to the hard disk first via the Record button at the upper right corner of the screen and then post-process them via [File]>[Open Frame by Frame]. It is possible to specify a frame overlap percentage in this case. Record Mode will be described later.



The Oscilloscope window always contains one data frame only. The data inside the data frame are normally continuous without any gap. However, this may or may not be true in Roll Mode where a data frame is formed by concatenating multiple individually-sampled data segments. The data inside each data segment are continuous without any gap, but there may be a gap between adjacent data segments depending on the sampling rate and system throughput, etc. The length of the data segment is called Roll Width which can be configured through [Setting]>[Display]> "Roll Mode"> "Roll Width". Roll Mode will be described later.

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1.6.2 Signal Flow Block Diagram

The following signal flow block diagram illustrates how the signal is processed and analyzed in the software. It helps you to understand the behavior of the software, and even configure your own signal processing and analysis algorithm for a particular application, without programming using development tools such as LabVIEW and MATLAB. For example, changing the oscilloscope's SINC interpolation setting for waveform rendering will not affect the Mean, Min., Max., RMS calculation of the signal as these values are calculated before the SINC interpolation. Inter-frame averaging and Intra-frame digital filtering in time domain (Oscilloscope) will affect the subsequent spectral analysis in frequency domain (Spectrum Analyzer).

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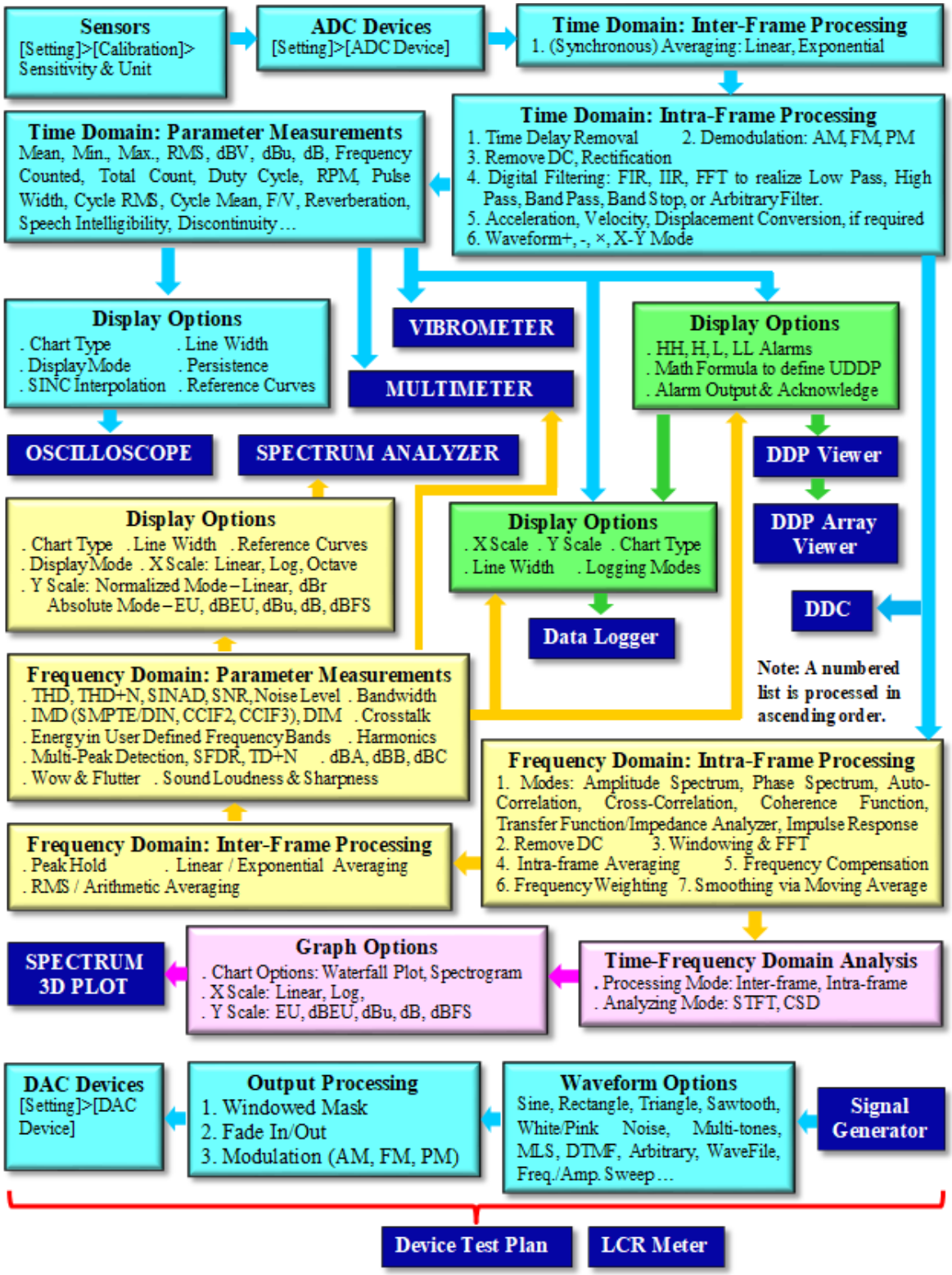
Sensors
[Setting]>[Calibration]>
Sensitivity & Unit
ADC Devices
[Setting]>[ADC Device]
Time Domain: Inter-Frame Processing
1. (Synchronous) Averaging: Linear, Exponential
Time Domain: Intra-Frame Processing
1. Time Delay Removal 2. Demodulation: AM, FM, PM
3. Remove DC, Rectification
4. Digital Filtering: FIR, IIR, FFT to realize Low Pass, High Pass, Band Pass, Band Stop, or Arbitrary Filter.
5. Acceleration, Velocity, Displacement Conversion, if required
6. Waveform+, -, ×, X-Y Mode
Time Domain: Parameter Measurements
Mean, Min., Max., RMS, dBV, dBu, dB, Frequency Counted, Total Count, Duty Cycle, RPM, Pulse Width, Cycle RMS, Cycle Mean, F/V, Reverberation, Speech Intelligibility, Discontinuity…
Display Options
. Chart Type . Line Width
. Display Mode . Persistence
. SINC Interpolation . Reference Curves
VIBROMETER
MULTIMETER
Display Options
. HH, H, L, LL Alarms
. Math Formula to define UDDP
. Alarm Output & Acknowledge
OSCILLOSCOPE
SPECTRUM ANALYZER
DDP Viewer
DDP Array Viewer
Display Options
. Chart Type . Line Width . Reference Curves
. Display Mode . X Scale: Linear, Log, Octave
. Y Scale: Normalized Mode – Linear, dBr
Absolute Mode – EU, dBEU, dBu, dB, dBFS
Display Options
. X Scale . Y Scale . Chart Type
. Line Width . Logging Modes
Data Logger
DDC
Note: A numbered list is processed in ascending order.
Frequency Domain: Parameter Measurements
. THD, THD+N, SINAD, SNR, Noise Level . Bandwidth
. IMD (SMPTE/DIN, CCIF2, CCIF3), DIM . Crosstalk
. Energy in User Defined Frequency Bands . Harmonics
. Multi-Peak Detection, SFDR, TD+N . dBA, dBB, dBC
. Wow & Flutter . Sound Loudness & Sharpness
Frequency Domain: Intra-Frame Processing
1. Modes: Amplitude Spectrum, Phase Spectrum, Auto-Correlation, Cross-Correlation, Coherence Function, Transfer Function/Impedance Analyzer, Impulse Response
2. Remove DC 3. Windowing & FFT
4. Intra-frame Averaging 5. Frequency Compensation
6. Frequency Weighting 7. Smoothing via Moving Average
Frequency Domain: Inter-Frame Processing
. Peak Hold . Linear / Exponential Averaging
. RMS / Arithmetic Averaging
SPECTRUM 3D PLOT
Graph Options
. Chart Options: Waterfall Plot, Spectrogram
. X Scale: Linear, Log,
. Y Scale: EU, dBEU, dBu, dB, dBFS
Time-Frequency Domain Analysis
. Processing Mode: Inter-frame, Intra-frame
. Analyzing Mode: STFT, CSD
DAC Devices
[Setting]>[DAC Device]
Output Processing
1. Windowed Mask
2. Fade In/Out
3. Modulation (AM, FM, PM)
Waveform Options
Sine, Rectangle, Triangle, Sawtooth, White/Pink Noise, Multi-tones, MLS, DTMF, Arbitrary, WaveFile, Freq./Amp. Sweep…
Signal Generator
Device Test Plan
LCR Meter

1.7 Precautions

Signals with high voltage can easily burn your sound card (or your ADC/DAC hardware) or even your computer. Be extremely careful and strictly follow your sound card (or ADC/DAC hardware) Manufacturer's Manual when connecting to external devices. Do not connect to signals with unknown amplitude. If the signal amplitude is high, attenuate it first before connecting it to your sound card (or ADC/DAC hardware).

To avoid personal injury, always follow the usual safety rules when working with electric circuits.

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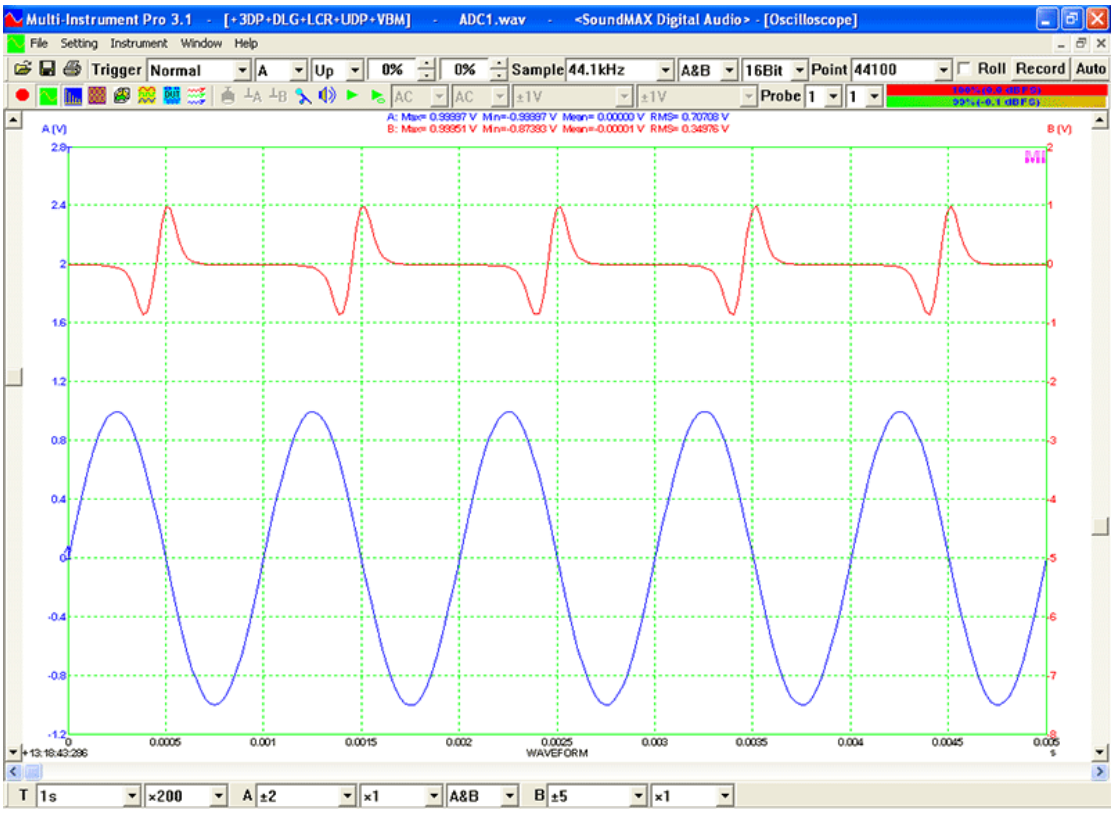
2 Oscilloscope

2.1 Overview



This is a dual channel Oscilloscope, providing five types of views:

- Waveform of Channel A and Channel B
- Waveform of Channel A + Channel B
- Waveform of Channel A - Channel B
- Waveform of Channel A × Channel B
- Lissajous Pattern for Channel A and Channel B

Statistical data such as Maximum, Minimum, Mean, Root Mean Square (RMS), Skewness and Kurtosis values are also calculated and displayed. Each frame of data is time stamped with an accuracy of 1 ms and the time stamp is shown at the lower left corner of the Oscilloscope view.

The acquired signal can be inter-frame averaged (synchronously averaged), time aligned, demodulated (AM, FM, PM), DC removed, half-wave or full-wave rectified, and digitally filtered before any other analyses. The types of digital filters supported are: Low Pass, High Pass, Band Pass, Band Stop and Arbitrary. The class of the filter can be FFT, FIR or IIR. Reverberation, speech intelligibility, waveform discontinuity, step response, echo, and damping ratio analyses are supported.

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The displayed waveform can be output directly via the default computer sound card (if the sampling rate and bit resolution of the waveform is compatible with it) by pressing the Play or Cyclic Play buttons in the Miscellaneous Toolbar.

The Oscilloscope also provides a Record Mode which can be used to record data to the hard disk continuously until the recording process is stopped manually or 2 gigabytes of data has been recorded, whichever is earlier. The oscilloscope display can also work in Roll mode, with which the screen will be updated in real time even if the sweep time is long.

The Oscilloscope can perform waveform conversion among acceleration, velocity and displacement when acceleration, velocity or displacement sensors are used.

The Oscilloscope can display both analog and digital signals in one window and thus can be used as a MSO (Mixed Signal Oscilloscope) when used with a hardware MSO.

The Oscilloscope supports digital persistence display mode, Equivalent Time Sampling (ETS) mode, SINC interpolation between samples.

2.2 Trigger Parameters

The above toolbar contains (from left to right):
Trigger Mode, Trigger Source, Trigger Edge, Trigger Level, Trigger Delay, Trigger Frequency Rejection.

The software supports both hardware trigger and software trigger. For hardware trigger, the triggering capacity is determined by the hardware. Different hardware may have different contents in the above parameters.

For sound card based data acquisition, it is possible to specify a software trigger condition for collecting a frame of data. There are two concurrent processes in this context, one is data sampling and collecting, the other is data analysis and display. Once a frame of data is collected into the PC memory, the software will start immediately to search for the next trigger event in order to collect the next frame of data. Meanwhile, the collected data will be analyzed and then displayed. With the fast PCs nowadays, normally data analysis and display will take less time than data sampling and collecting as the latter one is constrained by the sweep time and cannot benefit from the fast speed of the PCs. In case that the data sampling and collecting process is faster than the data analysis and display process, the latest frame of data will overwrite the previous one even if it has not yet been analyzed and displayed. This is to ensure that data analysis and display will always be performed on the latest collected frame of data.

In order to obtain a stable display for a periodic signal, the trigger parameters must be set properly such that there is only one such trigger condition in one cycle of the signal. The software also features a specially designed algorithm which effectively

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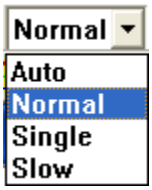
eliminates the lateral shaking of waveform display due to limited sampling rate compared with the signal frequency.

2.2.1 Trigger Mode



There are four trigger modes:

2.2.1.1 Auto

Depending on the ADC hardware used, the Auto mode here can be the "real" Auto mode or actually the Free Run mode.

The real Auto mode is similar to the Normal mode (see the section below) in that all trigger conditions specified are still in effect, except that a frame of data will be collected even if no trigger has been found after a certain timeout period. In this mode, the display might not be stable even if the signal under test is periodic, due to the fact that each frame of data may not be started at the same trigger position.

In a Free Run mode, frames of data are collected, analyzed and displayed continuously without a trigger. Very fast display refresh rate can be achieved in this mode, however, the display may not be stable even if the signal under test is periodic, due to the fact that each frame of data may not be started at the same trigger position. In this mode, the selection of Trigger Source, Trigger Edge, Trigger Level and Trigger Delay is disabled and these parameters are not used.

For sound card based data acquisition, the Auto mode means the Free Run mode.

2.2.1.2 Normal

A frame of data are collected, analyzed and displayed when the trigger condition specified is met. This process will keep going until the oscilloscope is stopped.

For sound card based data acquisition, when the sweep time (record length) is very short, the oscilloscope display may become unstable. It is recommended to use Slow Trigger Mode when the sweep time (record length) is less than 500 μs for internal sound cards, and 5 ms for external sound cards. Note that these recommended values may vary with the sound card used.

2.2.1.3 Single

One frame of data are collected, analyzed and displayed upon the first trigger event. The data acquisition process stops afterwards. This mode is ideal for transient signal capturing.

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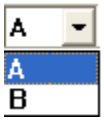
2.2.1.4 Slow

It is similar to the Normal mode, except that the data acquisition hardware will be reinitialized and restarted automatically every time a new frame of data is collected. The display refresh rate is thus slow in this mode.

For sound card based data acquisition, it is recommended to use this mode when the sweep time (record length) is very short and the display becomes unstable in Normal Trigger Mode.

2.2.2 Trigger Source



When only Channel A is sampled, the trigger source is fixed at Channel A and is not selectable.

When both Channel A and Channel B are sampled, the trigger source is selectable, either Channel A or Channel B.

The software also supports EXT (external) trigger and ALT (alternative) trigger if the hardware used supports them. The EXT trigger can be an external digital trigger or an external analog trigger with an adjustable trigger level. Under ALT trigger mode, both channels are triggered independently on their own. You can use ALT trigger mode to obtain stable waveform displays for two independent periodic signals.

2.2.3 Trigger Edge

Five types of Trigger Edge are available:

Up: When "Up" is selected, a trigger event is found when the signal is crossing the specified Trigger Level from below to above.

Dn (Down): When "Dn" is selected, a trigger event is found when the signal is crossing the specified Trigger Level from above to below.

UD (Up or Down): When “UD” is selected, a trigger event is found when the signal is crossing the specified Trigger Level from below to above, or from above to below.

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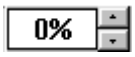
JP (Jump): When “JP” is selected, a trigger event is found when the amount of change of the signal is greater than the amount specified by the Trigger Level. When the Trigger Level is positive, the change must be “Jump Up”. When the Trigger Level is negative, the change must be “Jump Down”. For example, under this mode, “Trigger Level=100%” means that the signal magnitude must increase by 100% of half of the ADC full-scale voltage (1/2 Vpp) between two adjacent data points in order to be qualified for a trigger, and “Trigger Level = -100%” means that the signal magnitude must decrease by 100% of half of the ADC full-scale voltage (1/2 Vpp) between two adjacent data points in order to be qualified for a trigger.

DF (Differential): When “DF” is selected, a trigger event is found when the absolute amount of change of the signal is greater than the absolute amount specified by the Trigger Level. Under this mode, Trigger Level can only be adjusted from 0~100%. For example, “Trigger Level=100%” means that the signal magnitude must increase or decrease by 100% of half of the ADC full-scale voltage (1/2 Vpp) between two adjacent data points in order to be qualified for a trigger.

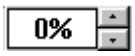
The last three trigger edges are generally only available with software trigger.

2.2.4 Trigger Level



Trigger Level is expressed as a percentage of half of the ADC full-scale voltage (1/2 Vpp). It is adjustable from -100% to 100%, except that when Trigger Edge is “DF”, Trigger Level can only be adjusted from 0 to 100%. The up and down arrows provide adjustment in a step of 1%. For finer adjustment, enter the value directly. Trigger Level can also be adjusted via the Trigger Level Marker described in Section “Trigger Marker”.

2.2.5 Trigger Delay



Trigger Delay spin box is located on the right hand side of the Trigger Level spin box. It is expressed as a percentage of the Record Length Per Sweep and is adjustable from -100% to 100%. A positive value means post-trigger while a negative value means pre-trigger. Trigger Delay can also be adjusted via the Trigger Delay Marker described in Section “Trigger Marker”. If you want the trigger point to be displayed at the center of the Oscilloscope rather than the default leftmost end (Trigger Delay = 0%), like an ordinary oscilloscope, set the Trigger Delay to -50%.

2.2.6 Trigger Frequency Rejection

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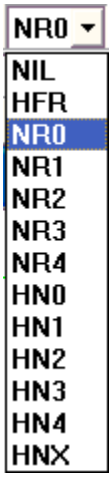


Trigger Frequency Rejection can be used to filter out noises from the trigger signal to prevent false triggering. Depending on the ADC hardware used, the available options can be: NIL (All-Pass), HFR (High Frequency Rejection), NR0~NR4 (Noise Rejection), HN0~HNX (High Frequency Rejection + Noise Rejection). There are different levels of noise rejection. Generally, for Levels 0~4, the hysteresis values are fixed while for Level X, it is user-configurable via [Setting]>[ADC Device]. Please refer to the respective hardware manual for details.

For sound cards (MME and ASIO) and software triggered NI DAQmx cards, the specification of each option is as follows:

Nil: No Rejection
HFR: High Frequency Rejection, cut off at 0.11×[Sampling Frequency]
NR0: Noise Rejection, hysteresis = 1% of half of full scale
NR1: Noise Rejection, hysteresis = 2% of half of full scale
NR2: Noise Rejection, hysteresis = 4% of half of full scale
NR3: Noise Rejection, hysteresis = 8% of half of full scale
NR4: Noise Rejection, hysteresis = 16% of half of full scale
HN0: HFR + NR0
HN1: HFR + NR1
HN2: HFR + NR2
HN3: HFR + NR3
HN4: HFR + NR4
HNX: selectable HFR + adjustable hysteresis = 0% ~ 25% of half of full scale
Note: The specified hysteresis may be modified internally to ensure [Trigger Level (%)] – [Hysteresis (%)] ≥ -100% at rising edge, or [Trigger Level (%)] + [Hysteresis (%)] ≤ 100% at falling edge.

2.3 Sampling Parameters

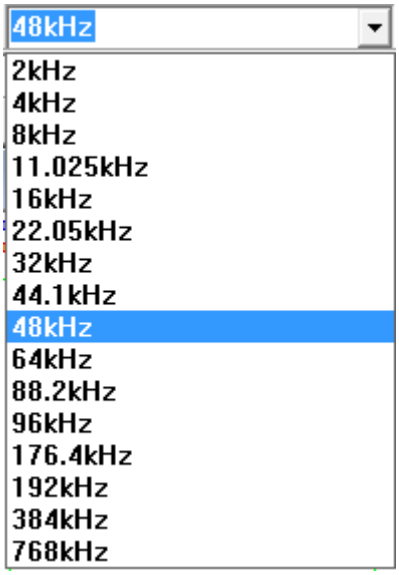
The above toolbar contains (from left to right):
Sampling Frequency, Sampling Channels, Sampling Bit Resolution, and Record Length Per Sweep.

Sampling Parameters together with the Trigger Parameters determine how the data are sampled and collected. The sampling capability is fully dependent on the ADC hardware used. Different hardware may have different contents in the above parameters.

Once the sampling parameter is specified and "Run" button is clicked, it will attempt to start sampling using the specified sampling parameters. An error message will pop up if the specified sampling parameters are not supported by the ADC hardware.

Note that some sound cards may not generate an error message even if the sampling frequency specified exceeds the limit. Please check your sound card manual before you use a sampling frequency greater than 44100 Hz, otherwise measurement error may be introduced.

2.3.1 Sampling Frequency



For a sound card based system, the following sampling frequencies can be selected: 2kHz, 4kHz, 8kHz, 11.025kHz, 16kHz, 22.05kHz, 32kHz, 44.1kHz, 48kHz, 64kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz, 384kHz, 768kHz. In addition, you can enter a frequency value directly when the Oscilloscope is not running.

2.3.2 Sampling Channels

For a sound card based system, two options are available:

- A
  Only Channel A is sampled.

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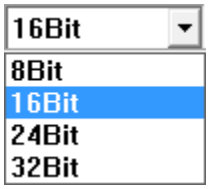
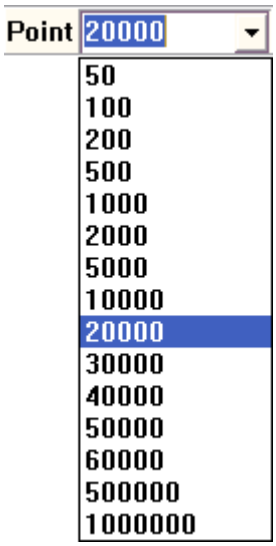
- A&B
  Both Channel A and Channel B are sampled.

2.3.3 Sampling Bit Resolution



For a sound card based system, four options are available: 8Bit, 16Bit, 24Bit, 32Bit, where the 32Bit option can be either in the 32-bit integer or 32-bit float format. “Float32” will be displayed at the bottom of the Oscilloscope window if the data are in 32-bit float format.

2.3.4 Record Length per Sweep



Record Length per Sweep (or per frame) determines how many points (samples) will be captured per sweep for each channel. Normally you do not need to specify the Record Length directly as it is implicitly determined by the Sweep Time and the Sampling Frequency (Record Length = Sweep Time × Sampling Frequency). When you change the Sweep Time or Sampling Frequency, the Record Length will be automatically updated accordingly. Under some circumstance, you may want to explicitly specify the record length, for example, you may want the Record Length to be an integer multiple of 10 or a power of 2. Then you can either select or enter the Record Length you want directly. The sweep time will then be automatically updated accordingly so that all data points can be accommodated in one sweep.

There are 15 options available for you to select: 50, 100, 200, 500, 1000, 2000, 5000, 10000, 20000, 30000, 40000, 50000, 60000, 500000, 1000000.

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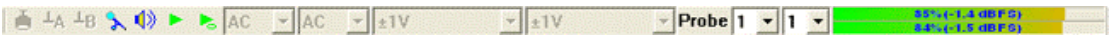
Alternatively, you can enter any number for the Record Length as long as the sweep time is less than 500 s and the computer memory allows.

Changing Record Length directly is disabled at the very beginning in order to avoid confusion to beginners. It can be enabled via [Setting]>[Display]>[Enable Record Length change via "Point" in Sampling Parameter Toolbar].

The maximum Record Length that can be set is limited by the ADC hardware buffer size. You should always keep this rule in mind when you change the Sweep Time or Sampling Frequency which will change the Record Length indirectly or when you change the Record Length directly. This rule is enforced by the software. For example, if the requested Record Length exceeds the ADC hardware buffer size when you change the Sweep Time, then the software will attempt to lower the Sampling Frequency in order to keep the Record Length within the ADC hardware buffer size. On the other hand, if the requested Record Length exceeds the ADC hardware buffer size when you change the Sampling Frequency, then the software will attempt to lower the Sweep Time in order to keep the Record Length within the ADC hardware buffer size. If the software is unable to enforce the rule, an error message will pop up.

The software also enforces a minimum Record Length when you change the Sweep Time or Sampling Frequency. This is to ensure that a sufficient number of data points are acquired in one frame. If the requested Record Length is lower than the minimum Record Length, then the software will adjust the Sampling Frequency or Sweep Time in order to keep the Record Length above the minimum value. It should be noted that changing the Record Length directly is considered as an intended action and thus is not affected by this rule. Different ADC hardware may have different minimum Record Length values which are set inside the software.

2.4 Miscellaneous Parameters



The above toolbar contains (from left to right):
Invert Input Signal button, Channel A Zeroing button, Channel B Zeroing button, Windows Recording Control, Windows Volume Control, Play button, Cyclic Play button, ADC Channel A Coupling Type, ADC Channel B Coupling Type, ADC Channel A Range, ADC Channel B Range, Probe A Current Switch Position, Probe B Current Switch Position, and Input Peak Level Indicators for Channel A and Channel B.

2.4.1 Invert Input Signal

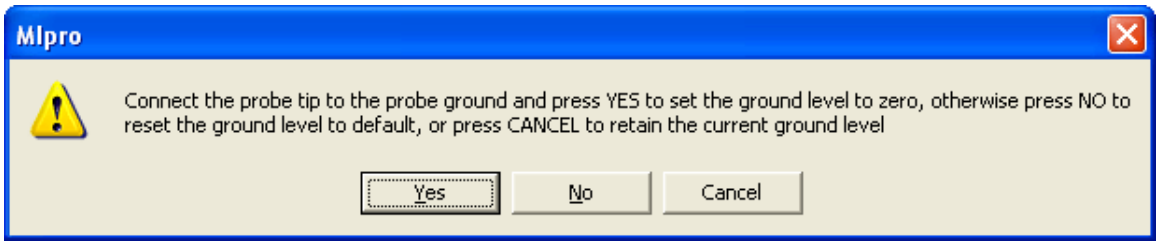
If the button is depressed, the input signal will be inverted by the software just after A/D conversion, e.g. +1V will become -1V after the inversion. All the subsequent processing such as triggering, data analysis and display will be performed based on the inverted signal.

The button is in released state by factory default.

2.4.2 Zeroing

If the input of a channel is connected to its ground, the oscilloscope should display a straight horizontal line at 0 V. However, this may not always be the case. Some ADC hardware, if not compensated by software, may display an offset voltage when their input is actually at the ground level. Therefore, there is a need to compensate this offset for ADC hardware.

: This button will only be enabled when the oscilloscope is in running state and the Trigger Mode is "Auto". Once pressed, a message box will pop up with three options: Yes, No, Cancel. You will need to connect the input for Channel A to the ground before you choose Yes, in order to set the ground level for Channel A to zero. If you select No, then the ground level will be reset to its default value (i.e. no compensation). You may choose Cancel to cancel the operation.



: This button will only be enabled when the oscilloscope is in running state, the Trigger Mode is "Auto" and the Sampling Channels is "A&B". Once pressed, a message box will pop up with three options: Yes, No, Cancel. You will need to connect the input for Channel B to the ground before you choose Yes, in order to set the ground level for Channel B to zero. If you select No, then the ground level will be reset to its default value (i.e. no compensation). You may choose Cancel to cancel the operation.

2.4.3 Windows Recording Control

2.4.3.1 Recording Control before Windows Vista

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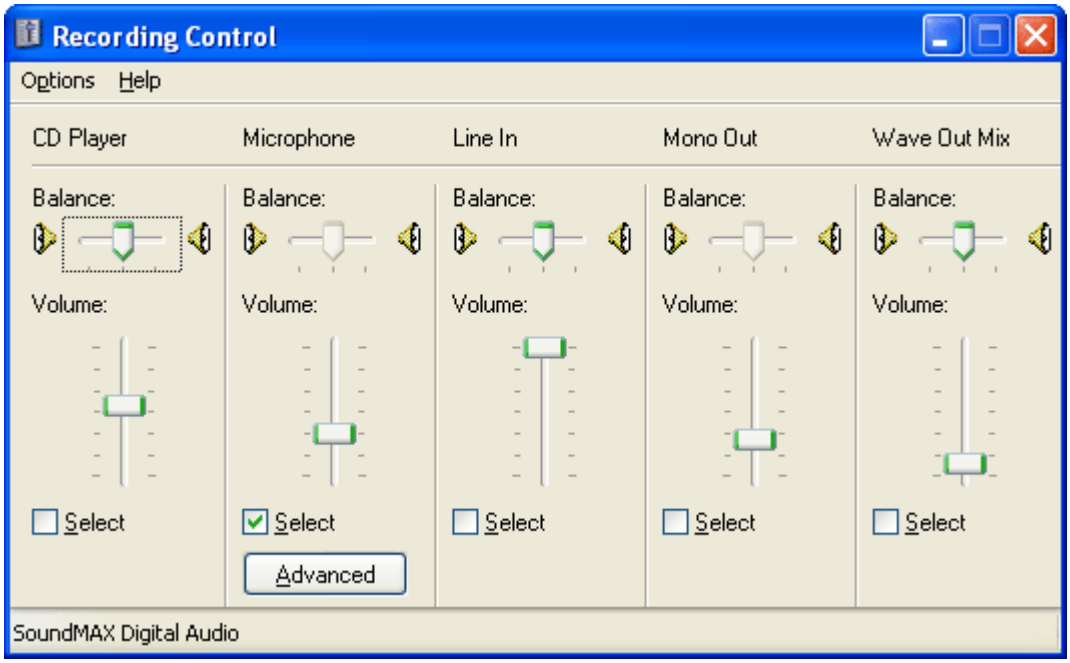


A typical Recording Control of Windows XP (or other Windows versions before Windows XP) is shown as above. Different sound cards may have different items in the Recording Control.

2.4.3.1.1 Sound Card Selection

When accessing the Windows Recording Control from the Windows Control Panel, you need to choose the sound card used for data acquisition first if multiple sound cards exist.

When pressing the button in the Miscellaneous Toolbar of the software, the Recording Control of the sound card used by the software for data acquisition will be opened.

For this software, the Recording Control is used to select the input source and adjust its internal gain. The selection of sound card is done via [Setting]>[ADC Device]>[Device No.].

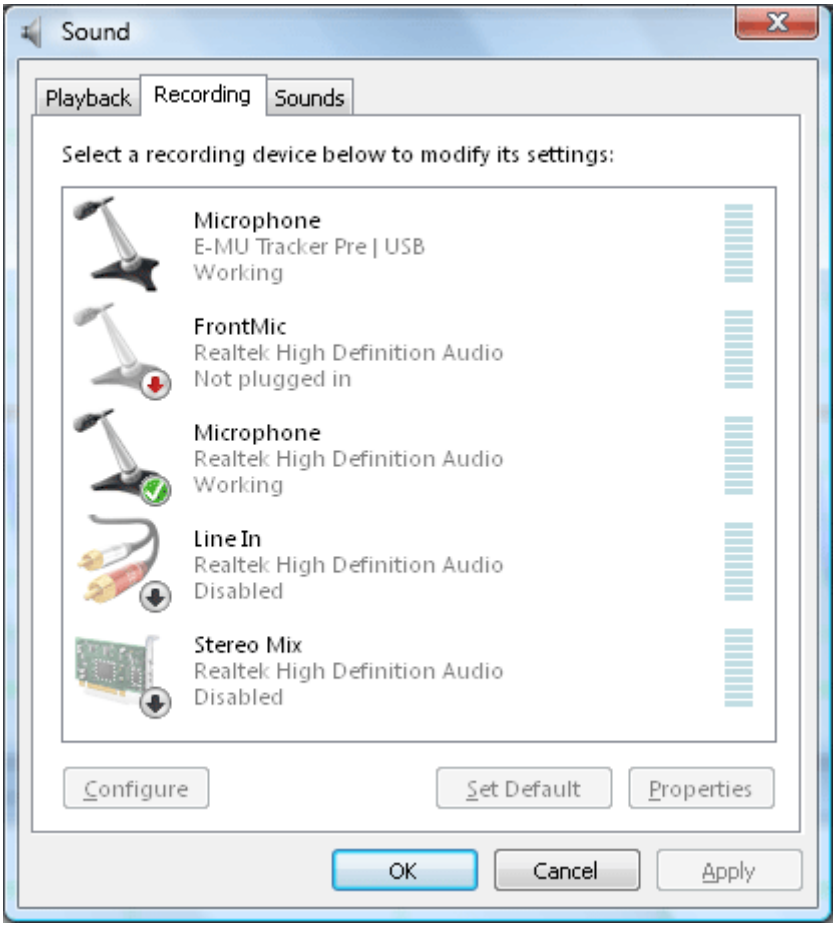
2.4.3.1.2 Input Source Selection

From the Recording Control, you can configure the input sources for data acquisition. The input source can be CD Player, Microphone, Line In, Wave Out Mix, etc, depending on the sound card used. To test an external electrical signal, either Mic Input or Line In should be used. Wave Out Mix (sometimes called “Stereo Mix”, “What U Hear” or something similar) can be used to get the signal being output by the sound card. You can select it as the input source for data acquisition in order to analyze and display what is being output by the Signal Generator. This, in fact, switches the software into simulation mode with the loopback at the sound card mixer level.

2.4.3.1.3 Input Gain Adjustment

The input gain can be adjusted by moving the volume slider corresponding to the input source selected for data acquisition. For Microphone, it is usually possible to further adjust the gain by selecting/removing Mic Boost in the Advanced Controls for Microphone as shown below. Normally selecting Mic Boost will increase the internal gain by 10 times (i.e. 20dB).

2.4.3.2 Recording Control from Windows Vista Onward



There are some changes of the Recording Control from Windows Vista onward. Under Windows Vista, you do not select a sound card first and then open its Recording Control which contains all its input sources and their gain controls. Instead, you select a so-called input endpoint (i.e. a particular input source on a particular sound card, for example, the microphone of a particular sound card) first in the Windows Sound Recording control panel (see the figure above) and then open its property window (see the figure below) to adjust its gain.

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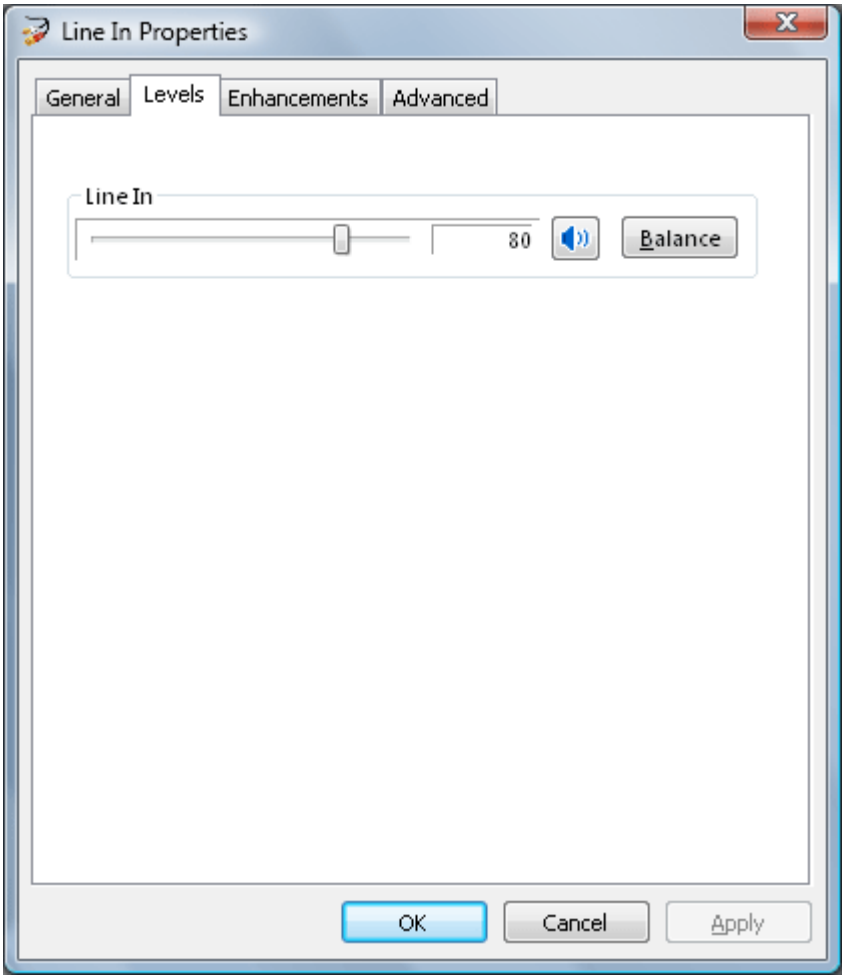

2.4.3.2.1 Input Endpoint Selection

When pressing the button in the Miscellaneous Toolbar of the software, the Windows Sound Recording control panel will be opened.

For this software, the Windows Sound Recording control panel is used to adjust the gain of the selected input endpoint. The selection of the input endpoint is done via [Setting]>[ADC Device]>[Device No.]. Note that only the enabled input endpoints will be available for selection, and any on-the-fly change (e.g. enable/disable, set as default) of the input endpoints when the Oscilloscope is running may cause the software to stop data acquisition. You will then need to restart the data acquisition if necessary.

2.4.3.2.2 Input Gain Adjustment

The gain of the selected input endpoint can be adjusted by selecting the input endpoints in the Windows Sound Recording control panel and then opening its property window to adjust its gain.

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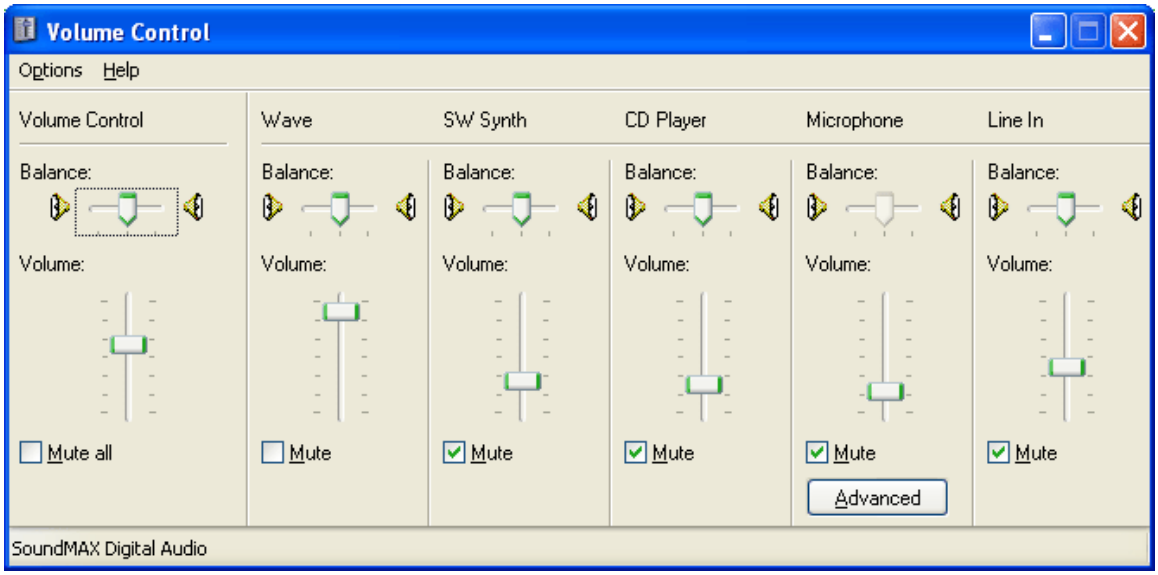
2.4.4 Windows Volume Control

2.4.4.1 Volume Control before Windows Vista



A typical Windows Volume Control is shown as above. Different sound cards may have different items in the Volume Control.

2.4.4.1.1 Sound Card Selection

When accessing the Windows Volume Control from the Windows Control Panel, you need to choose the sound card used for signal output first if multiple sound cards exist.

When pressing the button in the Miscellaneous Toolbar of the software, the Volume Control of the sound card used by the software for signal output will be opened.

For this software, the Volume Control is used to select the output source and adjust the output volume. The selection of sound card is done via [Setting]>[DAC Device]>[Device No.].

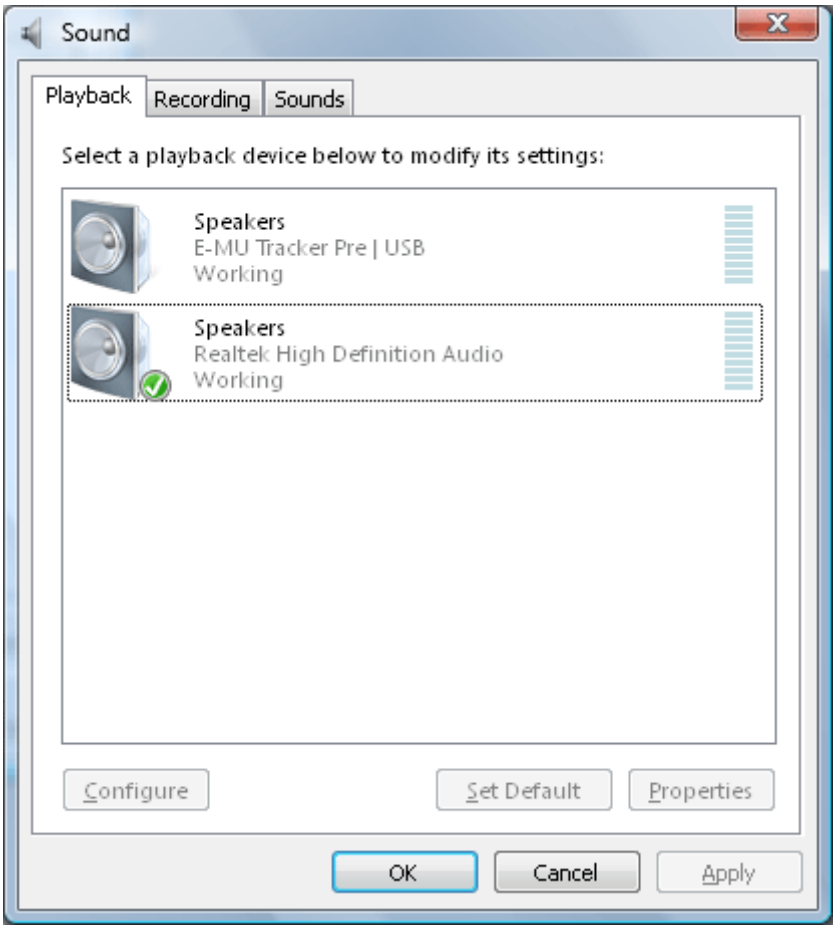
2.4.4.1.2 Output Source Selection

From the Volume Control, you can configure the output sources for signal output. For the Signal Generator of the software, all output sources for signal output should be muted except the Volume Control and Wave, in order to minimize the unwanted noises.

2.4.4.1.3 Output Volume Adjustment

The output volume can be adjusted via either the Volume Control slider or Wave slider.

2.4.4.2 Volume Control from Windows Vista Onward



Under Windows Vista, you select a so-called output endpoint (i.e. a particular output destination on a particular sound card, for example, the "Speakers" of a particular sound card) first in the Windows Sound Playback control panel (see the figure above) and then open its property window (see the figure below) to configure its output source and output volume.

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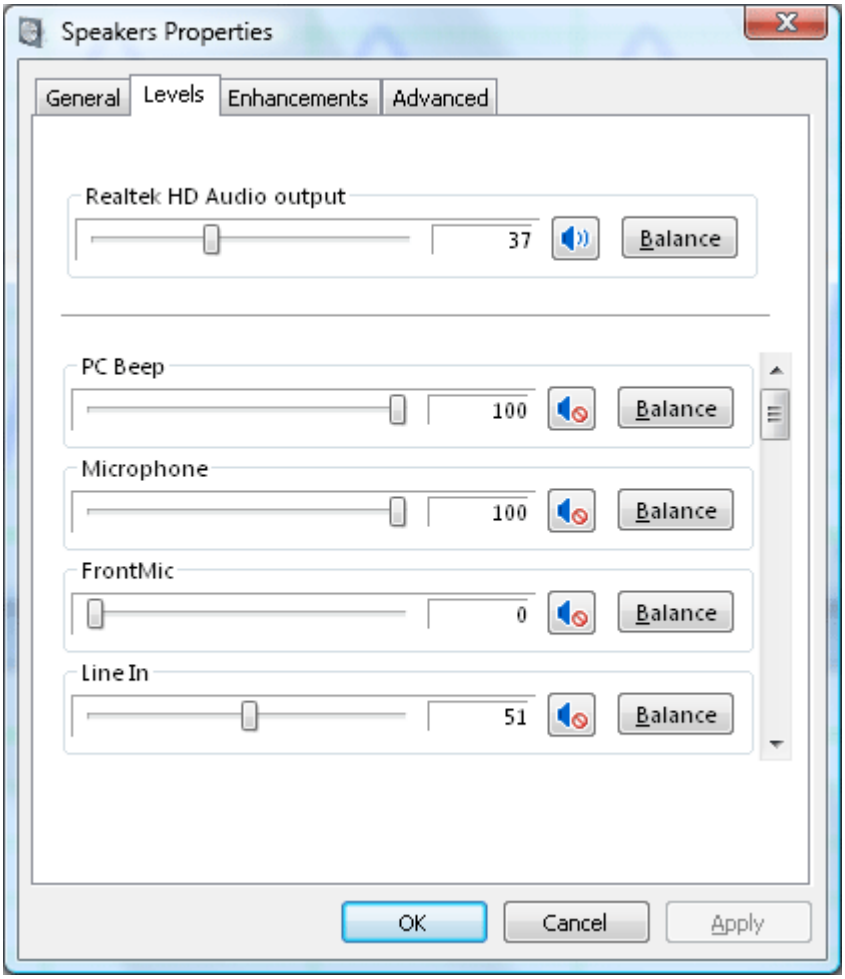


2.4.4.2.1 Output Endpoint Selection

When pressing the button in the Miscellaneous Toolbar of the software, the Windows Sound Playback control panel will be opened.

For this software, the Windows Sound Playback control panel is used to select the output source and adjust its output volume. The selection of the output endpoint is done via [Setting]>[DAC Device]>[Device No.].

2.4.4.2.2 Output Source Selection

From the property window of the selected output endpoint, you can configure the output sources for signal output. For the Signal Generator of the software, all output sources for signal output should be muted except the master Volume Control, in order to minimize the unwanted noises.

2.4.4.2.3 Output Volume Adjustment

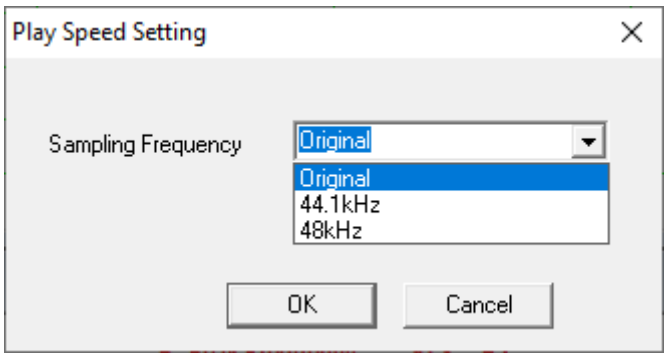
The output volume can be adjusted via the master Volume Control slider.

2.4.5 Waveform Play

The waveform displayed in the Oscilloscope can be played (output) through the default sound card by pressing the button in the Miscellaneous Toolbar. Note that this playback is possible only if the sampling rate and bit resolution of the waveform is compatible with the Windows default sound card.

2.4.5.1 Play Speed Changing and Pitch Shifting

The default play speed is the original sampling frequency. If there is a need to adjust this speed, click the button with the CTRL key down, the following dialog box will pop up.



The "Original" sampling frequency is always selected upon software restart. You can select or enter a new sampling frequency directly and click OK. The software will then create a temporary file "~temp.wav" in the software's installation directory and replay it. You can rename the file to keep a pitch-shifted and speed-changed WAV file, in order to convert an inaudible signal to an audible one.

2.4.6 Waveform Cyclic Play

The waveform displayed in the Oscilloscope can be cyclically played (output) through the default sound card by pressing the button in the Miscellaneous Toolbar. Releasing this toggle button will stop the cyclic playing. Note that this playback is possible only if the sampling rate and bit resolution of the waveform is compatible with the Windows default sound card.

2.4.6.1 Play Speed Changing and Pitch Shifting
Same as 2.4.5.1.

2.4.7 Coupling Type for ADC Channels A & B

The left one is for ADC Channel A and the right one is for ADC Channel B.

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Almost all sound cards are AC coupled. For other ADC hardware, the options can be AC, DC or GND, depending on the hardware used.

2.4.8 Range for ADC Channels A & B

The left one is for ADC Channel A and the right one is for ADC Channel B.

For sound cards, calibration is normally required in order to determine their ADC ranges, and their ADC ranges change with their input gain setting which is adjustable via the Windows Recording Control.

Calibration is normally not required for other ADC hardware with the ADC ranges explicitly specified. If the hardware supports multiple ADC ranges, the above combo boxes will become selectable.

2.4.9 Current Switch Position for Probes A & B

The two combo boxes on the right hand side of "Probe" in the Miscellaneous Toolbar allow you to select the probe attenuation factors corresponding to the current attenuation switch position on your probes or test leads. The left one is for Channel A and the right one is for Channel B. You can have at most 16 switch positions. The dedicated sound card oscilloscope probe supplied by Virtins Technology is best suited for sound card based systems.

Please refer to: *VIRTINS Sound Card Oscilloscope Probe Manual*.

Download link:
https://www.virtins.com/Virtins-Sound-Card-Oscilloscope-Probe-Manual.pdf

Note that these two combo boxes only allow you to select the corresponding attenuation factors so that the sampled data can be scaled properly in the software. They will not set the physical switch position on the probe for you. You have to set it manually. More generally, this function can be very useful to properly scale the sampled data for those ADC devices that have attenuation (gain) switches not controllable from Multi-Instrument.

The actual labels and attenuation factors for different switch positions can be configured via [Setting]>[Calibration]. This will be introduced later in this document.

2.4.10 Input Peak Level Indicator for ADC Channels A & B

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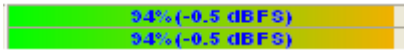
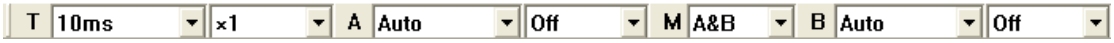


The above two Input Peak Level Indicators reflect the peak level of the frame of data acquired, the upper one is for Channel A and the lower one is for Channel B. It is expressed as a percentage of the ADC full-scale range. The color of the indicator changes gradually from green to orange as the percentage goes from 0% to 100% (i.e. 0 dBFS).

If the Input Peak Level is equal to 100%, the indicator will be fully filled with red color. Under this situation, it is recommended to lower the input gain of the hardware, increase external attenuation, or lower the signal under test directly, in order to avoid peak clipping from happening.

2.5 View Parameters



View Parameters determine how the collected data are displayed and analyzed.

2.5.1 Sweep Time (T)

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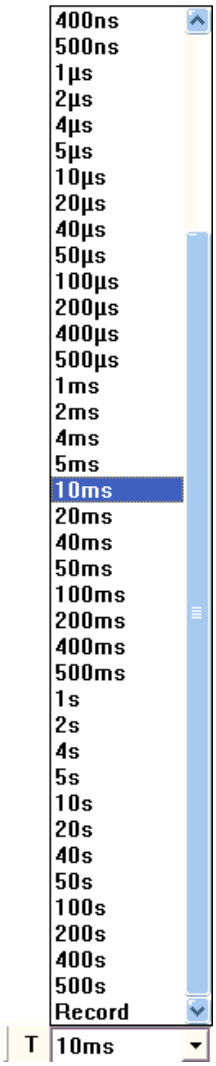

Unlike other view parameters, Sweep Time (T) is not only a view parameter, but also a sampling parameter. It determines the number of samples to acquire in one sweep (=[Sweep Time]×[Sampling Frequency]).

There are 49 options for Sweep Time. They are 1 ns, 2 ns, 4 ns, 5 ns, 10 ns, 20 ns, 40 ns, 50 ns, 100 ns, 200 ns, 400 ns, 500 ns, 1 µs, 2 µs, 4 µs, 5 µs, 10 µs, 20 µs, 40 µs, 50 µs, 100 µs, 200 µs, 400 µs, 500 µs, 1 ms, 2 ms, 4 ms, 5 ms, 10 ms, 20 ms, 40 ms, 50 ms, 100 ms, 200 ms, 400 ms, 500 ms, 1 s, 2 s, 4 s, 5 s, 10 s, 20 s, 40 s, 50 s, 100 s, 200 s, 400 s, 500 s, Record.

This parameter is applicable to all types of views in the Oscilloscope.

Note that if you select a longer sweep time, it will take a longer time for the acquired data and analyzed results to be shown on the screen. When the sweep time is long, you can tick the “Roll” checkbox in the Sampling Parameter Toolbar in order to get real time update of the screen. Please refer to the section for Roll Mode for details.

If “Record” is chosen, the oscilloscope will enter into Record Mode. Please refer to the section for Record Mode for details.

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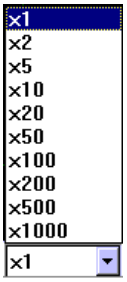
2.5.2 Sweep Time Multiplier



The Sweep Time Multiplier is the zooming factor for T axis. There are 16 options available: ×1, ×2, ×5, ×10, ×20, ×50, ×100, ×200, ×500, ×1000, ×2000, ×5000, ×10000, ×20000, ×50000, ×100000.

When "×1" is selected, the full range of the Sweep Time is displayed over the width of the view.

If you change the Sweep Time Multiplier to "×N" which is greater than 1, then only 1/N of the full range of the Sweep Time is displayed over the width of the view, with a horizontal scrollbar at the bottom which allows you to scroll over the full range of the Sweep Time.

This parameter is applicable to all types of views in the Oscilloscope except Lissajous Pattern display.

This multiplier can also be adjusted via the magnifying glass which will be shown if you put the mouse cursor just below T axis. Left click to zoom in and right click to zoom out. Zoom in/out can also be performed using the mouse wheel. Mouse wheel down & scroll will reset the Multiplier to "×1".

2.5.3 Channel A Display Range

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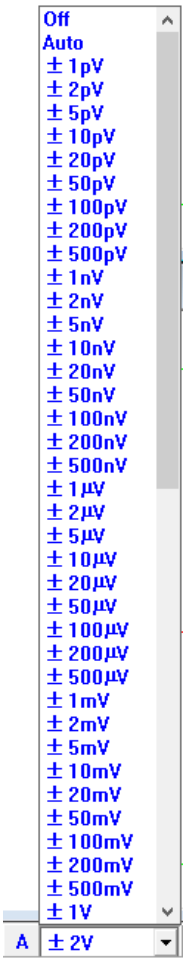

You can specify the Display Range for Channel A. Available options are: Off, Auto, ±1pV, ±2pV, ±5pV, ±10pV, ±20pV, ±50pV, ±100pV, ±200pV, ±500pV,±1nV, ±2nV, ±5nV, ±10nV, ±20nV, ±50nV, ±100nV, ±200nV, ±500nV, ±1µV, ±2µV, ±5µV, ±10µV, ±20µV, ±50 µV, ±100µV, ±200µV, ±500µV, ±1mV, ±2mV, ±5mV, ±10mV, ±20mV, ±50mV, ±100mV, ±200mV, ±500mV, ±1V, ±2V, ±5V, ±10V, ±20V, ±50V, ±100V, ±200V, ±500V, ±1kV, ±2kV, ±5kV, ±10kV, ±20kV, ±50kV, ±100kV, ±200kV, ±500kV, ±1MV, ±2MV, ±5MV, ±10MV, ±20MV, ±50MV, ±100MV, ±200MV, ±500MV, ±1GV, ±2GV, ±5GV, ±10GV, ±20GV, ±50GV, ±100GV, ±200GV, ±500GV. Note that the engineering unit of the above options is determined by the engineering unit of the sensor for Channel A, which can be set via [Setting]>[Calibration]> "Sensor"> "Unit". For example, if the unit is "g" instead of "V", then the unit in all the above options will be changed to "g".

When "Off" is selected, the signal in Channel A will not be displayed in the Oscilloscope. When "Auto" is selected, the Display Range for Channel A will be set automatically by the software based on the following formula:

Display Range = the nearest integer that is equal to or greater than the value of [ADC Range] / [Sensor Sensitivity]

where the sensor sensitivity is set via [Setting]>[Calibration]> "Sensor">"Sensitivity".

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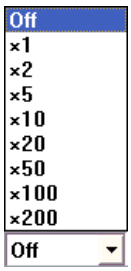
This parameter is applicable to all types of views in the Oscilloscope. It should be noted that in "Channel A + Channel B" view, this voltage display range is for "Channel A + Channel B". Similarly, In "Channel A – Channel B" view, it is for "Channel A – Channel B", and in "Channel A × Channel B" view, it is for "Channel A × Channel B".

2.5.4 Channel A Multiplier



The Multiplier for Channel A is the zooming factor for A axis. There are 9 options available: Off, ×1, ×2, ×5, ×10, ×20, ×50, ×100, ×200.

When "Off" is selected, the full Display Range for Channel A is displayed over the height of the view.

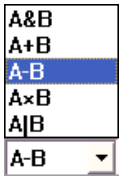
When "×1" is selected, initially the full range is displayed over the height of the view with a vertical scroll bar on the left of the view. You can use the scroll bar to move the data curve for Channel A up and down.

If you change the multiplier to "×N" which is greater than 1, then only 1/N of the full range is displayed over the height of the view, with a vertical scrollbar on the left of the view. You can use the scroll bar to scroll over the full Display Range.

This parameter is applicable to all types of views in the Oscilloscope except Lissajous Pattern display. It should be noted that in "Channel A + Channel B" view, this parameter is for "Channel A + Channel B". Similarly, In "Channel A – Channel B" view, it is for "Channel A – Channel B", and in "Channel A × Channel B" view, it is for "Channel A × Channel B".

This multiplier can also be adjusted via the magnifying glass which will be shown if you put the mouse cursor on the left side of A axis. Left click to zoom in and right click to zoom out. Zoom in/out can also be performed using SHIFT + mouse wheel. SHIFT + mouse wheel down & scroll will reset the Multiplier to "Off".

2.5.5 View Type



There are five types of views in the Oscilloscope: A&B, A+B, A-B, A×B, A|B。 Changing the View Type will affect the calculation of the following Derived Data Points (DDPs) of Channel A in the Oscilloscope:

(1) Max_A(EU): Maximum value for Channel A or combined
(2) Min_A(EU): Minimum value for Channel A or combined
(3) PP_A(EU): Peak-to-Peak value for Channel A or combined
(4) Mean_A(EU): Mean value for Channel A or combined
(5) RMS_A(EU): RMS value for Channel A or combined
(6) PWR_A(W): Power value (= RMS^2 / Load Factor) for Channel A or combined

It will also affect the following DDPs of Channel A in the Multimeter:
(1) RMSDBV_A(dBEU): RMS value in dBEU for Channel A or combined
(2) RMSDBu_A(dBu): RMS value in dBu for Channel A or combined
(3) RMSDB_A(dB): RMS value in dB for Channel A or combined

However, it does not affect the calculation of other DDPs of Channel A in the Oscilloscope, such as:

(1) Skewness_A: Skewness for Channel A
(2) Kurtosis_A: Kurtosis for Channel A
(3) PeakLevelPercent_A(%): Peak Level in Percentage for Channel A
(4) PeakLeveldBFS_A: Peak Level in dBFS for Channel A
(5) WaveformComparisonH_A: Waveform High or High-High Limit Alarm for Channel A
(6) WaveformComparisonL_A: Waveform Low or Low-Low Limit Alarm for Channel A

Channel B has all the above mentioned DDPs. The calculation of them is not affected by the selection of View Type.

2.5.5.1 A&B
Waveform display of Channel A and Channel B

2.5.5.2 A+B
Waveform display of Channel A + Channel B

2.5.5.3 A-B
Waveform display of Channel A - Channel B

2.5.5.4 A×B
Waveform display of Channel A × Channel B.

As an example, the following figure shows how to use this function to measure power factor. The instantaneous voltage and current are measured by Channel A and

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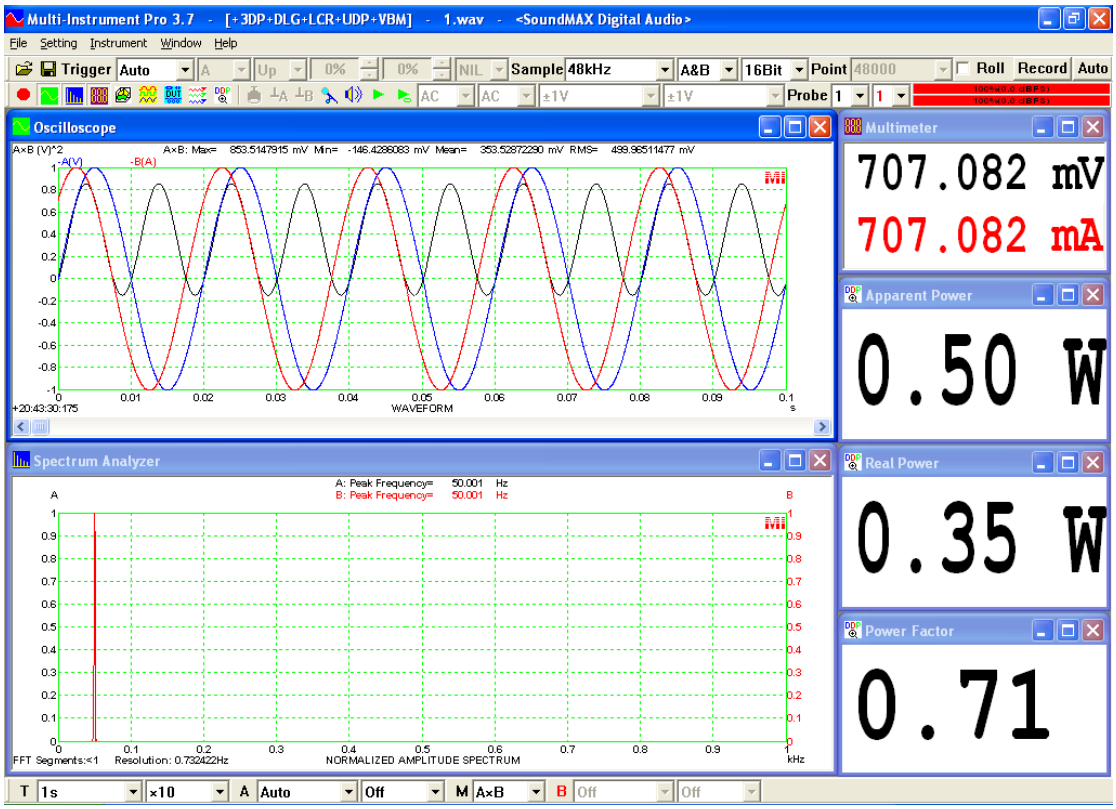
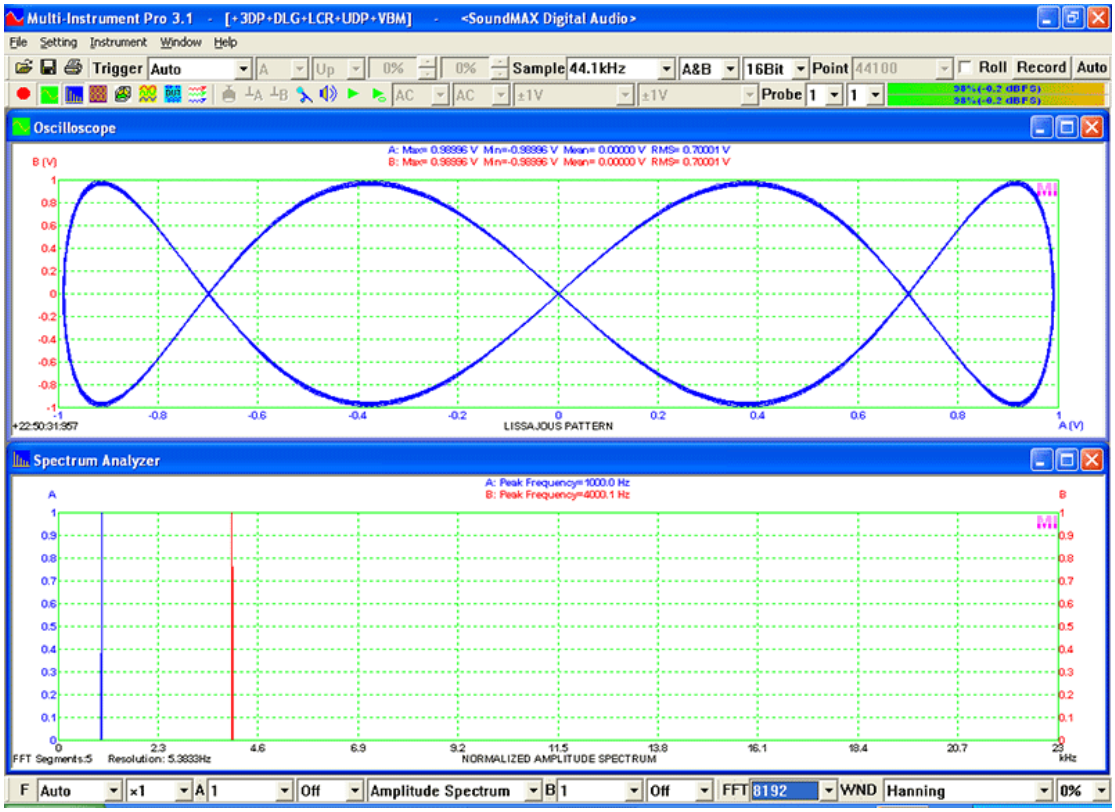
Channel B respectively. The measured RMS voltage in Channel A is 707 mV while the measured RMS current in Channel B is 707 mA. This results in an apparent power of 0.707(V)×0.707(A)=0.50(W). The real power is obtained by the mean value of A×B and reads 0.354W here. Therefore the power factor is: 0.354/0.50=0.71. In this example, the voltage and current are both sine waves and have a phase difference of 45°. Thus the power factor can also be calculated as: cos(45°)=0.71. The Panel Setting File of this measurement can be found in the software's installation directory\psf\PowerFactor_FFT32768_SR48000.psf.

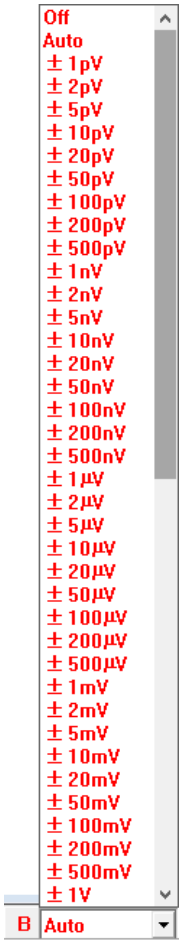


2.5.5.5 A|B (XY Mode, Lissajous Pattern)
Lissajous Pattern display for Channel A and Channel B.

The following figure illustrates the Lissajous Pattern of a 1 kHz sine wave in Channel A and 4 kHz sine wave in Channel B.



2.5.6 Channel B Display Range



You can specify the Display Range for Channel B. Available options are: Off, Auto, ±1pV, ±2pV, ±5pV, ±10pV, ±20pV, ±50pV, ±100pV, ±200pV, ±500pV, ±1nV, ±2nV, ±5nV, ±10nV, ±20nV, ±50nV, ±100nV, ±200nV, ±500nV, ±1µV, ±2µV, ±5µV, ±10µV, ±20µV, ± 50µV, ±100µV, ±200µV, ±500µV, ±1mV, ±2mV, ±5mV, ±10mV, ±20mV, ±50mV, ±100mV, ±200mV, ±500mV, ±1V, ±2V, ±5V, ±10V, ±20V, ±50V, ±100V, ±200V, ±500V, ±1kV, ±2kV, ±5kV, ±10kV, ±20kV, ±50kV, ±100kV, ±200kV, ±500kV, ±1MV, ±2MV, ±5MV, ±10MV, ±20MV, ±50MV, ±100MV, ±200MV, ±500MV, ±1GV, ±2GV, ±5GV, ±10GV, ±20GV, ±50GV, ±100GV, ±200GV, ±500GV. Note that the engineering unit of the above options is determined by the engineering unit of the sensor for Channel B, which can be set via [Setting]>[Calibration]> "Sensor">"Unit". For example, if the unit is "g" instead of "V", then the unit in all the above options will be changed to "g".

When "Off" is selected, the signal in Channel B will not be displayed in the Oscilloscope. When "Auto" is selected, the Display Range for Channel B will be set automatically by the software based on the following formula:

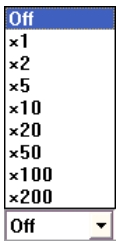
Display Range = the nearest integer that is equal to or greater than the value of [ADC Range] / [Sensor Sensitivity]

where the sensor sensitivity is set via [Setting]>[Calibration]> "Sensor">"Sensitivity".

For single channel data, this control will be disabled.

This parameter is only applicable to two types of views: Channel A & Channel B and Lissajous Pattern. It is disabled in the other two types of views.

2.5.7 Channel B Multiplier



The Multiplier for Channel B is the zooming factor for B axis. There are 9 options available: Off, ×1, ×2, ×5, ×10, ×20, ×50, ×100, ×200.

When "Off" is selected, the full Display Range for Channel B is displayed over the height of the View.

When "×1" is selected, initially the full range is displayed over the height of the view with a vertical scroll bar on the right of the view. You can use the scroll bar to move the data curve for Channel B up and down.

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If you change the multiplier to "×N" which is greater than 1, then only 1/N of the full range is displayed over the height of the view, with a vertical scrollbar on the right of the view. You can use the scroll bar to scroll over the full Display Range.

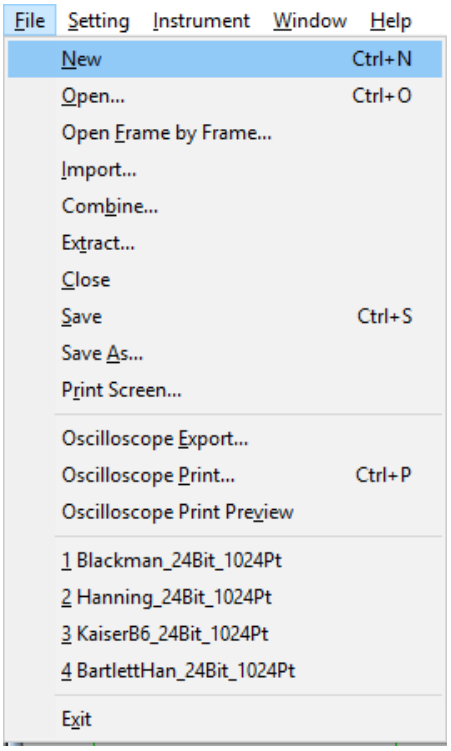
This parameter is only applicable to one types of view: Channel A & Channel B. It is disabled in other types of views.

This multiplier can also be adjusted via the magnifying glass which will be shown if you put the mouse cursor on the right side of B axis. Left click to zoom in and right click to zoom out. Zoom in / out can also be performed using CTRL + mouse wheel. CTRL + mouse wheel down & scroll will reset the Multiplier to "Off".

2.6 Menu

The Oscilloscope has its own menu and additional functions can be accessed through the menu items in each submenu. Click anywhere within the Oscilloscope window will switch the software's main menu to the Oscilloscope menu.

2.6.1 File SubMenu



This submenu provides access to the file operation and printing functions.

2.6.1.1 New (File SubMenu) (ALT-F-N, CTRL-N)

The command is used to create a new, blank document for measurement, with the system default panel setting fixed in the factory. The new document will be used to hold the latest collected frame of data.

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This command is also available through the button in the Instrument Toolbar. When there is no Oscilloscope, Spectrum Analyzer and Multimeter opened, click the above "Oscilloscope" button will open a new document without changing the current Trigger and Sampling Parameters.

2.6.1.2 Open (File SubMenu) (ALT-F-O, CTRL-O)

The command is used to open an existing document. Only standard and extensible WAV files in PCM format can be opened. An error message will pop up if the file format is not recognizable. WAV files with PCM format are widely supported by many software products. You can use a third-party software program such as "Sound Recorder" provided in Windows to record the data and then use Multi-Instrument to display and analyze the data.

This command is also available through the button in the Sampling Parameter Toolbar. Only one document can be opened at a time. It is possible to open a WAV file through “Drag and Drop” when the Oscilloscope is not running.

If the file to be opened is too big to be held in one Oscilloscope frame, the software will automatically open it frame by frame.

This command will reset the current panel setting after the file is loaded.

2.6.1.3 Open with Current Panel Setting (File SubMenu) (ALT-F-W)

The command is similar to the [File]>[Open] command, except that it will try to keep the current panel setting after the file is loaded.

2.6.1.4 Open Frame by Frame (File SubMenu) (ALT-F-F)

The command is the same as the Open command described previously, except that it opens a WAV file frame by frame. It is particularly useful for opening a long WAV file. The frame width (i.e. Record Length Per Sweep) should be set first. The following dialog box will pop up after the file to be opened is selected. You can either specify the frame width in millisecond or number of points. The total file length in millisecond and points are also displayed for reference.

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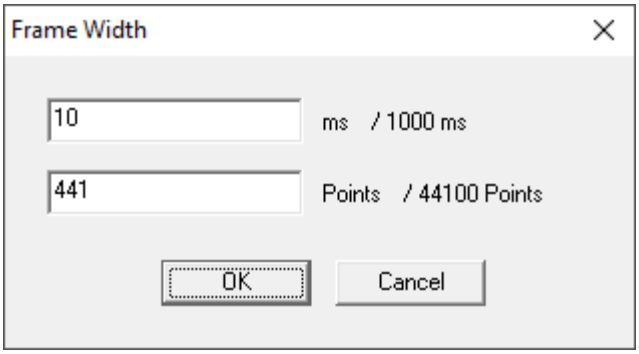
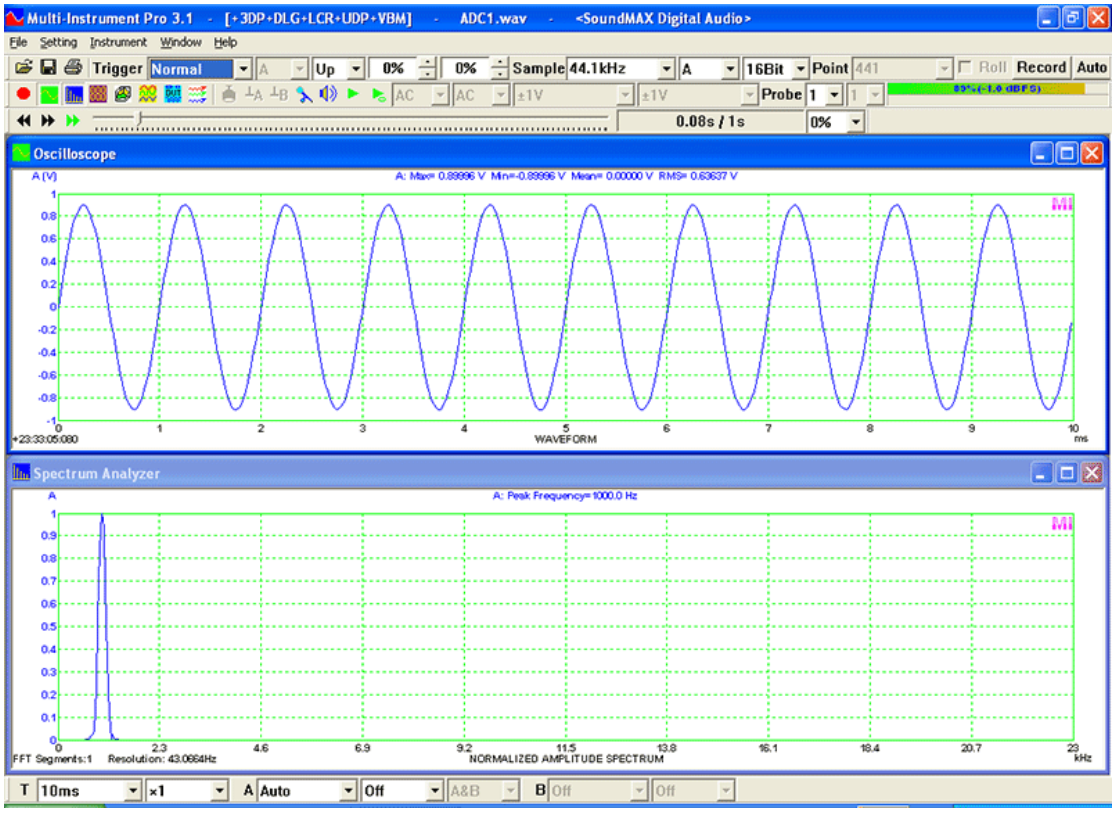


For example (see figure below), if the frame width is 441 points and the WAV file has 44100 points in total, then the file will be split into 100 frames with each frame contains 441 points. Under this mode, the Long Wave File Navigation Toolbar becomes visible. The meaning of each component in this toolbar is listed as follows:



: Frame Up, if pressed, the current frame position will move one frame backward if the beginning of the file has not been reached yet.

: Frame Down, if pressed, the current frame position will move one frame forward if the end of the file has not been reached yet.

: Frame Auto Scroll, a toggle button, if pressed down, the current frame position will move forward frame by frame with or without overlap automatically till the end of the file is reached. Every frame will be analyzed and displayed according to the current settings, and no frame will be skipped. This feature is very useful for performing post-analysis such as Spectrum 3D Plot and Data Logging automatically

on a long WAV file. During the auto scrolling process, the Frame Up and Frame Down buttons are disabled. If the Frame Auto Scroll button is released up, the auto scrolling process will stop, and the Page Up and Page Down buttons will be enabled again.

: Current Frame Position slider. It reflects the position of the current frame with regards to the length of the file. It can also be used to adjust the position of the current frame.

: Current Frame Position/Length of the File. For example, 0.08s/1s means that the current frame starts at 0.08 second and the length of the file is 1 second.

: Frame Overlap Percentage. This parameter specifies the inter-frame overlap percentage and has effect on Frame Down, Frame Up, Frame Auto Scroll commands. It can be set from 0% to 99.9%. As a comparison, the Window Overlap Percentage in the Spectrum Analyzer View Toolbar specifies the intra-frame overlap percentage of FFT segments. You can combine the two overlap percentages to achieve a certain overall overlap scheme for FFT analysis.

During frame navigation, it is possible to save the current frame of data as an individual wave file.

2.6.1.5 Import (File SubMenu) (ALT-F-I)

This command is used to import data from a properly formatted TXT file. The following illustrates the format of the TXT file for single channel and dual-channel cases.

(1) Format for Single Channel:

Example:

```
;Data Points
;Sampling Frequency (Hz) = 44100
;Sampling Bit Resolution (Bits) = 16
;Sampling Channels = 1
;A:Full-scale Voltage (V) = 1
;A:Sensor Sensitivity (V/V) = 1
;Total Data Points = 441
;Digital Channels = 0
1,0
2,0.141968
3,0.281097
……
439,-0.41449
440,-0.281097
441,-0.141968
```

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(2) Format for Dual Channels:

Example:

```
;Data Points
;Sampling Frequency (Hz) = 44100
;Sampling Bit Resolution (Bits) = 16
;Sampling Channels = 2
;A:Full-scale Voltage (V) = 1
;A:Sensor Sensitivity (V/V) = 1
;B:Full-scale Voltage (V) = 1
;B:Sensor Sensitivity (V/V) = 1
;Total Data Points = 441
;Digital Channels = 0
1,0,0
2,0.141968,-0.141968
3,0.281097,-0.281097
……
439,-0.41449,0.41449
440,-0.281097,0.281097
441,-0.141968,0.141968
```

The first a few lines (start with ";") form the header. The values on the right hand side of "=" are important, they specify the Sampling Frequency (Hz), Sampling Bit Resolution (Bits), Number of Sampling Channels (analog), Full-scale Voltage (V), Sensor Sensitivity (V/V), Total Number of Data Points, and Number of Digital Channels. Note that the unit of the Sensor Sensitivity is expressed as [ADC Engineering Unit] / [Sensor Engineering Unit]. The software supports three kinds of ADC Engineering Units, Voltage (V), Ampere (A), and Coulomb (C) depending on the ADC hardware used. The Sensor Engineering Unit can be configured freely.

Sampling Bit Resolution (Bits) can be 8, 16, 24, 32 or 32F where "32F" represents 32-bit floating format in contrast to the other four integer formats. These formats are used by the software to store data internally. The same format will be used if the imported data are saved as a WAV file.

The header lines are followed by the data lines. Each data line contains two (single channel) or three (dual channels) comma separated values. The first variable is the sequential number, and the second and third variables are the data in engineering units for Channel A and Channel B respectively. If an integer format is used in [Sampling Bit Resolution (Bits)], then the absolute values of all the data must be lower than the value of [Full-scale Voltage (Current / Charge)] / [Sensor Sensitivity] specified in the header lines. This is to prevent the "ADC" from being saturated. But this rule does not apply to the 32-bit floating format. The software will still function correctly if the sequential numbers are omitted. It is also possible to import bare data without header lines (see examples below). In this case, the 32-bit floating format will be used automatically with [Full-scale Voltage (V)] =1 and [Sensor Sensitivity (V/V)] = 1.

(3) Bare data for Single Channel (without header lines):

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Example:

0
0.141968
0.281097
……
-0.41449
-0.281097
-0.141968

(4) Bare data for Dual Channels (without header lines):

Example:

0,0
0.141968,-0.141968
0.281097,-0.281097
……
-0.41449,0.41449
-0.281097,0.281097
-0.141968,0.141968

(5) Bare Data with Sampling Time Information for Single Channels (without header lines):

Example:

0.000,0
0.001,0.141968
0.002,0.281097
……
0.997,-0.41449
0.998,-0.281097
0.999,-0.141968

(6) Bare Data with Sampling Time Information for Dual Channels (without header lines):

Example:

0.000,0
0.001,0.141968,0.141968
0.002,0.281097,0.281097
……
0.997,-0.41449,-0.41449
0.998,-0.281097,-0.281097
0.999,-0.141968,-0.141968

Under the above situation, the following dialog box will pop up, requesting additional information about the data, including sampling frequency, sampling bit resolution,

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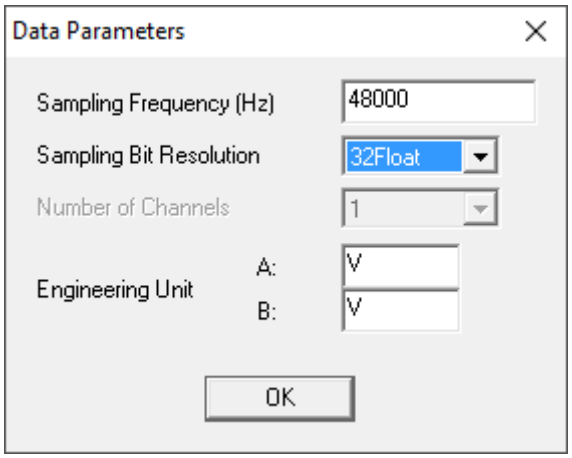
number of channels, and engineering unit. These fields will be automatically populated by the software based on its analysis of the data. The user can either confirm or modify them as needed.



In addition to the standard format described above, the TXT files exported by the Oscilloscope, Spectrum Analyzer (in Auto / Cross Correlation and Impulse Response Modes) and Signal Generator, which have a slightly different format than the aforementioned ones (i.e. one more column for time information or sequential number), can also be imported for analysis.

It is possible to import a TXT file through “Drag and Drop” when the Oscilloscope is not running.

This command will reset the current panel setting after the file is imported.

Some sample TXT files are provided in the WAV directory of the software and can be used as templates.

2.6.1.6 Import with Current Panel Setting (File SubMenu) (ALT-F-H)

This command is similar to the [File]>[Import] command, except that it will try to keep the current panel setting after the file is imported.

2.6.1.7 Combine (File SubMenu) (ALT-F-B)

This command is used to combine data from individual channels of two WAV files. One WAV file must be opened first. Then you can load the data from either channel of the second WAV file and use them to replace/combine the data in either channel of the currently opened WAV file.

WAV files that can be combined must have less than or equal to two channels' data, with the same sampling frequency and sampling bit resolution. The record length of the second WAV file must be less than or equal to that of the first WAV file.

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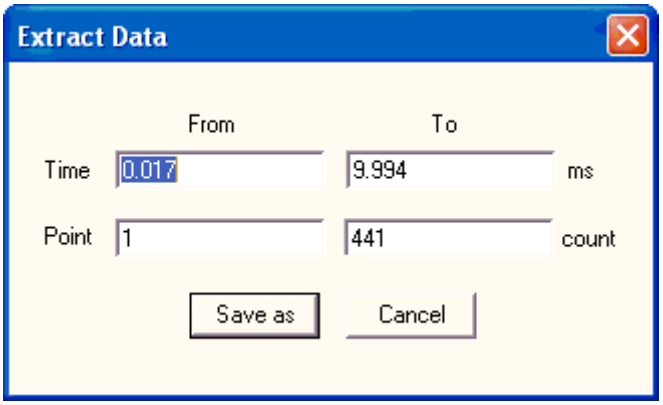
2.6.1.8 Extract (File SubMenu) (ALT-F-T)

This command is used to extract data from the currently opened WAV file and save them into a new WAV file.



You can either specify the time range or the point number range to extract data. The range can also be specified by placing two markers in the Oscilloscope. Pressing "Save as" to store the extracted data to a WAV file.

2.6.1.9 Close (File SubMenu) (ALT-F-C)

This command is used to close an opened document. If the document content has been changed and the change has not yet been saved, a message box will pop up to ask whether you want to save the change or not.

2.6.1.10 Save (File SubMenu) (ALT-F-S, CTRL-S)

This command is used to save an opened document. If the document is new, then you will be prompted to give a file name for the document.

This command is also available through the button in the Sampling Parameter Toolbar. This function is disabled when the document is empty.

2.6.1.11 Save As (File SubMenu) (ALT-F-A)

This command is used to save an opened document with a specified new file name. This function is disabled when the document is empty.

2.6.1.12 Print Screen (File SubMenu) (ALT-F-R)

This command allows you to print the currently displayed content in the main window of the software to a printer.

2.6.1.13 Oscilloscope Export (File SubMenu) (ALT-F-E)

This command is used to export either the measured data to a TXT file or the currently displayed graph to a BMP file. When clicked, a "Save As" window will pop up. You can specify whether you want to export as a TXT file or a BMP file by

selecting "Text File (*.txt)" or "Bitmap File (*.bmp)" in the "Save as type" combo box. The exported text files can be imported into third party software such as Microsoft Excel for further processing and analysis.

This function is disabled when the document is empty.

2.6.1.14 Oscilloscope Print (File SubMenu) (ALT-F-P, CTRL-P)

This command allows you to print the currently displayed graph in the Oscilloscope to a printer. This function is disabled when the document is empty.

2.6.1.15 Oscilloscope Print Preview (File SubMenu) (ALT-F-V)

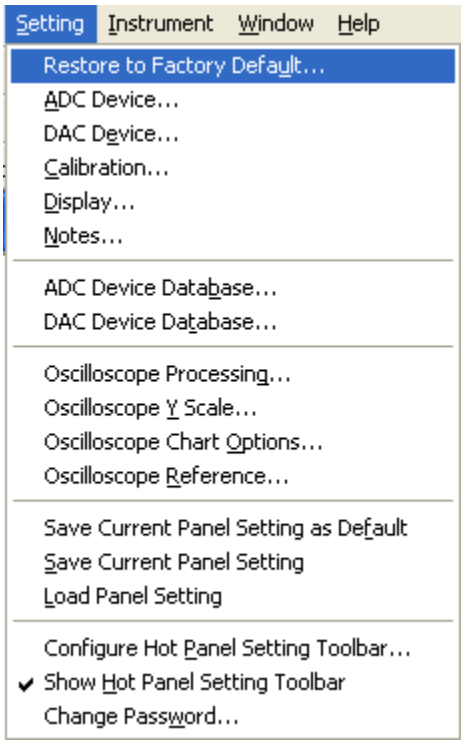
This command allows you to have a preview before printing. This function is disabled when the document is empty.

2.6.1.16 Recent File (File SubMenu)

Up to four recently opened files will be remembered. You can directly open them from the Recent File List.

2.6.1.17 Exit (File SubMenu) (ALT-F-X)
The program will exit upon this command.

2.6.2 Setting SubMenu

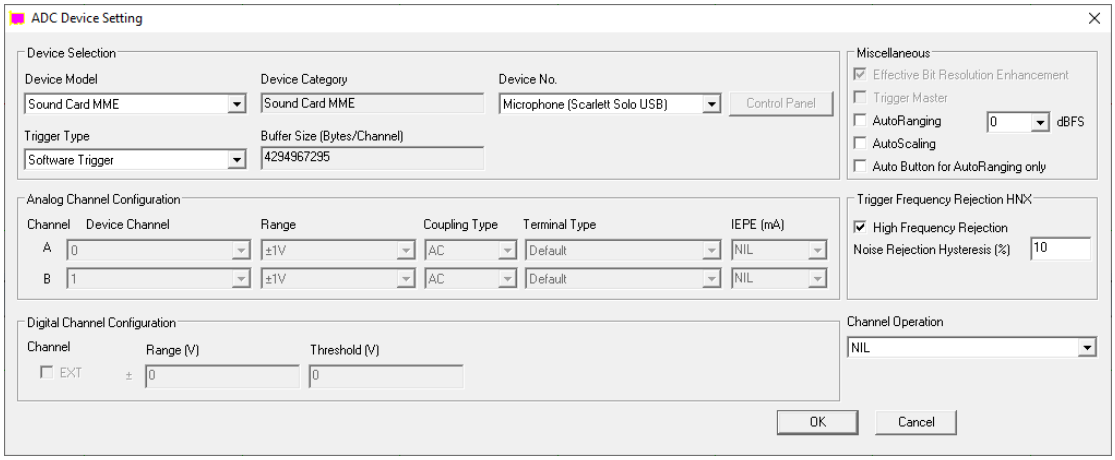


This submenu provides access to various setting functions.

2.6.2.0 Restore to Factory Default (Setting SubMenu) (ALT-S-U)

Refer to Section 1.3.1.1.

2.6.2.1 ADC Device (Setting SubMenu) (ALT-S-A)



This dialog is used to select and configure the current ADC device used by the software. You must make sure the device is already connected to the computer before selection.

The device models available for selection are configured via [Setting]>[ADC Device Database] which will be described later. Once you select a device model in the Device Model combo box, the rest of parameters will be updated accordingly. If multiple devices in the same device category exist in the system, then you can select one of them via the Device No. combo box. You may also need to select the Trigger Type, Device Channel, Range, Coupling Type and Terminal Type, etc., depending on the device model selected. Please refer to Section "ADC Device Database" for detailed description of these items. If the selected ADC device allows enabling / disabling IEPE excitation at the input, then the IEPE selection boxes will be enabled. Select "NIL" to disable IEPE excitation, and select or enter a non-zero excitation current value (in mA) to enable it at the specified value (if the hardware supports that value). Some ADC devices come with their own control panels, such as those with ASIO drivers. If the control panel is available for the selected device, the Control Panel button will be enabled. Clicking it will open the control panel.

The selection for Range and Coupling Type is also available in the Miscellaneous Toolbar in the main window.

Multi-Instrument supports two logic ADC channels: A and B. If the device has more than two physical ADC channels, then you need to assign two of them to the two logic channels.

In addition to the analog input channels, some ADC devices such as VT DSO-2810 and VT DSO-2810E, have digital input channels. You can choose to display the signals from the digital input channels in the oscilloscope view by checking the respective checkboxes. It may be possible to adjust the thresholds for digitizing the

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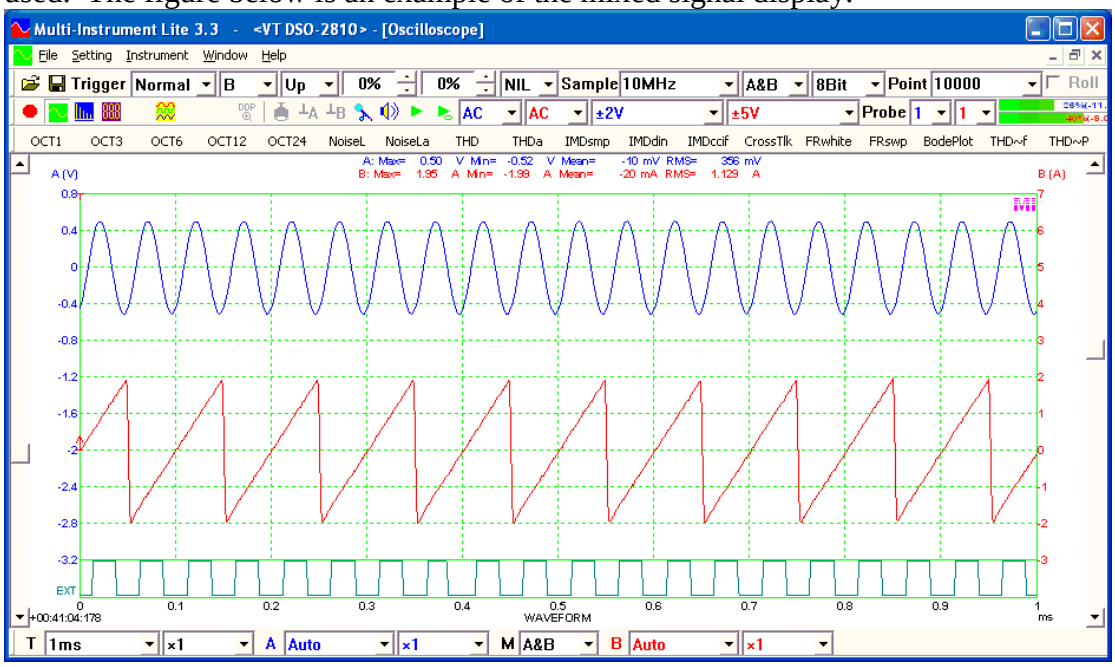
analog signals connected to the digital input channels, depending on the hardware used. The figure below is an example of the mixed signal display.



In the Miscellaneous section of the ADC Device Setting dialog, there are five options:

(1) Effective Bit Resolution Enhancement
(2) Trigger Master
(3) AutoRanging
(4) AutoScaling
(5) Auto Button for AutoRanging only

The above (1) and (2) are hardware-specific options, please refer to the respective hardware manual for details.

The AutoRanging checkbox will be available for selection for those devices that support multiple input ranges. If the AutoRanging checkbox is ticked, the software will monitor the peak level in the sampled data frame continuously and step up/down the input range based on the specified dBFS value. The specified dBFS value will also affect the autoranging function performed by the Auto button in the Sampling Parameter Toolbar in the main window. 0 dBFS is the default value, which is usually good for SNR, but may not be good for other parameters such THD.

If the AutoScaling checkbox is ticked, the software will automatically and continuously adjust the vertical scales of the Oscilloscope, Spectrum Analyzer and Spectrum 3D Plot based on the peak level in the sampled data frame. The difference between AutoRanging and AutoScaling is that AutoRanging changes the hardware gain settings while AutoScaling changes the software display settings.

By default, the Auto button in the Sampling Parameter Toolbar performs on-demand automatic adjustment of sampling rate, record length and input ranges. If the “Auto Button for AutoRanging only” checkbox is ticked, it will perform on-demand autoranging function only.

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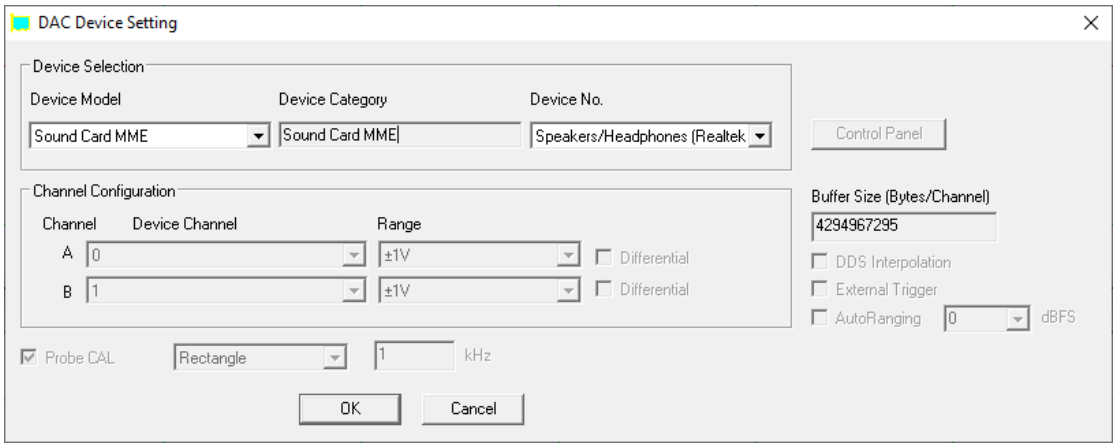
In the Trigger Frequency Rejection HNX section, you can customize the HNX option in the Trigger Frequency Rejection combo box in the Sampling Parameter Toolbar. You can specify (1) whether to incorporate the high frequency rejection (2) the noise rejection hysteresis in percentage. Please refer to the respective hardware manual for details.

Eight options for channel operation are available as follows. They are self-explanatory. These options are useful for some applications. For example, Option (2) can be used to make a real time waveform comparison between a sine wave and its residual harmonics (i.e. its fundamental is removed through digital filtering). Options (3) and (4) can be used to convert a single-ended measurement to a differential measurement and is useful in some I~V measurements. It should be noted that channel operation is performed before ADC range scaling and sensor sensitivity scaling, and thus the two scale factors in both channels must be the same in order for the operation to be meaningful.

(1) Nil
(2) A=iA, B=iA
(3) A=iA-iB, B=iB
(4) A=iA, B=iB-iA
(5) A= (iA+iB)/2, B=(iA-iB)/2
(6) A= (iA+iB)/2, B=iB
(7) A= iA, B=(iA+iB)/2
(8) A=iB, B=iB

The settings here will be applied and saved if the OK button is pressed. The corresponding axis titles in the Oscilloscope window will be heighted if channel operation is configured.

2.6.2.2 DAC Device (Setting SubMenu) (ALT-S-E)



This dialog is used to select and configure the current DAC device used by the software. You must make sure the device is already connected to the computer before selection.

The device models available for selection are configured via [Setting]>[DAC Device Database] which will be described later. Once you select a device model in the Device Model combo box, the rest of parameters will be updated accordingly. If multiple devices in the same device category exist in the system, then you can select one of them via the Device No. combo box. You may also need to select the Device Channel, Range, etc., depending on the device model selected. Please refer to Section "DAC Device Database" for detailed description of these items.

Multi-Instrument supports two logic DAC channels, A and B. If the device has more than two physical DAC channels, then you need to assign two of them to the two logic channels.

Some ADC devices have a Probe CAL output for oscilloscope probe calibration. The Probe CAL signal is normally a 1kHz ~ 10kHz square wave. During probe calibration, the Probe CAL signal is injected into the oscilloscope via the probe. One can fine tune the compensation trimmer on the probe such that the waveform displayed on the oscilloscope screen is "square" and not distorted, i.e. neither over compensated nor under compensated. You can choose to output either Rectangle or MLS signal. When MLS is selected, the frequency value here refers to the clock frequency of the MLS generator rather than the output signal frequency. Normally the Probe CAL signal output is enabled. You can disable the signal output by un-ticking the checkbox. Please refer to the respective hardware manual for details.

The DDS Interpolation and External Trigger options in the above dialog are also hardware specific. A DDS DAC device uses a lookup table (i.e. DDS buffer) to hold the shape of the signal to be generated. The DDS output suffers from the limited number of entries in the lookup table. The output value "jumps" when going from one entry to the next, introducing unwanted high frequencies in the output signal. This adverse effect may not be discernible when the output signal frequency is high (due to the output anti-aliasing low pass filtering and the fact that the limited entries in the DDS buffer are anyway not fully utilized under this situation), but becomes sensible as the output signal frequency goes down. DDS interpolation can be used to fix or alleviate this problem. Instead of using the value stored in the lookup table directly, it dynamically computes the output value through linear interpolation between two successive lookup table values. This effectively enlarges many times the DDS lookup table. Please refer to the respective hardware manual for details.

If the DAC device supports multiple output ranges, then the AutoRanging option will be enabled for selection. If it is selected, the output range will be automatically selected by the software based on the output amplitude specified in the Signal Generator and the specified dBFS value here. 0 dBFS is the default value, which is usually good for SNR, but may not be good for other parameters such THD.

Some DAC devices support both single-ended and differential outputs simultaneously. The Differential option is used to indicate whether the output ranges are specified in differential or single ended configuration. The output amplitudes in the Signal Generator will follow the same configuration.

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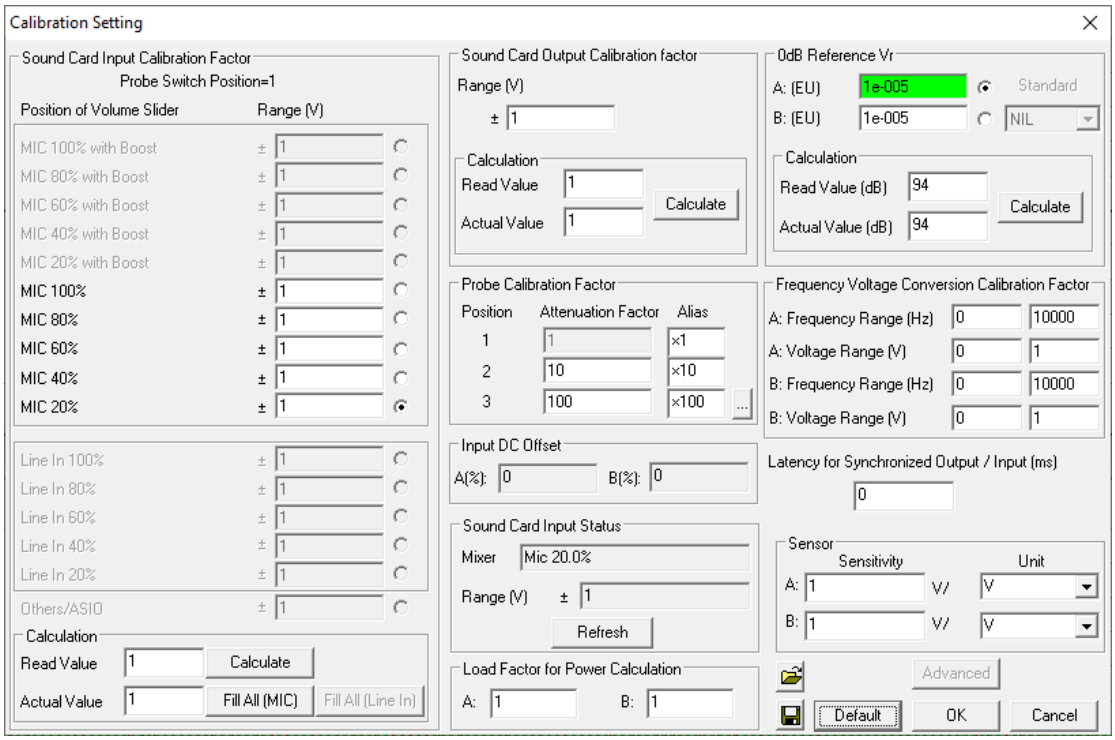
Some DAC devices come with their own control panels, such as those with ASIO drivers. If the control panel is available for the selected device, the Control Panel button will be enabled. Clicking it will open the control panel.

The settings here will be applied and saved if the OK button is pressed.

2.6.2.3 Calibration (Setting SubMenu) (ALT-S-C)



This dialog provides the following parameters/functions:

- Sound Card Input Calibration Factor
- Sound Card Output Calibration Factor
- Probe Calibration Factor
- Input DC Offset Status
- Sound Card Input Status
- 0dB Reference Vr
- Frequency Voltage Conversion Calibration Factor
- Latency for Synchronized Output / Input
- Sensor Sensitivity and Unit
- Load Factor for Power Calculation

During sound card calibration, Automatic Gain Control (AGC) (if any), Bass Boost (if any), Treble Adjustment (if any) and so forth must be disabled to ensure that no artificial modification on the input and output signal.

Most ADC devices measure voltage signals (unit: V), such as sound cards, VT DSOs, VT IEPE-2G05. Some ADC devices measure electric current signals (unit: A). There are also ADC devices that measure electric charge (unit: C), such as VT CAMP-2G05.

ADC devices normally come calibrated, such as VT DSOs, VT IEPE-2G05 and VT CAMP-2G05. Their measurement range can be selected in the ADC Range selection boxes in the Instrument & Miscellaneous Toolbar of the software. However, sound cards are normally not voltage calibrated, and their internal gain adjustment (or range selection, e.g. Mic 0%~100%, Line In 0%~100%, Mic Boost) is different from that of an ordinary ADC Device. This is why Sound Card Input Calibration Factors and internal gain selections are listed separately here.

Sometimes, there is a need to use an external attenuation or amplification circuit (e.g. a ×10 oscilloscope probe, a sound card hardware gain switch or knob) to expand the measurement range of an ADC device. The external circuit is not controllable from the software and thus it effects can only be taken into account by manually setting the Probe Switch Position selection box in the Instrument & Miscellaneous Toolbar. Each probe switch position has a probe attenuation factor associated with it. A value less than 1 represents amplification instead of attenuation.

When a sensor with a known sensitivity is connected to the ADC device or its external attenuation or amplification circuit, the data to be analyzed and displayed will be in the sensor's engineering unit. They will be calculated in the software automatically according to the following formula:

[ADC Device Measured Value] × [Probe Attenuation Factor] / [Sensor Sensitivity]

2.6.2.3.1 Sound Card Input Calibration Factor

You can calibrate the input channels of a sound card by specifying a calibrated ADC full-scale voltage (1/2 peak-to-peak) value, i.e. the "Range" in the Calibration Setting dialog. The ADC full-scale voltage varies with the sound card internal gain, which is continuously adjustable via the Recording Control described previously. It is not practical to calibrate for every possible value of the gain. Instead, the software divides the full range of the gain into five segments (0~20%, 20% ~ 40%, 40% ~ 60%, 60% ~ 80%, 80% ~ 100%) in each category (Microphone with Boost, Microphone, Line In), with an assumption that the gain within each segment is linear. Therefore, at most, you only need to perform calibration at 20%, 40%, 60%, 80%, 100% of the full gain in each category and the ADC full-scale voltage at a particular gain value within a gain segment can be calculated via interpolation or extrapolation without further calibration. The calibration procedure is as follows:

Normal calibration procedure:

(1) Set the sound card internal gain to be calibrated by clicking the corresponding radio button. The software will adjust the gain to that percentage for one time after every such click. The current full-scale voltage value should not be touched.

(2) Input a sine wave (e.g. 1 kHz) with a known amplitude or RMS value to the input channel to be calibrated. Adjust the signal source (not the sound card internal gain) such that the Input Peak Level Indicator shows a value in the range of 80%~95%. This is to ensure a sufficient calibration accuracy. The signal source can be the calibrated Signal Generator of the software itself.

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(3) In the Calculation pad, enter the amplitude/RMS value obtained from the Oscilloscope into the Read Value field, and enter the actual amplitude/RMS value of the signal into the Actual Value field. The actual value can be measured via other instrument such as a conventional multimeter or oscilloscope. Then, press the Calculate button ONCE. The corresponding full-scale voltage field (i.e. the Range field) will be filled automatically.

(4) Repeat (1)~(3) for other gain percentages.

For many sound cards, the gain is linear and Microphone with Boost provides exactly 20 dB's gain over Microphone without Boost. Under this situation or assumption, you only need to calibrate for one gain value and let the software to take care of the rest. This simplifies the calibration procedure as follows:

Simplified calibration procedure for sound cards with linear gain and 20 dB's Mic Boost:

(1) Set the sound card internal gain to be calibrated (i.e. 80%) by clicking the corresponding radio button. The software will adjust the gain to that percentage for one time after every such click. The current full-scale voltage value should not be touched.

(2) Input a sine wave (e.g. 1 kHz) with a known amplitude or RMS value to the input channel to be calibrated. Adjust the signal source (not the sound card internal gain) such that the Input Peak Level Indicator shows a value in the range of 80%~95%. This is to ensure a sufficient calibration accuracy. The signal source can be the calibrated Signal Generator of the software itself.

(3) In the Calculation pad, enter the amplitude/RMS value obtained from the Oscilloscope into the Read Value field, and enter the actual amplitude/RMS value of the signal into the Actual Value field. The actual value can be measured via other instrument such as a conventional multimeter or oscilloscope. Then, press the Calculate button ONCE. The corresponding full-scale voltage field will be filled automatically.

(4) In the Calculation pad, click "Fill All (MIC)" button if Microphone is being calibrated, or click "Fill All (Line In)" button if Line In is being calibrated. All the corresponding full-scale voltage fields (i.e. the Range fields) will be filled automatically.

If the software fails to detect the sound card's input status or ASIO driver is used, then you only need to calibrate the full-scale voltage value for the Other/ASIO field, and bear in mind that any change of the sound card's input gain after calibration may invalidate the calibration.

You can leave the input calibration factors at their factory default values if absolute magnitude measurement is not necessary.

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2.6.2.3.2 Output Calibration Factor

You can calibrate the output channels of a sound card by specifying a DAC full-scale voltage (1/2 peak-to-peak) value for a certain volume setting defined by yourself. The volume setting is adjustable via the Volume Control described previously. Note that the calibration become invalid if the volume setting changes after calibration. The calibration procedure is as follows.

(1) Set the sound card volume to be calibrated via the Volume Control. The current full-scale voltage value should not be touched.

(2) Output a sine wave (e.g. 1 kHz) from the Signal Generator with an amplitude within 80%~100% of the full-scale voltage (un-calibrated). This is to ensure a sufficient calibration accuracy.

(3) In the Calculation pad, enter the above un-calibrated amplitude value into the Read Value field, and enter the actual amplitude value of the output signal into the Actual Value field. The actual value can be measured via other instrument such as a conventional multimeter or oscilloscope. Then, press the Calculate button ONCE. The full-scale voltage field (i.e. the Range field) will be filled automatically. Note that normally a conventional multimeter only displays a RMS value, you need to convert it into an amplitude value via the formula: Amplitude = 1.414 × RMS.

2.6.2.3.3 Probe Calibration Factor

Up to 16 attenuation factors for external probe or attenuation switch positions can be defined. Three of them can be specified in the main calibration page. The rest can be accessed by clicking the "…" button to open the following page. An alias can be defined for each switch position. If the alias is "empty", then this switch position and those after it will not be available in the probe switch position selection boxes in the main window of the software. The configuration of these parameters helps to scale the sampled data properly if the ADC device have gain or attenuation switches not controllable from Multi-Instrument.

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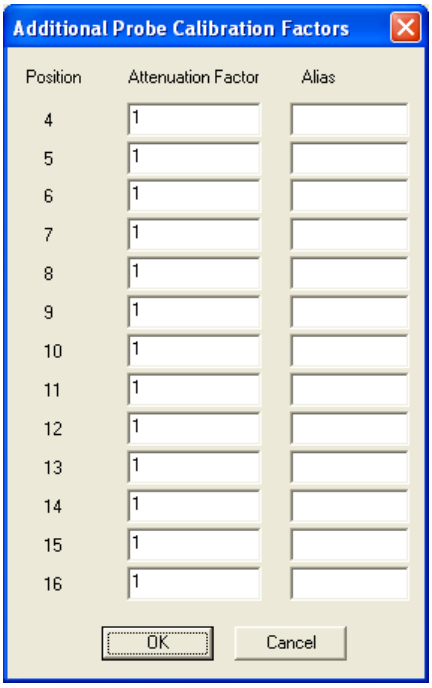

For sound card based systems, you can make the probe by yourself using Section "Input and Output Connection for Sound Card Based System" described previously as a reference. If you want to purchase the probe off-the-shelf, it is highly recommended to use the dedicated sound card oscilloscope probe supplied by Virtins Technology. Please refer to Virtins Technology's website for the specification and calibration procedure of the probe.

2.6.2.3.4 Input DC offset Status

The input DC offsets for Channel A and Channel B are displayed in two read-only edit boxes. They are expressed as a percentage of the ADC full-scale voltage (1/2 Vpp). Ideally, the input DC offsets should be zero. A positive value means that there exists a positive DC offset in that channel and it needs to be corrected by removing that offset from the raw data by the software. The input DC offset is measured using the "Zeroing" button in the Miscellaneous Toolbar. (refer to Section "Zeroing" described previously for details).

The display of this parameter is more for diagnostic purpose.

2.6.2.3.5 0dB Reference Vr

This parameter is used only when displaying dB, dB(A), dB(Z), dB(C) in the Spectrum Analyzer, Spectrum 3D Plot, and Multimeter. You can define your own 0dB reference or calibrate it according to a certain standard such as the standard for sound pressure level.

Sound pressure is the pressure deviation from the local ambient pressure caused by a sound wave. Sound pressure can be measured using a microphone in air and a

hydrophone in water. Sound Pressure Level (SPL) is a logarithmic measure of the RMS sound pressure relative to a reference value, that is:

$$SPL\ (dB) = 20\lg(p_{rms}/p_0)$$

where p_0 is the reference sound pressure and p_{rms} is the RMS sound pressure being measured. The commonly used reference sound pressure in air is 20 μPa (rms). In underwater acoustics, the reference sound pressure is 1 μPa (rms). The "0dB Reference Vr" to be calibrated is the RMS voltage corresponds to the reference sound pressure. It must be calibrated together with the microphone/hydrophone to be used. The calibration procedure is as follows:

(1) Open the Multimeter and set its view type to "dB".

(2) Put a calibrated conventional sound level meter at the same place with the microphone. Note that the weighting profile used by this meter must be the same as the one used by the software. (For better calibration accuracy, a sound level calibrator should be used.)

(3) Generate a reference sound signal (e.g. 1 kHz sine wave). Adjust the volume such that the Input Peak Level Indicator shows a value in the range of 80%~95%. This ensures sufficient calibration accuracy.

(4) In the Calculation pad, enter the sound level obtained from the Multimeter (in dB) into the Read Value field, and enter the actual sound level value obtained from the conventional sound level meter into the Actual Value field. Then, press Calculate button ONCE. The corresponding "0dB Reference Vr" field will be filled automatically.

It should be noted that the input channels should be calibrated first before performing sound level calibration.

In the procedure described above, the Sensor Sensitivity should be kept at 1V/V, and the "0dB Reference Vr" is the one to be calibrated without applying a standard reference value (i.e. "Nil" is selected in the "Standard" combo box).

If both the sensor sensitivity (e.g. in V/Pa) and the input channel (e.g. in V) have been calibrated separately (e.g. before leaving the factory), the "0dB Reference Vr" can be filled directly with a standard reference value without further calibration. For instance, for sound level in air, 0.00002 (Pa) should be used.

If the Sensor Sensitivity is set in V/Pa, the "Standard" combo box will be enabled. It is then possible to select a standard reference in air or water for the "0dB Reference Vr". In this case, the aforementioned procedure can be used to calibrate or recalibrate the sensor sensitivity instead.

2.6.2.3.6 Frequency Voltage Conversion Factor

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This parameter is used only when converting frequency back to voltage in the Multimeter. The software allows you to specify the frequency range and its corresponding voltage range. The relationship between them is linear.

2.6.2.3.7 Latency for Synchronized Output/Input

This parameter is only applicable for the following two synchronized output / input operation modes, i.e. “Sync. No Loopback” and “Sync. iB = oA”, in the Signal Generator.

The latency is the time delay between the time when the Signal Generator is commanded to start and the time when signal output actually starts. The latency needs to be calibrated so that accurate synchronization can be achieved between the Signal Generator and the Oscilloscope.

The calibration procedure is as follows:

(1) Set the latency value to zero during calibration.

(2) Loop back the output Channel A to the input Channel A via a cable with proper attenuation if necessary.

(3) In the Signal Generator, select “Sync. iB = oA” mode, set "Start OSC after (s)" to zero, prepare to output a 1 second’s sine wave at a frequency equal to roughly 1/10 of the sampling rate, e.g. 5kHz at a sampling rate of 48kHz.

(4) In the Oscilloscope, set “Trigger Mode” to “Single”, “Trigger Source” to “A”, “Trigger Edge” to “DF”, “Trigger Level” to about “1%”, “Trigger Delay” to “0%”. The measurement accuracy is directly affected by the “Sweep Time” of the Oscilloscope. The shorter the “Sweep Time”, the higher the accuracy. It should be set to about “400μs” if the sampling rate is 48 kHz. These parameters together with the sound card internal gain should ensure that the Oscilloscope is correctly triggered at the very beginning of the signal.

(5) Stop the Oscilloscope if it is running.

(6) Start the Signal Generator.

(7) In the Oscilloscope, Channel A will display the captured signal which has gone through the hardware (sound card output channel, loop back cable, sound card input channel), and Channel B will display the signal generated by the Signal Generator which has not gone through any hardware. On the bottom left corner of the Oscilloscope window, the time stamp of the first data point in Channel A will be displayed with an accuracy of 1 millisecond. On its right hand side, the time difference between the Channel A and Channel B will be displayed in 1/1000 of a millisecond. This time difference is normally a negative value, which means the data in Channel B is earlier than the data in Channel A. Change the sign of this value and assign it to the Latency being calibrated. Repeat the above procedure for a few times to get the average value.

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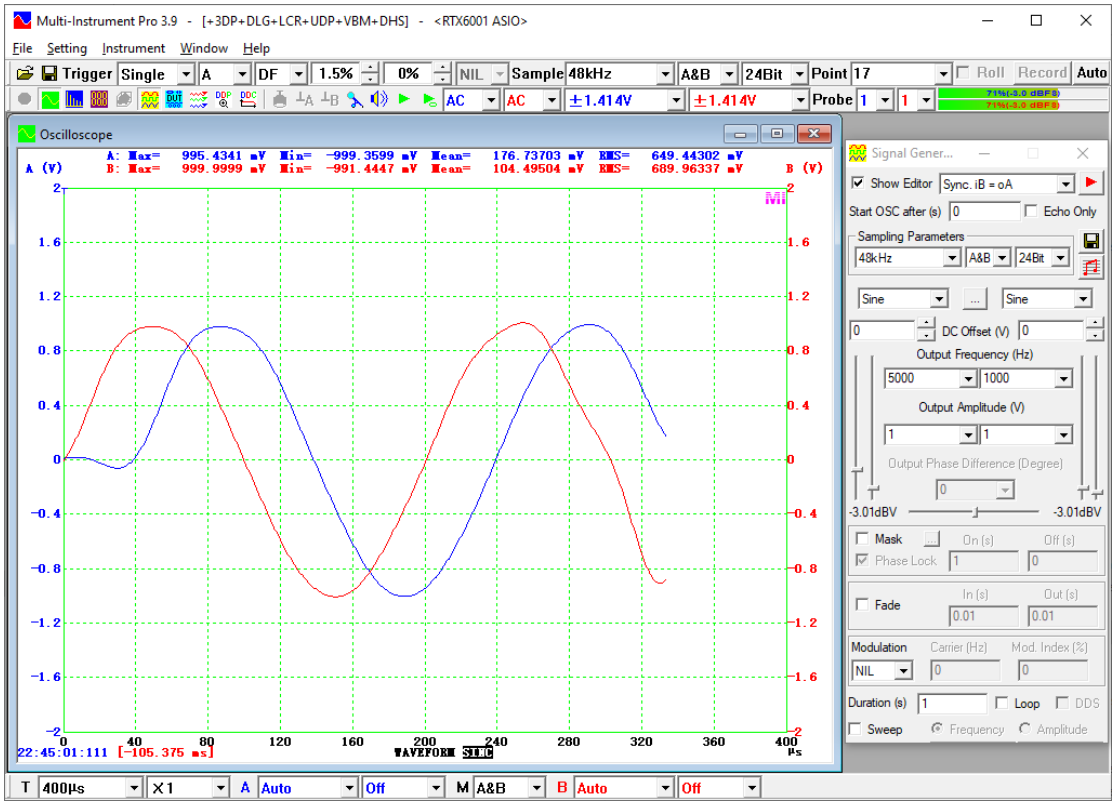
The following figure is an example of the above test.



2.6.2.3.8 Sound Card Input Status

Two statuses will be displayed here:

- Sound card input mixer status, including the input source as well as the gain percentage

- Sound card ADC full-scale voltage (Range) interpolated / extrapolated based on the sound card input calibration factors and the current sound card input mixer status

Note that if you change the sound card's input mixer setting while the Calibration Setting dialog is opened, you can click the Refresh button provided to capture those changes.

2.6.2.3.9 Sensor Sensitivity and Unit

The unit of the Sensor Sensitivity is expressed as [ADC Engineering Unit] / [Sensor Engineering Unit]. The software supports two kinds of ADC Engineering Units, Voltage (V) and Ampere (A), depending on the ADC hardware used. The Sensor Engineering Unit can be configured according to the sensor used. The software provides 12 pre-configured options for the Sensor Engineering Unit: V (Voltage), A (Ampere), g (for acceleration), m/s^2 (for acceleration), m/s (for velocity), m (for

displacement), i/s (for velocity in English unit), i (for displacement in English unit), Pa (for pressure), C (for electric charge), N (for force), lbf (for force in English unit). You can also enter your own sensor unit directly into the Unit edit box.

The value of the Sensor Sensitivity can be entered directly into the Sensitivity edit box.

2.6.2.3.10 Load Factor for Power Calculation

There are a few power related Derived Data Points in the software (Please refer to the chapter for Data Logger for details.). The power of the signal is calculated by:

$$\text{Power} = \text{RMS}^2 / [\text{Load Factor}]$$

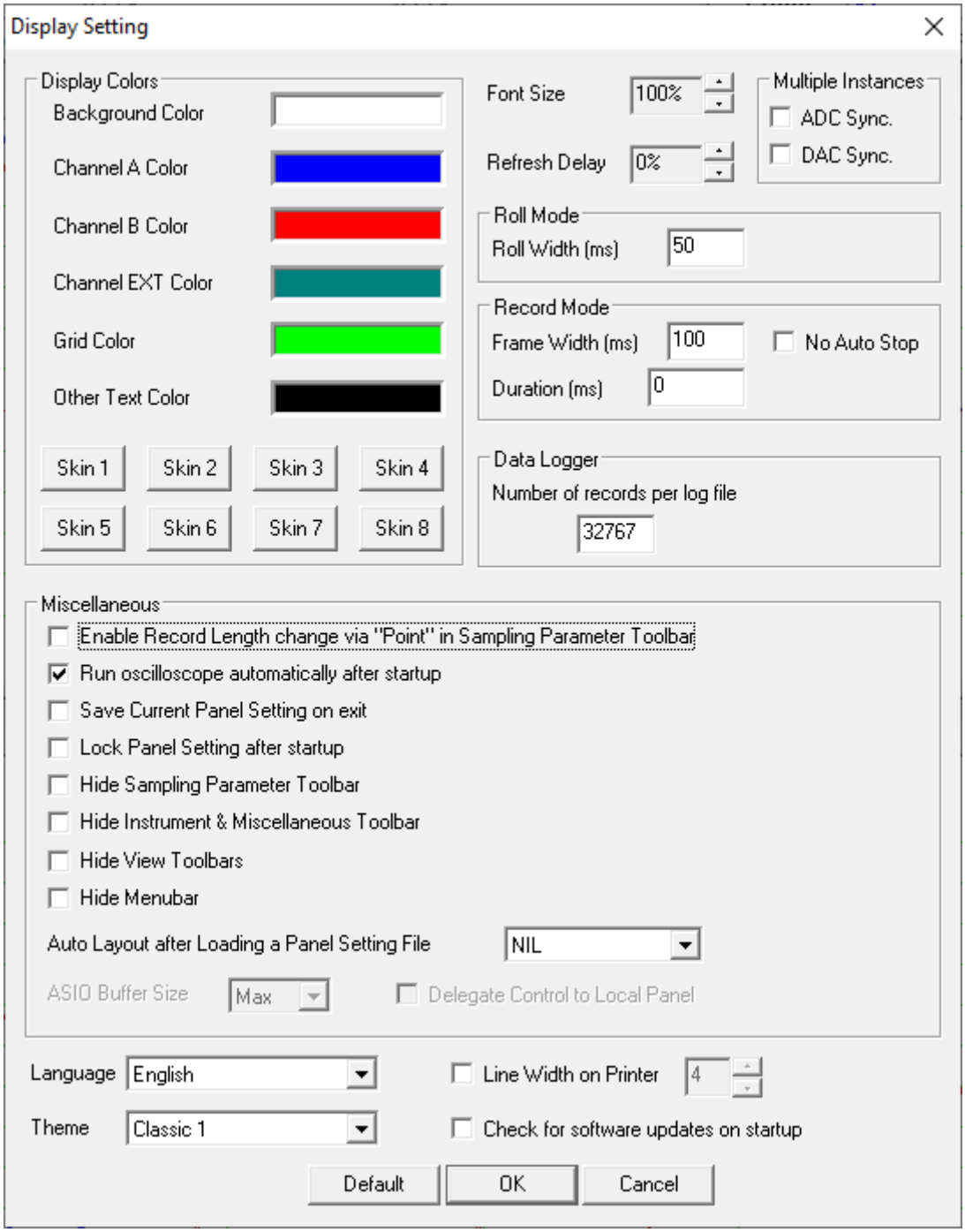
For example, if you measure the voltage across a resistor, then you can get the power consumed by the resistor from the corresponding Derived Data Point by entering the resistor's value into the load factor field in the dialog; If you measure the current through a resistor, then you can get the power consumed by the resistor from the corresponding Derived Data Point by entering the reciprocal of the resistor's value into the load factor field.

2.6.2.3.11 Others

The functions of the buttons in the dialog are as follows:

- : load a previously saved calibration file (*.cal).
- : Save the current calibration data into a calibration file (*.cal).
- Default: All parameters will be filled with the default values.
- OK: Apply and save the changes and close the dialog.
- Cancel: Cancel the changes and close the dialog.
- Advanced: Access to advanced hardware specific calibration function. Please refer to the respective hardware manual for details.

2.6.2.4 Display (Setting SubMenu) (ALT-S-D)



2.6.2.4.1 Display Colors

For all views, the following display parameters are configurable:

- Background Color
- Color of data curve for Channel A
- Color of data curve for Channel B
- Color of data curve for Channel EXT (digital input channel)
- Grid Color
- Other Text Color, such as the color for notes and horizontal axis label, etc.

Clicking on the color box will bring up a color selection window which allows you to select the color you want. Skin 1 ~ Skin 8 are preconfigured color schemes.

2.6.2.4.2 Miscellaneous

- Enable Record Length change via "Point" in Sampling Parameter Toolbar
  If this checkbox is ticked, it allows advanced user to change the Record Length directly. This is sometimes useful, e.g. when you want the Record Length to be an integer multiple of the FFT size.

  This checkbox is un-ticked by factory default.

- Run oscilloscope automatically after startup
  If ticked, the oscilloscope will run automatically with the default panel setting just after startup.

- Save Current Panel Setting on exit
  If ticked, the current panel setting will be saved automatically as the default panel setting when you exit the program.

- Lock Panel Setting after startup
  If ticked, the panel setting will be locked just after startup.

- Hide Sampling Parameter Toolbar
  If ticked, the Sampling Parameter Toolbar will be hidden.

- Hide Instrument & Miscellaneous Toolbar
  If ticked, the Instrument & Miscellaneous Toolbar will be hidden.

- Hide View Toolbars
  If ticked, all View Parameter Toolbars will be hidden.

- Hide Menubar
  If ticked, the menubar will be hidden. To access [Setting]>[Display] when the menu bar is hidden, press CTRL+BREAK or CTRL+B.

- Auto Layout after Loading a Panel Setting File
  Three options are available:
  (1) Nil: No auto layout, i.e. retain the original layout specified in the PSF file. A PSF file can be saved with Nil, Tile Horizontally, Tile Vertically.
  (2) Tile Horizontally: Equivalent to [Window]>[Tile Horizontally]
  (3) Tile Vertically: Equivalent to [Window]>[Tile Vertically]

- ASIO Buffer Size
  Selection of ASIO Buffer Size is enabled when ASIO driver is used and the ADC and DAC have not been started. Three options are available: Auto, Max, Min, Max should be used normally to avoid dropouts in the input and output signals.

- Delegate Control to Local Panel

Some devices such as RTX6001 have their own local control panels and support both local and remote (PC) control modes. This option is used to delegate the control from PC to local control panel. Under local control mode, Multi-Instrument will still be able to scale the input and output data properly according to the settings on the local panel, however, all the controls (such as range selection) from PC will not be disabled.

2.6.2.4.3 Language

You can change the language of the software user interface to your preferred language. The software supports Multilingual User Interface under Windows 2000, XP, 2003, Vista, 7, 8, 10 and above. Currently supported languages are English, French, German, Italian, Portuguese, Spanish, Russian, Simplified Chinese, Traditional Chinese, Japanese and Korean. The language change will take effect after your restart the program.

2.6.2.4.4 Font Size

This parameter is used to adjust the font size of texts displayed in each view.

2.6.2.4.5 Refresh Delay

This parameter is used to adjust the delay time after the data analysis and display of the current frame of data is finished and before the software starts to acquire the next frame of data. To obtain the fastest screen refresh rate, set the Refresh Delay to 0%.

2.6.2.4.6 Roll Width under Roll Mode

This parameter specifies the Roll Width under the Roll Mode. Depending on the ADC device used, it is expressed in either ms (millisecond) or pts (sampling points). Please refer to the section for Roll Mode for more information.

2.6.2.4.7 Frame Width, Duration, No Auto Stop under Record Mode

These three parameters specify the Frame Width, Duration and No Auto Stop option under the Record Mode. Depending on the ADC device used, the Frame Width is expressed in either ms (millisecond) or pts (sampling points) while the Duration is expressed in ms (millisecond) only. Frame Width specifies the buffer size of streaming mode and Duration specifies the total time length of data to be recorded per file. When Duration is 0, the recording process will only be stopped manually by pressing the Record button again or after 2G bytes has been recorded. If “No Auto Stop” is ticked, the recording process will record data into multiple files sequentially and can only be stopped manually. Please refer to the section for Record Mode for more information.

2.6.2.4.8 Number of Records per Log File

This parameter specifies the number of records in a Data Logger log file. (Refer to Section 6.2.4)

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2.6.2.4.9 Line Width on Printer

Printers usually have a finer resolution than computer screens and thus the line width on printer should usually be thicker than that on screen in order to make a graph look similar on both devices. If the option here is ticked, you can assign a fixed line width for all graphs plotted on printer. Otherwise, the line width on printer will be automatically determined by:

[Line Width on Printer] = [Line Width on Screen] × [Printer Horizontal Resolution] / [Graph Window Width]

The Line Width on Screen will be introduced later.

2.6.2.4.10 Theme

This selection box can be used to select a theme (visual style) for the software. The change will take effect after software restart.

2.6.2.4.11 Check for Software Updates on Startup

If this check box is ticked, the software will check online to see if a new software version is available for download during startup. If yes, it will prompt you for confirmation for download. Please refer to [Help]>[Check for Software Updates] for more information.

2.6.2.4.12 Synchronization among Multiple Instances

- ADC Sync.
  When there are multiple instances of the software on the same computer, those instances with this checkbox ticked will have their Oscilloscope synchronized to start and stop.

- DAC Sync.
  When there are multiple instances of the software on the same computer, those instances with this checkbox ticked will have their Signal Generator synchronized to start and stop.

2.6.2.4.13 Others

- Default: All parameters will be filled with the default values.
- OK: Apply and save the changes and close the dialog.
- Cancel: Cancel the changes and close the dialog.

2.6.2.5 Note (Setting SubMenu) (ALT-S-N)

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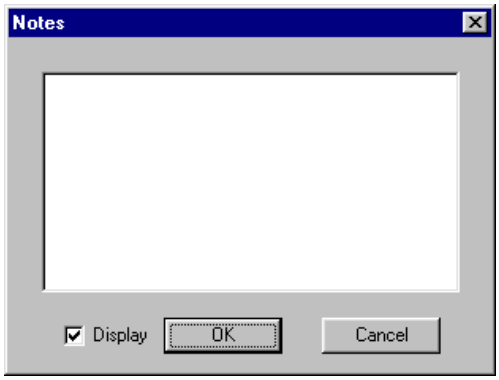
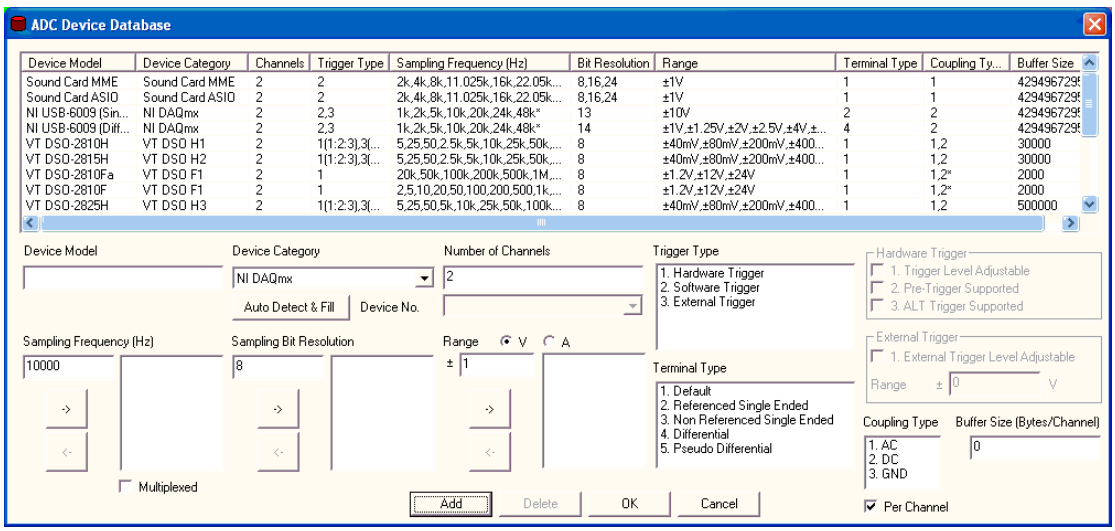


You can write down some notes for a measurement. The notes will be displayed if you tick the "Display" checkbox. The notes will persist in the WAV file if saved.

2.6.2.6 ADC Device Database (Setting SubMenu) (ALT-S-B)



Multi-Instrument is able to interface to many ADC and DAC devices including sound cards based on the standard data acquisition software interface specification developed by Virtins Technology (vtDAQ® for ADC and vtDAO® for DAC). For each category of devices, an intermediate interface DLL (dynamic link library) needs to be developed according to this standard interface specification to bridge Multi-Instrument and the device's original driver or software interface. The software can work with any device as long as the corresponding intermediate interface DLL is provided. One interface DLL should contain either the ADC functions or DAC functions, but not both if possible, even if all of these functions are supported by one single device. This is to ensure that the ADC and DAC devices can be selected independently in Multi-Instrument. For example, you can run a DSO (Digital Storage Oscilloscope) hardware for ADC and the sound card for DAC simultaneously.

For details of the vtDAQ and vtDAO interface specifications, please refer to: *vtDAQ and vtDAO_Interfaces*

Download link:

https://www.virtins.com/vtDAQ-and-vtDAO-Interfaces.pdf

The ADC Device database contains a list of the supported ADC device models as well as their specifications. You can add or remove the supported device models and modify their specifications using this database configuration dialog.

2.6.2.6.1 Device Category

The ADC interface DLLs for the following categories of devices are provided in the software package:

- Sound Cards with MME driver
  All Windows compatible sound cards fall into this category.

- Sound Cards with ASIO driver
  More and more sound cards, especially those used in the Pro-Audio field, support ASIO (Audio Stream Input / Output) driver in addition to MME driver. Some sound cards may have different functions / performance with their ASIO drivers than with their MME drivers. MME driver and ASIO driver are exclusive with each other. You can use only one of them at any time for a particular sound card. Unlike other device categories, a single interface DLL is provided for both the ADC and DAC functions for the ASIO driver due to the fact that the ADC and DAC for the same sound card must work synchronously within the ASIO driver. This implies that in Multi-Instrument, if you use the ASIO driver for ADC, you must NOT use the MME driver for DAC for the same sound card, and vice versa.

- NI DAQmx cards
  Many data acquisition cards from National Instruments fall into this category.

- VT DSO H1
  Type H1 DSO cards from Virtins Technology. Please refer to the hardware manual provided separately.

- VT DSO F1
  Type F1 DSO cards from Virtins Technology. Please refer to the hardware manual provided separately.

- VT DSO H2
  Type H2 DSO cards from Virtins Technology. Please refer to the hardware manual provided separately.

- VT DSO H3
  Type H3 DSO cards from Virtins Technology. Please refer to the hardware manual provided separately.

- VT DAQ 1
  Type 1 DAQ cards from Virtins Technology. Please refer to the hardware manual provided separately.

- VT DAQ 2

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Type 2 DAQ cards from Virtins Technology. Please refer to the hardware manual provided separately.

- My DAQ Device
  User definable DAQ device. If you want to allow Multi-Instrument to interface to your own DAQ device, then you can develop your own MyDAQ.dll according to vtDAQ interface specifications. Please refer to *vtDAQ and vtDAO Interfaces* provided separately.

2.6.2.6.2 Device Model

Each device category may contain one or more device models with possibly different hardware specifications:

- Device Model
  You can specify a Device Model name. Note that if a Device Model name is ended with "*", it means that it supports enabling / disabling IEPE excitation.

- Number of Channels

- Trigger Type
  Three types of trigger are supported:
  - ✓ Hardware Trigger
    You can further specify:
    - ➢ whether the Trigger Level is adjustable
    - ➢ whether Pre-Trigger is supported
    - ➢ whether ALT-Trigger is supported

  - ✓ Software Trigger
    Software Trigger is possible for those ADC devices that support continuous streaming, such as sound cards.

  - ✓ External Trigger
    You can further specify whether the External Trigger Level is adjustable and its range.

- Sampling Frequency
  You can add up to 32 sampling frequencies for each device model. You can further specify whether the maximum sampling frequency entered should be shared among the channels used (i.e. Multiplexed).

- Sampling Bit Resolution
  You can add up to 32 sampling bit resolutions for each device model. Note that the sampling bit resolution here refers to the original sampling bit resolution of the ADC device. It will be converted to 8, 16, 24, 32 bit by the interface DLL and thus only 8, 16, 24, 32 bit will be available for selection in the Sampling Parameter Toolbar.

- Range

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You can add up to 32 ADC ranges for each device model. Two types of ADC devices are supported: Analog Voltage to Digital Conversion, Analog Current to Digital Conversion, with respective engineering units V (Voltage) and A (Ampere). The ADC range must be symmetric with regards to zero, i.e. it must be ± xxx.

- Terminal Type
  Five options are available: Default, Referenced Single End, Non Referenced Single End, Differential, Pseudo Differential. You can choose more than one of them for each device model.

- Coupling Type
  Three options are available: AC, DC, GND. You can choose more than one of them for each device model. If the coupling type of each channel can be changed independently, then you should tick the Per Channel checkbox.

- Buffer Size
  You can specify the buffer size per channel for each device model. The unit of this parameter is Bytes/Channel. If the device supports continuous streaming, then the number of continuous raw data points the ADC device can provide is not limited by the physical buffer size of the device, and you should set the Buffer Size to the maximum value 4294967295.

Note that the above specifications must be filled according to the ADC device's hardware specifications.

2.6.2.6.3 Auto Detect & Fill

Some ADC devices can provide their hardware specification information through software interface when they are connected to the computer. The Auto Detect & Fill button is used to attempt to acquire the hardware information and fill the above hardware specification fields as many as possible when the device is connected to the computer. You still need to fill those blank fields (if any) manually and amend the auto-filled fields manually if necessary.

The Device No. combo box allows you to choose one device from a list of devices in the device category present in the system.

Once you select a device category in the Device Category combo box, the rest of parameters will be updated accordingly as if the Auto Detect & Fill button has been pressed once.

2.6.2.6.4 Others

- Add: Add a new device model with the set specifications into the ADC device database.
- Delete: Delete a selected device model from the ADC device database. Note that the device models labeled "Sound Card MME" and "Sound Card ASIO" are not allowed to be deleted.

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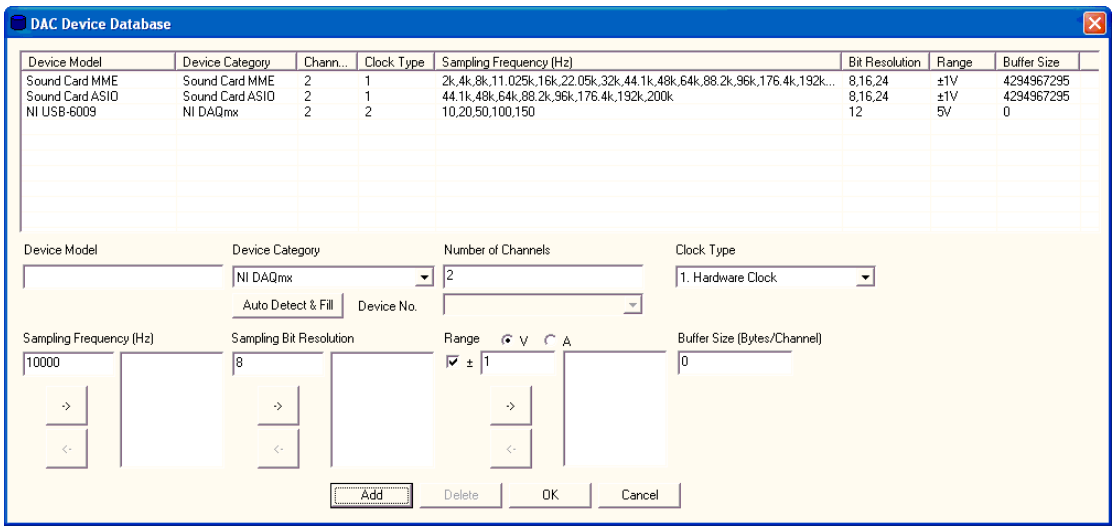
- OK: Save the changes to the ADC device database and close the dialog.
- Cancel: Cancel the changes to the ADC device database and close the dialog.

2.6.2.7 DAC Device Database (Setting SubMenu) (ALT-S-T)



The DAC Device database contains a list of the supported DAC device models as well as their specifications. You can add or remove the supported device models and modify their specifications using this database configuration dialog.

2.6.2.7.1 Device Category

The DAC interface DLLs for the following categories of devices are provided in the software package:

- Sound Cards with MME driver
  All Windows compatible sound cards fall into this category.

- Sound Cards with ASIO driver
  More and more sound cards, especially those used in the Pro-Audio field, support ASIO (Audio Stream Input / Output) driver in addition to MME driver. Some sound cards may have different functions / performance with their ASIO drivers than with their MME drivers. MME driver and ASIO driver are exclusive with each other, and you can use only one of them at any time for a particular sound card. Unlike other device categories, a single interface DLL is provided for both the ADC and DAC functions for the ASIO driver due to the fact that the ADC and DAC for the same sound card must work synchronously within the ASIO driver. This implies that in Multi-Instrument, if you use the ASIO driver for ADC, you must NOT use the MME driver for DAC for the same sound card, and vice versa.

- NI DAQmx cards
  Many data acquisition cards from National Instruments fall into this category.

- VT DAO 1

Type 1 DAO cards from Virtins Technology. Please refer to the hardware manual provided separately.

- My DAO Device
  User definable DAO device. If you want to allow Multi-Instrument to interface to your own DAO device, then you can develop your own MyDAO.dll according to vtDAO interface specifications. Please refer to *vtDAQ and vtDAO Interfaces* provided separately.

2.6.2.7.2 Device Model

Each device category may contain one or more device models with possibly different hardware specifications:

- Device Model
  You can specify a Device Model name.

- Number of Channels

- Clock Type
  Two clock types are supported:
  - ✓ Hardware Clock
    The sampling frequency is controlled by the hardware.

  - ✓ Software Timed
    The sampling frequency is controlled by the software.

- Sampling Frequency
  You can add up to 32 sampling frequencies for each device model.

- Sampling Bit Resolution
  You can add up to 32 sampling bit resolutions for each device model. Note that the sampling bit resolution here refers to the original sampling bit resolution of the DAC device. It will be converted to 8, 16, 24, 32 bit by the interface DLL and thus only 8, 16, 24, 32 bit will be available for selection in the Signal Generator.

- Range
  You can add up to 32 DAC ranges for each device model. Two types of DAC devices are supported: Digital to Analog Voltage Conversion, Digital to Analog Current Conversion, with respective engineering units V (Voltage) and A (Ampere). The DAC range can be either symmetric with regards to zero, i.e. ± xxx, or in the range from 0 to a positive value, i.e. xxx. You should tick the ± checkbox for the former case.

- Buffer Size
  You can specify the buffer size per channel for each device model. The unit of this parameter is Bytes/Channel. If the device supports continuous streaming, then the number of continuous raw data points the DAC device can provide is not limited by the physical buffer size of the device, and you should set the Buffer Size to the maximum value of 4294967295.

Note that the above specifications must be filled according to the DAC device's hardware specifications.

2.6.2.7.3 Auto Detect & Fill

Some DAC devices can provide their hardware specification information through software interface when they are connected to the computer. The Auto Detect & Fill button is used to attempt to acquire the hardware information and fill the above hardware specification fields as many as possible when the device is connected to the computer. You still need to fill those blank fields (if any) manually and amend the auto-filled fields manually if necessary.

The Device No. combo box allows you to choose one device from a list of devices in the same device category present in the system.

Once you select a device category in the Device Category combo box, the rest of parameters will be updated accordingly as if the Auto Detect & Fill button has been pressed once.

2.6.2.7.4 Others

- Add: Add a new device model with the set specifications into the DAC device database.
- Delete: Delete a selected device model from the DAC device database. Note that the device models labeled "Sound Card MME" and "Sound Card ASIO" are not allowed to be deleted.
- OK: Save the changes to the DAC device database and close the dialog.
- Cancel: Cancel the changes to the DAC device database and close the dialog.

2.6.2.8 Oscilloscope Processing (Setting SubMenu) (ALT-S-G)

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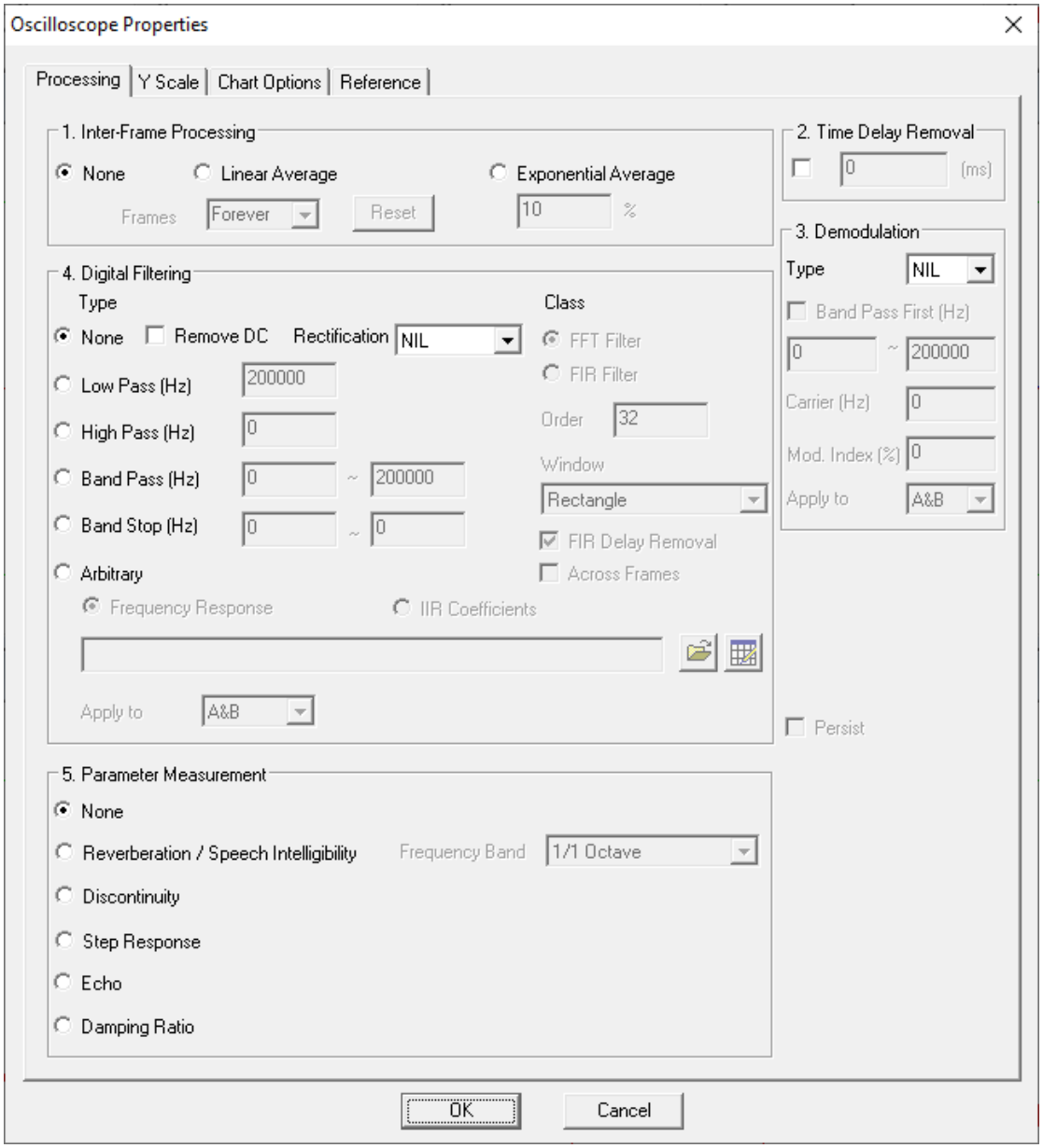

A signal is acquired by the Oscilloscope frame by frame. The sweep time of the Oscilloscope is sometimes called Frame Width or Record Length. A data frame contains multiple samples. These samples are normally continuous within the data frame. But there may be a gap between two adjacent data frames. The acquired frame of data will undergo the following processes in time domain sequentially:

(1) Inter-frame processing
(2) Intra-frame processing
(3) Parameter measurement

Then the processed signal is passed to the Spectrum Analyzer for further processing and analysis in frequency domain.

"Persist" must be chosen if you want to persist the above inter-frame and intra-frame processing results in the raw data. Affected commands are: File Save, File Save as, Oscilloscope Export, File Extract, Play, Cyclic Play.

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2.6.2.8.1 Inter-frame Processing

Inter-frame processing in time domain is performed before intra-frame processing. There are three options: None, Linear Average and Exponential Average. Averaging is useful for removing noises from a repetitive signal. Inter-frame Averaging in time domain must be synchronized in order to obtain meaningful results. Synchronization can be achieved through proper triggering during data acquisition. The number of frame averaged is displayed at the bottom left corner of the Oscilloscope window. It can be accessed through DDP: NumberOfFrames.

2.6.2.8.1.1 Linear Average

If Linear Average is selected, the Oscilloscope will keep track of each data frame acquired and only display the averaged waveform of a specified number of data frames acquired most recently. The number of frames averaged will be displayed at the bottom-left corner of the Oscilloscope view.

You can specify the number of contiguous frames to be processed. The available options are: 2, 3, 4, 5, 6, 7, 8, 9, 10, 20, 30, 40, 50, 60, 70, 80, 90, 100, 120, 140, 160, 180, 200 and forever. You can also enter any number between 1 and 200 manually. When "Forever" is selected, you can reset the process using the Reset button when necessary.

2.6.2.8.1.2 Exponential Average

Unlike Linear Average where all data frames used for average are given equal weights, in Exponential Average, the weighting factor for each data frame decreases with time exponentially, giving much more importance to recent observations while still not discarding older observations entirely. The degree of weighting decrease is expressed as a constant α in percentage. The greater the α, the faster the decrease. Alternatively, α may be expressed in terms of N, where $\alpha = 2/(N+1)$, and $N \times$ [time interval between the start times of two adjacent data frames] is called the time constant.

2.6.2.8.2 Intra-frame Processing

Intra-frame processing in time domain is performed after inter-frame processing. The frame of data undergoes the following processes sequentially:

(1) Time Delay Removal
(2) Demodulation: Nil, AM, FM or PM
(3) Remove DC, Rectification (or Detection) and then Digital Filtering

2.6.2.8.2.1 Time Delay Removal

Some measurements require the acquired signals in the two input channels to be time aligned. This is usually not an issue if the signals travel within electric circuits without any intentional digital delay, due to the lightning fast travelling speed of electric signal in circuits. Time Delay Removal may be required for those systems

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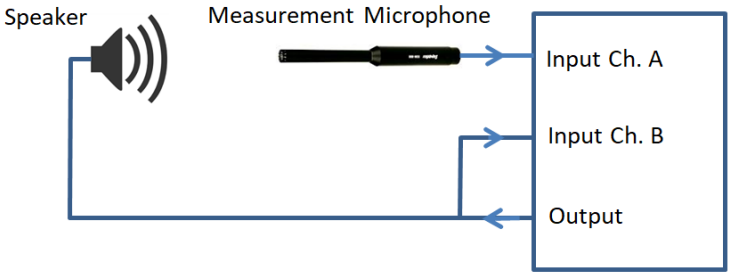
that contain transducers which convert electric signals to other forms of energy (e.g. sound) or vice versa. These systems usually involve a signal travelling path in which the signal travelling time cannot be neglected in the measurements where time alignment between the two input channels is required. This is the case for acoustic transfer function measurement using dual-channel FFT (see figure below). In this measurement, the signal output by the signal generator will go through two different paths: one is connected directly to the input Channel B, the other contains an acoustic path from the speaker to the microphone before going into the input Channel A. As the sound travels much slower than the electric signal, the signal received in Channel A will be delayed considerably compared with that in Channel B. Thus, it is normally required to remove this delay in acoustic transfer function measurement as well as Coherence / Non-Coherence function measurement.



To remove the time delay in Channel A, tick the Time Delay Removal option and enter the time delay value (in ms) in Channel A with respect to Channel B. A positive value means that the signal in Channel A is delayed and vice versa. If the time delay is positive, after the time delay removal, the data in Channel A will be shifted left by the time delay value so as to be time aligned with the data in Channel B. Then the right most part of the data in Channels A & B with a length equal to the time delay will be reset to zero. On the other hand, if the time delay is negative, after the time delay removal, the data in Channel B will be shifted left by the time delay value so as to be time aligned with the data in Channel A. Again, the right most part of the data in Channels A & B with a length equal to the time delay will then be reset to zero. Keep this in mind when configuring your own signal processing algorithm in the software.

In case the time delay between the two input channels is not readily available through calculation, it can be measured using quite a few methods. Cross-correlation using white noise, MLS or pink noise as stimulus is recommended. Cross-correlation function will be introduced later.

The following two figures show the time delay measured using cross-correlation with white noise before and after the time delay removal, respectively. It should be noted that in the cross-correlation measurement, a positive time delay value means that the signal in Channel A is delayed and vice versa. In this example, 2.979 ms is measured in the cross correlation and thus 2.979 ms should be entered in the Time Delay Removal edit box here. After this time delay removal, the cross correlation measures a time delay of 0 μs meaning that the data in both channels are perfectly time aligned.

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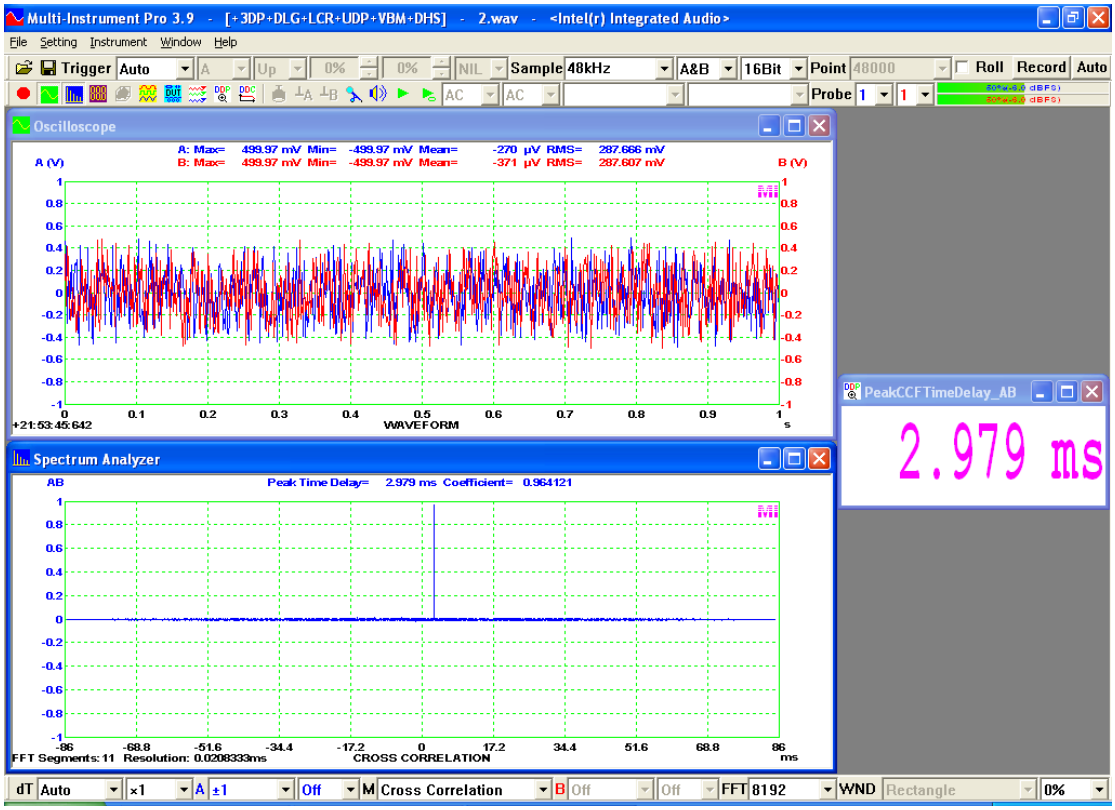
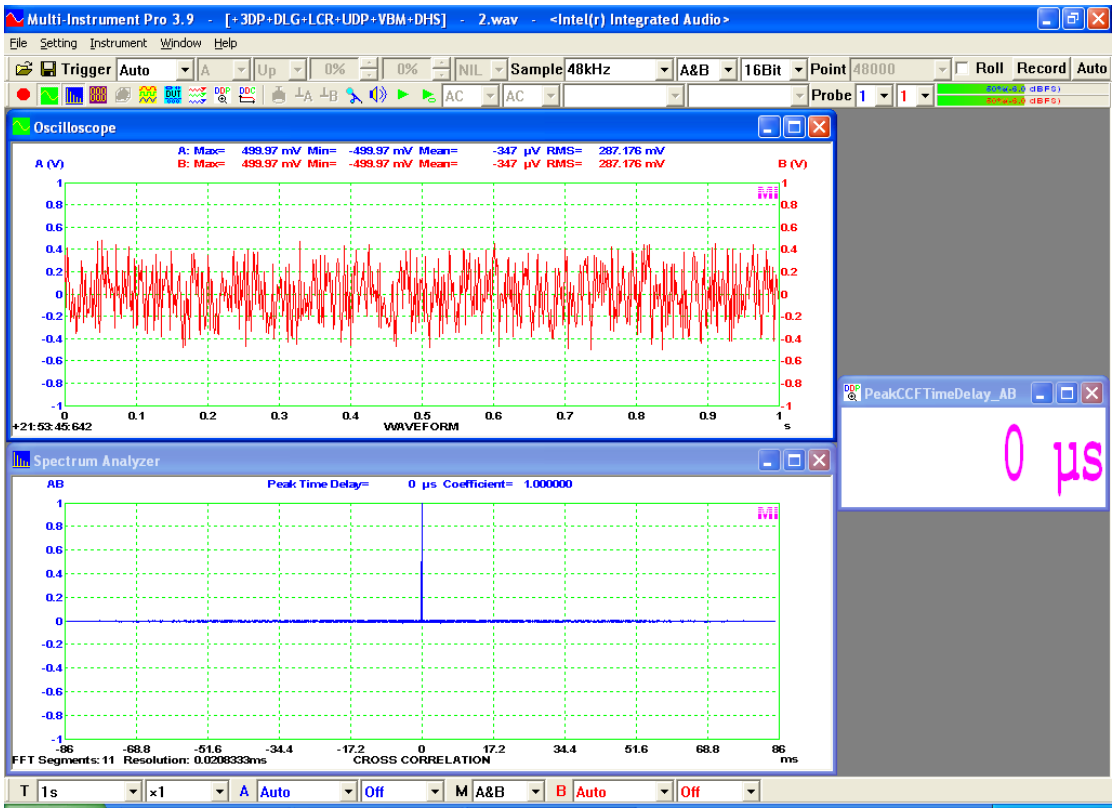




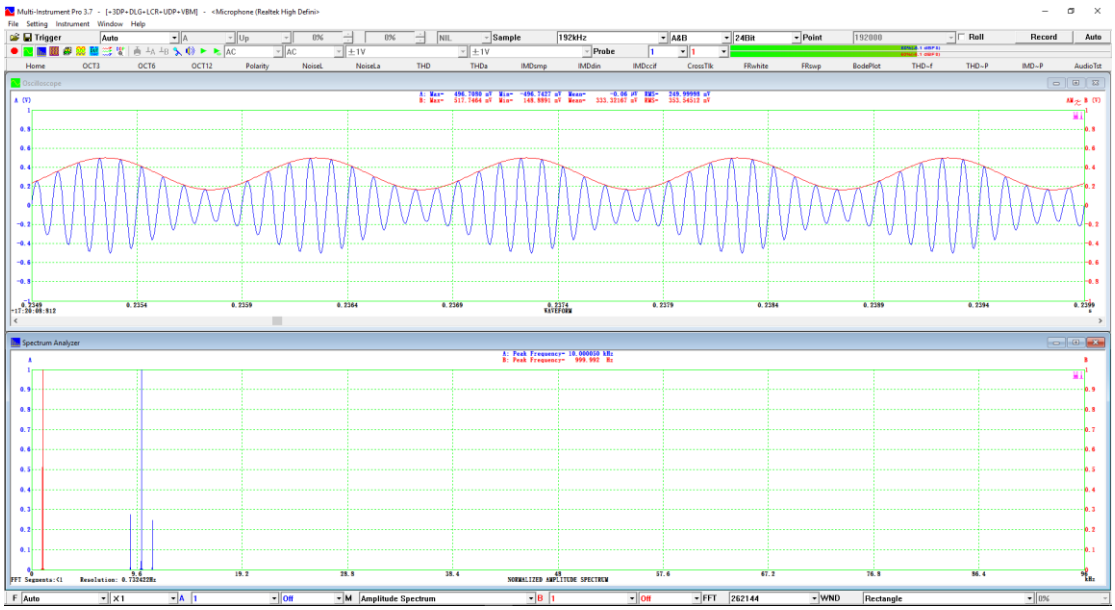
2.6.2.8.2.2 Demodulation

Demodulation is performed after Time Delay Removal. It is processed within a frame and thus there may be some "boundary effects" at the two ends of the frame. There

are four demodulation options: None, AM, FM and PM. It is possible to let the signal to pass through a bandpass filter first before demodulation. This simulates the frequency tuning process in a radio system and is sometime important in order to achieve better demodulation quality. The lower and upper frequency limits of the band pass filter can be specified. The AM, FM, PM demodulation methods used here are based on Hilbert Transform. Demodulation can be applied to Channel A, Channel B, or both of them.

2.6.2.8.2.2.1 AM Demodulation

In Amplitude Modulation (AM), the amplitude of the carrier signal varies in proportion to the instantaneous amplitude of the modulating signal. An AM signal is demodulated here by detecting its envelope. The following is an example of an AM signal (in blue) and its demodulated signal or envelop (in red). The carrier frequency is 10 kHz and modulating frequency is 1 kHz. The modulation index is 50%.



"Remove DC" or a high pass filter (described later) can be used to remove the DC component in the demodulated signal in the above figure. When AM demodulation is applied to a channel, the symbol "AM" will be displayed beside the corresponding axis label in the Oscilloscope view.

Demodulation of an AM signal can also be done through rectification followed by low pass filtering (described later).

2.6.2.8.2.2.2 FM Demodulation

In Frequency Modulation (FM), the instantaneous frequency deviation from the carrier frequency of the modulated signal varies in proportion to the instantaneous amplitude of the modulating signal. A FM signal is demodulated here by calculating the instantaneous frequency deviation through Hilbert Transform. The instantaneous frequency deviation is then converted to the instantaneous amplitude of the

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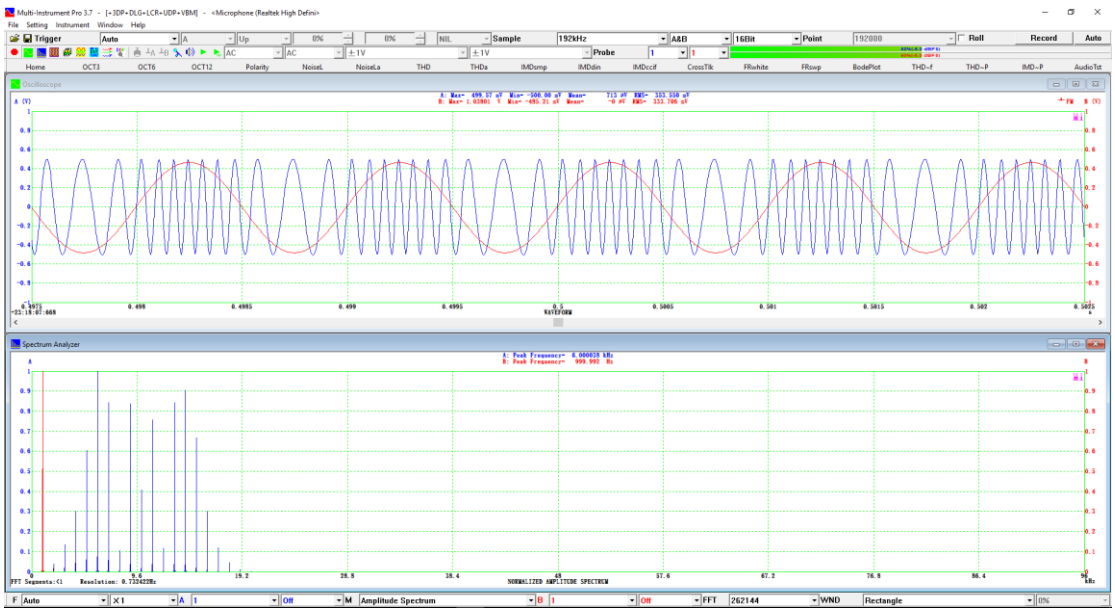
modulating signal using the frequency modulation sensitivity which has a unit of Hz/EU, where EU is the sensor's unit which is usually Volt. For FM demodulation, two parameters need to be manually entered by the users: the carrier frequency and the frequency modulation sensitivity. When FM demodulation is applied to a channel, the symbol "FM" will be displayed beside the corresponding axis label in the Oscilloscope view.

The following is an example of a FM signal (in blue) and its demodulated signal (in red). The FM signal has a carrier frequency of 10 kHz, a modulating frequency of 1 kHz, a maximum frequency deviation of 5 kHz, and a frequency modulation sensitivity of 10 kHz/V. Thus the amplitude of the demodulated signal is ±0.5V. "Remove DC" is used to remove the DC residual in the demodulated signal in the figure below.

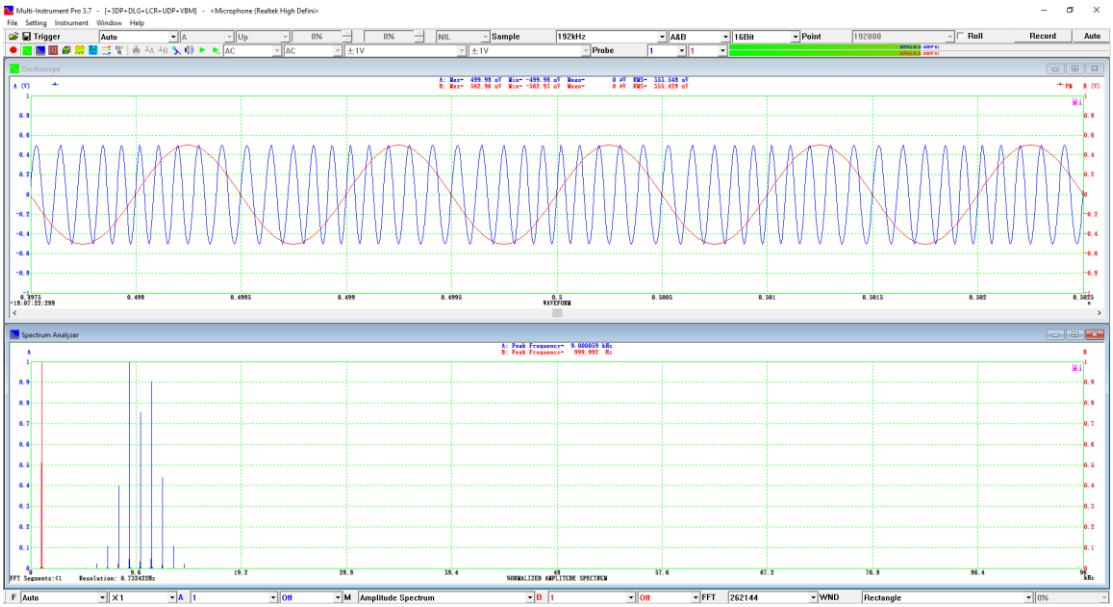


Multi-Instrument features unique FM demodulation algorithm which is able to demodulate the FM signal accurately even when the sampling rate is not very high compared with the FM signal spectrum.

2.6.2.8.2.2.3 PM Demodulation

In Phase Modulation (PM), the instantaneous phase deviation from the carrier phase (i.e. the phase if no modulation on the carrier signal is applied) of the modulated signal varies in proportion to the instantaneous amplitude of the modulating signal. A PM signal is demodulated here by calculating the instantaneous phase deviation through Hilbert Transform. The instantaneous phase deviation is then converted to the instantaneous amplitude of the modulating signal through the phase modulation sensitivity which has a unit of °/EU, where EU is the sensor's unit which is usually Volt. For PM demodulation, two parameters need to be manually entered by the users: the carrier frequency and the phase modulation sensitivity. When PM demodulation is applied to a channel, the symbol "PM" will be displayed beside the corresponding axis label in the Oscilloscope view.

The following is an example of a PM signal (in blue) and its demodulated signal (in red). The PM signal has a carrier frequency of 10 kHz, a modulating frequency of 1 kHz, a maximum phase deviation of 90°, and a phase modulation sensitivity of 180°/V. Thus the amplitude of the demodulated signal is ±0.5V. "Remove DC" is used to remove the DC residual in the demodulated signal in the figure below.



2.6.2.8.2.3 Digital Filtering

Digital filtering belongs to intra-frame processing and is performed after demodulation. There may be some "boundary effects" at the two ends of the frame if digital filtering is applied.

Six options are available: None, Low Pass, High Pass, Band Pass, Band Stop, Arbitrary. The class of filter can be selected from FFT, FIR and IIR. These options are performed after "Remove DC" and Rectification.

A Finite Impulse Response (FIR) filter, as the name suggested, has a finite impulse response. It has no feedback and therefore is always stable. It is usually designed to have symmetrical filter coefficients in order to achieve a linear phase response. It should be noted that this filter will cause the signal to delay [FIR order]/2/[Sampling Rate] seconds. It is possible to remove this delay by ticking the FIR Delay Removal checkbox.

An Infinite Impulse Response (IIR) filter has an impulse response function which is non-zero over an infinite length of time. It uses feedbacks from the output to the input and thus may lead to instability. It does not have a linear phase response usually.

Unlike FIR and IIR filters, the filtering function of a FFT filter is not done in time domain. Instead, the input signal is transformed from time domain to frequency domain using FFT. Its spectrum is then multiplied with the filter's frequency response and the result is transformed back to time domain using inverse FFT. FFT filter has a linear phase response and causes no delay (i.e. zero-phase response). Normally

Rectangle window function should be used for FFT filter. If there is a need to suppress the boundary effect (e.g. overshoot at both ends of the waveform), one of the 7 Tukey windows can be applied.

The software will force itself to use the FIR filter when the record length of data is greater than ½ of the maximum allowable FFT size (i.e. 4194304), even if a FFT filter is specified.

On the other hand, the software will force itself to use the FFT filter when the record length of data is less than twice of the FIR filter order, even if a FIR filter is specified.

The software has a built-in FIR filter designer based on Window or Fourier Transform method. The filter order must be even and in the range of 2~32766. Different window functions can be used to fulfill different requirements for stopband attenuation and transition bandwidth. The filter order determines the width of the transition band. The higher the order, the narrower the transition between the passband and stopband, and the slower the processing speed.

The software also supports the import of IIR or FIR filter coefficients designed externally.

Digital filtering is performed frame by frame independently. It may cause discontinuities at frame boundaries even if the data there are contiguous before the filtering. For FIR and IIR filters, it is possible to perform continuous filtering across frames if the "Across Frames" option is ticked. The prerequisite is that the original data are continuous across frames. Auto trigger mode together with a low sampling rate or Record mode can be used to ensure that.

Digital filter can be applied to Channel A, Channel B, or both of them.

2.6.2.8.2.3.1 Remove DC

"Remove DC" is performed before rectification (detection) and any digital filtering. When this option is applied to a channel, the symbol will be displayed beside the corresponding axis label in the Oscilloscope view.

2.6.2.8.2.3.2 Rectification (Detection)

Rectification (Detection) is performed right after "Remove DC" and before any digital filtering. Three options are available: Nil, Half Wave, and Full Wave. When rectification is applied to a channel, the symbol (Half Wave) or (Full Wave) will be displayed beside the corresponding axis label in the Oscilloscope view.

The following figure shows a sine wave before and after half-wave rectification.

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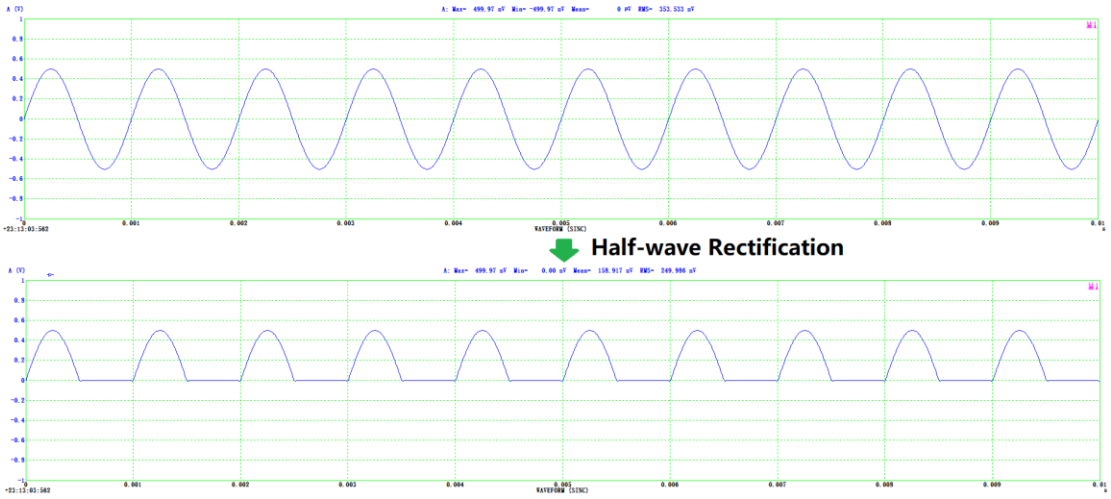
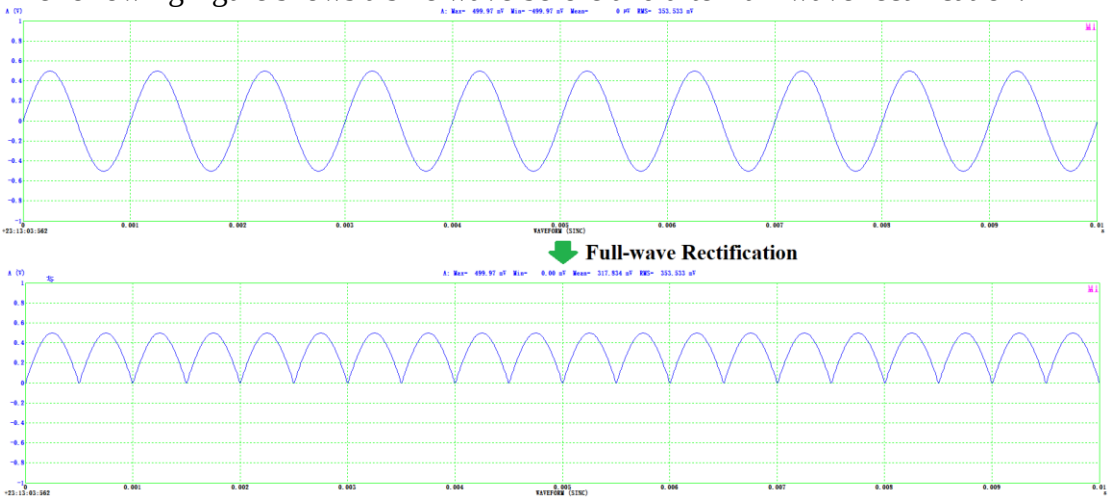


The following figure shows a sine wave before and after full-wave rectification.



2.6.2.8.2.3.3 None

No filtering will be performed.

2.6.2.8.2.3.4 Low Pass

A low pass filter is a filter that passes low frequencies and rejects (attenuates the amplitude of) frequencies higher than the cutoff frequency. Two classes of low pass filters are supported: FFT and FIR.

When a Low Pass filter is applied to a channel, the symbol ≈ will be displayed beside the corresponding axis label in the Oscilloscope view.

2.6.2.8.2.3.5 High Pass

A high pass filter is a filter that passes high frequencies and rejects (attenuates the amplitude of) frequencies lower than the cutoff frequency. Two classes of high pass filters are supported: FFT and FIR.

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When a High Pass filter is applied to a channel, the symbol will be displayed beside the corresponding axis label in the Oscilloscope view.

2.6.2.8.2.3.6 Band Pass

A band pass filter is a filter that passes frequencies within a certain range and rejects (attenuates the amplitude of) frequencies outside that range. Two classes of band pass filters are supported: FFT and FIR.

When a Band Pass filter is applied to a channel, the symbol will be displayed beside the corresponding axis label in the Oscilloscope view.

2.6.2.8.2.3.7 Band Stop

A band stop filter is a filter that rejects (attenuates the amplitude of) frequencies in a certain range and passes the rest of frequencies. Two classes of band stop filters are supported: FFT and FIR.

When a Band Stop filter is applied to a channel, the symbol will be displayed beside the corresponding axis label in the Oscilloscope view.

2.6.2.8.2.3.8 Arbitrary

An arbitrary filter is a filter that has an arbitrary frequency response. There are two ways to define an arbitrary filter. One is to define its frequency response (magnitude). The other is to define its filter coefficients.

- Use Frequency Response File (*.frf) to define an arbitrary filter

  A Frequency Response File is a Comma Separated Value (CSV) TXT file that defines the magnitude frequency response of a filter. It has the following format:

  Example:

  1,0,0
  2,1000,0
  3,1000.1,-1000
  4,100000,-1000
  ……

  Each line contains three comma separated values. The first variable is the sequential number. The second one is the frequency value in Hz. And the third one is the corresponding gain value in dB. Note that 0 dB represents a unit gain. Any frequencies that fall outside the defined range will be given a

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gain of –1000 dB. The example shown above is a low pass filter with a cutoff frequency of 1000 Hz.

Some sample files are provided in the FRF directory of the software and can be used as templates, such as:

(1) Recording RIAA
(2) Reproducing RIAA
(3) SpeechMale (IEC 60268-16)
(4) SpeechFemale (IEC 60268-16)
(5) AES6-2008-W&F (Wow & Flutter)
(6) ISO-2631-1-Wc (horizontal whole-body vibration x-axis)
(7) ISO-2631-1-Wd (horizontal whole-body vibration x-axis or y-axis)
(8) ISO-2631-1-We (rotating whole-body vibration)
(9) ISO-2631-1-Wf (low-frequency vertical whole-body vibration z-axis)
(10) ISO-2631-1-Wj (vertical head vibration)
(11) ISO-2631-1-Wk (vertical whole-body vibration z-axis)
(12) ISO-2631-2-Wm (whole-body vibration on non-specific posture)
(13) ISO-2631-4-Wb (vertical whole-body vibration z-axis)
(14) ISO-5349-1-Wh (hand-arm vibration)

- Use IIR Coefficient File (*.iir) to define an arbitrary filter

An IIR filter is defined as:

$y[n]=b_0x[n]+b_1x[n-1]+\ldots +b_Mx[n-M]- a_1y[n-1]- a_2y[n-2]-\ldots- a_Ny[n-N];$

where:
x[n] is the input signal,
y[n] is the output signal,
b_i is the so-called feedforward filter coefficients,
a_i is the so-called feedback filter coefficients,
M is the feedforward filter order,
N is the feedback filter order.

An IIR Coefficient File is a Comma Separated Value (CSV) TXT file that contains the coefficients of the IIR filter. It has the following format (assuming M>N):

Example:

0, b_0, a_0
1, b_1, a_1
2, b_2, a_2
……
N, b_N, a_N
……
M-1, b_{M-1}, 0
M, b_M, 0

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where a0 (usually assigned a value of "1") is just a placeholder and will always be ignored by the software. For IIR filters, usually M and N are designed to be equal. In the cases that they are not equal, zeros must be used to fill up the gaps. The maximum order supported is 32766. If N = 0, then the filter becomes a FIR filter.

It should be noted that the software is built with a limiter which will prevent the output of any digital filter from going outside of the range defined by the input ADC full-scale voltage.

Some sample files are provided in the IIR directory of the software and can be used as templates.

When an Arbitrary filter is applied to a channel, the symbol will be displayed beside the corresponding axis label in the Oscilloscope view.

The button in this dialog can be used to open the selected Frequency Response File or the IIR Coefficient File for viewing or editing via Microsoft Notepad.

2.6.2.8.3 Parameter Measurement

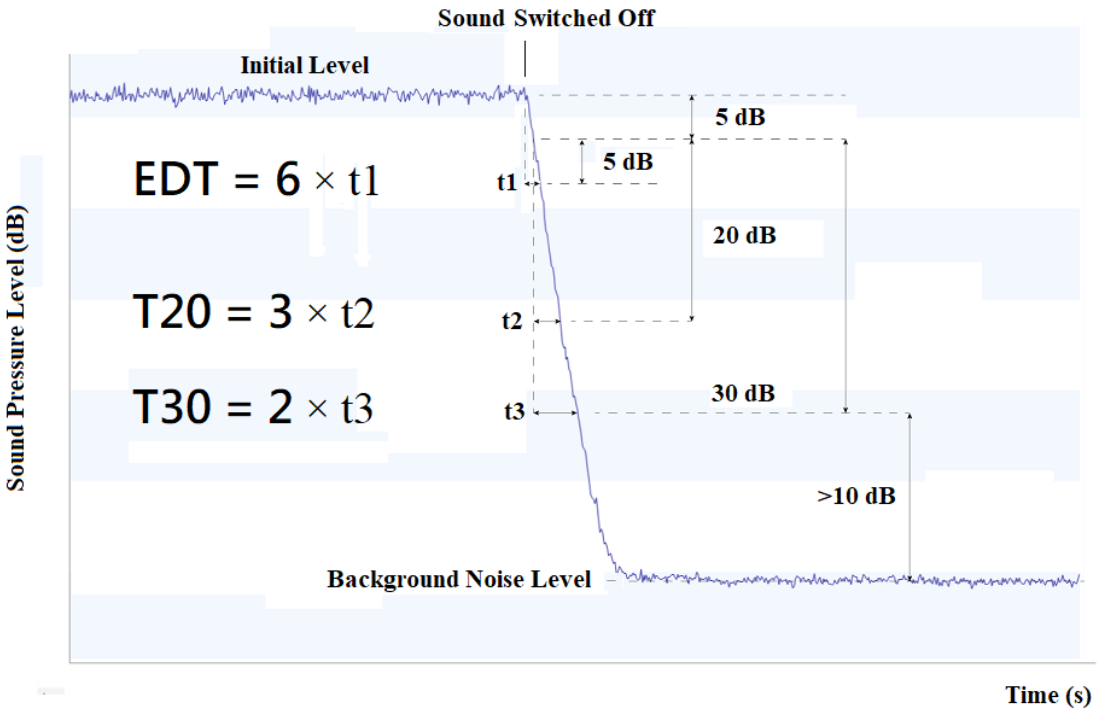
Parameter measurement is performed after inter-frame and intra-frame processing. The following parameters can be measured.

2.6.2.8.3.1 Reverberation and Its Related Acoustic Parameters

Reverberation is the phenomenon of persistence of sound after it has been produced, as a result of multiple reflections from surfaces such as walls, ceilings, floors, furniture, and people in an enclosure. These reflections build up with each reflection and decay gradually as they are absorbed by the reflecting surfaces. Reverberation can enhance the perception of music but it can reduce speech intelligibility.

2.6.2.8.3.1.1 Reverberation Time

Reverberation can be quantitatively characterized by Reverberation Time. The time taken for the sound pressure level to decay 60 dB after the source emission has stopped is called the Reverberation Time (RT, or RT60). It is the most important parameter describing the acoustic quality of a room or space. In a real environment, it can be difficult to achieve a dynamic range of more than 60 dB to measure RT60 directly, due to the noisy background and other constraints. Therefore, RT60 is usually evaluated based on a smaller dynamic range and extrapolated to a decay time of 60 dB. It is then labelled accordingly. ISO3382 defines T20, T30 and Early Decay Time (EDT) as follows.



T20 is the RT60 derived from the decay curve between 5dB and 25dB below the initial level. The initial level must be at least 35dB higher than the background noise level.

T30 is the RT60 derived from the decay curve between 5dB and 35dB below the initial level. The initial level must be at least 45dB higher than the background noise level.

If the reverberation sound field is fully diffused after multiple reflections, the energy decay rate will become exponential if linear scale is used, or linear if dB (logarithmic) scale is used. This is implicitly assumed in the concept of Reverberation Time and is the basis for the above extrapolation. However, the first few reflections (i.e. the earliest, lowest-order reflections), which are often good facsimiles of the direct sound, are likely to decay at a different rate. This "early" decay is strongly affected by the directivity of the sound source and surface materials of the objects near it.

Early Decay Time (EDT) is derived from the decay curve between 0 dB and 10 dB below the initial level. Similar to T20 and T30, it is also extrapolated to a decay time of 60 dB.

2.6.2.8.3.1.2 Clarity C50 and C80, Definition D50, Center Time Ts

The reverberant characteristics of an enclosure can also be quantitatively described by early-to-late energy ratio or Clarity. Clarity is defined as the ratio of the energy arriving before the time t to the energy arriving after the time t within the reverberant decay period, expressed in dB. The two most commonly used ones are C50 (for speech) and C80 (for music), in which the energy is split at 50 ms and 80 ms after the direct sound stops.

Another parameter in this category is Definition D50. It is an early-to-total energy ratio defined as the ratio of the energy in the first 50 ms to the total energy within the reverberant decay period, expressed in percentage.

Center Time Ts is the time of the "center of gravity" of the energy within the reverberant decay period.

2.6.2.8.3.1.3 Frequency Bands Selection

Reverberation is frequency dependent. Although reverberation time is often stated as a single value if measured as a broadband signal (20Hz ~ 20kHz), it is more precisely described in terms of frequency bands. Two options are available in the software: 1/1 octave and 1/3 octave. In 1/1 octave band analysis, the acquired data will be filtered by 1/1 octave band pass filters centered at 63Hz, 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz and 8kHz first before reverberation analysis. If "1/3 octave" is chosen, the acquired data will be filtered by 1/3 octave band pass filters centered at: 50Hz, 63Hz, 80Hz, 100Hz, 125Hz, 160Hz, 200Hz, 250Hz, 315Hz, 400Hz, 500Hz, 630Hz, 800Hz, 1kHz, 1.25Hz, 1.6Hz, 2kHz, 2.5kHz, 3.15kHz, 4kHz, 5kHz, 6.3kHz, 8kHz, 10kHz instead. In both cases, the original data without filtering will also be analyzed and the results will be tagged under "broadband".

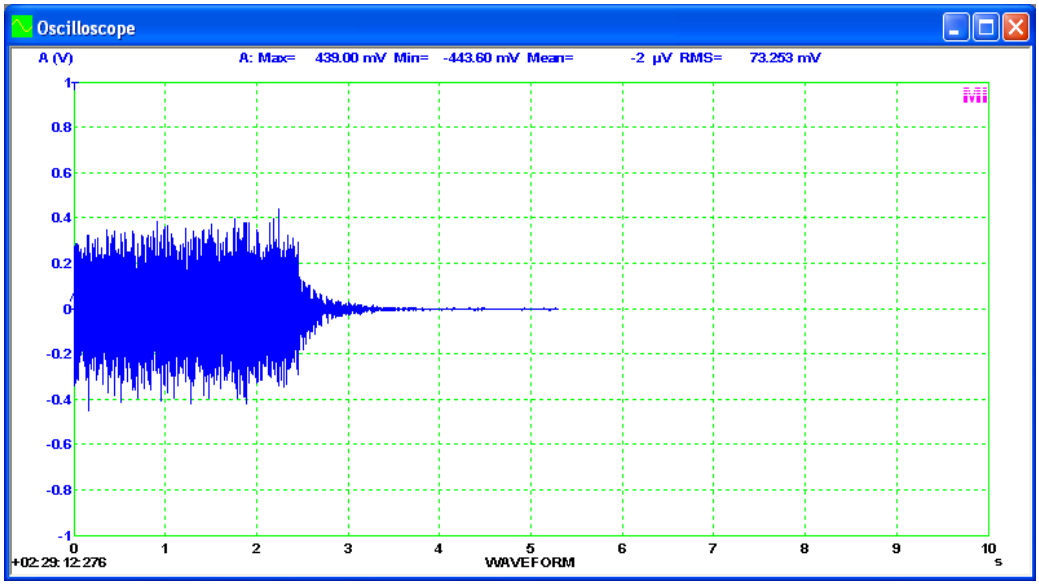
2.6.2.8.3.1.4 Measurement Methods

Both the sound source and the measurement microphone should be omnidirectional. The measurements should be made with different microphone positions which are at least 1.5 m apart, 2 m from any sound source and 1 m from any reflecting surfaces.

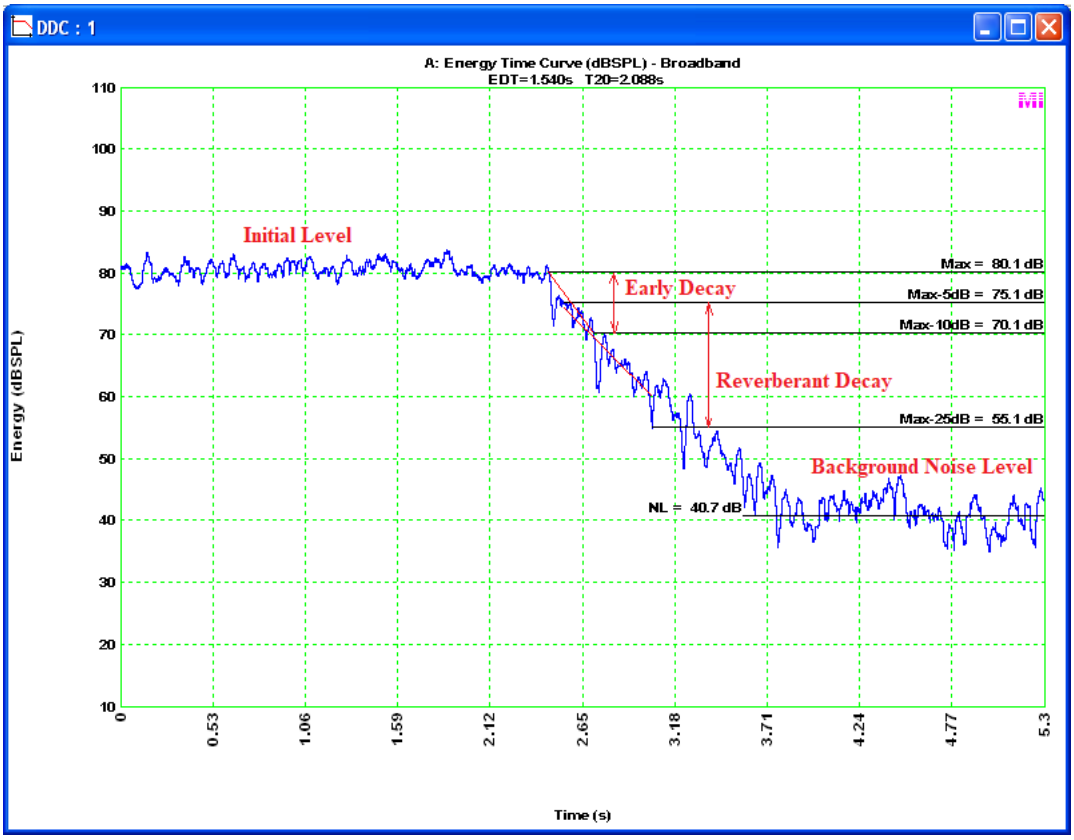
There are generally two methods to measure the reverberant decay curve: interrupted noise method and integrated impulse response method. Multi-Instrument is able to automatically differentiate these two based on the overall shape of the data curve in the Oscilloscope. Therefore in the software there is no need to explicitly specify which method is used. Beside the time domain analysis methods described here, it is also possible to analyze reverberation in time-frequency domain, this will be introduced later in the Chapter of Spectrum 3D Plot.

(1) Interrupted noise method
The decay curve can be obtained by direct recording of the decay of the sound pressure level after exciting an enclosure with broadband or band-limited noise. Pink noise is usually used in this method. The excitation signal should be sufficiently long to produce a steady-state sound pressure level in all frequency bands or a particular frequency band before it is switched off. In order to obtain steady-state conditions, the excitation time shall be at least half of the estimated reverberation time. Synchronized signal generator and oscilloscope operation of Multi-Instrument can be used to facilitate the capture of the sound pressure decay process. A Panel Setting File example can be found at \psf\Pro\RT60InterruptedNoise5s.psf.

The following figure shows an example of the instantaneous sound pressure variation recorded using this method.



During reverberation analysis, the directly recorded instantaneous sound pressure variation will be band-pass filtered for each frequency band of interest and converted to sound pressure level (dBSPL) with the integration time determined by the software automatically (see figure below). EDT, T20 and T30 will then be calculated automatically according to ISO3382. Note that the latter two are obtained through linear regression.



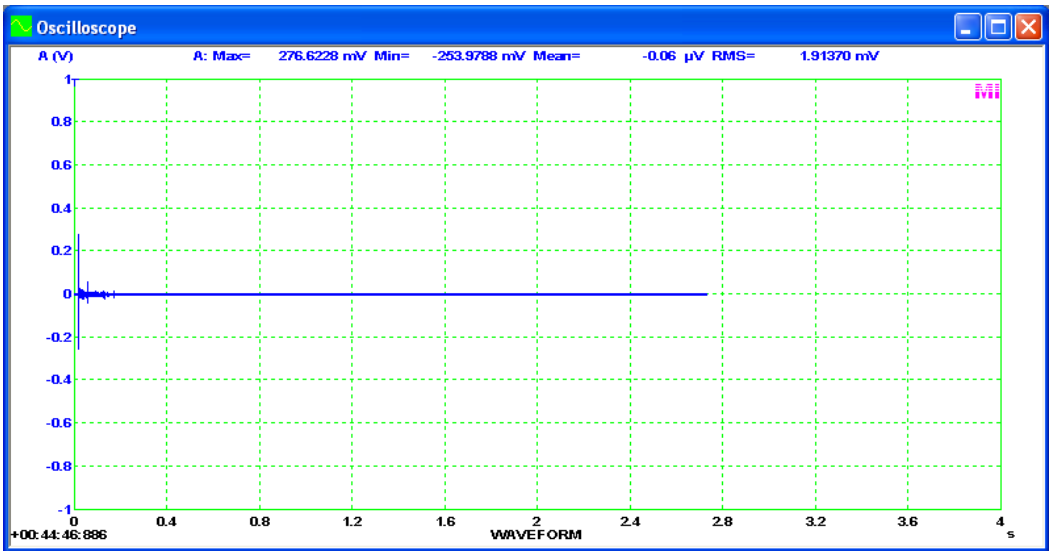
The decay curve measured with this method may vary from one measurement to another with no change in the characteristics of the enclosure and transmitting /

receiving position. This is due to the randomness of the excitation signal (different initial amplitudes and phases). Averaging 3 or more decay curves or reverberation times measured at one transmitting / receiving position is mandatory in order to obtain a suitable statistical repeatability.

A sample panel setting file for this method can be found at: psf\pro\ RT60InterruptedNoise5s.psf.

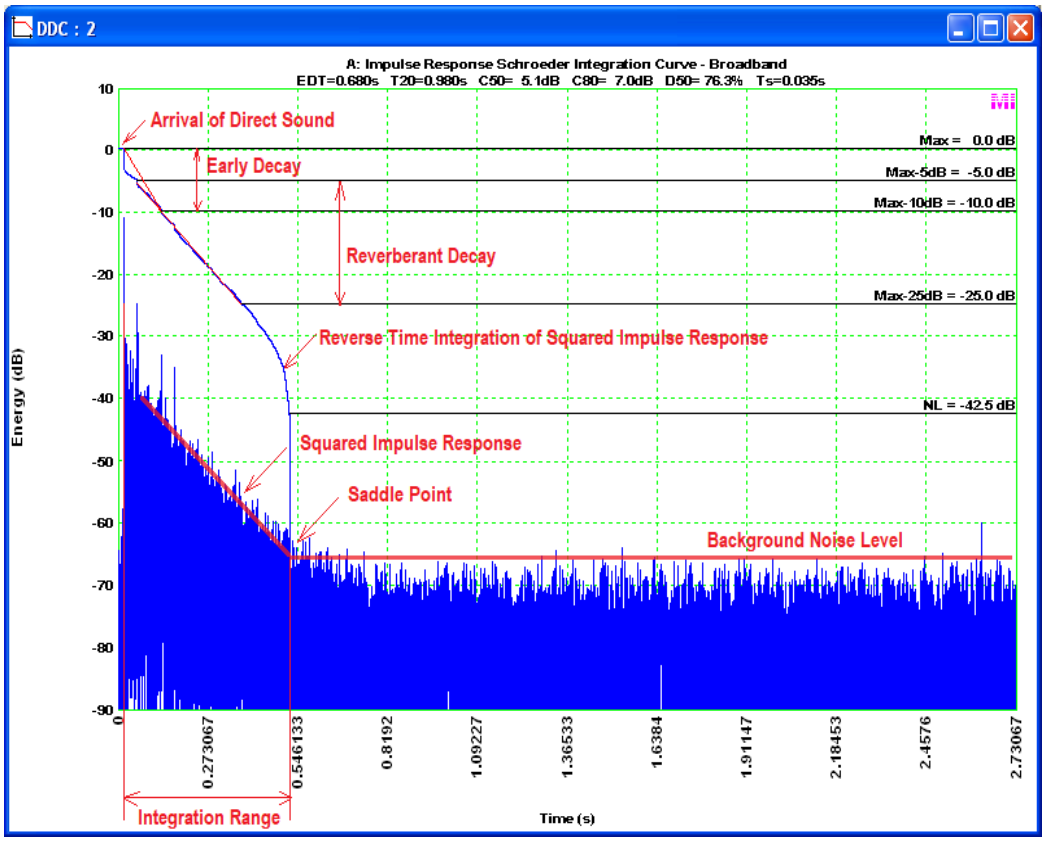
(2) Integrated impulse response method

Impulse response is the temporal evolution of the sound pressure observed at a point in an enclosure as a result of the emission of a Dirac impulse at another point in the enclosure. The following figure shows an impulse response example.



The decay curve can be obtained by reverse-time integration of the squared impulse response (see figure below). This method is also known as Schroeder Integration method. Dr. Manfred Schroeder has shown that the ensemble average of decay curves from multiple measurements using the interrupted noise method can be obtained from the backward integrated squared impulse response through one single measurement. This method smooths the decay and provides a very clean maximum level. Theoretically, the integration should start at the end of the time record and work back to the beginning. However, due to the existence of the background noise in the actual measurement, the integration curve will flatten out as the reverberant decay slope runs into the noise floor. This will lead to overestimation of the reverberation time if the impulse response has a limited dynamic range and a long noise tail. To solve this problem, the integration should start at the point where the decay slope meets the noise floor. This point is sometimes called “saddle point”. Similarly, the backward integration should end at the point where the direct sound arrives, excluding any background noise before that time. This ensures the correct detection of the maximum level. The above integration range will be automatically detected by Multi-Instrument. To standardize the rendering of the integration curve, the Schroeder integration is normalized so that it peaks at 0 dB. Unlike the case of the interrupted noise method in which the energy decay curve has a platform of the initial level, the energy decay curve of an impulse response only has a peak at the time when the direct sound arrives. This characteristic is used by the software to identify an

impulse response. In case an impulse response has its peak at or very close to the beginning of the time record, then the software will fail to recognize it as an impulse response and it will then be treated as if it is measured with the interrupted noise method.



The impulse response and frequency response of a LTI (Linear Time-Invariant) system are a Fourier transform pair. They are the most important characteristics of a LTI system. As the impulse response is a deterministic function and not prone to statistical deviations, no averaging is necessary in this method. Beside the reverberation time EDT, T20 and T30, Clarity C50 and C80, Definition D50 and Center Time Ts will also be calculated in this method. Again, T20 and T30 are obtained through linear regression.

There are direct and indirect methods to obtain the impulse response.

(i) Direct Method
It is impossible in practice to create and radiate a true Dirac impulse but short transient sounds (e.g. a gunshot) can offer close enough approximations for practical measurements.

The impulse response can be recorded directly using an impulse source such as a pistol shot, balloon burst, spark gap, fire cracker, hand clap, or any other sound source that produces an impulse with sufficient bandwidth and energy.

(ii) Indirect Method
The impulse response can be measured indirectly using broadband excitation signals such as white noise, pink noise, linear / logarithmic swept sine, and MLS (Maximum

Length Sequence). This will be introduced later in the impulse response section as well as cross-correlation section in the Spectrum Analyzer chapter. The measured impulse response can be exported as a TXT file from the Spectrum Analyzer and then imported into the Oscilloscope via [File]>[Import] for reverberation analysis.

A sample panel setting file for this method can be found at: psf\pro\RT60-STIIImpulseResponse5s.psf.

2.6.2.8.3.1.5 Analysis Results

If the "Reverberation / Speech Intelligibility" option is ticked, the software will perform reverberation and speech intelligibility analysis and store the results in two DDP arrays: AcousticParameter_A_Array[?] for Channel A and AcousticParameter_B_Array[?] for Channel B. When fetching a particular parameter, "?" should be replaced by its array index. The following table shows the content of each array element relevant to reverberation under 1/1 octave band analysis. For example, AcousticParameter_A_Array[18] should be used to read T20 in broadband in Channel A.

	Broadband	63Hz	125Hz	250Hz	500Hz	1kHz	2kHz	4kHz	8kHz
EDT(s)	0	1	2	3	4	5	6	7	8
Reserved	9	10	11	12	13	14	15	16	17
T20(s)	18	19	20	21	22	23	24	25	26
rT20	27	28	29	30	31	32	33	34	35
T30(s)	36	37	38	39	40	41	42	43	44
rT30	45	46	47	48	49	50	51	52	53
C50(dB)	54	55	56	57	58	59	60	61	62
C80(dB)	63	64	65	66	67	68	69	70	71
D50(%)	72	73	74	75	76	77	78	79	80
Ts(s)	81	82	83	84	85	86	87	88	89

The following table shows the content of each array element under 1/3 octave band analysis.

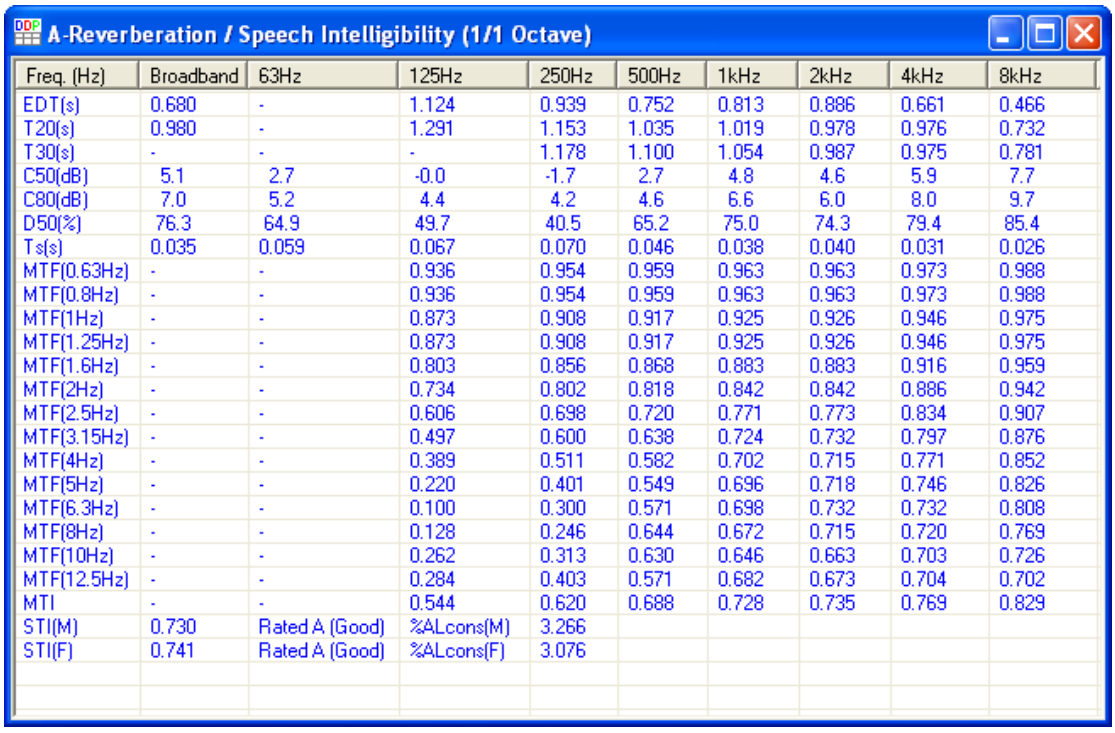
	EDT(s)	Reserved	T20(s)	rT20	T30(s)	rT30	C50(dB)	C80(dB)	D50(%)	Ts(s)
Broadband	300	325	350	375	400	425	450	475	500	525
50Hz	301	326	351	376	401	426	451	476	501	526
63Hz	302	327	352	377	402	427	452	477	502	527
80Hz	303	328	353	378	403	428	453	478	503	528
100Hz	304	329	354	379	404	429	454	479	504	529
125Hz	305	330	355	380	405	430	455	480	505	530
160Hz	306	331	356	381	406	431	456	481	506	531
200Hz	307	332	357	382	407	432	457	482	507	532
250Hz	308	333	358	383	408	433	458	483	508	533
315Hz	309	334	359	384	409	434	459	484	509	534
400Hz	310	335	360	385	410	435	460	485	510	535
500Hz	311	336	361	386	411	436	461	486	511	536
630Hz	312	337	362	387	412	437	462	487	512	537
800Hz	313	338	363	388	413	438	463	488	513	538
1kHz	314	339	364	389	414	439	464	489	514	539
1.25kHz	315	340	365	390	415	440	465	490	515	540
1.6kHz	316	341	366	391	416	441	466	491	516	541
2kHz	317	342	367	392	417	442	467	492	517	542
2.5kHz	318	343	368	393	418	443	468	493	518	543
3.15kHz	319	344	369	394	419	444	469	494	519	544
4kHz	320	345	370	395	420	445	470	495	520	545
5kHz	321	346	371	396	421	446	471	496	521	546
6.3kHz	322	347	372	397	422	447	472	497	522	547
8kHz	323	348	373	398	423	448	473	498	523	548
10kHz	324	349	374	399	424	449	474	499	524	549

rT20 and rT30 are the linear regression correlation coefficients for T20 and T30 respectively. If the result in a particular array element is invalid, a super large negative value -10^{20} will be stored there. A result can be invalid if the dynamic range of the decay curve is not enough, the linear regression correlation coefficient is outside the range of -1~-0.4, or the interrupted noise method is used instead of the integrated impulse response method.

The above parameters can be viewed through DDP viewer individually. They can also be viewed together in the following four reports using DDP array viewer:

(1) A-Reverberation / Speech Intelligibility (1/1 Octave)
(2) B-Reverberation / Speech Intelligibility (1/1 Octave)
(3) A-Reverberation / Speech Intelligibility (1/3 Octave)
(4) B-Reverberation / Speech Intelligibility (1/3 Octave)

The following figure shows the 1/1 octave band results of an impulse response. Note that it also contains the speech intelligibility analysis results which will be described later.



A-Reverberation / Speech Intelligibility (1/1 Octave)

Freq. (Hz)	Broadband	63Hz	125Hz	250Hz	500Hz	1kHz	2kHz	4kHz	8kHz
EDT(s)	0.680	-	1.124	0.939	0.752	0.813	0.886	0.661	0.466
T20(s)	0.980	-	1.291	1.153	1.035	1.019	0.978	0.976	0.732
T30(s)	-	-	-	1.178	1.100	1.054	0.987	0.975	0.781
C50(dB)	5.1	2.7	-0.0	-1.7	2.7	4.8	4.6	5.9	7.7
C80(dB)	7.0	5.2	4.4	4.2	4.6	6.6	6.0	8.0	9.7
D50(%)	76.3	64.9	49.7	40.5	65.2	75.0	74.3	79.4	85.4
Ts(s)	0.035	0.059	0.067	0.070	0.046	0.038	0.040	0.031	0.026
MTF(0.63Hz)	-	-	0.936	0.954	0.959	0.963	0.963	0.973	0.988
MTF(0.8Hz)	-	-	0.936	0.954	0.959	0.963	0.963	0.973	0.988
MTF(1Hz)	-	-	0.873	0.908	0.917	0.925	0.926	0.946	0.975
MTF(1.25Hz)	-	-	0.873	0.908	0.917	0.925	0.926	0.946	0.975
MTF(1.6Hz)	-	-	0.803	0.856	0.868	0.883	0.883	0.916	0.959
MTF(2Hz)	-	-	0.734	0.802	0.818	0.842	0.842	0.886	0.942
MTF(2.5Hz)	-	-	0.606	0.698	0.720	0.771	0.773	0.834	0.907
MTF(3.15Hz)	-	-	0.497	0.600	0.638	0.724	0.732	0.797	0.876
MTF(4Hz)	-	-	0.389	0.511	0.582	0.702	0.715	0.771	0.852
MTF(5Hz)	-	-	0.220	0.401	0.549	0.696	0.718	0.746	0.826
MTF(6.3Hz)	-	-	0.100	0.300	0.571	0.698	0.732	0.732	0.808
MTF(8Hz)	-	-	0.128	0.246	0.644	0.672	0.715	0.720	0.769
MTF(10Hz)	-	-	0.262	0.313	0.630	0.646	0.663	0.703	0.726
MTF(12.5Hz)	-	-	0.284	0.403	0.571	0.682	0.673	0.704	0.702
MTI	-	-	0.544	0.620	0.688	0.728	0.735	0.769	0.829
STI(M)	0.730	Rated A (Good)	%ALcons(M)	3.266					
STI(F)	0.741	Rated A (Good)	%ALcons(F)	3.076					

The following figure shows the 1/3 octave band results of an impulse response. It contains the reverberation analysis results only.

A-Reverberation / Speech Intelligibility (1/3 Octave)

Freq. (Hz)	Broadband	50Hz	63Hz	80Hz	100Hz	125Hz	160Hz	200Hz	250Hz	315Hz	400Hz	500Hz	630Hz	800Hz	1kHz	1.25kHz	1.6kHz	2kHz	2.5kHz	3.15kHz	4kHz
EDT(s)	0.680	-	-	1.063	0.971	1.260	1.099	1.106	0.826	0.702	0.823	0.917	0.978	0.873	0.935	0.867	0.895	0.842	0.955	0.823	0.745
T20(s)	0.980	-	-	0.820	1.897	1.566	1.315	1.179	1.193	1.203	1.333	1.170	1.123	1.095	1.036	1.051	1.005	0.943	1.061	1.002	0.979
T30(s)	-	-	-	-	-	-	-	-	1.120	1.235	1.446	1.492	1.424	1.189	1.037	1.082	0.980	0.953	1.036	0.974	1.000
C50(dB)	5.1	7.4	1.2	3.4	0.2	0.0	0.4	-0.5	1.3	-0.6	2.5	1.4	-0.9	4.1	1.4	3.5	5.8	4.1	3.0	3.1	4.9
C80(dB)	7.0	15.2	4.1	4.0	2.9	2.8	1.3	1.8	4.6	5.9	4.8	3.9	1.4	5.6	4.7	5.3	6.5	5.8	5.0	5.7	6.8
D50(%)	76.3	84.7	56.8	68.4	50.9	50.0	52.1	46.9	57.2	46.6	63.9	58.2	44.6	71.9	58.1	68.9	79.3	72.1	66.5	67.2	75.6
Ts(s)	0.035	0.028	0.065	0.060	0.077	0.080	0.080	0.082	0.064	0.059	0.049	0.059	0.072	0.044	0.057	0.046	0.035	0.042	0.051	0.047	0.036

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It should be noted that for the interrupted noise method, only EDT, T20 and T30 will be shown under both 1/1 and 1/3 octave band analyses. For T20 and T30, if the corresponding linear regression correlation coefficient is greater than -0.6 for the case of the interrupted noise method, or -0.95 for the case of the integrated impulse response method, "~" will be displayed before the measured value to indicate that the value may not be very reliable.

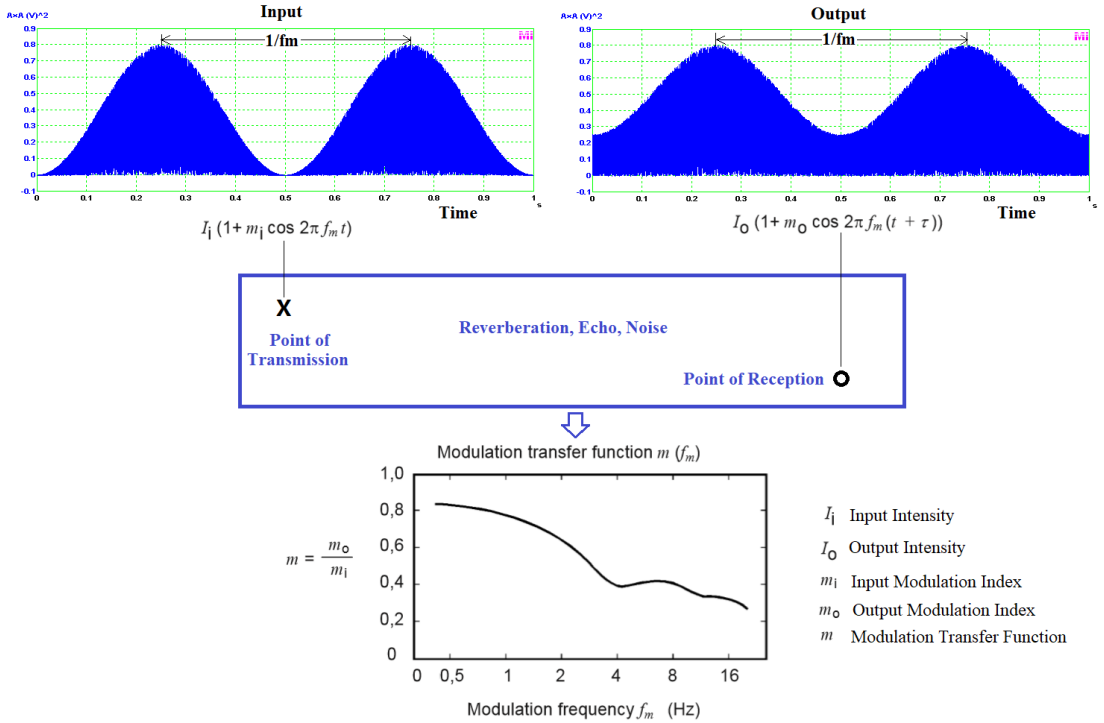
The reverberation and speech intelligibility analysis here calculates the acoustic parameters in each frequency band one by one automatically and thus may take a little while to complete. The data in the Oscilloscope should be kept small in size and contain only the information relevant to reverberation decay in order to cut down the computation time. If you are only interested in the acoustic parameters in a particular frequency band, you can use the Derived Data Curve (DDC) function instead, where Energy Time Curve (dBSPL) can be used for the interrupted noise method, and Impulse Response Schroeder Integration Curve can be used for the integrated impulse response method. These DDCs can also be used to manually verify the results of the automatic parameter measurement in the Oscilloscope. DDC function will be described later.

2.6.2.8.3.2 Speech Intelligibility

A transmission channel transmits speech from a talker to a listener through electrical, acoustic, optical or other signal paths. The speech intelligibility characteristics of the transmission channel can be quantified through either subjective listening tests or objective measurements. The latter are faster and normally used. Speech Transmission Index (STI) is commonly used as an objective measure to predict the intelligibility of speech transmitted by the transmission channel. It was originally proposed by Houtgast and Steeneken, and standardized most recently by IEC 60286-16:2011 in its fourth edition. The concept of STI is based on the empirical finding that in speech signals the most relevant information to speech intelligibility is the intensity fluctuations (or modulations), which result from the acoustic separation of sentences, words, and phonemes, the fundamental elements of speech. Modulation transfer function (MTF) is used to quantify how well the transmission channel preserves this intensity modulation. A single-number STI value can be calculated after MTFs with 14 modulation frequencies (from 0.63Hz to 12.5Hz, 1/3-octave apart) and in seven 1/1 octave bands (from 125Hz to 8kHz) are obtained.

2.6.2.8.3.2.1 Modulation Transfer Function

In objective measurement, speech can be modeled as a noise carrier, with a speech shaped frequency spectrum, intensity-modulated by a low-frequency sine wave. The intensity here is equal to the square of the sound pressure. Modulation Transfer Function (MTF) is defined as the ratio of the modulation index (or modulation depth) of the received signal to that of the transmitted signal. Any reduction of the MTF by the transmission channel, due to reverberation, echo, linear and nonlinear distortion, and noise, is considered to result in some deterioration of speech intelligibility. The definition of MTF is illustrated as follows.



Modulation Transfer Function - Input / Output Comparison

MTF is a function of the modulation frequency fm and the spectral content of the carrier. To calculate STI, MTF is evaluated for 14 modulation frequencies: 0.63Hz, 0.8Hz, 1Hz, 1.25Hz, 1.6Hz, 2Hz, 2.5Hz, 3.15Hz, 4Hz, 5Hz, 6.3Hz, 8Hz, 10Hz, 12.5Hz. The lower modulation frequencies mimic sentence and word separation while the higher modulation frequencies mimic phoneme separation within words. The spectral content of speech itself is also important. Seven octave bands: 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz, 8kHz, within which the carrier's overall spectral shape matches the long-term averaged speech spectrum, are evaluated for each modulation frequency. This gives a total of 7 × 14 = 98 combinations and thus 98 MTFs. The seven octave bands do not contribute equally to intelligibility. Although low speech frequencies (vowels) contribute most to the power of a speech signal, it is the higher speech frequencies (consonants) which contribute the most to speech intelligibility. This is reflected in the frequency band weighting factors applied when calculating the single-number STI from a matrix of MTFs. IEC 60286-16:2011 stipulates different weighting profiles for males and females. Both STI(M) for males and STI(F) for females are calculated in Multi-Instrument.

2.6.2.8.3.2.2 Speech Transmission Index

FULL STI method requires 98 test signals to be applied sequentially, with each measurement taking about 10 seconds. The overall measurement time is approximately 15 minutes. This method is considered to be the most accurate of the STI methods, but it is rarely used today because simplified methods which were subsequently developed are faster and more practical. Three simplified methods are popular: STIPA (STI for public address systems), STITEL (STI for telecommunication systems), RASTI (Room Acoustical STI).

STIPA consists of a test signal with a predefined set of two modulations (1.6Hz & 8Hz, 1Hz & 5Hz, 0.63Hz & 3.15Hz, 2Hz & 10Hz, 1.25Hz & 6.25Hz, 0.8Hz & 4Hz,

2.5Hz & 12.5Hz) per octave band (125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz, 8kHz) that are generated simultaneously giving a total of 14 modulation indices. One measurement takes between 10s and 15s.

STITEL consists of a test signal with a predefined set of seven modulation frequencies (1.12Hz, 11.33Hz, 0.71Hz, 2.83Hz, 6.97Hz, 1.78Hz and 4.53Hz), one per octave band (125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz, 8kHz), that are generated simultaneously giving a total of seven modulation indices. One measurement takes approximately 12s.

RASTI consists of a test signal with a predefined set of nine modulation frequencies that are generated simultaneously, five (0.7Hz, 1.4Hz, 2.8Hz, 5.6Hz, 11.2Hz) for the 2000Hz octave band and four (1.0Hz, 2.0Hz, 4.0Hz, 8.0Hz) for the 500Hz octave band. One measurement takes approximately 30s. RASTI is considered obsolete in IEC-60286-16:2011.

The direct FULL STI method is generally now only used for background STI research. However, indirect FULL STI method exists and is popular.

2.6.2.8.3.2.3 Measurement Methods

MTF is the basis of STI. There are two MTF measurement methods:

(1) Direct Method
This method uses modulated speech-shaped noises to measure MTF directly. It accounts for non-linear distortion and can be used for both linear and nonlinear systems.

It is possible to measure the modulation index and then MTF directly using User Defined Data Point (UDDP) in Multi-Instrument. This will be described later in the UDDP section. The derivation of STI from a matrix of MTFs obtained through the direct method is however not directly supported in the software.

It is also possible to generate the modulated noise through the Signal Generator in Multi-Instrument. Although the amplitude modulation function of the Signal Generator supports only a sinusoidal carrier and thus cannot be used here, the window-shaped mask function can be used instead to generate the modulated signal using an arbitrary signal as the carrier, and one of the available window functions as the modulation function. In particular, true or pseudo pink or white noise, MLS, or speech-like noise can be used as the carrier and Cosine 1.0 window can be used as the intensity modulation function. If amplitude modulation rather than intensity modulation is required, Hann window (=Cosine 2.0) should be used instead.

(2) Indirect Method
This method uses the system's impulse response to derive MTF. It should only be used for linear time-invariant systems. Dr. Manfred Schroeder has shown that MTF can be derived from the Fourier transform of the squared impulse response as follows.

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$$m_k(f_m)=\frac{\left|\int_0^{\infty} h_k(t)^2 e^{-j2\pi f_m t}dt\right|}{\int_0^{\infty} h_k(t)^2 dt}$$

where:

$m_k(f_m)$ is the MTF at modulation frequency f_m in octave band k;
$h_k(t)$ is the impulse response of octave band k;
f_m is the modulation frequency;
t is the integration variable for time;

$h_k(t)$ can be obtained by filtering the broadband impulse response with the respective octave band filter. The length of the impulse response must not be less than half of the reverberation time and 1.6s to ensure a reliable calculation of the modulation indices for the lowest modulation frequency of 0.63 Hz. MTF is noise dependent. It should be noted that the measuring process of the impulse response should not involve averaging which would otherwise reduce artificially the background noise. On the other hand, if the impulse response contains unrealistically low or no noise, the measured MTF is called "Noise-free" MTF and the noise effect can be taken into account subsequently as follows:

$$m_k'(f_m)=m_k(f_m)\frac{1}{1+10^{\frac{-SNR_k}{10}}}$$

where SNR_k is the signal-to-noise ratio in dB for octave band k.

The above formula shows that, when SNR_k is greater than 15 dB, the reduction of MTF due to noise is less than 3%.

During impulse response measurement, if the SNR in each octave band is at least 20 dB, no matter whether averaging is used or not, the impulse response is considered "noise-free". Impulsive signals such as the Dirac function are not generally suitable here when background noise, pass-band limiting and non-linear distortion are significant, since the average spectral distribution of typical speech is not represented in the test signal. If the impulse response is measured through deconvolution of a deterministic signal, such as a MLS or sweep signal, the measurement takes far less time than with the modulated noise method for the same reproducibility (some 5 s on average). If the measured impulse response is considered "noise-free", and it is necessary to take into account the noise effect, the background noise can be measured separately. Compensation for the noise effect through a separate background noise measurement is not currently supported in the software.

MTF can be further extended to include auditory effects such as level dependent auditory masking and absolute speech reception threshold, as defined in IEC 60268-16: 2011. These effects should only be taken into account when test signals are obtained acoustically. Compensation for the auditory effects is currently not implemented in the software.

Normally FULL STI is calculated in the indirect method, as the processing time is quite short, unlike the direct method. There is no advantage to calculate other reduced versions of STI in the indirect method. This method is adopted in the software. A

sample panel setting file for this method can be found at: psf\pro\RT60-STIImpulseResponse5s.psf.

2.6.2.8.3.2.4 Deriving STI from MTF

Once MTFs are obtained, either through direct method or through indirect method, STI can be calculated using the same method as follows:

(1) $MTF_{k,fm}$ is transformed into an effective signal-to-noise ratio $SNR_{eff\ k,fm}$ (dB) as given by:

$$SNR_{\text{eff } k,f_m} = 10 \times \log \frac{m'_{k,f_m}}{1 - m'_{k,f_m}}$$

$SNR_{eff\ k,fm}$ (dB) is limited to the range of -15dB ~ +15dB.

(2) $SNR_{eff\ k,fm}$ (dB) is converted to a transmission index $TI_{k,fm}$ using:

$$TI_{k,f_m} = \frac{SNR_{\text{eff } k,f_m} + 15}{30}$$

(3) All $TI_{k,fm}$ in each octave band are arithmetically averaged to obtain a Modulation Transfer Index (MTI_k) for that octave band.

(4) All MTI_k are combined using the following formula to get the STI(M) and STI(F).

$$STI = \sum_{k=1}^{7} \alpha_k \times MTI_k - \sum_{k=1}^{6} \beta_k \times \sqrt{MTI_k \times MTI_{k+1}}$$

where α_k is the weighting factor for octave band k and β_k the redundancy factor between octave band k and octave band k+1.

STI is limited to the range of 0~1. Males and females have different weighting and redundancy factors and thus different STI. Both STI(M) for males and STI(F) for females are calculated.

STI can be labelled as “Excellent”, “Good”, “Fair”, “Poor” and “Bad” based on the demarcation values: 0.75, 0.60, 0.45, 0.30. This relates it to the subjective ratings on speech intelligibility. It can also be categorized under “A+”, “A”, “B”, “C”, “D”, “E”, “F”, “G”, “H”, “I”, “J”, “U” according to the demarcation values: 0.76, 0.72, 0.68, 0.64, 0.60, 0.56, 0.52, 0.48, 0.44, 0.40, 0.36.

2.6.2.8.3.2.5 Articulation Loss of Consonants (%ALcons)

The %ALcons is originally based on the reception of words by listeners. It can also be derived from STI through a widely used approximation formula:

$$\%AL_{cons} = 170.5405 \cdot e^{-5.419 \cdot STI}$$

The %ALcons values normally range from 0 (corresponding to an excellent speech intelligibility) to 100 (corresponding to an extremely bad speech intelligibility), but

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the %ALcons value calculated from the abovementioned approximation formula, will exceed 100 at a very low STI. In Multi-Instrument, %ALcons(M) for males and %ALcons(F) for females will also be reported after the corresponding STIs are obtained.

2.6.2.8.3.2.6 Analysis Results

If the "Reverberation / Speech Intelligibility" option is ticked, the software will perform reverberation and speech intelligibility analysis and store the results in two DDP arrays: AcousticParameter_A_Array[?] for Channel A and AcousticParameter_B_Array[?] for Channel B. When fetching a particular parameter, "?" should be replaced by its array index. The following table shows the content of each array element relevant to speech intelligibility under 1/1 octave band analysis. Note that if IN/IR>=0, it reflects that the data in the Oscilloscope have been automatically detected as an impulse response and the speech intelligibility related parameters have been calculated. There are no applicable parameters in the shaded cells in the table.

	Broadband	63Hz	125Hz	250Hz	500Hz	1kHz	2kHz	4kHz	8kHz
MTF(0.63Hz)	90	91	92	93	94	95	96	97	98
MTF(0.8Hz)	99	100	101	102	103	104	105	106	107
MTF(1Hz)	108	109	110	111	112	113	114	115	116
MTF(1.25Hz)	117	118	119	120	121	122	123	124	125
MTF(1.6Hz)	126	127	128	129	130	131	132	133	134
MTF(2Hz)	135	136	137	138	139	140	141	142	143
MTF(2.5Hz)	144	145	146	147	148	149	150	151	152
MTF(3.15Hz)	153	154	155	156	157	158	159	160	161
MTF(4Hz)	162	163	164	165	166	167	168	169	170
MTF(5Hz)	171	172	173	174	175	176	177	178	179
MTF(6.3Hz)	180	181	182	183	184	185	186	187	188
MTF(8Hz)	189	190	191	192	193	194	195	196	197
MTF(10Hz)	198	199	200	201	202	203	204	205	206
MTF(12.5Hz)	207	208	209	210	211	212	213	214	215
MTI	216	217	218	219	220	221	222	223	224
STI(M)	225								
STI(F)	226								
%ALcons(M)	227								
%ALcons(F)	228								
IN/IR	229								

The above parameters can be viewed through DDP viewer individually. They can also be viewed together in the following two reports using DDP array viewer:

(1) A-Reverberation / Speech Intelligibility (1/1 Octave)
(2) B-Reverberation / Speech Intelligibility (1/1 Octave)

2.6.2.8.3.3 Discontinuity

In Multi-Instrument, discontinuity is defined as the instantaneous abruptness on a waveform. It is a function of time and rated in percentage. If the "Discontinuity" option is ticked, the software will calculate the discontinuity at every sample of the waveform and store the maximum value together with its occurring time in the following DDPs: MaxDiscontinuityValue_A(%) and MaxDiscontinuityTime_A(s) for Channel A, and MaxDiscontinuityValue_B(%) and MaxDiscontinuityTime_B(s) for Channel B. The following three figures show a sine sweep with a normal

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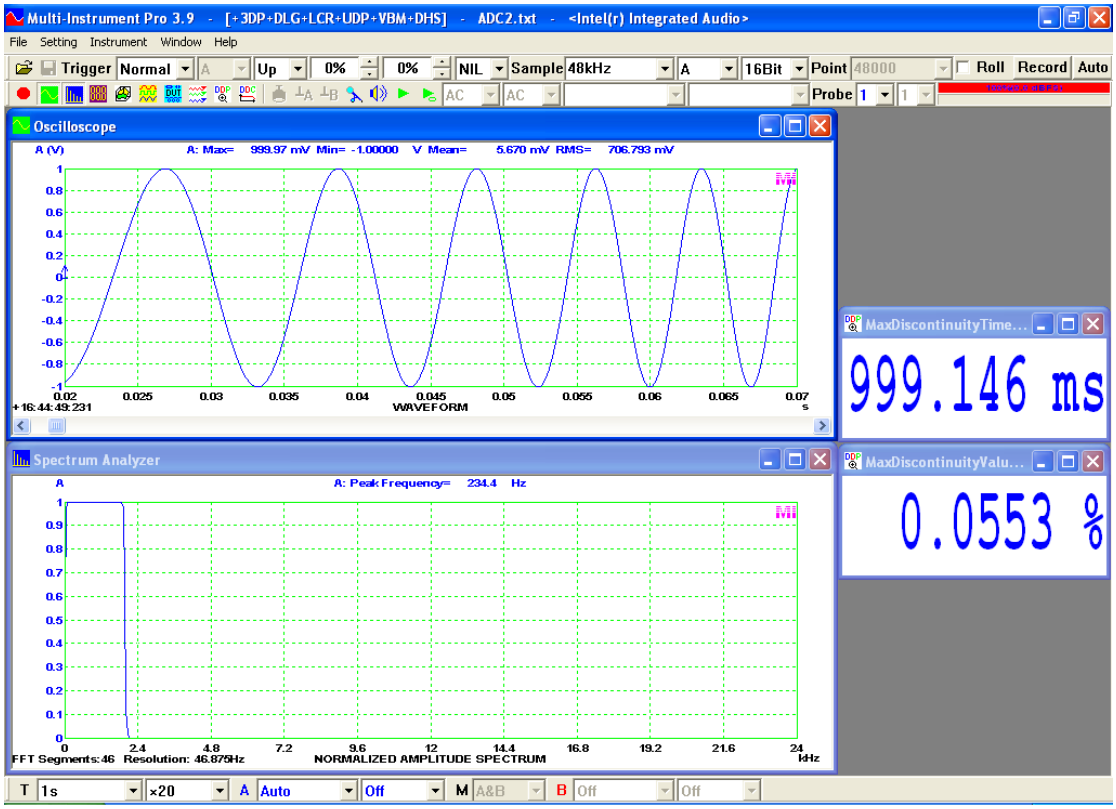
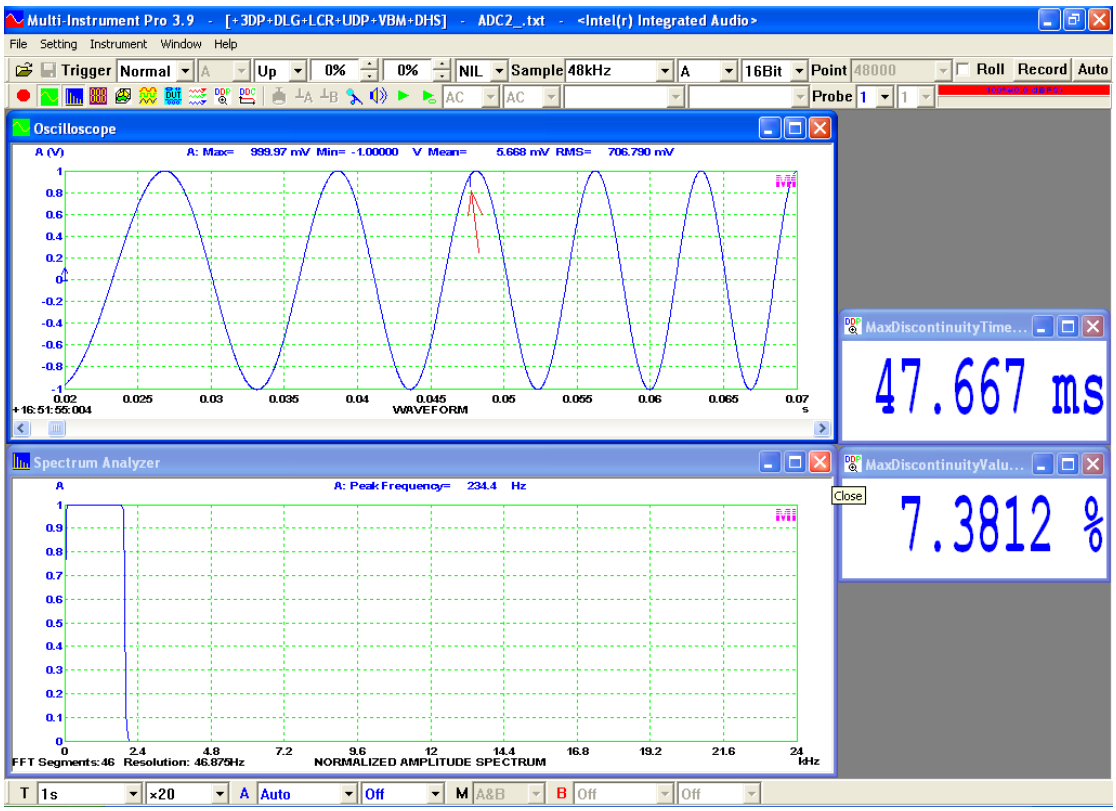
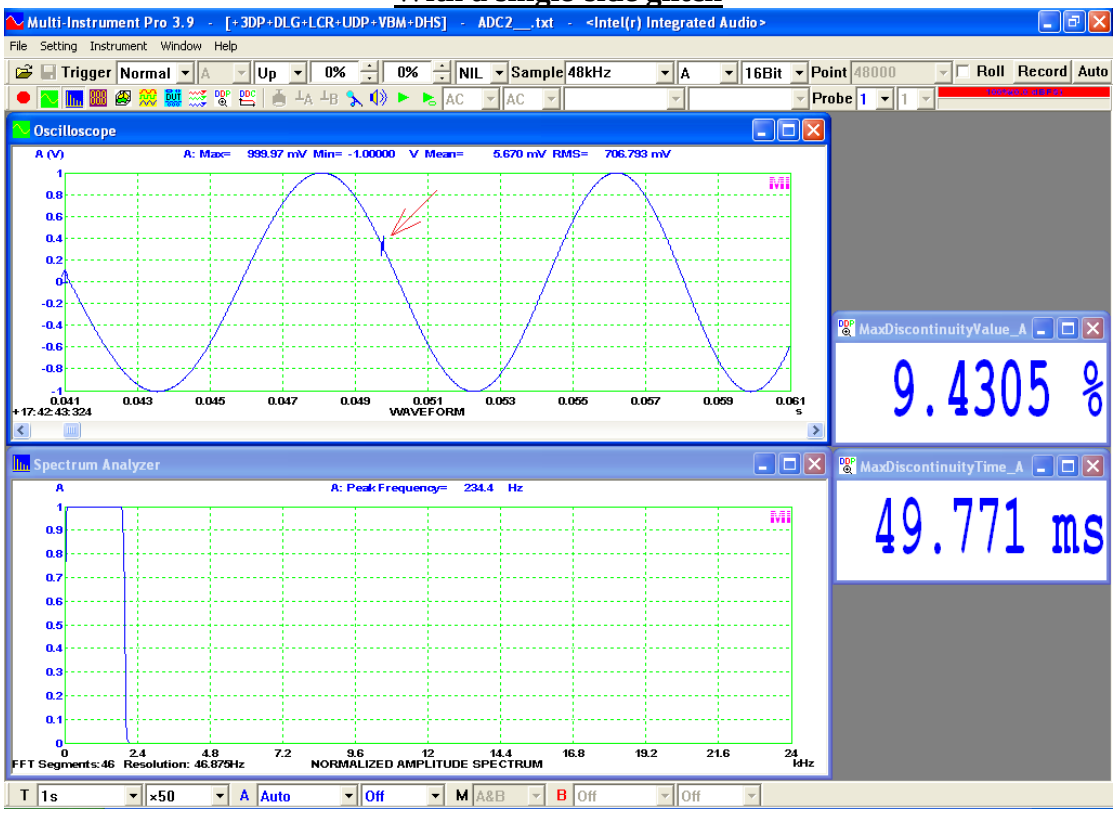
waveform, a waveform with a single-side glitch, and a waveform with a double-side glitch, respectively. It can be seen that the two glitches are detected correctly with their discontinuity values (7.3812% and 9.4305%) much higher than the maximum discontinuity value (0.0553%) of the normal case.



Normal



With a single-side glitch



With a double-side glitch

Discontinuity measurement has its applications in detection of digital audio dropouts and speaker rub & buzz, etc.. A dropout in digital audio is a brief loss of data that causes a discontinuity in a data stream. It usually occurs during data transmission. When it comes to analog domain, the discontinuity refers to the abruptness along a data curve.

An ideal sine wave is perfectly smooth and has a discontinuity value of 0% everywhere. A continuous sine sweep is very smooth and has very small discontinuity values along its waveform. These values depend on the sweep speed. The longer the sweep time, the narrower the sweep range, the lower the discontinuity values. If the sweep speed changes linearly, the discontinuity is a constant. Digital audio dropouts can be checked using a sine wave stimulus, while the speaker rub & buzz is usually tested using a continuous sine sweep.

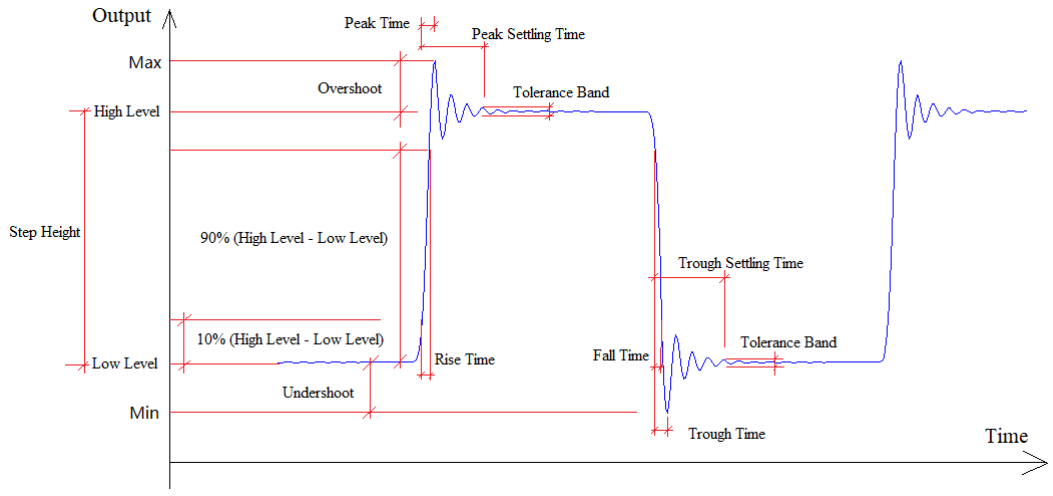
When a device under test (DUT) is excited with a sine wave or continuous sine sweep, its output waveform will be very smooth if its regular harmonic distortions are low and there is no irregular distortion occurs. Whether the regular harmonic distortions will cause abrupt changes of waveform depends on their orders and amplitudes as well as the phase response of the DUT. Higher order harmonic distortions contribute more to waveform abruptness than lower ones. Regular harmonic distortions can be quantized in frequency domain and this will be introduced later. On the other hand, the distortions caused by digital audio dropouts and speaker rub & buzz, are usually irregular and impulsive. They carry very little energy and thus do not show up very discernibly in frequency domain that contains only time averaged information. Traditional frequency domain methods like RMS-FFT and Total Harmonic Distortion

(THD) are thus not useful. In many cases, it is easier and better to detect them directly in time domain where the irregular distortions contribute much more to the discontinuity peaks than the regular harmonic distortions.

A sample panel setting file can be found at: psf\pro\ DiscontinuityTest.psf.

2.6.2.8.3.4 Step Response

Step response is the time evolution of the outputs of a system when its input changes from zero (low level) to one (high level) or one (high level) to zero (low level) instantly. Ten parameters of a step response are measured in the software: Low Level, High Level, Percentage Overshoot, Percentage Undershoot, Rise Time, Fall Time, Peak Time, Peak Settling Time, Trough Time and Trough Settling Time. They can be accessed through the following DDPs:
Channel A
StepResponseHighLevel_A(EU),
StepResponseHighLevel_A(EU),
StepResponseOvershoot_A(%),
StepResponseUndershoot_A(%),
StepResponseRiseTime_A(s),
StepResponseFallTime_A(s),
StepResponsePeakTime_A(s),
StepResponsePeakSettle_A(s),
StepResponseTroughTime_A(s),
StepResponseTroughSettle_A(s)
and a similar set of DDPs for Channel B. The following figure illustrates their definitions.



Percentage Overshoot is defined as (Max-HighLevel)/(HighLevel-LowLevel)×100%.

Percentage Undershoot is defined as (LowLevel-Min)/(HighLevel-LowLevel)×100%.

Rise Time is the time for the signal to rise from 10% of (HighLevel-LowLevel) to 90% of (HighLevel-LowLevel).

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Fall Time is the time for the signal to fall from 90% of (HighLevel-LowLevel) to 10% of (HighLevel-LowLevel).

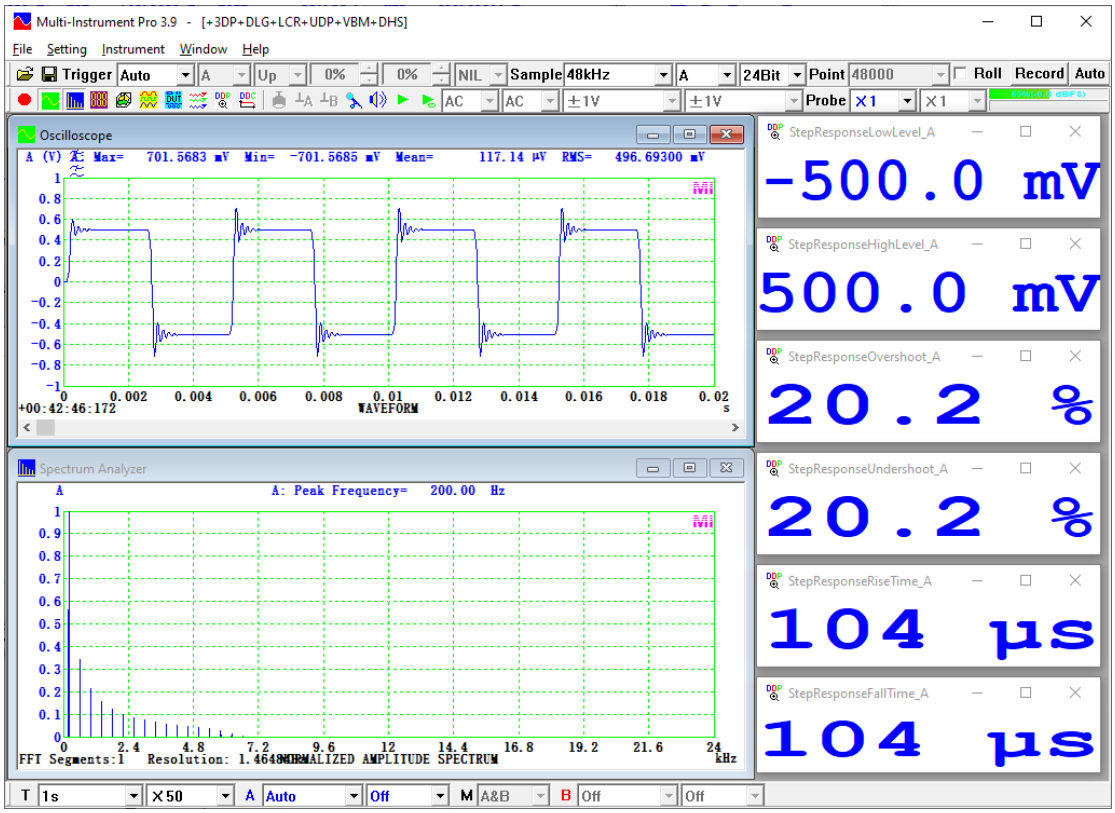
Peak Time is the time for the signal to rise from 10% of (HighLevel-LowLevel) to its peak value.

Peak Settling Time is the time for the signal to reach and remain within a given tolerance band of the HighLevel from 10% of (HighLevel-LowLevel). The tolerance used in the software is ±2% of (HighLevel-LowLevel).

Trough Time is the time for the signal to fall from 90% of (HighLevel-LowLevel) to its trough value.

Trough Settling Time is the time for the signal to reach and remain within a given tolerance band of the LowLevel from 90% of (HighLevel-LowLevel). The tolerance used in the software is ±2% of (HighLevel-LowLevel).

HighLevel-LowLevel is called step height. For the step response analysis algorithm to work properly, the waveform in the Oscilloscope must contain a rising edge, a falling edge, both a rising edge and a falling edge, or a multiple of periods of them. The following figure shows a measurement example. Percentage Overshoot is a function of damping ratio. It could be used to derive the damping ratio directly using User Defined Data Point (UDDP) introduced later.



A sample panel setting file can be found at: psf\VTDSO\StepResponseSR100MHz .psf.

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2.6.2.8.3.5 Echo

This option is used to analyze the data captured in the Oscilloscope intelligently and measure the time delay between emission and echoes, if any. Together with the signal generator function of the software, it can be used as an echo sounder for depth or distance measurement, or time-domain reflectometer for cable discontinuity localization.

The data should be long enough to contain the entire emission and one or multiple echoes. Normally, the emission should be a short pulse such that the echoes can be clearly separated from it in the time domain. It should be noted that the system under test may alter the emitted waveform along the way and the echo waveform may become totally different. One advantage of the pulse method is that it does not require the emission and the echoes are similar in waveform. However, due to the short duration of the pulse, the energy fed into the system is limited. As a result, this method is not very immune to noises. In some systems where the linear and nonlinear distortions are small and thus the emission and echoes have similar waveforms, it is possible to use a continuous excitation signal with a broadband spectrum, such as white noise, swept sine, and MLS. In this case, the emission and echoes may be superimposed together and cannot be separated by eyes in the Oscilloscope, but the time delay between them can still be accurately measured by the algorithm here. The advantage of this method is that it is very immune to noises due to the large energy pumped into the system. The measured echo times are reflected in DDPs: EchoTime_A(s) for Channel A, and EchoTime_B(s) for Channel B. It is possible to convert them to depth or distance using User Defined Data Point (UDDP) introduced later.

The following screenshot shows an example of the pulse method where the emission and the echoes are quite similar.

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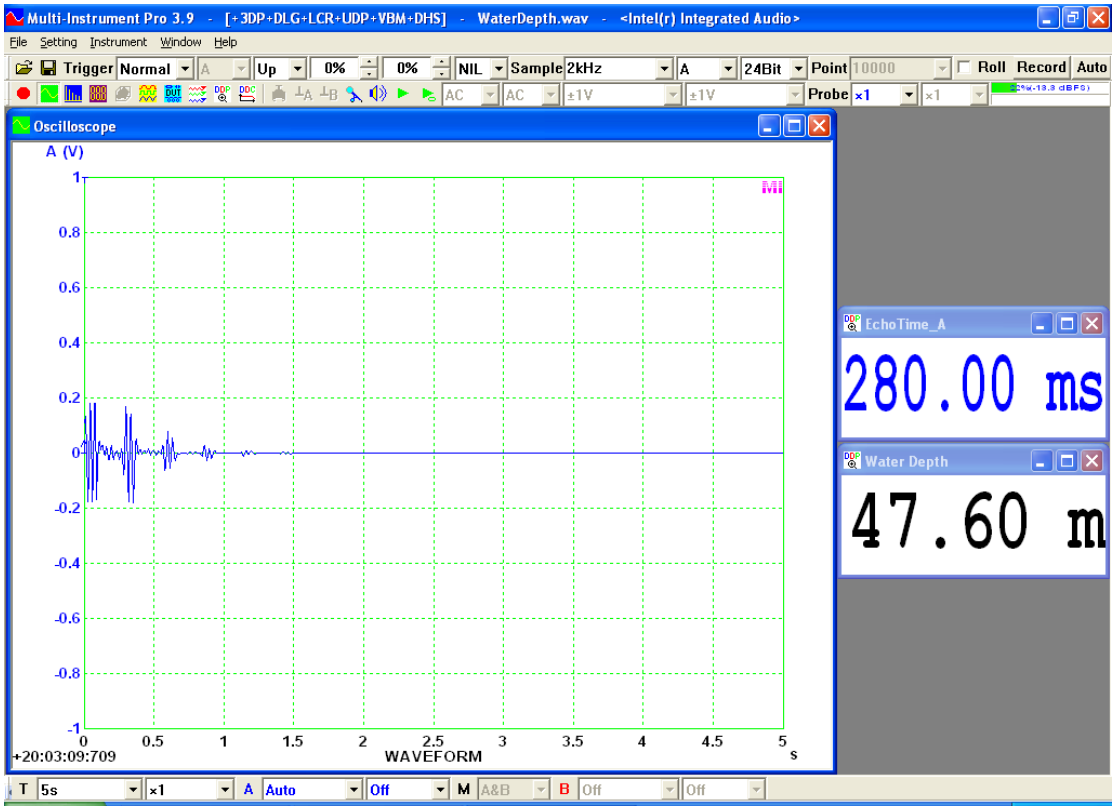
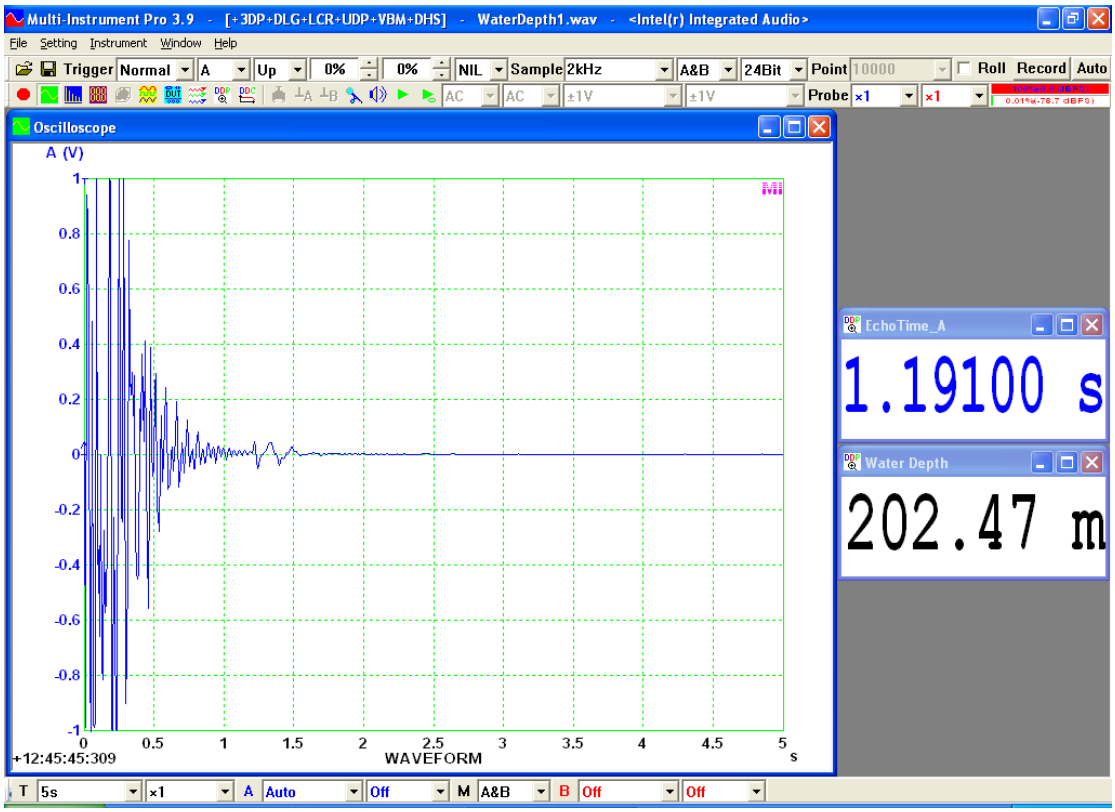
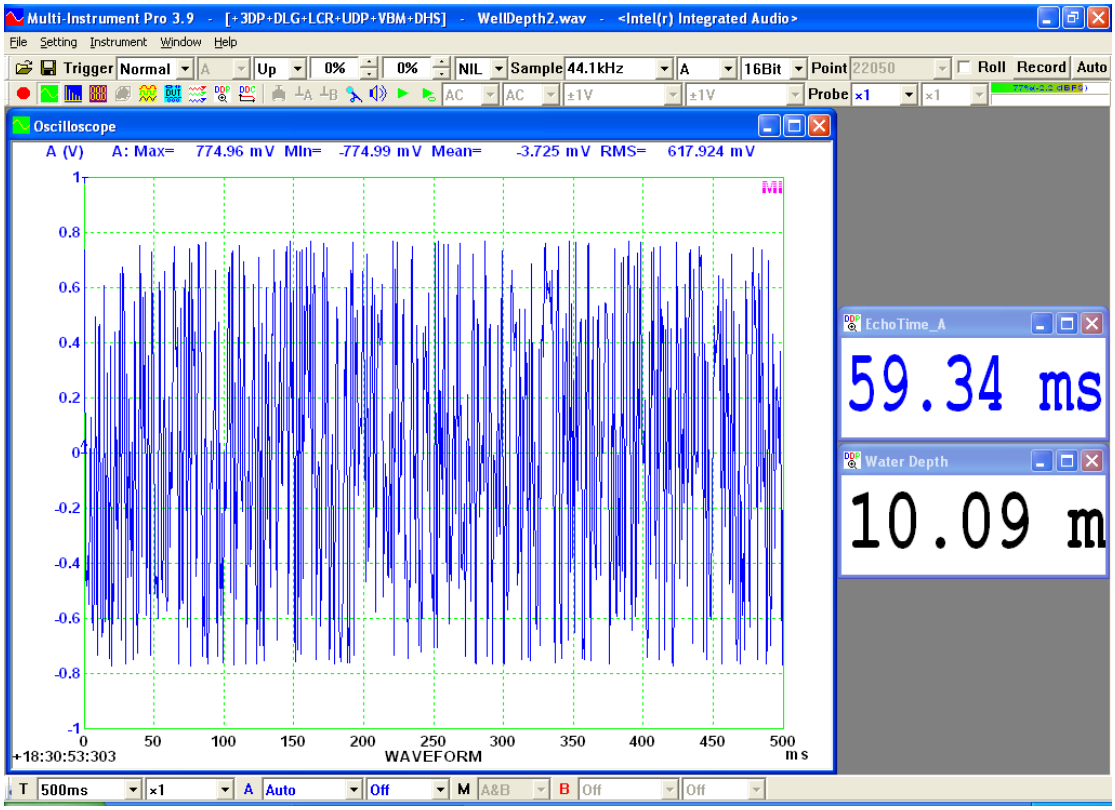


The following screenshot shows an example of the pulse method where the emission and the echoes are not similar.



The following screenshot shows an example of the continuous excitation method using MLS.



2.6.2.8.3.6 Damping Ratio

Damping ratio is a dimensionless measure describing how oscillations in a system decay after a disturbance. It is a system parameter which can vary from undamped (= 0), underdamped (< 1), through critically damped (= 1), to overdamped (>1). It is measured here using free vibration decay curve in time domain. This method is applicable to vibration with single degree of freedom. The measured values are reflected in DDPs: DampingRatio_A for Channel A, and DampingRatio_B for Channel B. It is possible to convert them to Q factor by the formula:

[Q factor] = 0.5 / [Damping Ratio]

Another common method to measure the damping ratio is the so-called half-power method. It measures the magnitude frequency response of the system first, and then derives the damping ratio based on the following formula:

[Damping Ratio] ≈ [-3dB Bandwidth] / ([Peak Frequency] × 2)

In Multi-Instrument, the above formula can be used to define a UDDP for damping ratio measurement (for Channel A) in frequency domain as follows:

([BandWidthHighLimit_A(Hz)]-[BandWidthLowLimit_A(Hz)])/2/[f1Freq_A(Hz)]

The Spectrum Analyzer should be set to bandwidth measurement mode in order for the above formula to work. The half-power method, although strictly applicable only to lightly damped single degree of freedom systems, is frequently applied to well-

separated modes of multi-degree of freedom systems. It should be noted that the frequency resolution must be set fine enough in order to achieve sufficient measurement accuracy in this method.

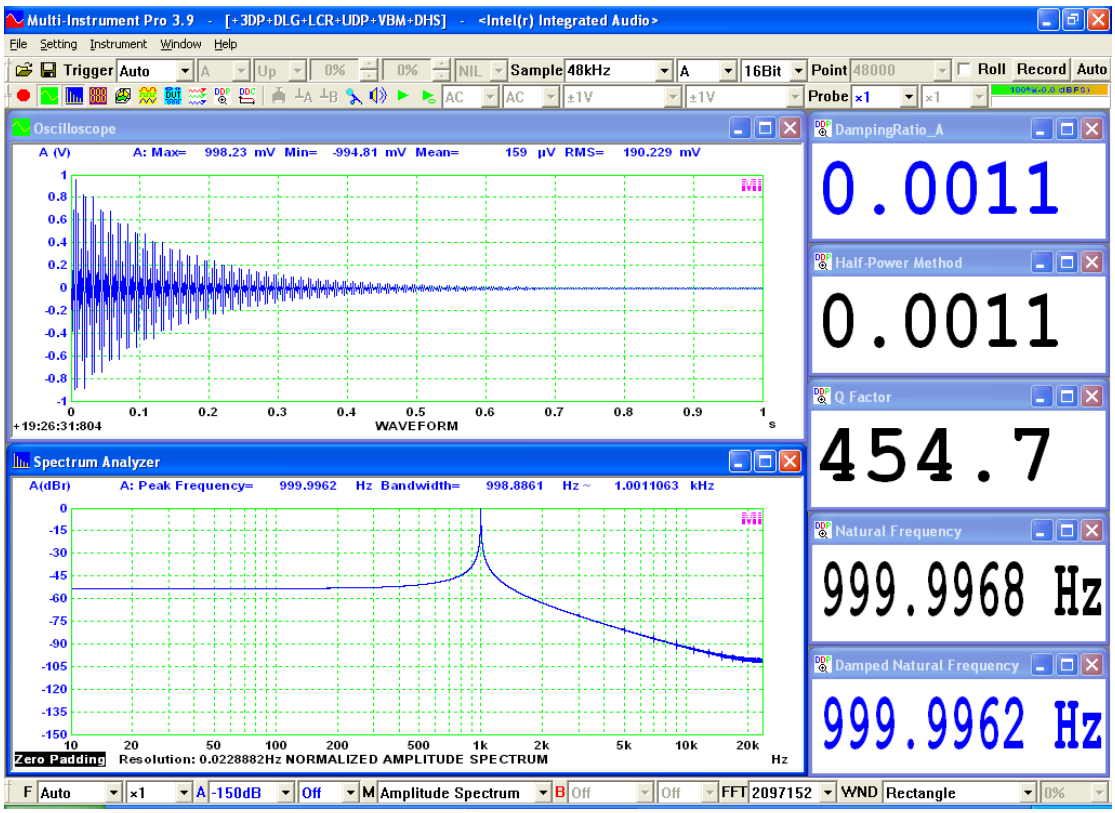
With the damping ratio and damped natural frequency (i.e. peak frequency) measured, the natural frequency of an underdamped system can be derived as follows:

$$[\text{Natural Frequency}] = [\text{Damped Natural Frequency}] / (1-[\text{Damping Ratio}]^2)^{1/2}$$

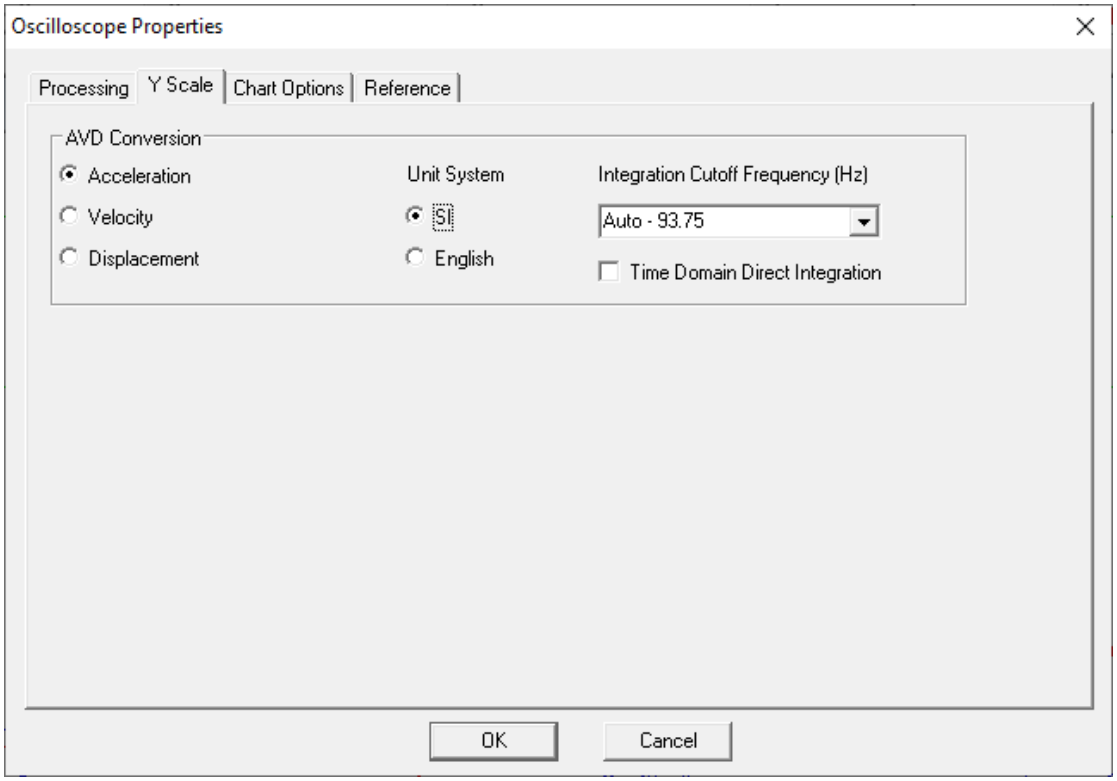
Using the above formula to define a UDDP for natural frequency measurement (for Channel A) in Multi-Instrument, we have:

[f1Freq_A(Hz)]/SQRT(1- POW([DampingRatio_A],2))

The following screenshot shows an impulse response in the Oscilloscope and thus the magnitude frequency response in the Spectrum Analyzer. It demonstrates that the damping ratios measured using the free vibration decay method and the half-power method match very well. The measured Q factor and natural frequency are also shown.



2.6.2.9 Oscilloscope Y Scale (Setting SubMenu) (ALT-S-Y)



The options for Y Scale allow waveform conversion among acceleration, velocity and displacement, when the raw data are acquired from acceleration (engineering units: g or m/s^2), velocity (engineering unit: m/s or i/s) or displacement (engineering unit: m or i) sensors. The statistical parameters of them are summarized in the vibrometer function of the Multimeter. The Oscilloscope in the following figure shows the velocity waveform obtained by integrating the raw acceleration data over time. A high pass filter with a very low cut-off frequency (configurable) is used to remove the errors accumulated during the integration process. The cutoff frequency for the "Auto" option varies based on the sampling rate, and its current value is shown on the right hand side. The high pass filter may cause some unwanted "boundary effects" at one or both ends of the data. These data are disregarded by the vibrometer and marked with a cross.

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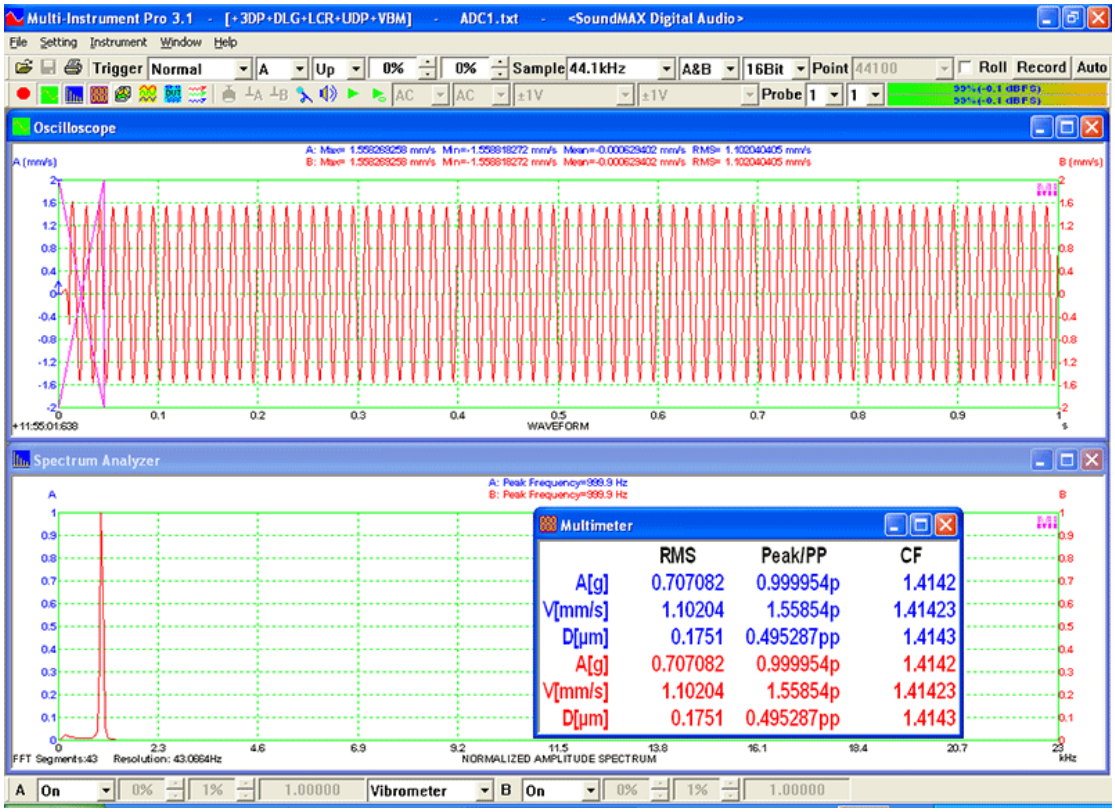
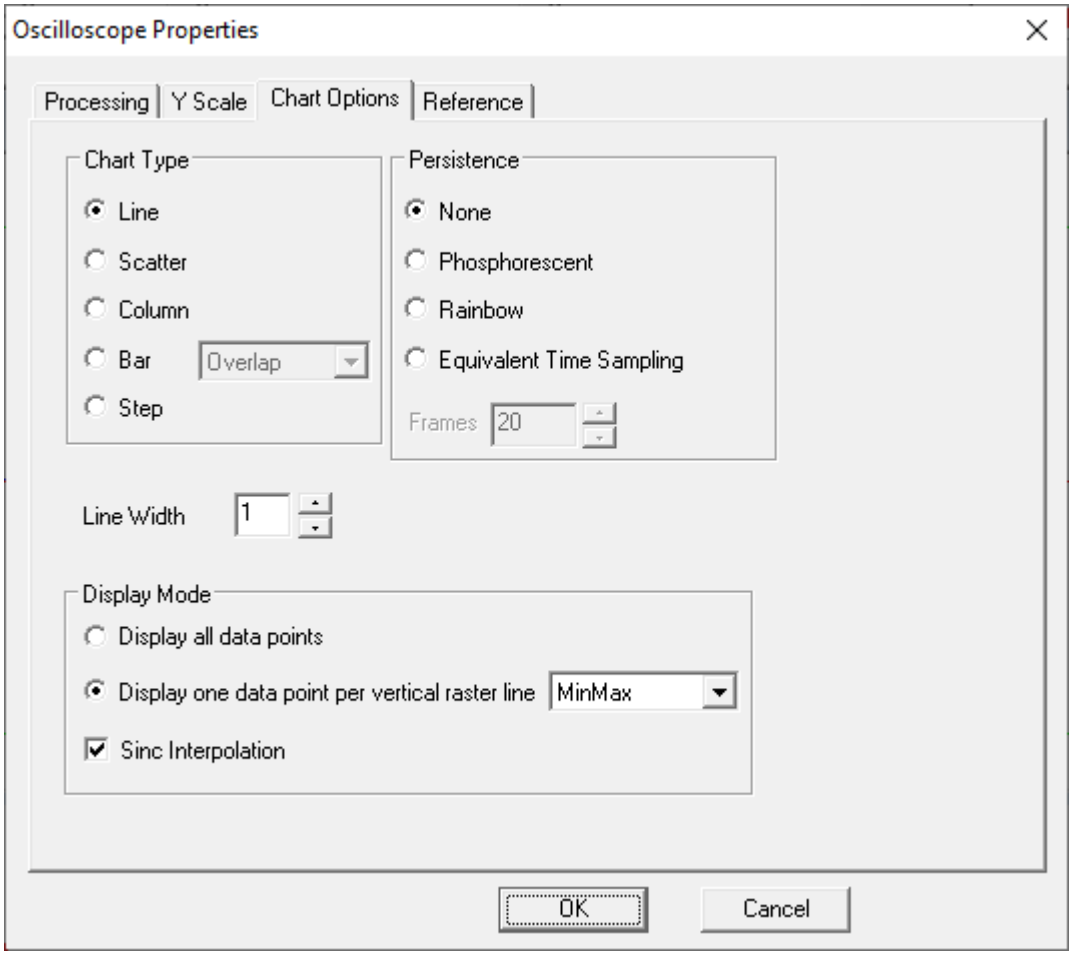


Both SI and English unit systems are supported. The software will automatically choose the unit system according to the sensor sensitivity units entered via [Setting]>[Calibration]>"Sensor">"Unit". Manual switching between the two unit systems is only allowed when acceleration sensors are used.

Conversion from acceleration to velocity and subsequently from velocity to displacement requires integration. Two integration methods are available: time domain integration and frequency domain integration. Generally, frequency domain integration should be employed. It always integrates a signal with its DC component removed first. It is the default setting in the software. Under some special cases, for instance, dealing with a short-time signal with DC and very-low-frequency components, such as a half-sine shock pulse, time domain direct integration may be more appropriate. It does not remove DC component first. The high-pass filtering is only applied after the first-time integration. When "0" is manually entered for "Integration Cutoff Frequency (Hz)", time domain direct integration will be enforced, and no high-pass filtering will performed.

2.6.2.10 Oscilloscope Chart Options (Setting SubMenu) (ALT-S-O)



2.6.2.10.1 Chart Type

Five chart types are supported: Line, Scatter, Column, Bar and Step. This function can also be accessed by double clicking in the area on the top of the plot region. The "Column" chart has two subtypes: Overlap and Interleave, while the "Bar" chart includes three subtypes: Overlap, Interleave, and Overlay. These subtypes control how the data from multiple channels are superimposed.

2.6.2.10.2 Line Width

Line width can be adjusted from 1 to 10. The default value is 1.

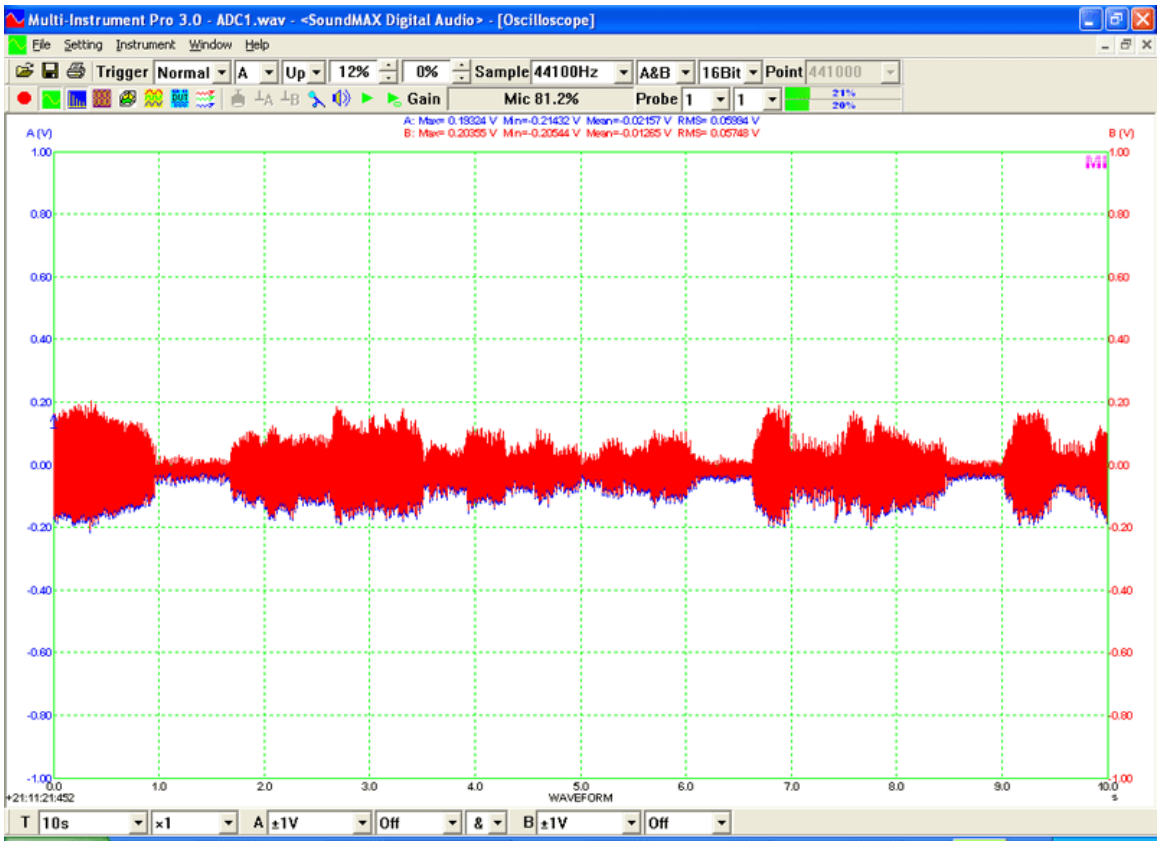
2.6.2.10.3 Display Mode

There are two ways to display a frame of data:

- Display all data points
  In this mode, all data points will be plotted. When the total number of data points to be plotted is greater than the total number of the horizontal pixels of the plot region, multiple points may be plotted per vertical raster line. For example, if the total number of data points to be plotted is 10000 and the horizontal length of the plot region is 1000 pixels, then 10 points will be plotted per vertical raster line on

average. The advantage of this display mode is that you are able to see all peaks and troughs in the data. However, the screen refresh becomes slow and the data curve becomes clustered when the total number of data points is large.

The following figure illustrates a ten-second sound record with a sampling frequency of 44100 Hz under this display mode.



- Display one data point per vertical raster line
  In this mode, one data point is plotted per vertical raster line at most. Thus the screen refreshes very fast even if the total number of data points to be presented is very large. There are three methods to determine the vertical value for each vertical raster line: Interpolation, Average and MinMax.

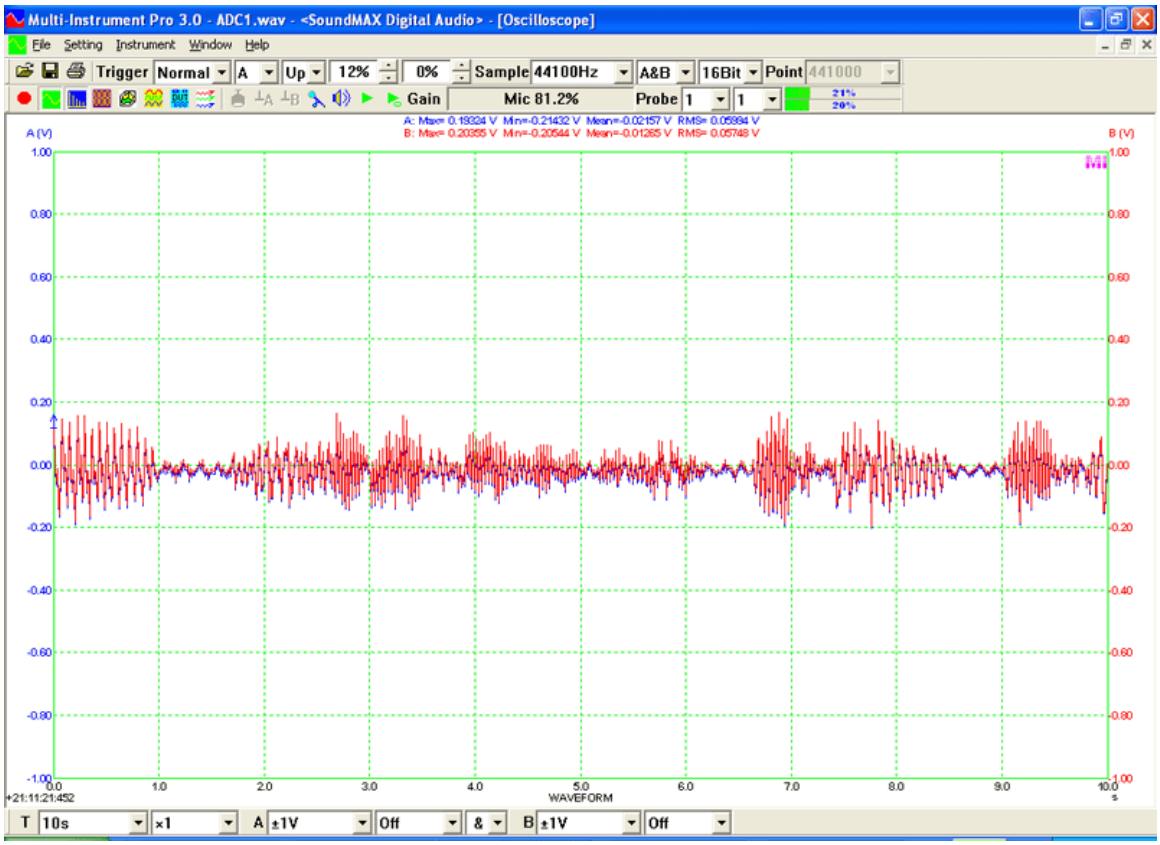
  With the Interpolation method, the vertical value for each vertical raster line is interpolated or uniformly down-sampled from the original data. The disadvantage is that it might miss some characteristics of the data such as peak values, and under some circumstances gives you a wrong impression that something is wrong with the waveform. This phenomenon is called screen aliasing.

  With the Average method, the vertical value for each vertical raster line is the average value of the original data that fall within that vertical raster line. It has a similar disadvantage to the Interpolation method, but it reduces noise along the curve.

  With the MinMax method, the vertical values for every two consecutive vertical raster lines are the minimum and maximum values of the original data that fall

within the two vertical raster lines. This approach highlights the vertical range of the original data, showing variability more effectively than the previous two methods.

The following figure shows the same sound record as the previous example under this display mode with the Interpolation method.



It should also be noted that Lissajous Pattern View will always display all collected data points irrespective of the display mode setting.

The software has an automatic display mode switching feature. In "Display one data point per vertical raster line" mode, the software will switch to "Display all data points" mode internally when the total number of data points in time domain is less than twice of the horizontal length of the plot region in the Oscilloscope. In the Spectrum Analyzer, the software will switch to "Display all data points" mode internally when the total number of data points in frequency domain is less than 4 folds of the horizontal length of the plot region. The threshold value in the Spectrum Analyzer is higher because the data curve in frequency domain tends to change more abruptly (thus more likely to miss the peak values) than that in time domain. Note that the horizontal length of the plot region is increased by multiple times with the corresponding multiplier. Therefore, you are always able to see all data points if you zoom in sufficiently irrespective of the display mode you set.

Due to the above automatic display mode switching feature, it is recommended that the "Display one data point per vertical raster line" mode be used as the normal setting. You can switch to "Display all data points" mode manually when necessary.

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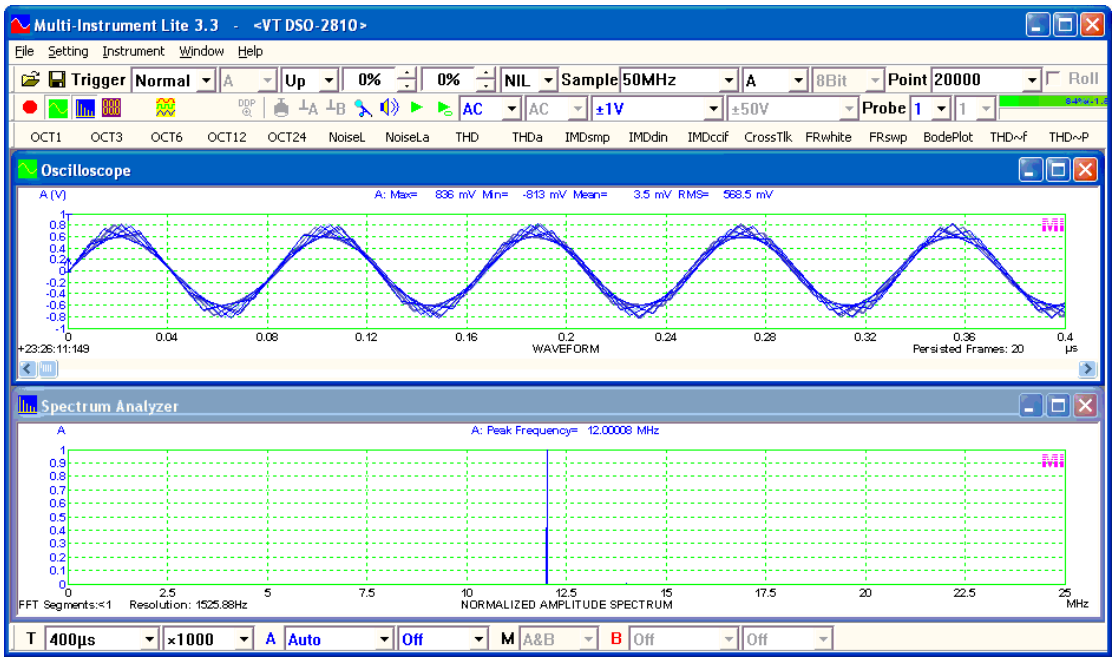
By factory default, the Oscilloscope, Spectrum Analyzer and Spectrum 3D Plot are all set to “Display one data point per vertical raster line” mode with the MinMax method.

2.6.2.10.4 Persistence

Under persistence mode, up to 200 most recent waveforms are kept in the computer memory. These waveforms can be superimposed / drawn in the oscilloscope view in three ways as follows. Note that the persistence mode affects the graphic rendering of these waveforms in the oscilloscope view only and does not have impact on any other aspects such as RMS values, spectrum analysis, data storing, etc..

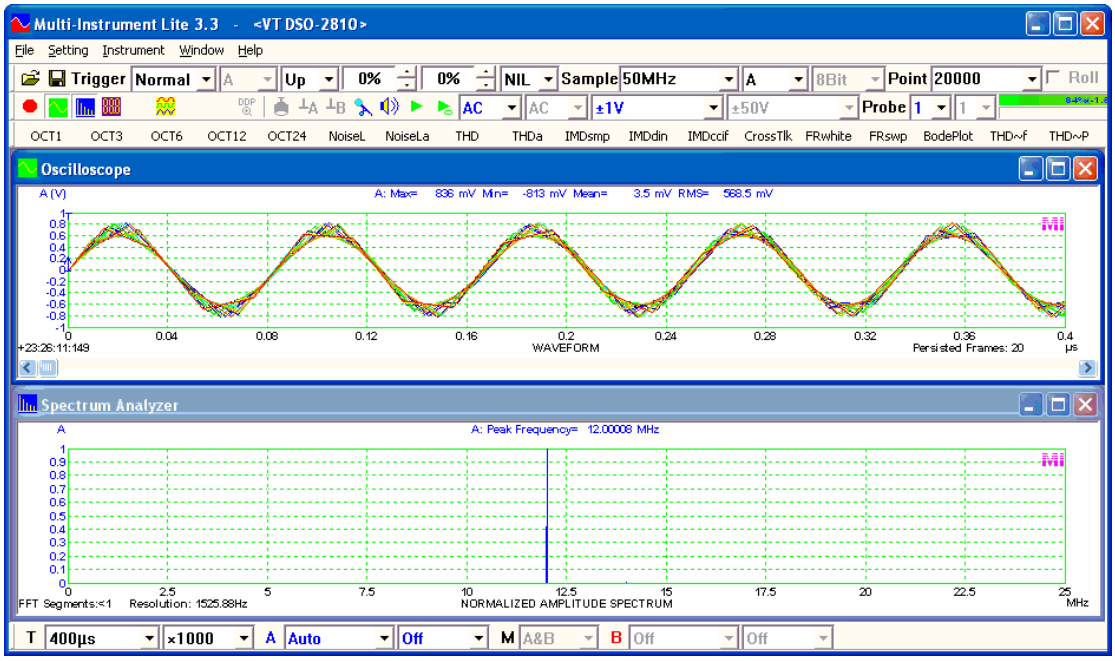
(1) Phosphorescent

The most recent waveform is drawn in the selected color for that channel, the least recent waveform is drawn in the background color, and the rest of waveforms are drawn in colors interpolated between these two according to their ages. This emulates the phosphor display of a conventional analog oscilloscope whereby the trace decays over time.



(2) Rainbow

Waveforms are drawn in colors derived from a predefined rainbow color palette according to their ages. Unlike the phosphorescent mode, the trace does not decay over time.



(3) Equivalent Time Sampling (ETS)

While a spectrum analyzer can analyze a signal correctly as long as the sampling frequency is at least twice the highest frequency contained in the signal (Nyquist Sampling Theorem), an oscilloscope, under linear interpolation mode (refer to the next section), normally requires the sampling frequency to be at least 10 times the signal frequency in order to acquire sufficient samples in one cycle to draw the waveform in reasonably good shape. ETS can be used to greatly increase the number of samples per cycle by accumulating samples over many cycles. It is most useful when the number of samples per cycle acquired using real-time sampling is inadequate. One prerequisite to use ETS is that the signal under test must be repetitive and the sampling is properly triggered. The following two figures show a comparison between real time sampling and ETS. The first one samples a 12MHz sine wave with a 50MHz clock in real time sampling mode while the second one samples the same signal with a 50MHz clock in ETS mode. The latter looks much better.

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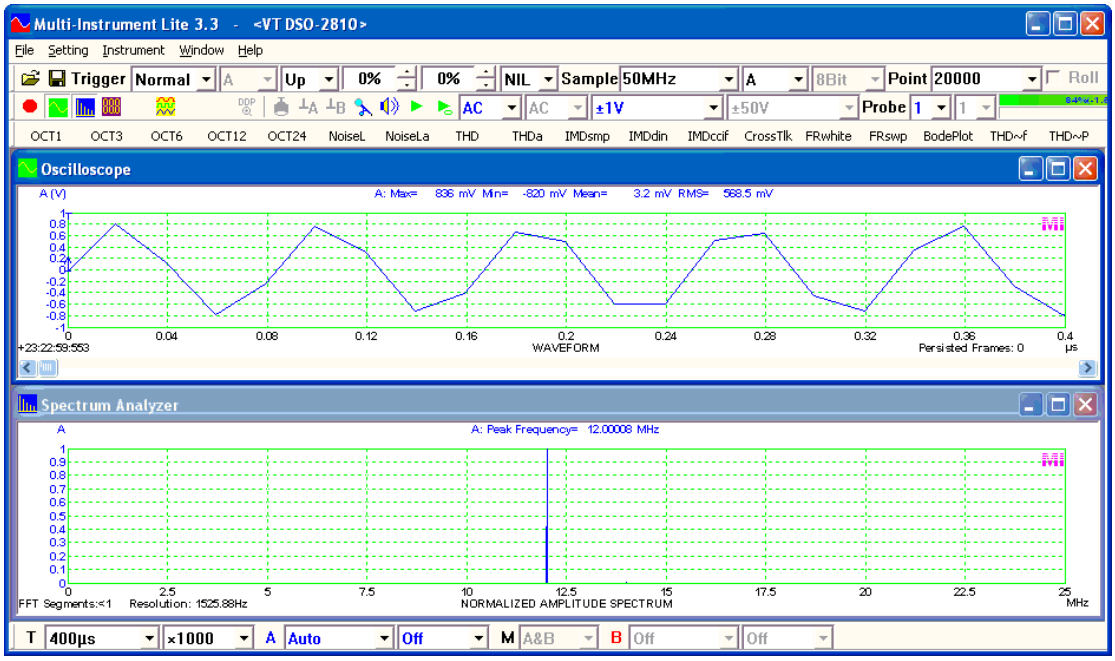
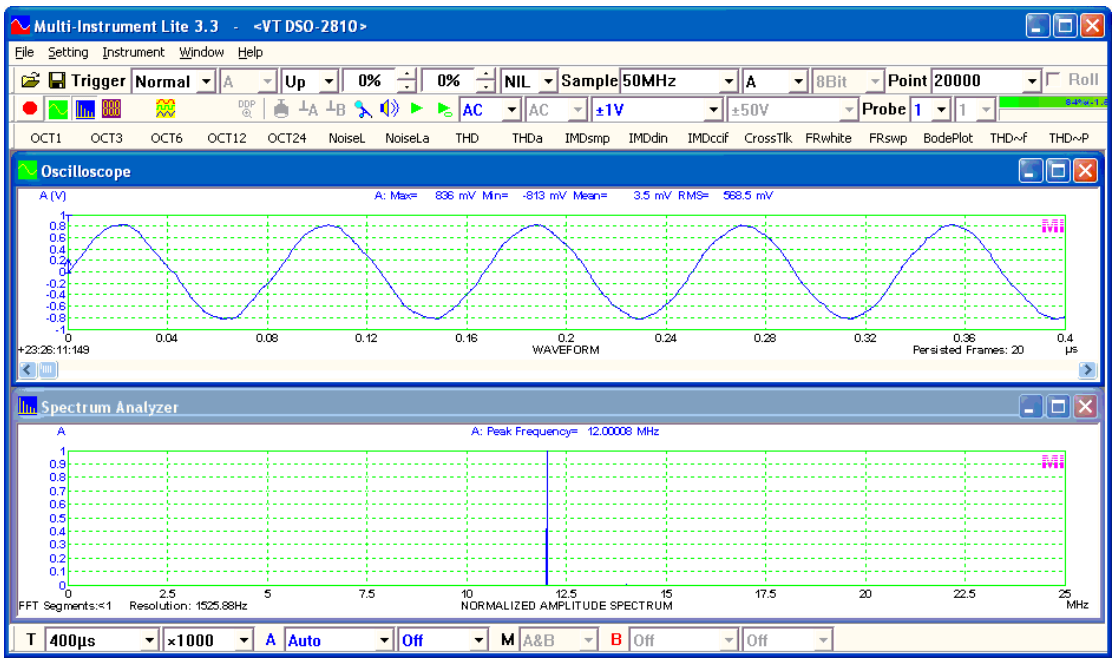




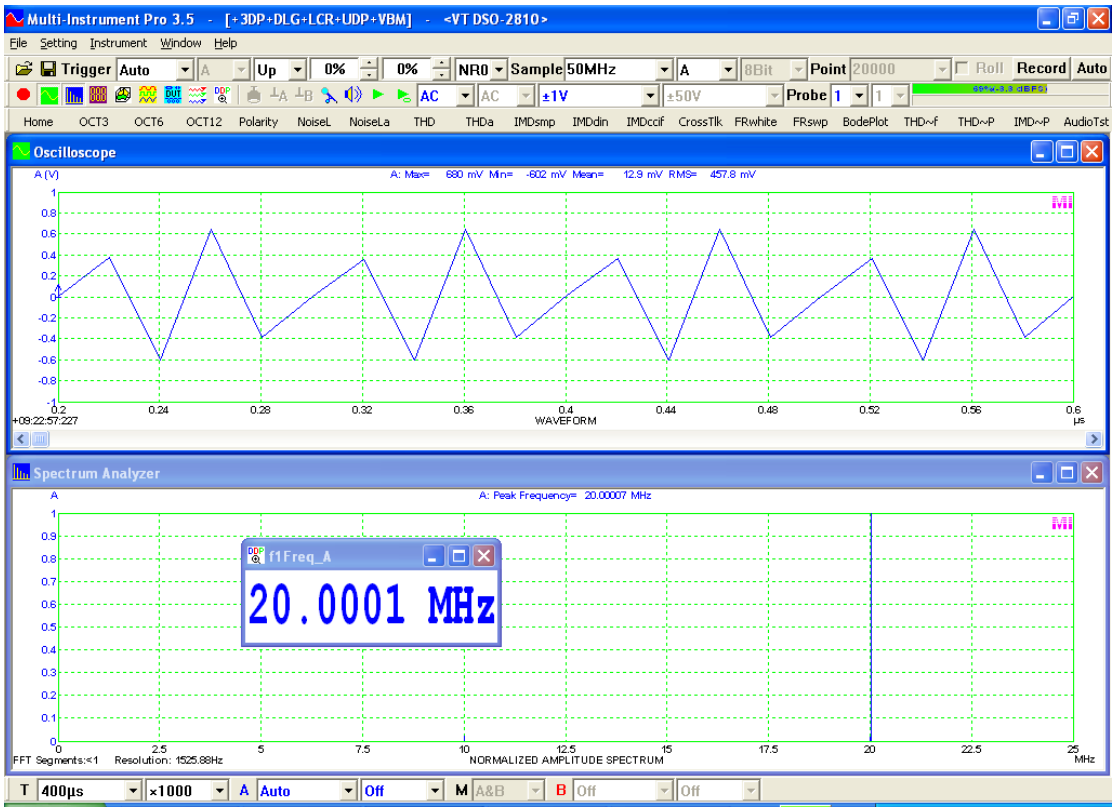
2.6.2.10.5 Interpolation Mode

Interpolation is used when two adjacent samples span more than two pixels in horizontal direction on the screen. That is, there is at least one pixel in between the two samples horizontally. Interpolation mode determines the way to connect two adjacent samples on the screen when plotting a waveform. It affects the rendering of a waveform on the screen only and does not have impact on any other aspects such as RMS values, spectrum analysis, data storing, etc.. Two methods are available: Linear Interpolation and SINC Interpolation.

(1) Linear Interpolation

Linear interpolation is the most straightforward method. It connects two adjacent samples using a straight line on the screen. When the sampling frequency is at least

5~10 times the signal frequency, linear interpolation can gives reasonably good waveforms. When the ratio of the sampling frequency to the signal frequency is less than 5~10 (but still higher than 2 as dictated by Nyquist Sampling Theorem, i.e. the spectral analysis results are still absolutely correct), the waveform plotted using linear interpolation can be severely distorted. The following example samples a 20MHz sine wave with a 50MHz clock in real time and then renders the waveform using linear interpolation. The sampling-frequency-to-signal-frequency ratio is 2.5. The waveform in the oscilloscope looks very distorted.

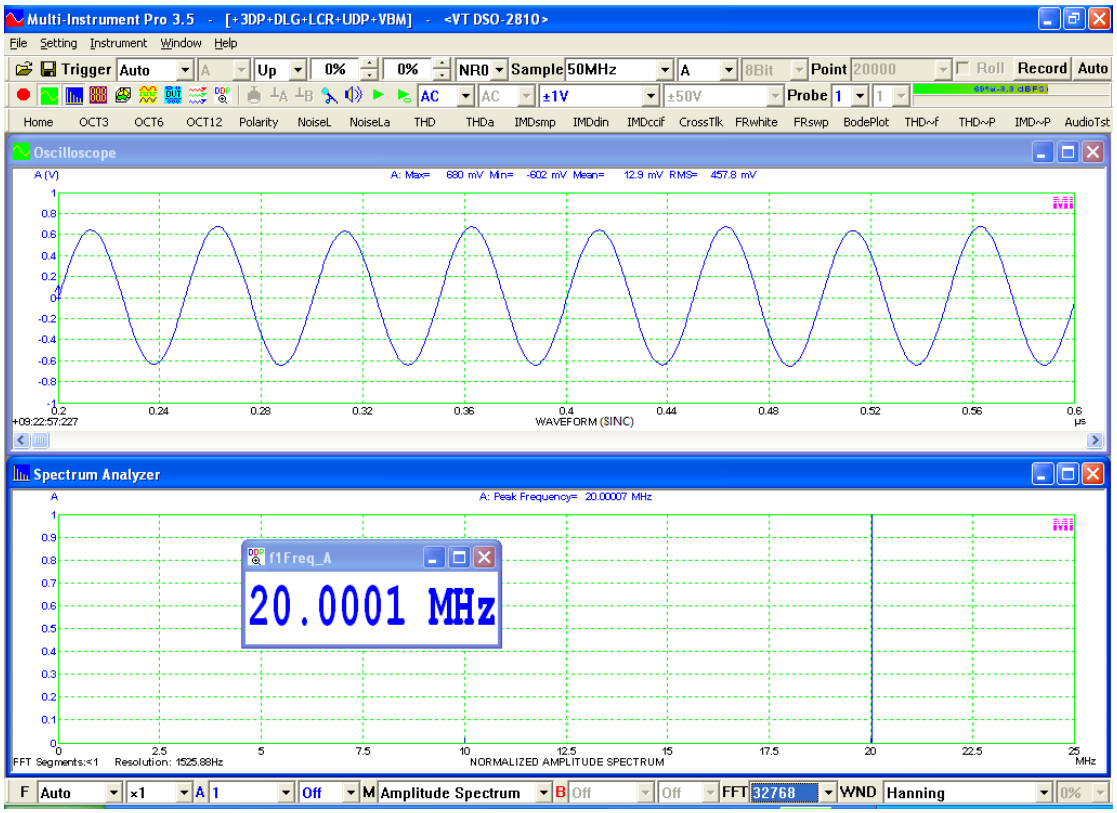


(2) SINC Interpolation

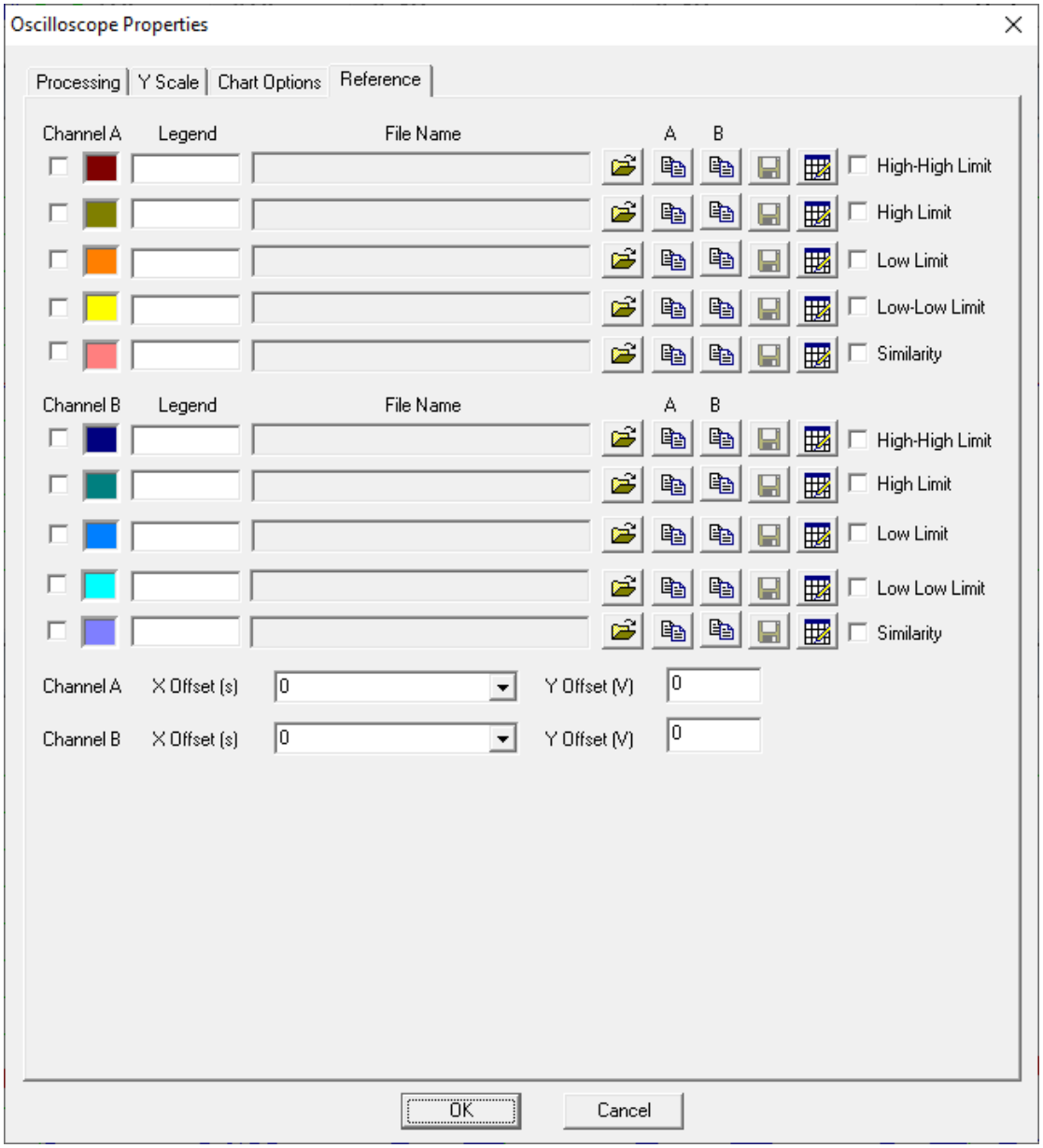
Nyquist-Shannon Sampling Theorem states that an analog signal that has been sampled can be perfectly reconstructed from the samples if the sampling frequency is greater than twice of the highest frequency in the original signal. If we know that the signal is band-limited within the Nyquist frequency (1/2 of the sampling frequency), then we can use SINC interpolation to reconstruct the original signal waveform on the screen. A sampling-frequency-to-signal-frequency ratio greater than 2.5 would give nearly perfect results while a ratio in the range of 2.1~2.5 would still give reasonably good results. Many DAQ devices such as VT DSOs and sound cards are equipped with anti-aliasing filters and satisfy the requirements of Nyquist-Shannon Sampling Theorem, SINC interpolation is thus preferred as it produces much better curve fitting results than linear interpolation. SINC interpolation is selected by default in the software. **If you want to display signals such as square waves, which ideally contain frequencies higher than the Nyquist frequency, then de-selecting it to enable linear interpolation will give the square waveforms a better look.** When SINC interpolation is applied, "SINC" will be displayed at the bottom middle of the Oscilloscope window. It should be noted that SINC interpolation may have some

"boundary effects" within a very short range (less than 32 samples) at both ends of the waveform.

The following example samples a 20MHz sine wave with a 50MHz clock in real time and then renders the waveform using SINC interpolation. The sampling-frequency-to-signal-frequency ratio is 2.5. The waveform in the oscilloscope looks nearly perfect.



2.6.2.11 Oscilloscope Reference (Setting SubMenu) (ALT-S-R)



Reference curves are very useful for data comparison. The software allows you to configure up to 5 reference curves for each channel. Reference curves can be configured by copying the currently displayed data curve from Channel A or Channel B, or loading a reference curve file from the hard disk.

Each line in the above figure consists of the parameters/controls for one reference curve. From left to right, they are:

- Display/Hide Checkbox
  It is used to display/hide the respective reference curve. If the reference curve has not been saved to the hard disk, hiding it will clear it from the computer memory.

- Color Selection Box
  Clicking on it will bring up a color selection window which allows you to select the color you want for that reference curve.

- Legend
  The software will automatically assign a textual label for the reference curve, such as “RefA0”, “RefB0”. You can overwrite it with your own preference.

- File Name
  If the reference curve is configured by copying the currently displayed data curve from Channel A or Channel B and it has not yet been saved to the hard disk, then it will be named as “Memory A0”, “Memory B0”….. If you save the reference curve to the hard disk, then the saved file name will be displayed here.

  If the reference curve is loaded from the hard disk, then the file name will be reflected here.

- File Open
  Use this command to load a reference file from the hard disk.

- Copy from Channel A
  Configure the reference curve by copying the currently displayed data curve from Channel A.

- Copy from Channel B
  Configure the reference curve by copying the currently displayed data curve from Channel B.

- File Save
  Use this command to save the reference curve data from the computer memory to the hard disk.

- File Edit
  Use this command to edit an existing reference file or create a new reference file.

- High-High Limit, High Limit, Low Limit, Low-Low Limit, Similarity
  If ticked, the reference curves are assigned as the respective limits for the real-time waveform. Whether the waveform breaks these limits and how much is exceeded can be checked through the respective DDPs:

  WaveformComparisonH_A:
  0: waveform in Ch. A is normal.
  <0: waveform in Ch. A exceeds the High Limit only. The absolute value indicates how much is exceeded.
  >0: waveform in Ch. A exceeds the High-High Limit. The absolute value indicates how much is exceeded.

  WaveformComparisonL_A:
  0: waveform in Ch. A is normal.
  >0: waveform in Ch. A falls below the Low Limit only. The absolute value indicates how much is exceeded.

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<0: waveform in Ch. A falls below the Low-Low Limit. The absolute value indicates how much is exceeded.

Similarly, WaveformComparisonH_B and WaveformComparisonL_B are used for Ch. B. "Similarity" function and its respective DDPs: WaveformSimilarity_A and WaveformSimilarity_B are reserved for future development.

The reference file (*.ref) is a Comma Separated Value (CSV) TXT file with the following format:

Example:

1, 0.000108441, 0.103241
2, 0.000131117, 0.0823364
…….

Each row contains the coordinates of a point of the reference curve with three variables: sequential number, X value, and Y value. Minimum two points must be specified per reference curve.

Some sample files are provided in the REF directory of the software and can be used as templates.

It is possible to enter a generic X offset value and a generic Y offset value for the reference curves in each channel. Two additional options are provided particular for X offset: (1) Align to Peak; (2) Align to Trough. If selected, each reference curve will be shifted automatically such that its peak or trough is aligned to that of the captured waveform. These two options are useful when there is only one peak or trough on the reference curve and the captured waveform.

2.6.2.12 Save Current Panel Setting as Default (Setting SubMenu) (ALT-S-F)

When you start the software at the very first time, system default panel setting set in the factory will be loaded. You can subsequently change the panel setting to whatever you want, and then save the panel setting as default via this command. This default setting will be loaded at startup next time. You can always go back to the system default panel setting by clicking [File]>[New] command.

The panel setting here includes the following parameters:

- Trigger Parameters
- Sampling Parameters
- Miscellaneous Parameters
- Oscilloscope Parameters
- Spectrum Analyzer Parameters
- Multimeter Parameters
- Spectrum 3D Plot Parameters
- Signal Generator Parameters
- Data Logger Parameters

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- LCR Meter Parameters
- Device Test Plan Parameters
- DDP Viewer Parameters
- DDP Array Viewer Parameters
- DDC Parameters
- Screen Layout, including which instrument to be shown and the position of each instrument on the screen.

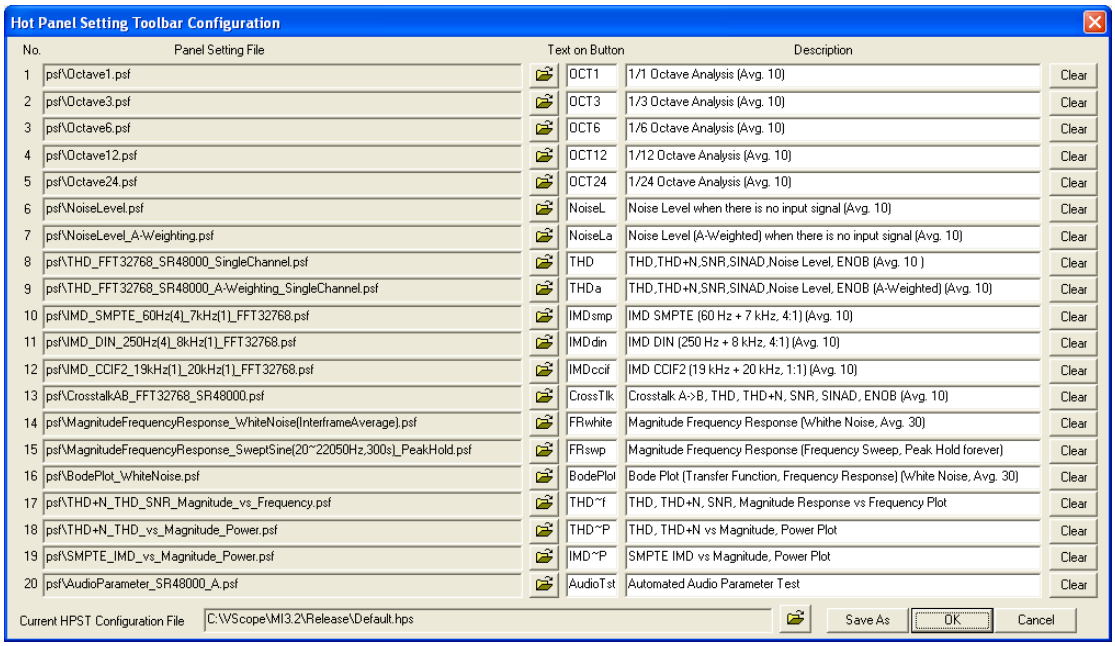
2.6.2.13 Save Current Panel Setting (Setting SubMenu) (ALT-S-S)

You can save the current panel setting into a Panel Setting File (*.psf) using this command. The saved panel setting can be loaded at a later time. This will free you from re-adjusting the panel setting for a particular type of tests again and again.

2.6.2.14 Load Panel Setting (Setting SubMenu) (ALT-S-L)

You can load a panel setting from a previously saved Panel Setting File (*.psf) using this command. This will free you from re-adjusting the panel setting for a particular type of tests again and again.

2.6.2.15 Configure Hot Panel Setting Toolbar (Setting SubMenu) (ALT-S-P)



You can configure up to 20 most frequently used panel settings in the Hot Panel Setting Toolbar so that these settings can be loaded by just a single mouse click on their respective buttons.

Each line in the above figure defines one button in the Hot Panel Setting Toolbar. The parameters from left to right are:

- Button No.
  The buttons are numbered from 1 to 20 from left to right in the Hot Panel Setting Toolbar.

- Panel Setting File
  The path and file name of the panel setting file assigned to the button using the File Load button below.

- File Load
  It is used to locate the panel setting file to be assigned to the button.

- Text on Button
  The text entered in this edit box will be shown on the button. The text should preferably contain 3~5 characters and reflect concisely the function of the button.

- Description
  The description entered in this edit box will be shown as a tooltip when you rest your mouse cursor on the button.

- Clear
  It is used to clear the button definition.

You can save the above configuration into a Hot Panel Setting Toolbar configuration file (*.hps) by clicking the "Save As" button at the bottom of the above dialog. You can load a Hot Panel Setting Toolbar configuration file by clicking the File Open button at the bottom of the dialog. The current Hot Panel Setting Toolbar configuration file used by the software is also displayed at the bottom.

The Hot Panel Setting Toolbar configuration file (*.hps) is a TXT file which can be edited directly using Windows Notepad. Direct editing allows the configuration of some special functions usually used for software customization. Panel Setting File field supports the following predefined texts for special functions:

STARTSTOP	Toggle button	Start / Stop the Oscilloscope
SETTING	Push button	Different setting windows should be developed for different customized software applications.
EXIT	Push button	Exit the software
HELP	Push button	Display a help file. The help file should be placed in the software's root directory. The file name should be put in the Description field.
PRINTSCREEN	Push button	Equivalent to [File]>[Print Screen]
RECORD	Toggle button	Equivalent to the Record button in the Sampling Parameter Toolbar
MONITOR	Toggle button	Monitor the signal captured by the selected SoundCardMME ADC device in the software through the system's default playback sound card. It is

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		independent of the sampling process of the Oscilloscope.
WAVFILEPLAY	Push button	Prompt the user to select a WAV file and play it back through the system's default playback sound card.
SYNCSTART	Toggle button	Set the Signal Generator in "Sync. No Loopback" mode and start it. The Oscilloscope will be set to Single Trigger mode and started afterwards.
TIMEDSYNCSTART	Toggle button	SYNCSTART command will be fired at a regular interval specified in the Description field in seconds.

In Pro edition and above, it is possible to change the placement of the Hot Panel Setting Toolbar: Top, Bottom, Left, Right. It is also possible to change the button appearance: Text (only a text is displayed), Text & Image (both a text and a predefined icon are displayed), Image (only a predefined icon is displayed), Custom Image (only a customized icon is displayed). These changes will take effect after software restart.

The ordinary buttons for panel setting file loading are all push buttons when the panel setting is not locked, otherwise they are interlocked toggle buttons and only one of them will be in "down" state at any time. In this case, the loaded panel setting file name will be indicated in the title bar of the main window. When the panel setting is locked via [Help]>[Lock Panel Setting], there is no way to modify the loaded panel setting, except that you can load another preconfigured panel setting file through another button in the Hot Panel Setting Toolbar. This arrangement is usually used by users with access right at operator level. Those with engineer-level access right are normally allowed to modify various settings.

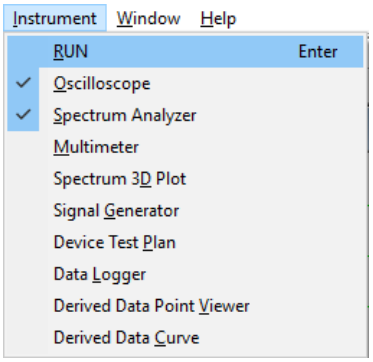
2.6.2.16 Show Hot Panel Setting Toolbar (Setting SubMenu) (ALT-S-H)

It is a toggle command to show or hide the Hot Panel Setting Toolbar.

2.6.2.17 Change Password (Setting SubMenu) (ALT-S-W)

You can change the password for unlocking the panel setting here. No password is set initially. If you want to add a password, you can leave the Current Password field empty and enter and confirm the new password. For changing the existing password, you need to enter the current password, enter and confirm the new password. To delete the password, just enter the current password and leave the New Password and New Password (Confirm) fields empty.

2.6.3 Instrument SubMenu



This submenu provides access to opening / closing each individual instrument provided.

2.6.3.1 Run (Instrument SubMenu) (ALT-I-R, CTRL-R or Enter)

This command will toggle between Run and Stop for data acquisition. The command is also available through the button in the Instrument Toolbar. The color of the button will toggle between red and green to indicate "Stop" and "Run" status.

2.6.3.2 Oscilloscope (Instrument SubMenu) (ALT-I-O)

This command will toggle between Open and Close for the Oscilloscope. The command is also available through the button in the Instrument Toolbar.

2.6.3.3 Spectrum Analyzer (Instrument SubMenu) (ALT-I-S)

This command will toggle between Open and Close for the Spectrum Analyzer. The command is also available through the button in the Instrument Toolbar.

2.6.3.4 Multimeter (Instrument SubMenu) (ALT-I-M)

This command will toggle between Open and Close for the Multimeter. The command is also available through the button in the Instrument Toolbar.

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2.6.3.5 Spectrum 3D Plot (Instrument SubMenu) (ALT-I-D)

This command will toggle between Open and Close for the Spectrum 3D Plot. The command is also available through the button in the Instrument Toolbar.

2.6.3.6 Signal Generator (Instrument SubMenu) (ALT-I-G)

This command will toggle between Open and Close for the Signal Generator. The command is also available through the button in the Instrument Toolbar.

2.6.3.7 Device Test Plan (Instrument SubMenu) (ALT-I-P)

This command will toggle between Open and Close for the Device Test Plan. The command is also available through the button in the Instrument Toolbar.

2.6.3.8 Data Logger (Instrument SubMenu) (ALT-I-L)

This command will launch a Data Logger window. Up to 8 Data Logger windows can be opened. The command is also available through the button in the Instrument Toolbar.

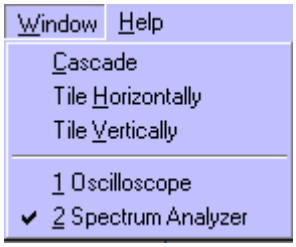
2.6.3.9 Derived Data Point Viewer (Instrument SubMenu) (ALT-I-V)

This command will launch a DDP viewer window. Up to 24 DDP viewer windows can be opened. The command is also available through the button in the Instrument Toolbar.

2.6.3.10 Derived Data Curve (Instrument SubMenu) (ALT-I-C)

This command will launch a DDC window. Up to 8 DDC windows can be opened. The command is also available through the button in the Instrument Toolbar.

2.6.4 Window SubMenu



The Window submenu provides the access to window arrangement functions.

2.6.4.1 Cascade (Window SubMenu) (ALT-W-C)

This command will cascade the opened views and dialog boxes.

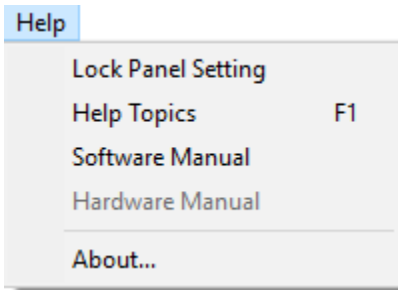
2.6.4.2 Tile Horizontally (Window SubMenu) (ALT-W-H)

This command will tile the opened views and dialog boxes horizontally. For example, if both the Oscilloscope and the Spectrum Analyzer are opened, then this command will make each of them take up half of the screen in vertical direction.

2.6.4.3 Tile Vertically (Window SubMenu) (ALT-W-V)

This command will tile the opened views and dialog boxes vertically. For example, if both the Oscilloscope and Spectrum Analyzer are opened, then this command will make each of them take up half of the screen in horizontal direction.

2.6.5 Help SubMenu



The Help SubMenu provides access to the help function and software version and license information. It also provides the function for locking/unlocking the panel setting.

2.6.5.1 Lock Panel Setting (Help SubMenu) (ALT-H-L or Ctrl-K)

This command will toggle between locking and unlocking panel setting. If the panel setting is locked, then only the following operations are allowed by the software:

- Access to the Help SubMenu
- Run / Stop the Oscilloscope and the Signal Generator.
- Load a panel setting file via a button in the Hot Panel Setting Toolbar
- Save a file
- Invert Input Signal, Channel A Zeroing, Channel B Zeroing, Windows Recording Control, Windows Volume Control, Waveform Play, Waveform Cyclic Play

You can add a password for unlocking the panel setting via [Setting]>[Change Password], to allow only authorized persons to unlock the panel setting if it has been

locked. The software will start as locked if you tick “Lock Panel Setting after startup” via [Setting]>[Display].

You can establish two levels of operation of the software by combining the lock panel setting function with the Hot Panel Setting Toolbar. The two levels are: Engineer Level and Operator Level. The engineers will be given the password to unlock the panel setting, and they are able to access all functions provided by the software. The engineers should assign the frequently used panel setting files to the Hot Panel Setting Toolbar so that the operators can load them when the panel setting is locked.

2.6.5.2 Help Topics (Help SubMenu) (ALT-H-H or F1)

This command will open the Help window. You can use the Content Tab, Index Tab or Search Tab to locate the help topic you want.

2.6.5.3 Software Manual (Help SubMenu) (ALT-H-S)

This command will open the software manual.

2.6.5.4 Hareware Manual (Help SubMenu) (ALT-H-M)

This command will open the hardware manual of the ADC device. If the ADC device is not a VT hardware device but the DAC device is, then the hardware manual of the DAC device will be opened instead. If both the ADC and DAC devices are not a VT hardware device, then this option will grey out.

2.6.5.5 Application Notes and Videos (Help SubMenu) (ALT-H-N)

This command will open the Application Notes and Videos page on the official website of the software.

2.6.5.6 Check for Software Updates (Help SubMenu) (ALT-H-U)

This command will check online to see if a new software version is available for download. If yes, it will prompt you for confirmation for download.

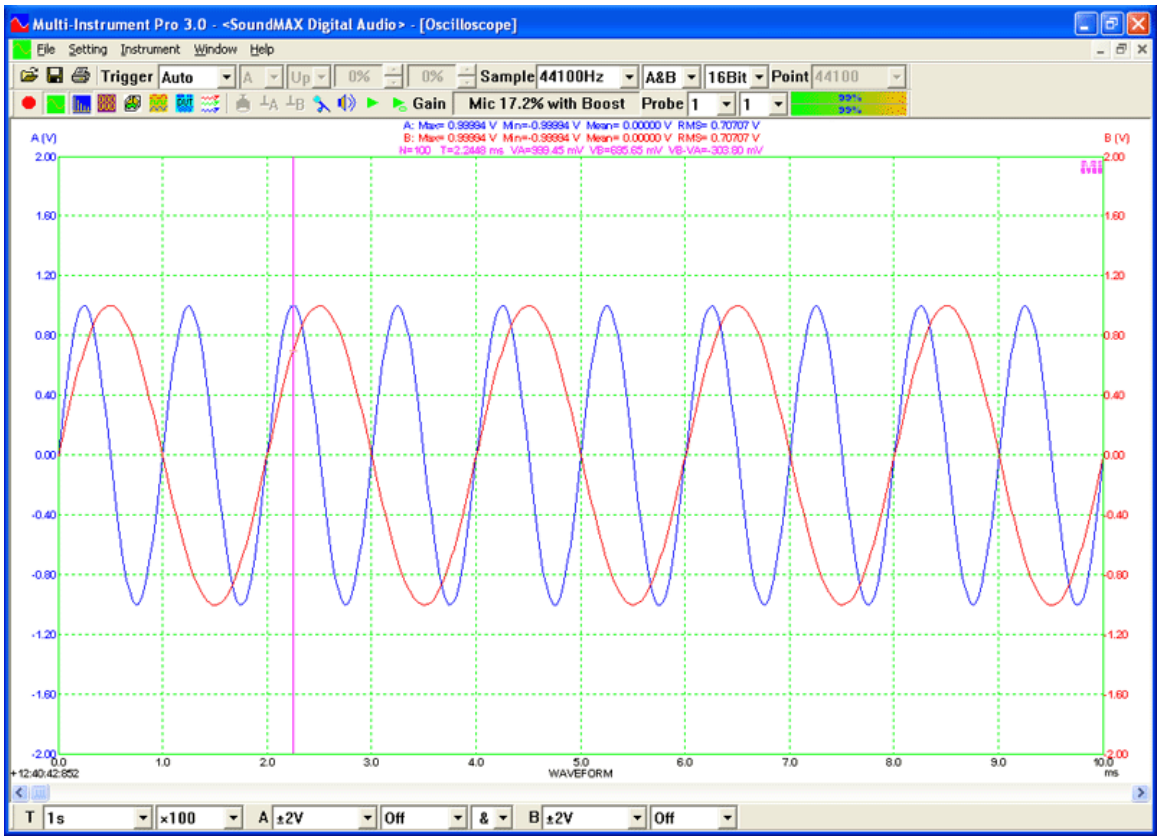
After download, you will need to install the software by yourself with the admin right. If the software is installed into the same directory of an old installation without performing uninstallation first, the old configuration such as the color scheme and calibration data, etc. will be inherited. It is also possible for a brand-new installation to inherit an old configuration if the old configuration file Scins.cfg, before uninstallation, is kept and renamed to Scins.cfg.old and placed under the new installation directory before the very first run of the new version.

2.6.5.7 About (Help SubMenu) (ALT-H-A)

This command will open the About window which displays the versions of the software, firmware and driver as well as the license information.

2.7 Cursor Reader and Markers

2.7.1 Cursor Reader



The cursor reader is shown when you left click and hold anywhere within the plot region of a view. It will stick to the actual measurement point nearest to the cursor in the horizontal direction and show its X and Y readings for both channels in the top region of the view. The cursor reader moves with the cursor and remains active until the left mouse button is released.

In Lissajous Pattern display, the cursor reader will show the X and Y readings at exactly the cursor position and will not stick to the actual measurement point.

2.7.2 Marker

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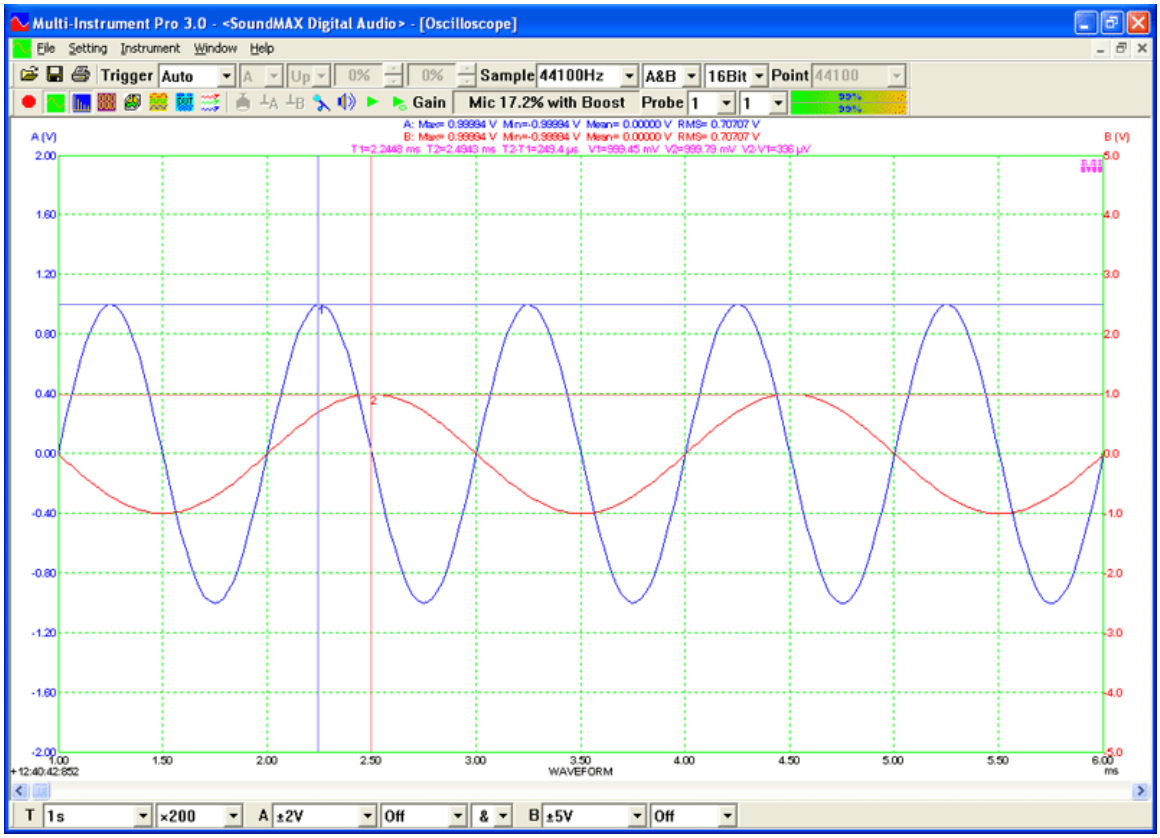

Maximum two markers can be placed in the plot region of a view.

Double clicking the left mouse button anywhere within the plot region will place a marker on the screen at the actual measurement point nearest to the position of mouse click in the horizontal direction. Left double clicking places a marker for the left vertical axis while CTRL-Left double clicking places a marker for the right vertical axis. No marker will be placed if there is no corresponding vertical axis.

The second marker can be placed by Shift-Left double clicking or Shift-CTRL-Left double clicking.

Clicking anywhere outside the plot region in the view will remove all markers placed previously.

The horizontal and vertical readings of a marker will be displayed in the top region of the view. If two markers are placed, the reading difference between them will also be shown.

2.7.3 Combined Use of Marker and Cursor Reader

You can place a marker as a fixed reference point and then use the cursor reader to read out the difference between the cursor and the reference point as the cursor moves. This is very useful, for example, when you want to measure the pulse width, the rise time or the fall time.

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2.8 Maximum, Minimum, Mean and RMS Values

The Oscilloscope will display the Maximum, Minimum, Mean and RMS values of each frame of data. These parameters can also be accessed through the following DDPs: Max_A(EU), Min_A(EU), Mean_A(EU), RMS_A(EU) for Channel A and Max_B(EU), Min_B(EU), Mean_B(EU), RMS_B(EU) for Channel B.

2.9 Time Stamp

The time stamp displayed at the lower left corner of the Oscilloscope view represents the time (with an accuracy of 1 millisecond) of the first data point (i.e. the leftmost data point) of the current frame of data. In most cases, the two channels of data are sampled at the same time and thus only one time stamp is displayed. However, there are cases whereby Channel A is used for live input and Channel B is fed by the Signal Generator directly at software level (refer to the manual of the Signal Generator). In these cases, the time stamp for Channel A will be displayed and on its right hand side, the time difference between Channel A and Channel B will also be displayed, with an accuracy of one sampling interval between the raw data. A negative value of the time difference means that the data of Channel B arrives earlier than that of Channel A.

2.10 Trigger Marker

Two markers will be displayed in the oscilloscope view to reflect the trigger parameters. One marker will be displayed on the left vertical axis to reflect the current Trigger Level (by the location of the marker on the vertical axis), Trigger Edge (by the arrow direction of the marker) and Trigger Source (by the color of the marker). The other marker will be displayed on the horizontal axis on the top to reflect the Trigger Delay (by the location of the marker on the horizontal axis) as well as the Trigger Source (by the color of the marker). Only a negative Trigger Delay (i.e. pre-trigger) can be displayed. The trigger position on a waveform can be found based on these two markers.

These markers are displayed only in one type of view: waveform display of Channel A and Channel B. They will not be displayed when Trigger Edge is “JP” or “DF”.

When the trigger level marker is shown, if you rest the cursor on it, the cursor will change from the Windows default one to . If you then press down and hold the left mouse button, you can adjust the Trigger Level by moving the marker. You can adjust the Trigger Delay in a similar way. Some ADC hardware devices support ALT trigger mode. Under ALT trigger mode, the trigger level for Channel B can be adjusted by moving the trigger level marker on the right vertical axis.

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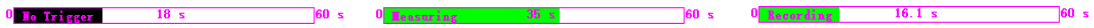
2.11 DAQ Progress Bar



A DAQ progress bar will be displayed at the lower left corner of the Oscilloscope view under either of the following conditions:

- The Oscilloscope is started in non-Record mode and the specified trigger condition has not been met for at least 5 seconds.
- The Oscilloscope is started in Record mode.

Black color is used to fill up the progress bar when no trigger event has been found, the time elapsed since the start of the Oscilloscope will be displayed. The progress bar will resets and changes to green color after a trigger event is found in the returned DAQ buffer, the time elapsed since then will be displayed. The DAQ progress bar will be hidden again after a frame of data is collected, analyzed and displayed.

2.12 Record Mode

There are two ways to enter into the Record Mode:

- Press the above “Record” button in the Sampling Parameter Toolbar, or
- Select “Record” in the Sweep Time combo box and then press the “Run” button of the Oscilloscope.

Under the Record Mode, raw ADC data will be written into the hard disk continuously in WAV file format. Meanwhile, data analysis and display will still be performed in order to keep the screen updated in real time. Priority is given to the former process in order to try the best to ensure uninterrupted data recording. Whether the recorded data are continuous (i.e. the adjacent frames of data are connected smoothly without missing any data in between) or not depends on whether the ADC hardware supports continuous streaming, the system throughput, sampling frequency, bit resolution, number of sampling channels, etc. It is generally possible to record signals continuously without any interruption using sound cards. For VT DSOs using USB2.0, the data rate limit would be a sampling rate of 10~20MHz, 8 bits and single channel. Beyond this limit, discontinuity may occur. When the recorded data are continuous, the DAQ progress bar at the bottom left corner of the Oscilloscope window will be filled with green color. Once a discontinuity is detected, red color will be used instead.

(Recorded data are continuous)

(Recorded data are not continuous)
(Discontinuity is indicated by red color)

The sweep time (for screen refresh purpose only) under this mode is defaulted to 100 ms (for the case of a sound card) and is adjustable via [Setting]>[Display]>"Record Mode">"Frame Width". The record duration is set via [Setting]>[Display]>"Record Mode">"Duration". By default, the Duration is set to zero, meaning that the recording process will stop automatically only after 2 gigabytes of data have been recorded. The recording process can also be stopped manually by pressing the Run / Stop button in the Instrument Toolbar or the Record button in the Sampling Parameter Toolbar. You can specify a duration value greater than zero so that the recording process will stop automatically according to it. Only Auto Trigger Mode and Normal Trigger Mode are allowed under the Record mode. Sampling parameters are prevented from being changed during the recording process by the software. Upon finishing recording, the recorded file will be opened automatically by the software in one-frame mode if the length of the data is not too long, or in frame-by-frame mode otherwise. It may take a little while if the file is big.

If you start recording without opening a file (in this case, no file name will be displayed in the title bar of the main window of the software), then the software will automatically use "Record1.wav" as the file name for this record. If you repeat the recording process by stopping and starting the recording again and again, then the file name will be incremented each time, e.g. Record2.wav, Record3.wav…. This file name incrementing process will restart only if you restart the program.

If you start recording on an opened file which does not have a file name as "Recordxxx.wav", then the original file will be overwritten by the recorded data, and the file name will not change even if you repeat the recording process.

If [Setting]>[Display]>"Record Mode">"No Auto Stop" is ticked, then the recording process can only be stopped manually. One WAV file will be recorded automatically per Record Duration, or 2G bytes' data if the duration is 0. The WAV file names will be assigned automatically based on the time stamp of the first sample in the file in the format of YYYY-MM-DD-HH-MM-SS-XXX.wav, with date and time in accuracy of milliseconds. This function can be used to log the raw data continuously as long as the storage space is allowed.

2.13 Roll Mode

Roll Mode is activated by ticking the above "Roll" checkbox in the Sampling Parameter Toolbar. Under this mode, the Oscilloscope frame is split into many segments with the length of each segment equal to the Roll Width, which is set via [Setting]>[Display]> "Roll Mode">"Roll Width". The Roll Width is set to 50 ms (for the case of a sound card) by default. Under the Roll Mode, the size of data acquisition is based on the Roll Width rather than the Record Length per sweep. The data displayed in the Oscilloscope view will shift left at a step of one Roll Width each time when a new segment of data arrives. The newly arrived data will be shown in the right most portion of the graph. You can consider using Roll Mode if the sweep time is too long (e.g. greater than 1 s) to avoid long time waiting for screen update. You

must use Roll Mode (if it is available) when the Record Length per sweep is greater than the size of the ADC hardware buffer.

The Roll Mode checkbox is enabled when the Record Length per sweep is at least four times as long as the Roll Width.

Under the Roll Mode, whether the acquired data are continuous (i.e. the adjacent segments of data are connected smoothly without missing any data in between) or not depends on whether the ADC hardware supports continuous streaming, the system throughput, sampling frequency, bit resolution, number of sampling channels, etc.

2.14 Sampling Parameter Auto Setting

When a periodic signal is connected, you can let the software to automatically choose proper sampling parameters such as Sweep Time and ADC Range by clicking the above “Auto” button in the Sampling Parameter Toolbar. You are not allowed to change any sampling parameters during this auto setting process. The auto setting process will be stopped automatically after the proper sampling parameters are found for the signal under test or timeout. Some detailed functions of this button can be configured via [Setting]>[ADC Device]>“Auto Ranging” and “Auto Button for Auto Ranging only”. Please refer to the respective sections introduced previously.

2.15 Magnifying Glass

The magnifying glass will show up when you put the cursor on the outer side of an axis, if the axis has a multiplier associated with it. The magnifying glass has three states:

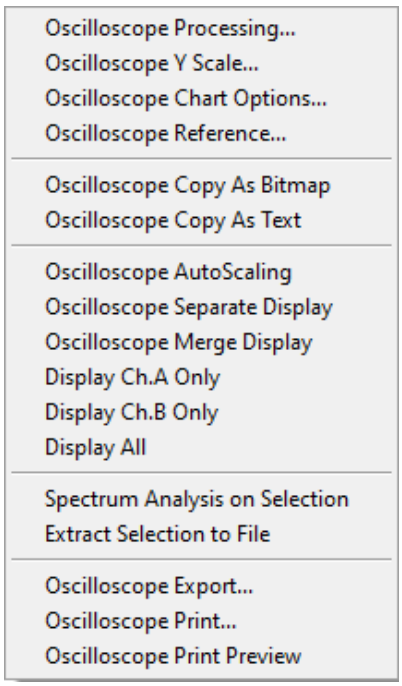
: Only zoom in is available.

: Both zoom in and zoom out is available.

: Only zoom out is available.

Pressing the left mouse button will zoom in one step and pressing the right mouse button will zoom out one step. Zoom in / out can also be performed using the mouse wheel. Mouse wheel, SHIFT + mouse wheel, CTRL + mouse wheel can be used to zoom in / out X, A, B axes respectively. Mouse wheel down & scroll, SHIFT + mouse wheel down & scroll, CTRL + mouse wheel down & scroll will return X, A, B axes to their default values respectively.

2.16 Context Menu



The above context menu will be shown when right clicking anywhere within the Oscilloscope view. It provides additional convenience to you. All menu items in the context menu can also be found in the main menu of the Oscilloscope, except the following items:

- Copy As Bitmap

  It is similar to the Oscilloscope Export (as Bitmap) function. Instead of saving the bitmap image of the oscilloscope view to the hard disk, it copies the image to the clipboard which can be subsequently pasted out into other programs such as Microsoft Word.

- Copy As Text

  It is similar to the Oscilloscope Export (as Text) function. Instead of saving the texts of the oscilloscope data to the hard disk, it copies the data to the clipboard which can be subsequently pasted out into other programs such as Microsoft Excel. Note the data in the clipboard is Tab separated instead of comma separated.

- AutoScaling

  It adjusts the vertical display scales based on the peak values in the sampled data frame.

- Separate Display

  It separates the display of Channel A from that of Channel B vertically.

- Merge Display

  It superimposes the displays of Channel A and Channel B.

- Display Ch. A Only

  It will display Ch. A only and hide other channels in all views.

- Display Ch. B Only

  It will display Ch. B only and hide other channels in all views.

- Display All

  It will display all channels in all views.

- Spectrum Analysis on Selection

  This menu item will be enabled when two markers have been placed on the acquired waveforms. It will display the spectra of the waveforms between the two markers in one (single channel) or two (dual channels) Derived Data Curve (DDC) windows.

- Extract Selection to File

  This menu item will be enabled when two markers have been placed on the acquired waveforms. It can be used to extract the waveforms between the two markers and save them as a separate WAV file, via [File]>[Extract].

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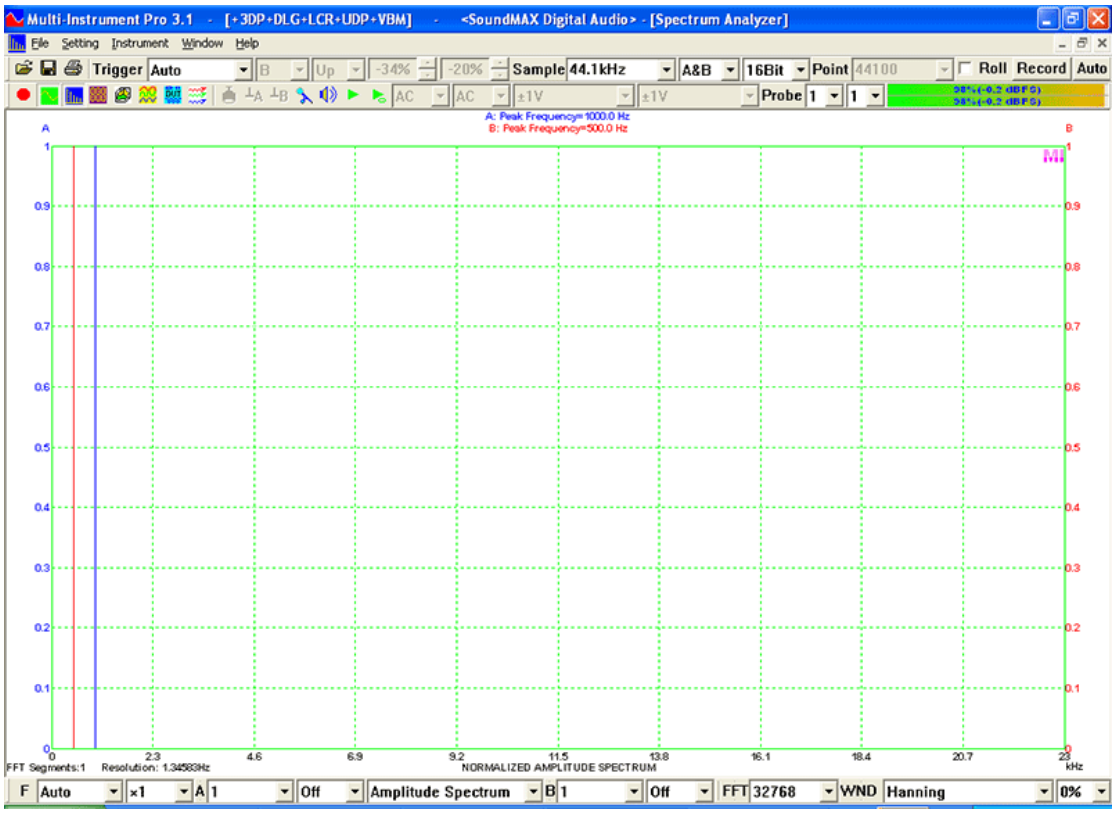
3 Spectrum Analyzer

3.1 Overview



This is a dual channel Spectrum Analyzer, providing seven types of views:

- Real time Amplitude Spectrum / Power Spectrum Density / Impedance Spectrum
- Real time Phase Spectrum
- Real time Auto Correlation Function (Linear or Circular)
- Real time Cross Correlation Function (Linear or Circular, Original or Generalized)
- Real time Coherence / Non-Coherence Function
- Real time Transfer Function (Bode Plot, Frequency Response, or Gain and Phase Plot) / Impedance Analyzer
- Real time Impulse Response

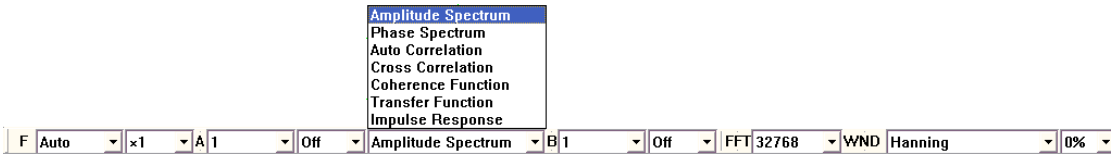
with adjustable FFT size ranging from 128 to 4194304, adjustable window overlap percentage (0%~99.9%), and 69 selectable window functions such as Rectangle, Triangle, Hann, Hamming, Blackman, Kaiser and so on. It supports display in logarithmic scale for both X axis and Y axis (dBr, dBV, dBu, dBFS, dB), octave analysis (1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96), frequency compensation, frequency weighting (Flat, A weighting, B weighting, C weighting, ITU-R 468 weighting), moving average smoothing (linear or 1/1/~1/96 octave), DC removal, peak hold, linear average, exponential average, cross power spectrum vector average ("cross

correlation average"), and parameter measurements (THD, THD+N, SINAD, SNR, Noise Level, IMD-SMPTE/DIN, IMD-CCIF2, IMD-CCIF3, DIM, Crosstalk, Bandwidth, Harmonics, Peaks, Energy in user defined frequency bands, Wow & Flutter, Sound Loudness & Sharpness, Total Non-Coherent Distortion + Noise, GedLee Metric, Total Distortion + Noise, Noise Rating & Noise Criterion, Sound Intensity), etc.

The Spectrum Analyzer shares the same Trigger Parameters, Sampling Parameters and Miscellaneous Parameters with the Oscilloscope. Please refer to the relevant sections in the Oscilloscope for details.

3.2 View Parameters

View parameters determine how the collected data are analyzed and displayed.



There are seven types of views in the Spectrum Analyzer:

- Amplitude Spectrum
- Phase Spectrum
- Auto-Correlation
- Cross-Correlation
- Coherence
- Transfer Function
- Impulse Response

Spectrum Analyzer generates many DDPs. For a complete list of these DDPs, please refer to Section 6.2.1.

3.2.1 View Parameters for Amplitude Spectrum Display

Selecting Amplitude Spectrum in the above View Type selection box switches the Spectrum Analyzer to amplitude spectrum display mode, which shows the amplitude vs frequency graph of the measured signal. It can also be used to show impedance vs frequency graph. This will be described later.

The following figure illustrates the amplitude spectrum of a 1 kHz square wave with X axis in linear scale and Y axis in normalized linear scale. It shows that a square wave consists of a fundamental frequency and an infinite number of odd harmonics, each of which has an amplitude equal to 1/N of that of the fundamental frequency, where N is the order of the harmonic.

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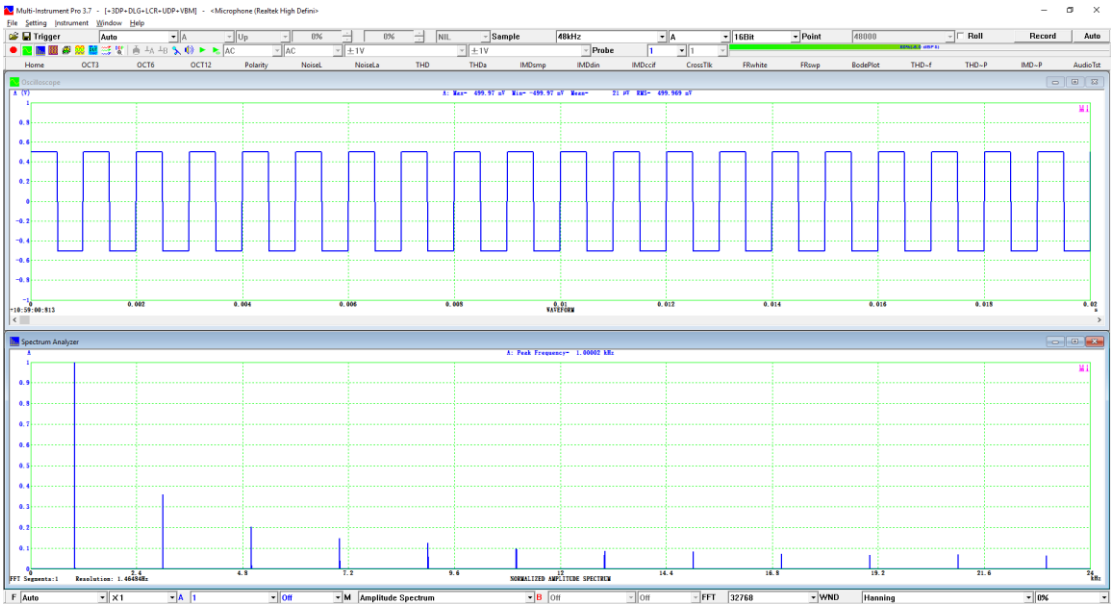
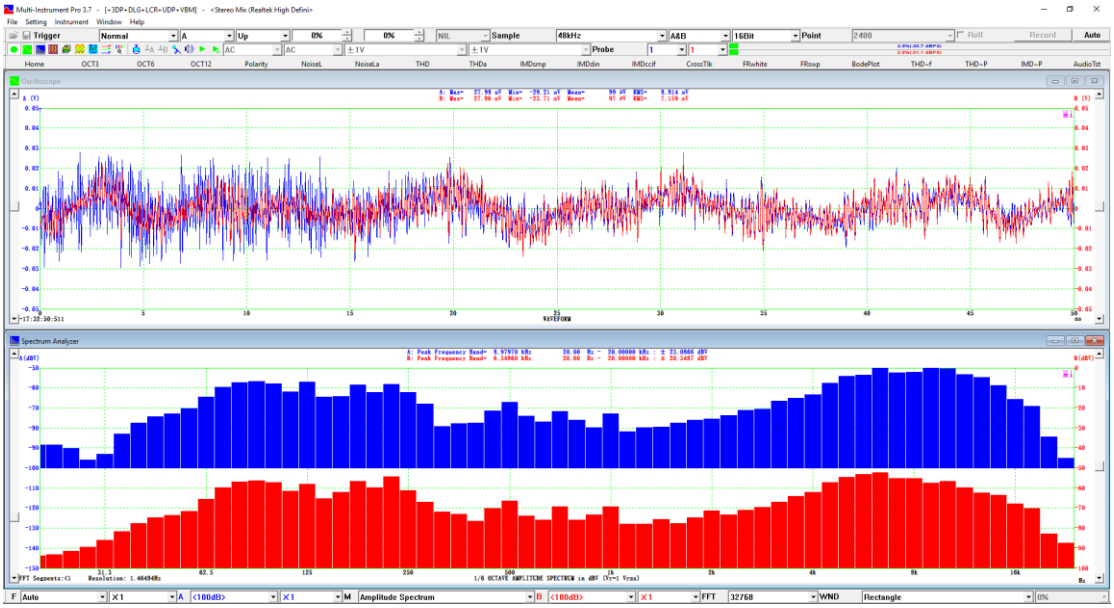
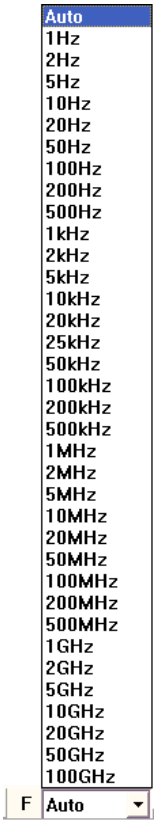


The following figure illustrates the 1/6 octave amplitude spectrum of a 50-ms stereo pop song replayed and then captured by a sound card. The X axis is in 1/6 octave scale and the Y axis is in dBV scale.



3.2.1.1 Frequency Range (F)



36 options are available for Frequency Range (F). They are: Auto, 1Hz, 2Hz, 5Hz, 10Hz, 20Hz, 50Hz, 100Hz, 200Hz, 500Hz, 1kHz, 2kHz, 5kHz, 10kHz, 20kHz, 25kHz, 50kHz, 100kHz, 200kHz, 500kHz, 1MHz, 2MHz, 5MHz, 10MHz, 20MHz, 50MHz, 100MHz, 200MHz, 500MHz, 1GHz, 2GHz, 5GHz, 10GHz, 20GHz, 50GHz, 100GHz.

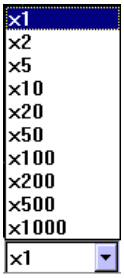
When "Auto" is selected, the Frequency Range will be set automatically by the software based on the following formula:

Frequency Range = the nearest integer that is equal to or greater than 1/2 of [Sampling Frequency]

An important principle in digital signal processing is the "Nyquist-Shannon Sampling Theorem" which states that an analog signal that has been sampled can be perfectly reconstructed from the samples if the sampling frequency is greater than twice of the highest frequency in the original signal. This means that if you wish to measure a 3,000 Hz signal, the sampling rate must be greater than 6,000 Hz, otherwise aliasing will occur.

In Amplitude Spectrum Display, the horizontal axis can be displayed in linear, logarithmic, 1/1, 1/3, 1/6, 1/12, 1/24, 1/48, or 1/96 octave scale, which can be selected via [Setting]>[Spectrum Analyzer X Scale].

3.2.1.2 Frequency Multiplier



The Frequency Multiplier is the zooming factor for the horizontal axis. There are 16 options available: ×1, ×2, ×5, ×10, ×20, ×50, ×100, ×200, ×500, ×1000, ×2000, ×5000, ×10000, ×20000, ×50000, ×100000.

When "×1" is selected, the full Frequency Range is displayed over the width of the view.

If you change the Frequency Multiplier to "×N" which is greater than 1, then only 1/N of the full Frequency Range will be displayed over the width of the view, with a horizontal scrollbar at the bottom which allows you to scroll over the full range of the Frequency.

This multiplier can also be adjusted via the magnifying glass which will be shown if you put the mouse cursor just below the horizontal axis.

3.2.1.3 Channel A Display Range

In Amplitude Spectrum Display, there are two modes for the vertical axis, which can be selected via [Setting]>[Spectrum Analyzer Y Scale]:

3.2.1.3.1 Absolute Display Mode

The vertical axis is scaled in engineering unit. All data points are plotted based on their absolute values, in Vrms, dBV, dBu, dB, or dBFS. Note that by definition, the reference voltages for dBV and dBu are 1 Vrms and 0.775 Vrms respectively. 1 dBu in amplitude spectrum is equivalent to 1 dBm in power spectrum when the load is 600 ohms. The reference levels for dB for both channels are user definable. For example, you can calibrate them to sound pressure level. Please refer to the section for 0dB Reference Vr described previously. The reference voltage for dBFS is the ADC full-scale voltage (1/2 Vpp). More detailed explanation will be given later in the section for Spectrum Analyzer Y Scale.

3.2.1.3.1.1 Linear Scale

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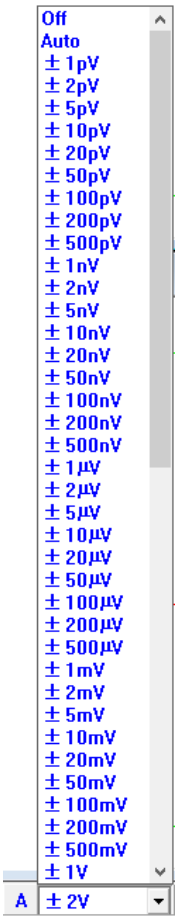


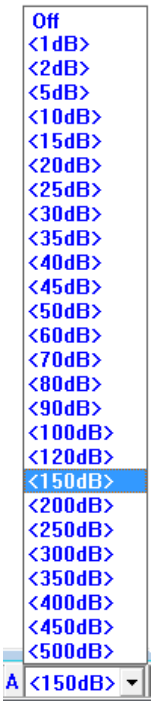
For linear scale such as Vrms, the available Display Range options are: Off, Auto, 1pV, 2pV, 5pV, 10pV, 20pV, 50pV, 100pV, 200pV, 500pV, 1nV, 2nV, 5nV, 10nV, 20nV, 50nV, 100nV, 200nV, 500nV, 1µV, 2µV, 5µV, 10µV, 20µV, 50µV, 100µV, 200µV, 500µV, 1mV, 2mV, 5mV, 10mV, 20mV, 50mV, 100mV, 200mV, 500mV, 1V, 2V, 5V, 10V, 20V, 50V, 100V, 200V, 500V, 1kV, 2kV, 5kV, 10kV, 20kV, 50kV, 100kV, 200kV, 500kV, 1MV, 2MV, 5MV, 10MV, 20MV, 50MV, 100MV, 200MV, 500MV, 1GV, 2GV, 5GV, 10GV, 20GV, 50GV, 100GV, 200GV, 500GV. Note that the engineering unit of the above options is determined by the engineering unit of the sensor for Channel A, which can be set via [Setting] > [Calibration] > "Sensor" > "Unit". For example, if the unit is "g" instead of "V", then the unit in all the above options will be changed to "g".

When "Off" is selected, the data in Channel A will not be displayed. When "Auto" is selected, the Display Range for Channel A will be set automatically by the software based on the following formula:

> *Display Range = the nearest integer that is equal to or greater than the value of [ADC Range] / [Sensor Sensitivity]*

where the sensor sensitivity is set via [Setting] > [Calibration] > "Sensor" > "Sensitivity".

3.2.1.3.1.2 Logarithmic Scale



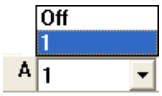
For logarithmic scale such as dBV, the available Display Range options are: 1dB, 2dB, 5dB, 10dB, 15dB, 20dB, 25dB, 30dB, 35dB, 40dB, 45dB, 50dB, 60dB, 70dB, 80dB, 90dB, 100dB, 120dB, 150dB, 180dB, 200dB, 250dB, 300dB, 350dB, 400dB, 450dB, 500dB. Note that the bracket "<>" means that the value represents only the span of the axis, the upper limit of the axis is determined by the ADC Range as well as the Sensor Sensitivity, and the lower limit is equal to [Upper Limit] - [Span]. A Display Range value without the bracket "<>" represents a range from 0 to the Display Range value, which is the case for dBFS.

When "Off" is selected, the data in Channel A will not be displayed.

3.2.1.3.2 Relative Display Mode

The vertical axis is scaled in relative values, in either linear or dBr (logarithmic) scale. All data points are plotted based on their relative values with regard to the maximum value (or the value at a specified frequency) in the current measurement.

3.2.1.3.2.1 Linear Scale

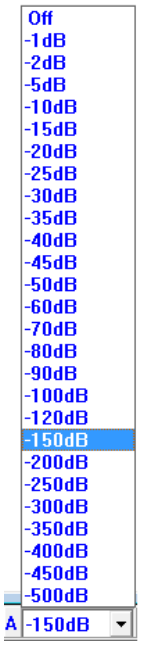


In this mode, the vertical axis ranges from 0 to 1 where 1 corresponds to the highest absolute vertical value in the measurement.

When "Off" is selected, the data in Channel A will not be displayed.

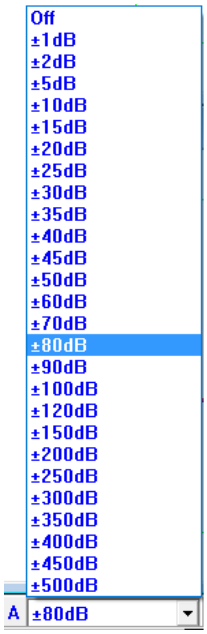
3.2.1.3.2.2 Logarithmic Scale (dBr)

(1) 0dB reference Vr = Peak



In this mode, the vertical axis ranges from 0 to the Display Range value selected, where 0 dB corresponds to the highest absolute vertical value in the measurement.

(2) 0dB reference Vr at a specified frequency



In this mode, the vertical axis covers ± [Display Range value] selected, where 0 dB corresponds to the measured value at the specified frequency.

When "Off" is selected, the data in Channel A will not be displayed.

3.2.1.4 Channel A Multiplier

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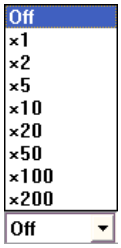

The Multiplier for Channel A is the zooming factor for A axis. There are 9 options available: Off, ×1, ×2, ×5, ×10, ×20, ×50, ×100, ×200.

When "Off" is selected, the full Display Range for Channel A is displayed over the height of the View.

When "×1" is selected, initially the full range is displayed over the height of the view with a vertical scroll bar on the left of the view. You can use the scroll bar to move the data curve for Channel A up and down.

If you change the multiplier to "×N" which is greater than 1, then only 1/N of the full range is displayed over the height of the view, with a vertical scrollbar on the left of the view. You can use the scroll bar to scroll over the full Display Range.

This multiplier can also be adjusted via the magnifying glass which will be shown if you put the mouse cursor on the left side of A axis.

3.2.1.5 Channel B Display Range

For single channel measurement, this control is disabled.

3.2.1.5.1 Absolute Display Mode

3.2.1.5.1.1 Linear Scale

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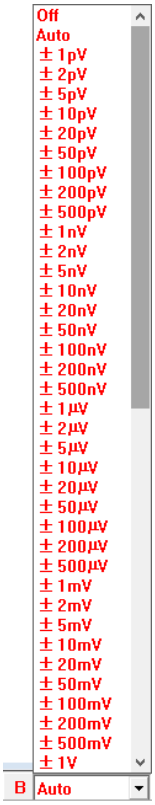

For linear scale such as Vrms, the available Display Range options are: Off, Auto, 1pV, 2pV, 5pV, 10pV, 20pV, 50pV, 100pV, 200pV, 500pV, 1nV, 2nV, 5nV, 10nV, 20nV, 50nV, 100nV, 200nV, 500nV, 1µV, 2µV, 5µV, 10µV, 20µV, 50µV, 100µV, 200µV, 500µV, 1mV, 2mV, 5mV, 10mV, 20mV, 50mV, 100mV, 200mV, 500mV, 1V, 2V, 5V, 10V, 20V, 50V, 100V, 200V, 500V, 1kV, 2kV, 5kV, 10kV, 20kV, 50kV, 100kV, 200kV, 500kV, 1MV, 2MV, 5MV, 10MV, 20MV, 50MV, 100MV, 200MV, 500MV, 1GV, 2GV, 5GV, 10GV, 20GV, 50GV, 100GV, 200GV, 500GV. Note that the engineering unit of the above options is determined by the engineering unit of the sensor for Channel B, which can be set via [Setting]>[Calibration]>"Sensor">"Unit". For example, if the unit is "g" instead of "V", then the unit in all the above options will be changed to "g".

When "Off" is selected, the data in Channel B will not be displayed. When "Auto" is selected, the Display Range for Channel B will be set automatically by the software based on the following formula:

> *Display Range = the nearest integer that is equal to or greater than the value of [ADC Range] / [Sensor Sensitivity]*

where the sensor sensitivity is set via [Setting]>[Calibration]>"Sensor">"Sensitivity".

3.2.1.5.1.2 Logarithmic Scale

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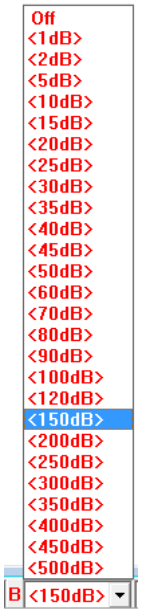


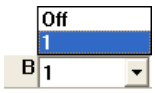
For logarithmic scale such as dBV, the available Display Range options are: 1dB, 2dB, 5dB, 10dB, 15dB, 20dB, 25dB, 30dB, 35dB, 40dB, 45dB, 50dB, 60dB, 70dB, 80dB, 90dB, 100dB, 120dB, 150dB, 180dB, 200dB, 250dB, 300dB, 350dB, 400dB, 450dB, 500dB. Note that the bracket "<>" means that the value represents only the span of the axis, the upper limit of the axis is determined by the ADC Range as well as the Sensor Sensitivity, and the lower limit is equal to [Upper Limit] - [Span]. A Display Range value without the bracket "<>" represents a range from 0 and the Display Range value, which is the case for dBFS.

When "Off" is selected, the data in Channel B will not be displayed.

3.2.1.5.2 Relative Display Mode

The vertical axis is scaled in relative values, in either linear or dBr (logarithmic) scale. All data points are plotted based on their relative values with regard to the maximum value (or the value at a specified frequency) in the current measurement.

3.2.1.5.2.1 Linear Scale



In this mode, the vertical axis ranges from 0 to 1 where 1 corresponds to the highest absolute vertical value in the measurement.

When "Off" is selected, the data in Channel B will not be displayed.

3.2.1.5.2.2 Logarithmic Scale (dBr)

(1) 0dB reference Vr = Peak

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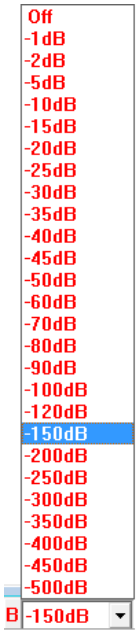
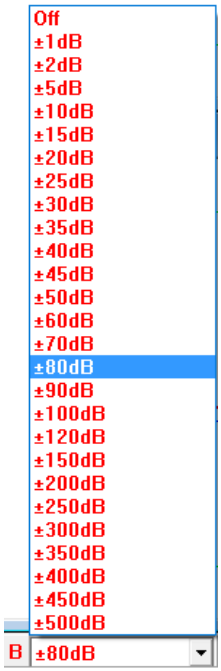


In this mode, the vertical axis ranges from 0 to the Display Range value selected, where 0 dB corresponds to the highest absolute vertical value in the measurement.

(2) 0dB reference Vr at a specified frequency



In this mode, the vertical axis covers ± [Display Range value] selected, where 0 dB corresponds to the measured value at the specified frequency.

When "Off" is selected, the data in Channel B will not be displayed.

3.2.1.6 Channel B Multiplier

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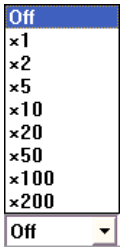


The Multiplier for Channel B is the zooming factor for B axis. There are 9 options available: Off, ×1, ×2, ×5, ×10, ×20, ×50, ×100, ×200.

When "Off" is selected, the full Display Range for Channel B is displayed over the height of the View.

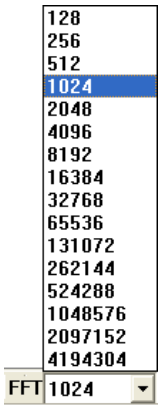
When "×1" is selected, initially the full range is displayed over the height of the view with a vertical scroll bar on the right of the view. You can use the scroll bar to move the data curve for Channel B up and down.

If you change the multiplier to "×N" which is greater than 1, then only 1/N of the full range is displayed over the height of the view, with a vertical scrollbar on the right of the view. You can use the scroll bar to scroll over the full Display Range.

For single channel measurement, this control is disabled.

This multiplier can also be adjusted via the magnifying glass which will be shown if you put the mouse cursor just on the right side of B axis.

3.2.1.7 FFT Size



This parameter is applicable to all types of views in the Spectrum Analyzer. 16 options are available: 128, 256, 512, 1024, 2048, 4096, 8192, 16384, 32768, 65536, 131072, 262144, 524288, 1048576, 2097152, 4194304.

The selected FFT size directly affects the resolution of the resulting spectrum. The number of spectral points is always 1/2 of the selected FFT size plus one. Thus a 1024-point FFT produces 513 spectral points.

The frequency resolution of each spectral point is equal to [Sampling Frequency]/[FFT Size]. For instance, if the FFT size is 1024 and the Sampling Frequency is 44100 Hz, the resolution of each spectral point would be:

44100 / 1024 = 43.07 Hz

A larger FFT size provides a finer spectral resolution but takes a longer time to compute.

If the FFT size is greater than the number of data points per sweep (Record Length) in the Oscilloscope, then zeros will be padded at the end of the actual measurement data during FFT computation. It should however be noted that the real frequency resolution is equal to [Sampling Frequency]/[Number of data points], although the apparent FFT frequency resolution is determined by [Sampling Frequency]/[FFT Size]. In other words, zero padding does not improve the real frequency resolution although it does provide more spectral points via interpolation.

If the FFT size is less than the number of data points per sweep (record length) in the Oscilloscope, then the measurement data will be split into different segments with the size of each segment equal to the FFT size. The last segment of data will be dropped if its size is not equal to the FFT size. The final result will be obtained by averaging the FFT results from all segments. It should be noted that this approach is used for Amplitude Spectrum, Auto Correlation Function, Cross Correlation Function, Coherence Function, Transfer Function, and Impulse Response, except Phase Spectrum where only the first segment of data is used. The average method here is referred to as Intra-Frame Average in contrast to the Inter-Frame Average set through [Setting]>[Spectrum Analyzer Processing]>[Inter-Frame Processing]. The following table lists the average method used for each type of analysis:

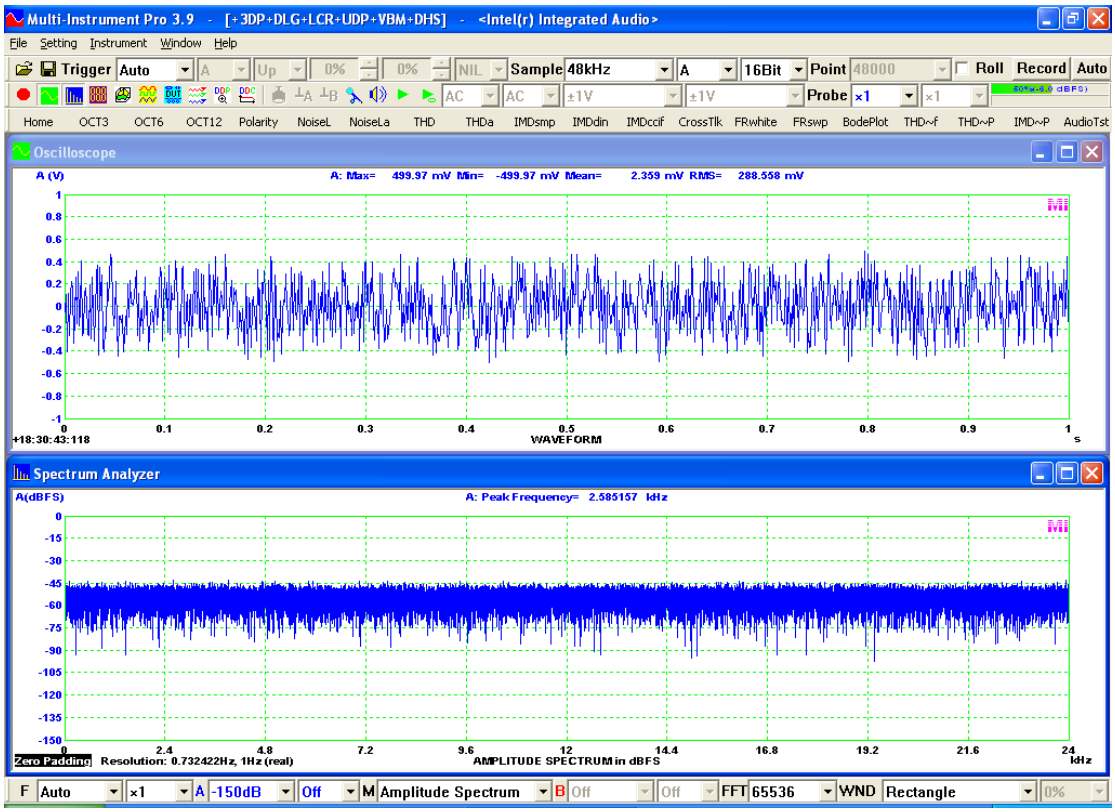
	Amplitude Spectrum		Phase Spectrum	Auto Correlation	Cross Correlation	Coherence Function	Transfer Function	Impulse Response
	Independent Channels	Dual Channel Cross Correlation						
Intra-Frame Average	Auto Power Average	Cross Power Vector Average	No	Auto Power Average	Cross Power Vector Average	Cross Power Vector Average	Cross Power Vector Average	Cross Power Vector Average
Inter-Frame Average	Auto Power Average	Cross Power Vector Average	Arithmetic Average	Arithmetic Average	Arithmetic Average	Arithmetic Average	Arithmetic Average	Arithmetic Average

Note: Auto power average reduces the fluctuation of data but does not reduce the noise level. Cross power vector average reduces the uncorrelated noise in the two channels.

You can adjust the number of data points per sweep (Record Length) before data sampling such that it equals to an integer multiple of the FFT size, if necessary. When both the Oscilloscope and Spectrum Analyzer are opened, the FFT Size and Record Length are changed independently. However, when only the Spectrum Analyzer is opened, changing the FFT size will change the Record Length and make it equal to the FFT size.

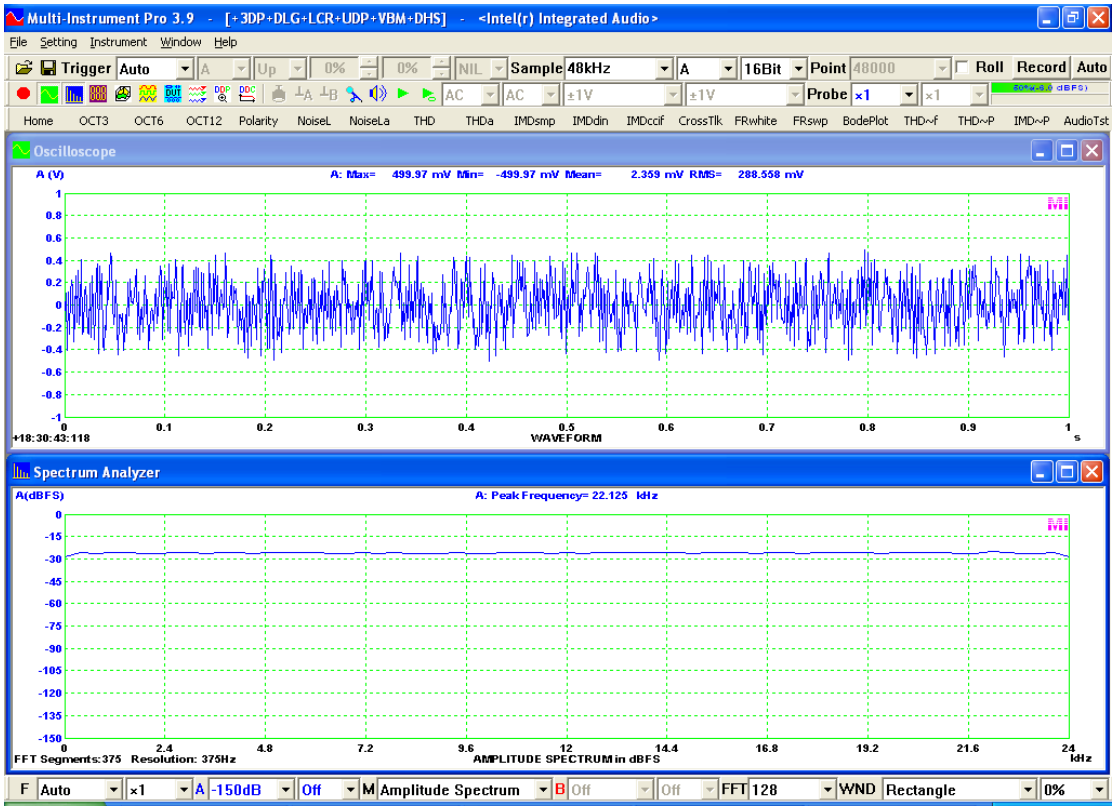
Both the number of FFT segments and the FFT frequency resolution are displayed at the lower left corner of the Spectrum Analyzer view. If zero-padding is used, then the real frequency resolution will also be displayed.

The following figure shows a 1-second white noise sampled at 48 kHz. Thus the Record Length is 48000. The FFT size is 65536 and Rectangle window function (i.e. No window function) is applied. The apparent FFT resolution is 48000/65536 = 0.732422 Hz and the real frequency resolution is 48000/48000 = 1 Hz. 65536-48000=17536 zeros are padded at the end of the originally sampled data during FFT. "Zero Padding, Resolution: 0.732422 Hz, 1Hz (real)" is thus displayed at the lower left corner of the Spectrum Analyzer view to indicate the zero padding operation, and the apparent and real frequency resolutions.



The following figure shows the amplitude spectrum of the same data as above. The only difference is that the FFT size is changed from 65536 to 128. Thus the apparent and real FFT resolutions are both 48000 Hz / 128 = 375 Hz. The number of FFT segments is 48000 samples / 128 samples = 375 (segments). "FFT Segments: 375 Resolution: 375 Hz" is thus displayed at the lower left corner of the Spectrum Analyzer view to indicate the intra-frame average over 375 FFT segments, and the FFT frequency resolution. The average method here is power average. Due to this averaging process and the coarser frequency resolution, the amplitude spectrum in the figure below is much smoother than that in the figure above.

A white noise has a flat amplitude spectrum. Its power distributes uniformly across all FFT bins. The amplitude level is around -26 dBFS in the figure below, which is about 27 dB higher than that in the figure above. The width of a FFT bin in the figure below is 375 Hz while that in the figure above is 0.732422 Hz. Therefore each FFT bin in the figure below contains 10×lg(375/0.732422) = 27 dB 's power more if the power is distributed uniformly.



To know more about the basics of FFT, please refer to: *FFT Basics and Case Study using Multi-Instrument*.

Download link:

https://www.virtins.com/doc/D1002/FFT_Basics_and_Case_Study_using_Multi-Instrument_D1002.pdf

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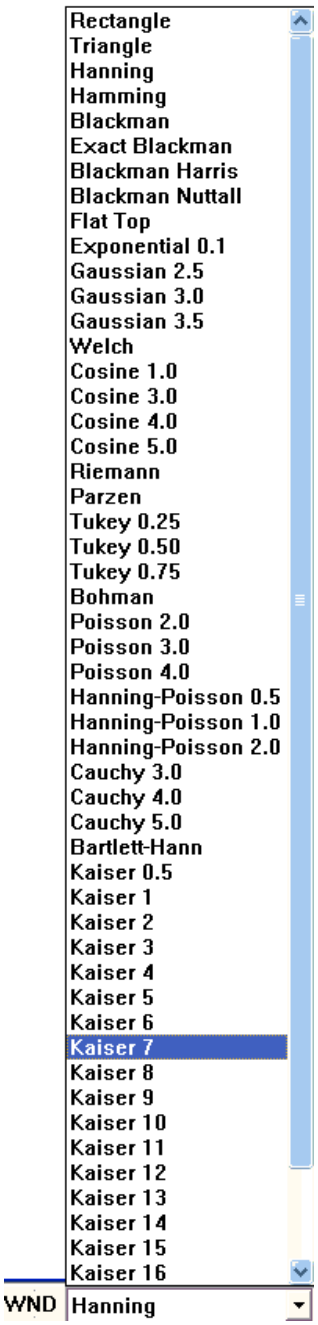
3.2.1.8 Window Function



A window function can be used to suppress spectral leakage caused by a non-infinite length of data in time domain. 69 window functions are supported: Rectangle, Triangle (or Fejer), Hann, Hamming, Blackman 4, Exact Blackman, Blackman Harris 4, Blackman Nuttall, Flat Top, Exponential (Exponential 0.1), Gaussian (Gaussian 2.5, Gaussian 3.0, Gaussian 3.5), Welch (or Riesz), Cosine (Cosine 1.0, Cosine 3.0, Cosine 4.0, Cosine 5.0), Riemann (or Lanczos), Parzen, Tukey (Tukey 0.25, Tukey 0.50, Tukey 0.75), Bohman, Poisson (Poisson 2.0, Poisson 3.0, Poisson 4.0), Hann-Poisson (Hann-Poisson 0.5, Hann-Poisson 1.0, Hann-Poisson 2.0), Cauchy (Cauchy 3.0, Cauchy 4.0, Cauchy 5.0), Bartlett-Hann, Kaiser (Kaiser 0.5, Kaiser 1, Kaiser 2, Kaiser 3, Kaiser 4, Kaiser 5, Kaiser 6, Kaiser 7, Kaiser 8, Kaiser 9, Kaiser 10, Kaiser 11, Kaiser 12, Kaiser 13, Kaiser 14, Kaiser 15, Kaiser 16, Kaiser 17, Kaiser 18,

Kaiser 19, Kaiser 20, Blackman Harris 7, CosineSum 220, CosineSum 233, CosineSum 246, CosineSum 261, Tukey 0.10, Tukey 0.05, Tukey 0.02, Tukey 0.01, Dolph-Chebyshev 80, Dolph-Chebyshev 100, Dolph-Chebyshev 150, Dolph-Chebyshev 200, Dolph-Chebyshev 250). The value behind the window name is the parameter value of that window. Please refer to relevant reference books for the definitions of these window functions.

Hann window is used by default. It should be noted that except Rectangle window, the rest of window functions are not applicable to Auto Correlation Function and Cross Correlation Function and thus are disabled accordingly under these two modes.

A 1024-point 24-bit WAV file is provided for each window function in the WAV\window directory of the software. They can be used to evaluate the behavior of these window functions in frequency domain. For this purpose, the following changes to the system default settings for the Spectrum Analyzer are required after loading a WAV file:

- [Window]: Rectangle;
- [FFT size]: >1024;
- [Setting]>[Spectrum Analyzer Processing]>"Intra-Frame Processing"> "Remove DC": Unchecked;

The following changes to the system default settings for the Spectrum Analyzer are recommended:

- [Setting]>[Spectrum Analyzer Y Scale]: dBr
- [Horizontal Axis Multiplier]: × 20

The following figure shows the spectrum of a Rectangle window.

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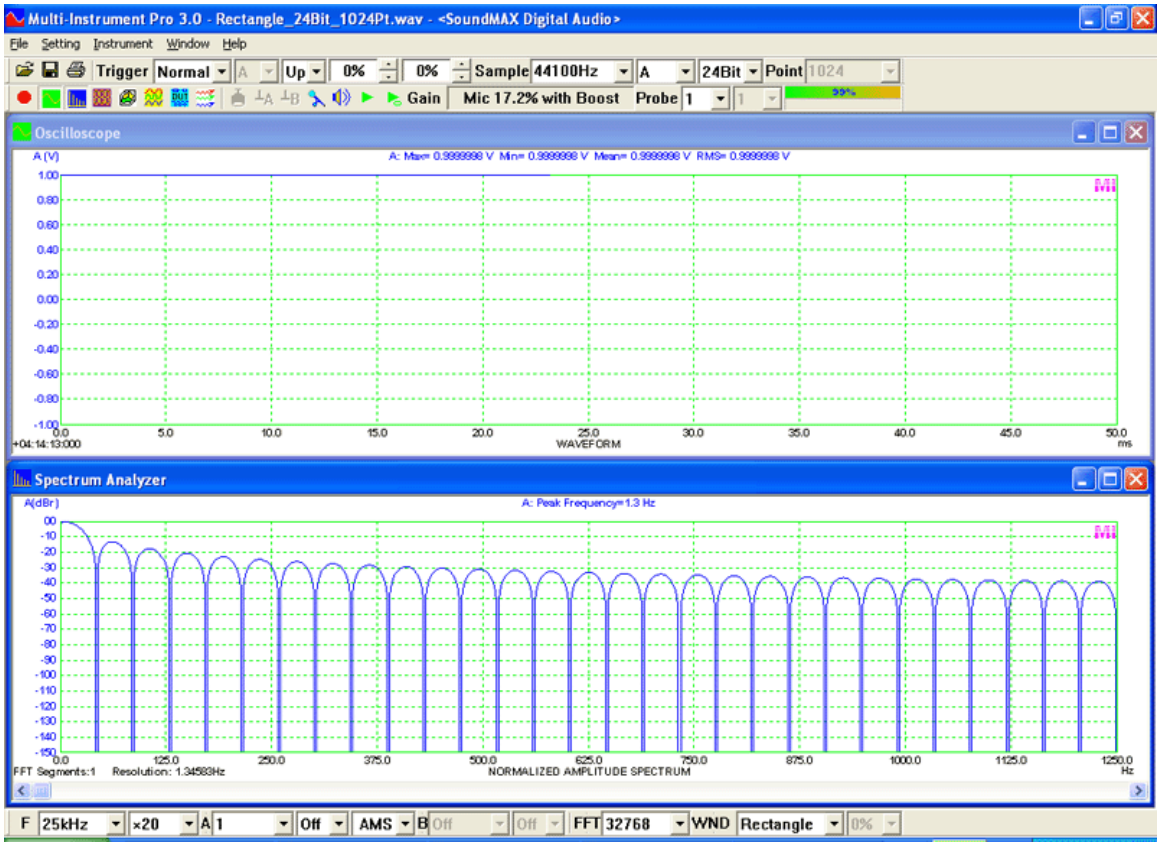
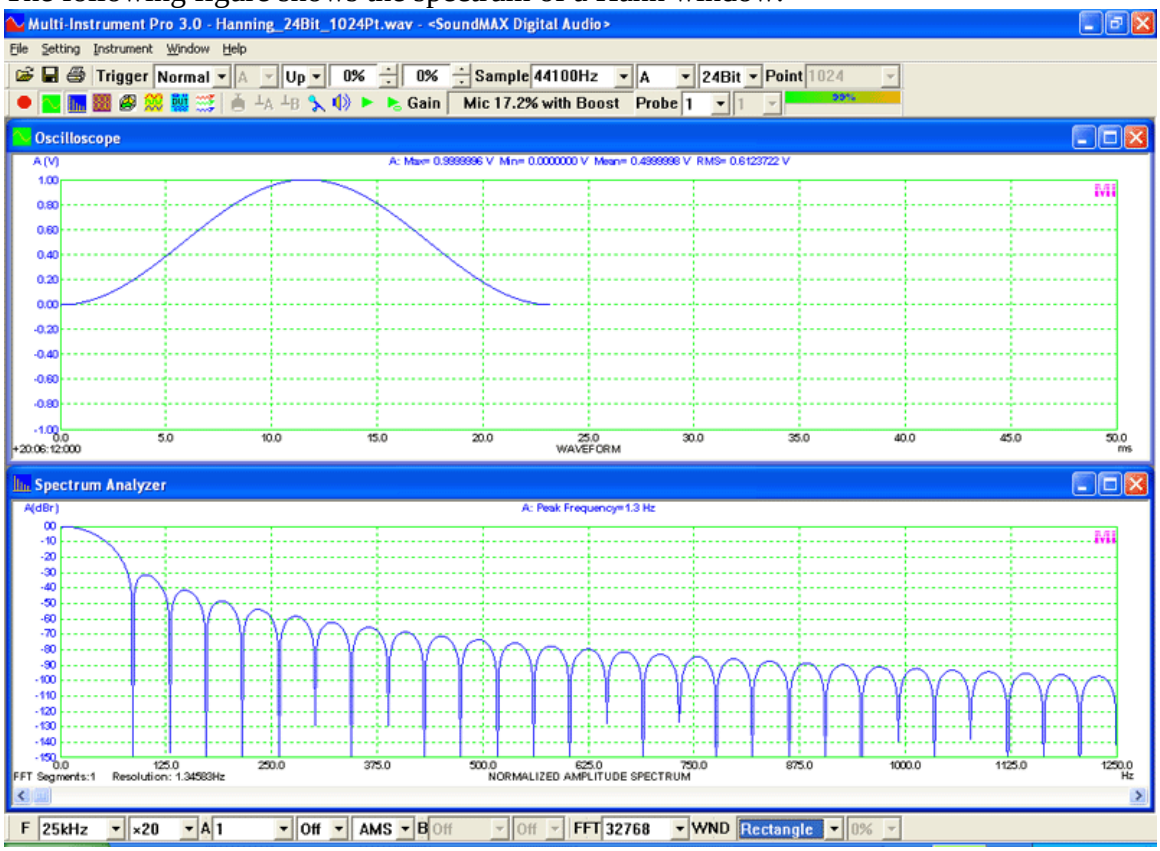


The following figure shows the spectrum of a Hann window.



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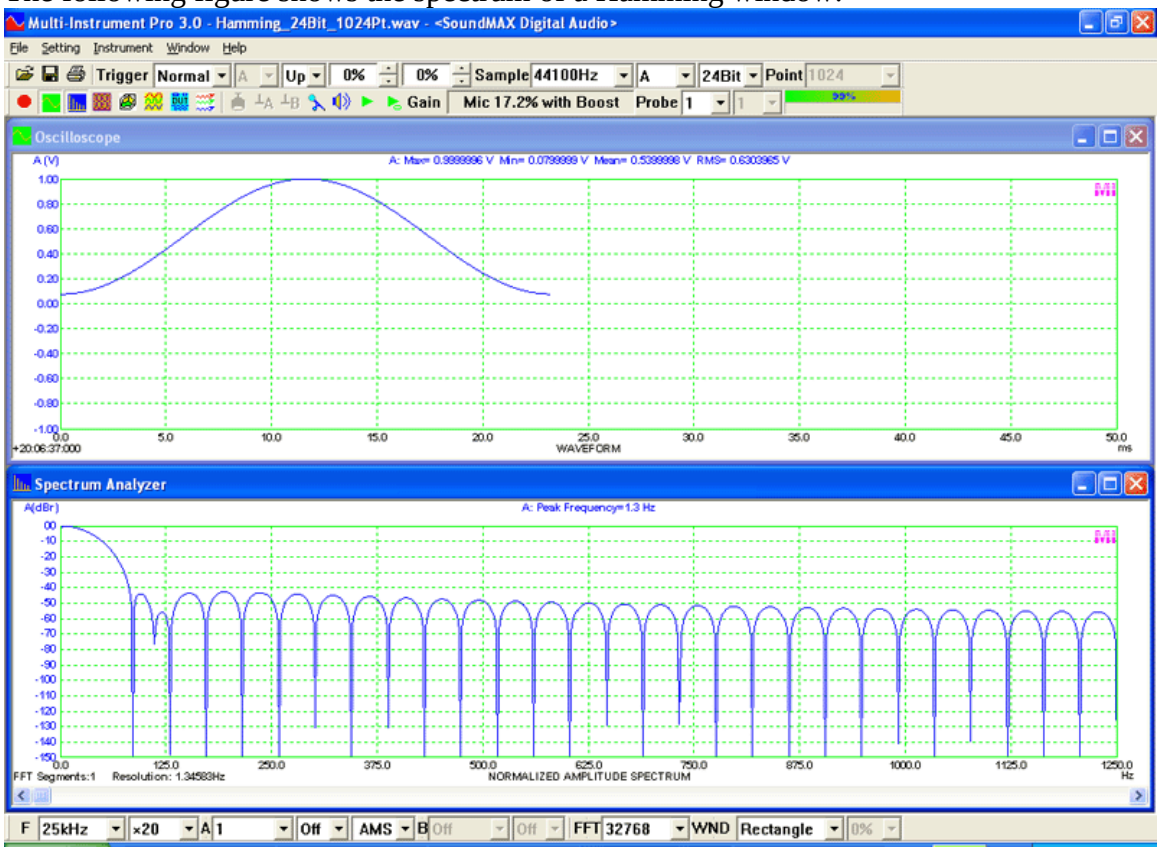
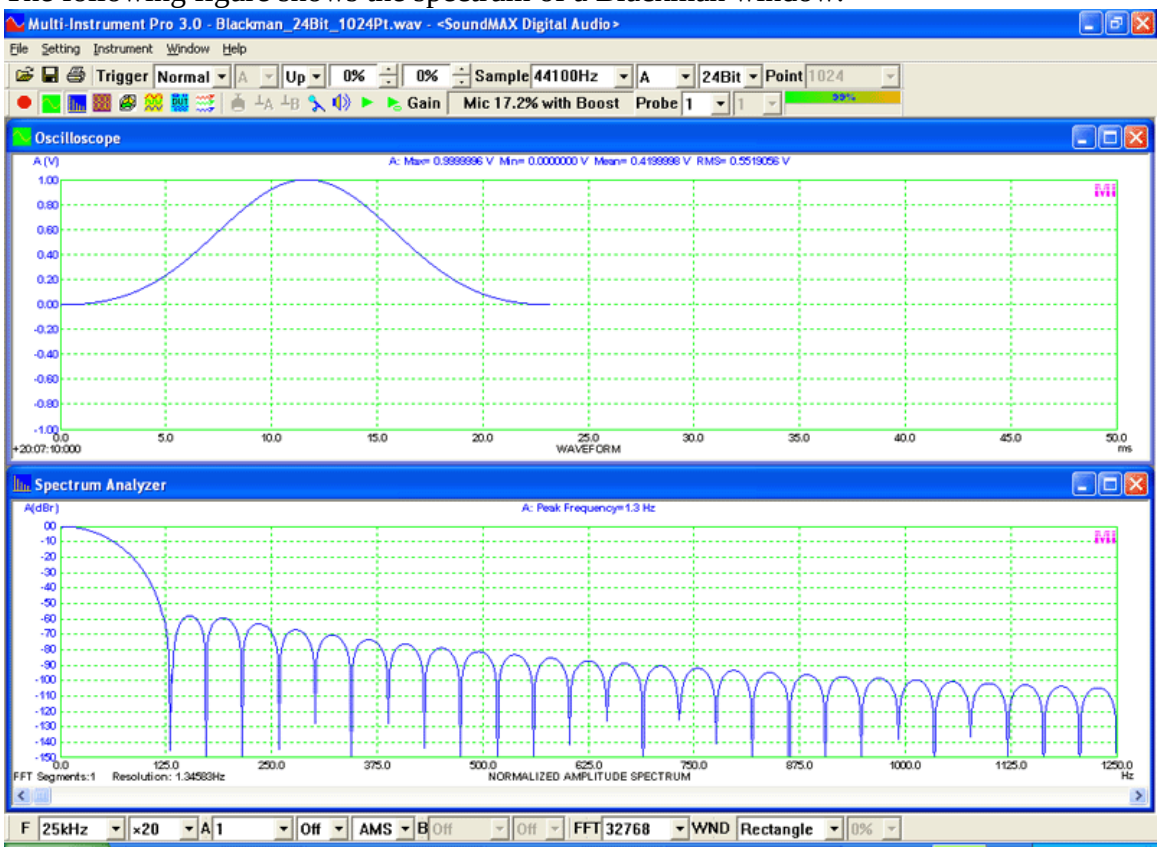
The following figure shows the spectrum of a Hamming window.



The following figure shows the spectrum of a Blackman window.



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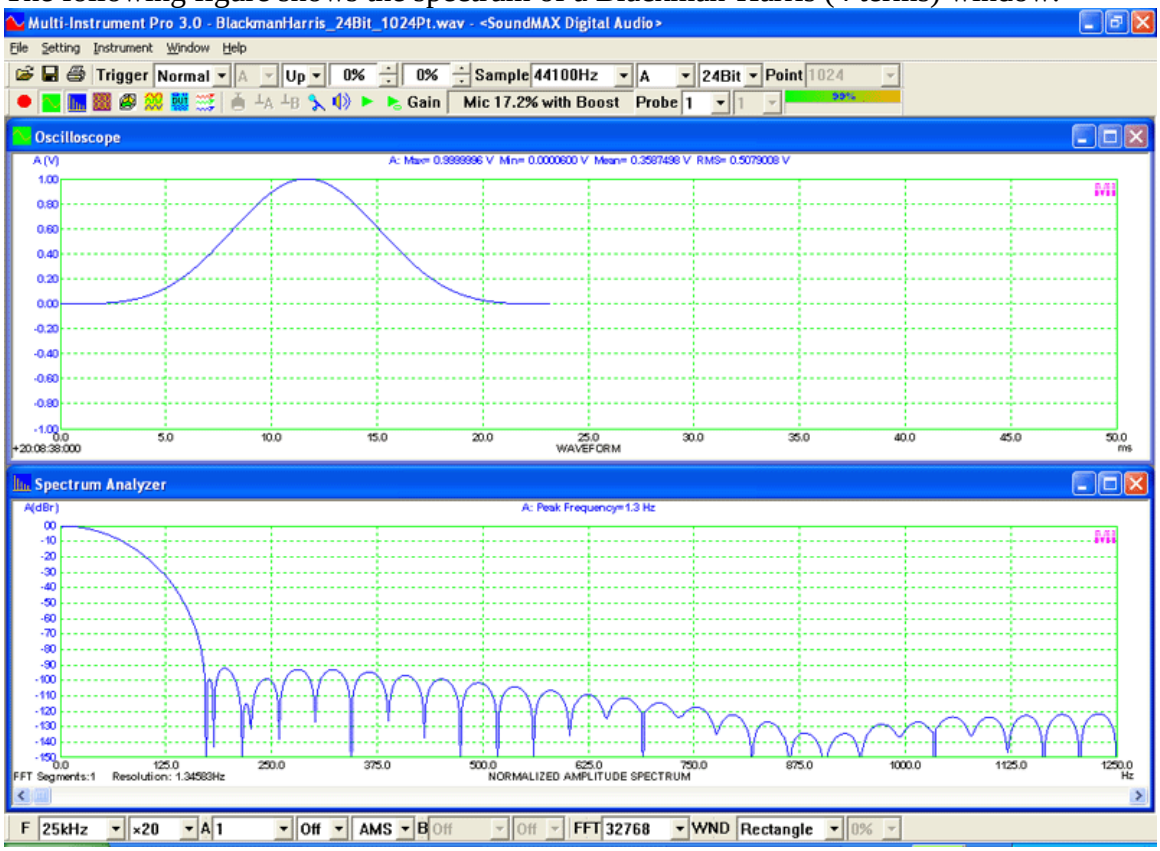
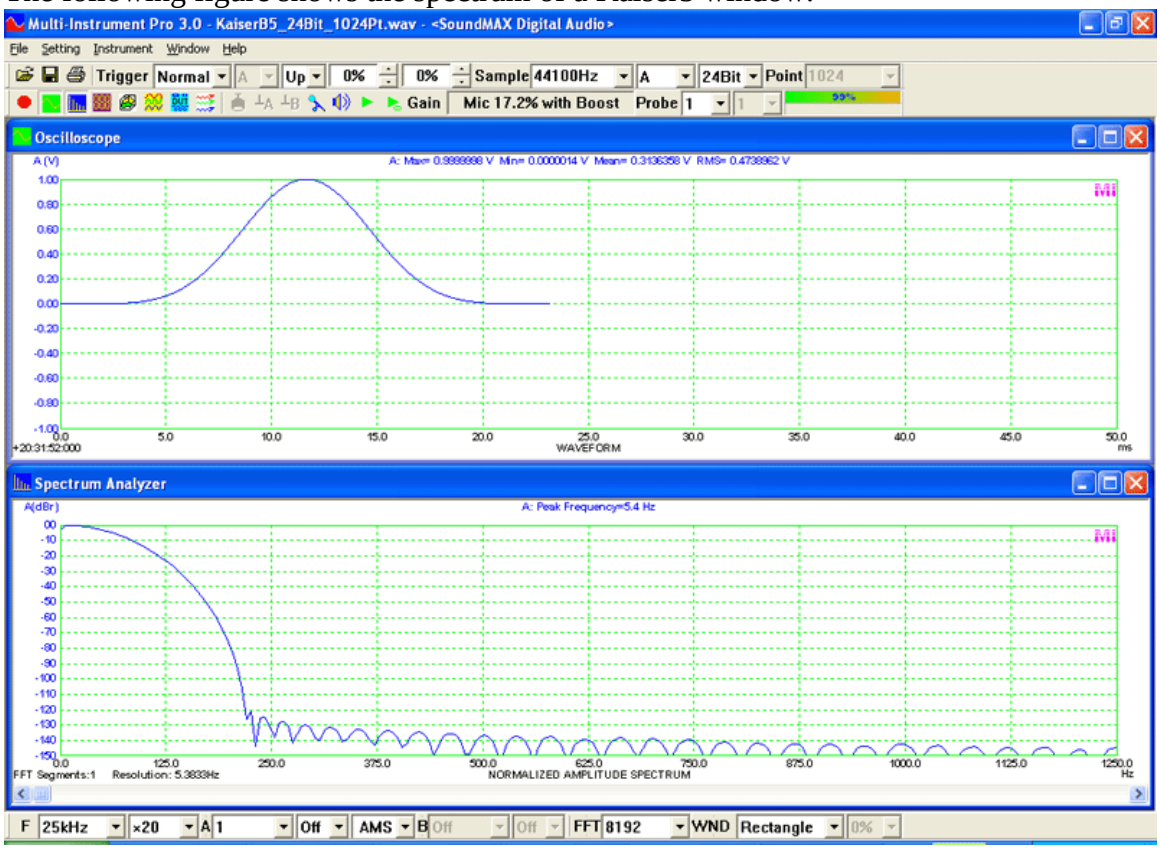
The following figure shows the spectrum of a Blackman-Harris (4 terms) window.



The following figure shows the spectrum of a Kaiser5 window.



From the above six figures, some important characteristics of the six windows can be readily obtained.

Window Function	Highest Side Lobe Level (dB)	Side Lobe Fall Off Rate (dB/Octave)	-3dB Main Lobe Width (bins)	-6dB Main Lobe Width (bins)	Scallop Loss (dB)	Coherent Gain	Equivalent Noise Bandwidth (bins)
Rectangle	-13	-6	0.88	1.21	3.92	1	1
Hann	-32	-18	1.44	2.00	1.42	0.5	1.50
Hamming	-43	-6	1.30	1.81	1.75	0.54	1.36
Blackman	-58	-18	1.64	2.29	1.10	0.42	1.73
Blackman-Harris (4 terms)	-92	-6	1.90	2.66	0.83	0.36	2.00
Kaiser5	-120	-6	2.16	3.03	0.64	0.31	2.28

The following figure summarizes the behavior of the above six windows in both time domain and frequency domain. The figure is obtained via the following steps:

1. Use “File Open” to open the WAV file of Rectangle window.
2. Use “File Combine” to open the WAV file of another window.
3. Set the settings for the Spectrum Analyzer properly as mentioned before
4. Copy both curves in the Oscilloscope and Spectrum Analyzer As Reference Curves
5. Use “File Combine” to import another two WAV files
6. Copy both curves in the Oscilloscope and Spectrum Analyzer As Reference Curves
7. Repeat 5~7.

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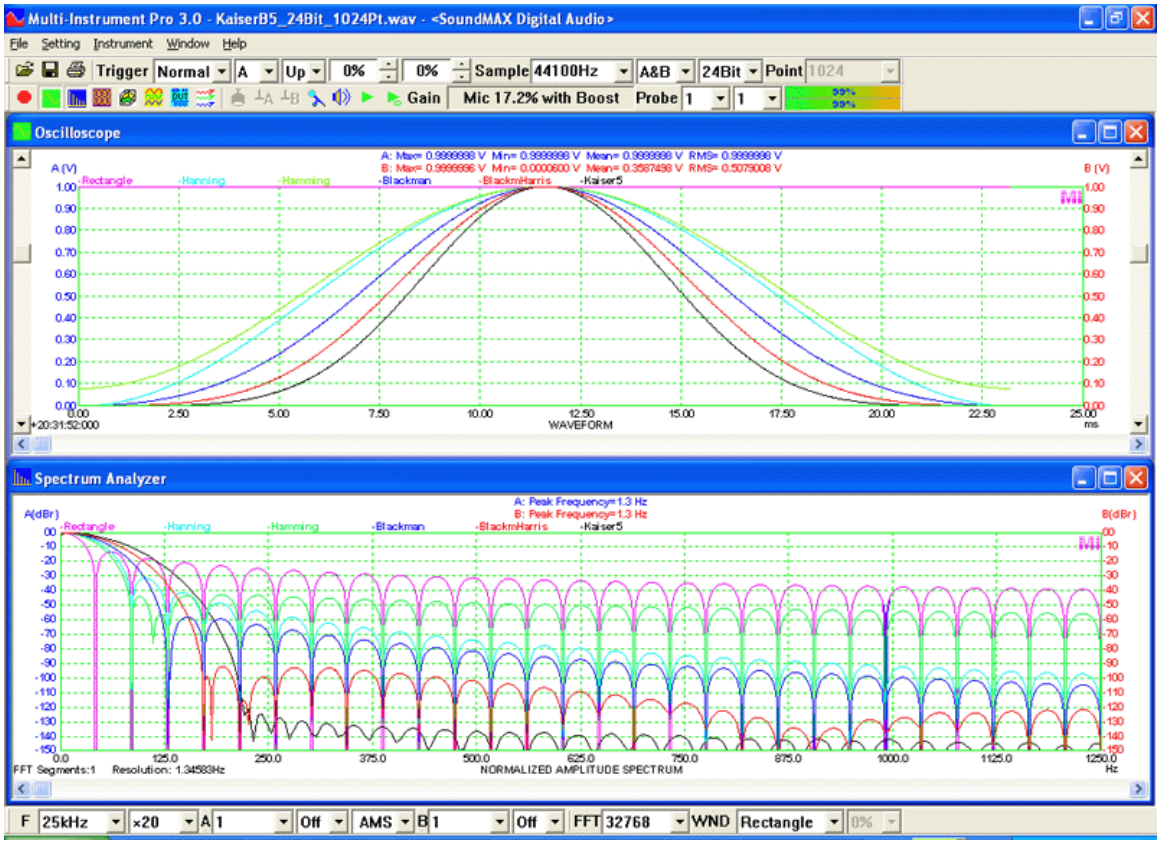


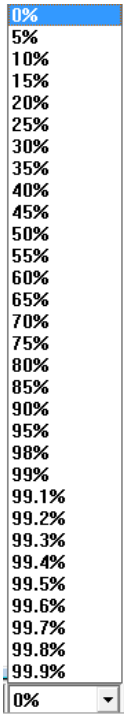
Spectral leakage is the result of the assumption in the FFT algorithm that the time record in a FFT segment is exactly repeated throughout all time and that signals contained in a FFT segment are thus periodic at intervals that correspond to the length of the FFT segment. If the time record in a FFT segment has a non-integer number of cycles, this assumption is violated and spectral leakage occurs. Spectral leakage distorts the measurement in such a way that energy from a given frequency component spreads to adjacent frequency lines or bins. In most cases, you cannot guarantee that you are sampling an integer number of cycles in a FFT segment. Choosing a window function correctly to suppress the spectral leakage for a certain measurement is thus critical. To choose a window function, you must estimate the signal frequency content first. If the signal contains strong interfering frequency components distant from the frequency of interest, choose a window with a high side lobe roll-off rate. If there are strong interfering signal near the frequency of interest, choose a window with a low highest side lobe level. If the frequency of interest contains two or more signals very near to each other, then frequency resolution is very important. It is best to choose a window with a very narrow main lobe. If the amplitude accuracy of a single frequency component is more important than the exact location of the component in a given frequency bin, choose a window with a wide main lobe. If the signal spectrum is rather flat or broadband in frequency content, use the Rectangle window. In general, the Hann window has a good frequency resolution and reduces spectral leakage. It is satisfactory in 95% of the cases.

To know the characteristics of more window functions, please refer to: *Evaluation of Various Window Functions using Multi-Instrument.*

Download link:

https://www.virtins.com/doc/D1003/Evaluation_of_Various_Window_Functions_using_Multi-Instrument_D1003.pdf

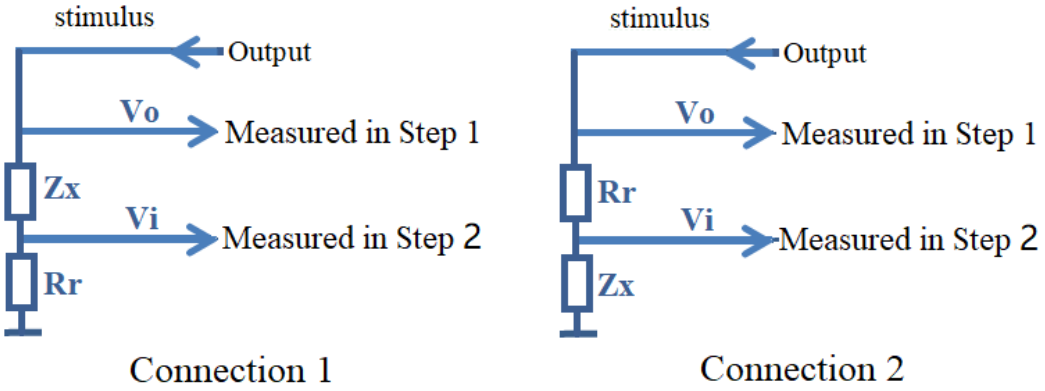
3.2.1.9 Window Overlap Percentage



The window function suppresses greatly the data at the edges of the window. As a result, these data contribute much less to the analysis result than the data at the center of the window. To make full use of the acquired data, FFT segments can be overlapped. The overlap ratio can be set via the Window Overlap Percentage combo box, from 0% to 99.9%. This combo box is enabled only when the number of data points per sweep (record length) is greater than the FFT size.

3.2.1.10 Impedance Measurement Mode

Amplitude Spectrum mode can be used to measure impedance variation with frequency using the single-channel method described below. It is possible to use two input channels of the measuring device to measure two impedances simultaneously. Two connection methods are supported in the software (see figure below).



where Zx is the impedance to be measured, Rr the reference resistor, Vo the output voltage from the signal generator and Vi the input voltage fed by the voltage divider formed by Zx and Rr. Both Vo and Vi are ground referenced, so measuring devices with either ground referenced inputs or differential inputs can be used. The input impedance of the measuring device must be much higher than Zx and Rr so that its effect on the measurement accuracy can be ignored. The output impedance and overall frequency response of the measuring device do not affect the measurement accuracy directly, as the same input channel is used to measure Vo and Vi, but it is still recommended for the measuring device to have a low output impedance and a flat overall frequency response.

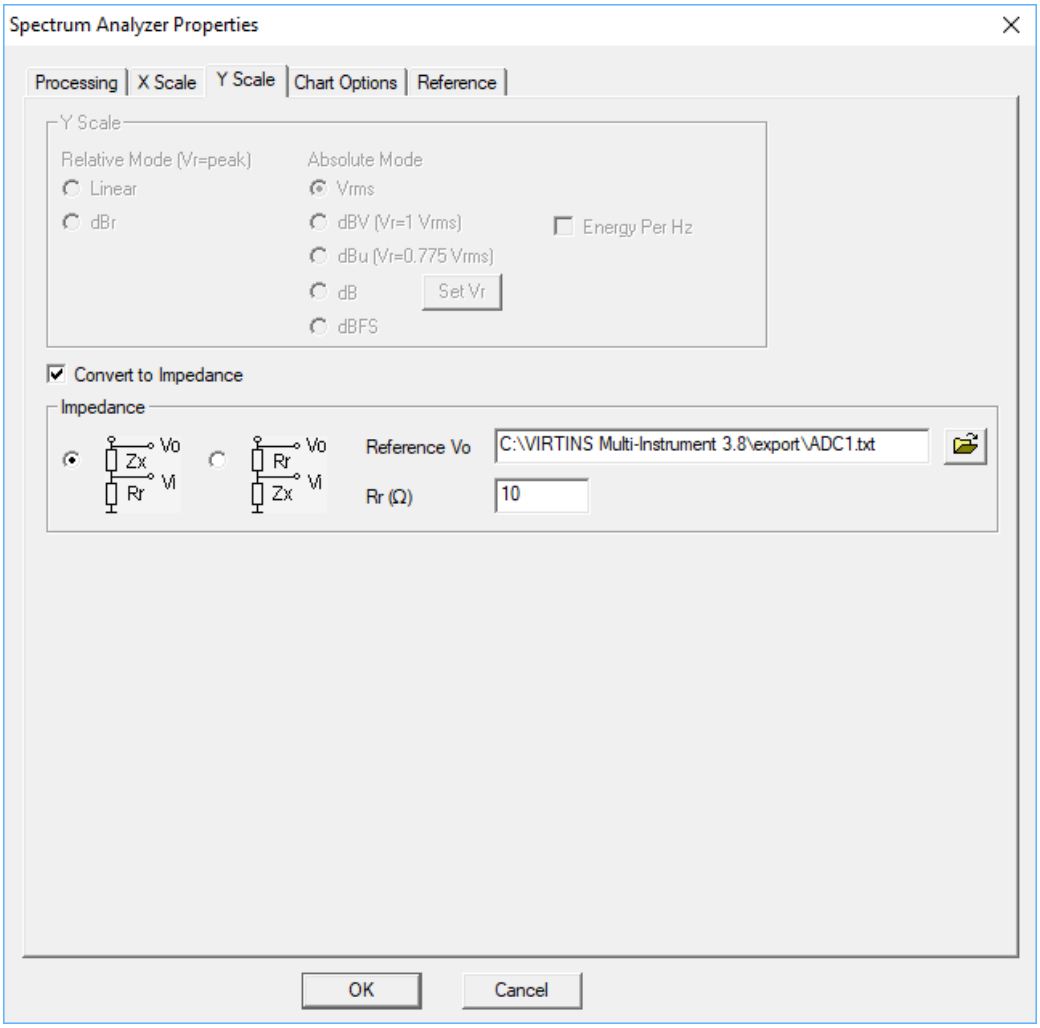
Rr is an external reference resistor. You need to find the resistor and make the connection by yourself. It is recommended to have Rr's value comparable to that of Zx. The actual value of Rr should be entered into the Rr(Ω) edit box in [Spectrum Analyzer Y Scale] dialog box. The actual connection used also needs to be selected there. In Connection 1, Zx = Rr × (Vo-Vi) / Vi while in Connection 2, Zx = Rr × Vi / (Vo-Vi). To achieve a good measurement accuracy, significant stimulus energy must be present in the frequency range of interest. Commonly used signals are chirp signal (frequency linearly or logarithmically swept sine) and multitones capable of simultaneously stimulating all FFT/octave frequency bands in the frequency range of interest. Unlike white and pink noises, they have a repeatable waveform. This is important when Vo and Vi are measured in two separate steps. Various averaging methods can be used to smoothen the result except the inter-frame peak-hold, linear average with the "forever" option and exponential average.

Two steps are required:

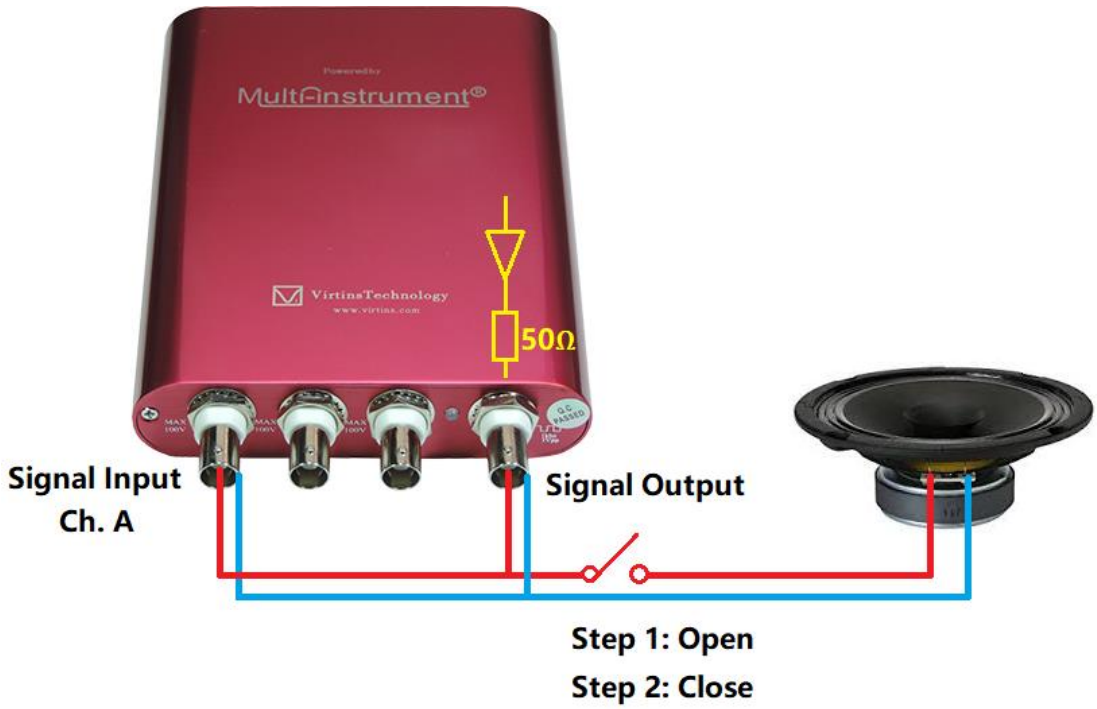
(1) Step 1
Vo is measured and the Spectrum Analyzer Y scale MUST be set to Vrms. After Vo's amplitude spectrum (in Vrms) is obtained, right click anywhere within the Spectrum Analyzer window and select "Spectrum Analyzer Export" to export it as a TXT file. Then right click anywhere within the Spectrum Analyzer window again and select "Spectrum Analyzer Y Scale". Tick "Convert to Impedance" checkbox and load the TXT file saved previously as the Reference Vo file (see figure below).

(2) Step 2

Vi is measured with exactly the same sampling, processing and analysis settings as Step 1, except that the voltage measurement range can be changed if the measuring device is voltage calibrated. In case the settings in Step 2 is different from those in Step 1, the reference Vo file will be invalid and the "Convert to Impedance" checkbox will be unticked automatically.



It should be noted that in most of the cases, Step 1 needs only to be conducted once as long as the signal generator has a sufficiently low output impedance so that its output voltage Vo does not virtually vary with Zx + Rr. For example, many audio power amplifiers have an output impedance approaching 0 Ω, therefore they are very suitable for this kind of tests. It is also possible to use the signal generator's internal resistance as the reference resistor Rr so as to simplify the connection. For example, the signal generator of VT DSO-2810E, 2820E, 2A10E and 2A20E have an output impedance (resistance) of 50Ω, thus they can be readily used to measure the impedance of a Device Under Test such as a speaker. In this case, the reference Vo can be measured through measuring Vi without the connection of Zx (refer to the above Connection 2) given that the input impedance of the measuring device is high enough. In other words, Step 1 is to measure Vi without the connection of Zx and Step 2 is to measure Vi with the connection of Zx. This method is applicable to Connection 2 only. The following figure shows the connection of this method.



Under the impedance measurement mode, Channel A Display Range and Channel B Display Range will be updated to display impedance in Ω. The display range of the "Auto" option is about 5×Rr. The following DDPs are generated by this mode:

(1) f1Freq_A(Hz): Frequency at the peak impedance in Ch. A
(2) f1RMS_A(EU): Peak impedance in Ch. A
(3) InBandMinF_A(Hz): Frequency at which the minimum impedance is located in the specified frequency band in Ch. A
(4) InBandMinRMS_A(EU): The minimum impedance in the specified frequency band in Ch. A
(5) InBandMaxF_A(Hz): Frequency at which the maximum impedance is located in the specified frequency band in Ch. A
(6) InBandMaxRMS_A(EU): The maximum impedance in the specified frequency band in Ch. A
(7) oBandRMS_A(EU)_Array[?]: Impedance value in a particular octave band in Channel A. "?" should be replaced by the actual band number starting from 0.

A similar set of DDPs are also generated for Channel B: f1Freq_B(Hz), f1RMS_B(EU), InBandMinF_B(Hz), InBandMinRMS_B(EU), InBandMaxF_B(Hz), InBandMaxRMS_B(EU), oBandRMS_B(EU)_Array[?].

The following examples use different stimuli and configurations to measure a 2.5" 8Ω speaker's impedance. The first three examples use VT DSO-2A20E as the ADC device and a sound card as the DAC device, with an external 25Ω reference resistor Rr using Connection 1. The latter three examples use VT DSO-2A20E as both the ADC and DAC device, and its 50Ω output impedance (resistance) as the reference resistor Rr, based on Connection 2. To avoid the interference from the automatically generated oscilloscope probe calibration signal, switch it off via [Setting]>[DAC Device]> "Probe CAL" before tests.

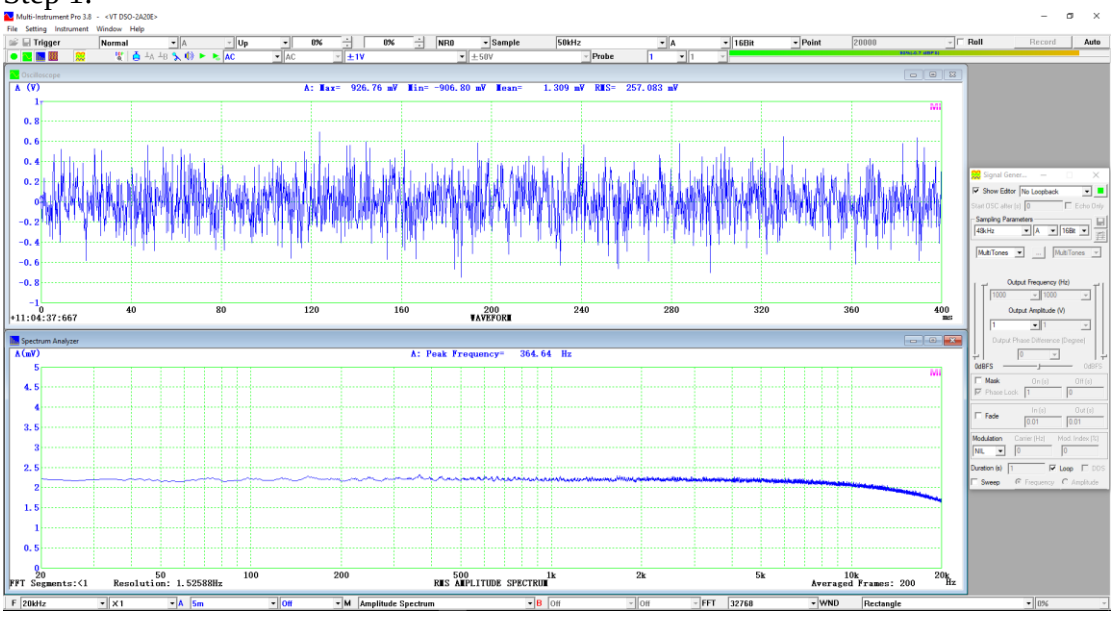
(1) Multitone stimulus aligned to FFT bands (Periodic white Noise)

Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 400 ms (20000 samples)
Trigger Mode: Normal, Trigger Level: 0%

Spectrum Analyzer:
FFT size: 32768, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: 200-frame Linear Average
X Scale: Log, Y Scale: Vrms,
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

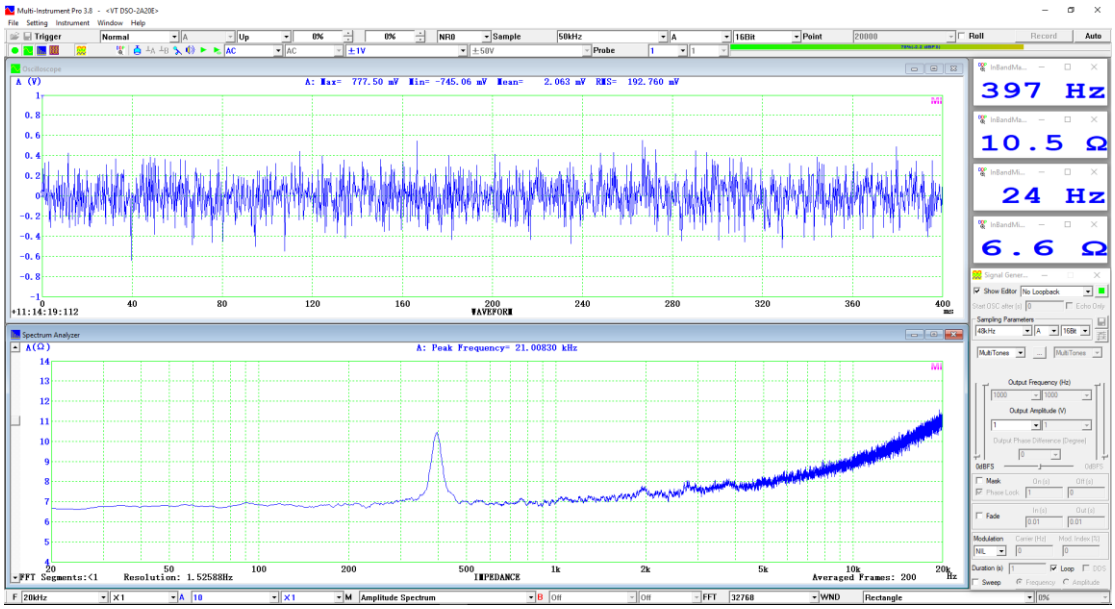
Signal Generator:
DAC Device: Sound Card Headphone Output, Duration: Loop
Waveform: Multitones aligned to FFT, Frequency Range: 10 Hz ~ 21 kHz
Sampling Frequency: 48 kHz

Step 1:



The above figure actually shows the overall frequency response of the setup. It mainly reflects the fact that VT DSO-2A20E's anti-aliasing filter has a cutoff frequency at 0.443 × [Sampling Frequency] = 0.443 × 50 kHz = 22.15 kHz under its 16-bit mode. Note that there is no need to compensate the frequency response in this test.

Step 2:



The measured resonant frequency is 397 Hz at the impedance peak 10.5 Ω. The lowest impedance is 6.6 Ω at the lower frequency end.

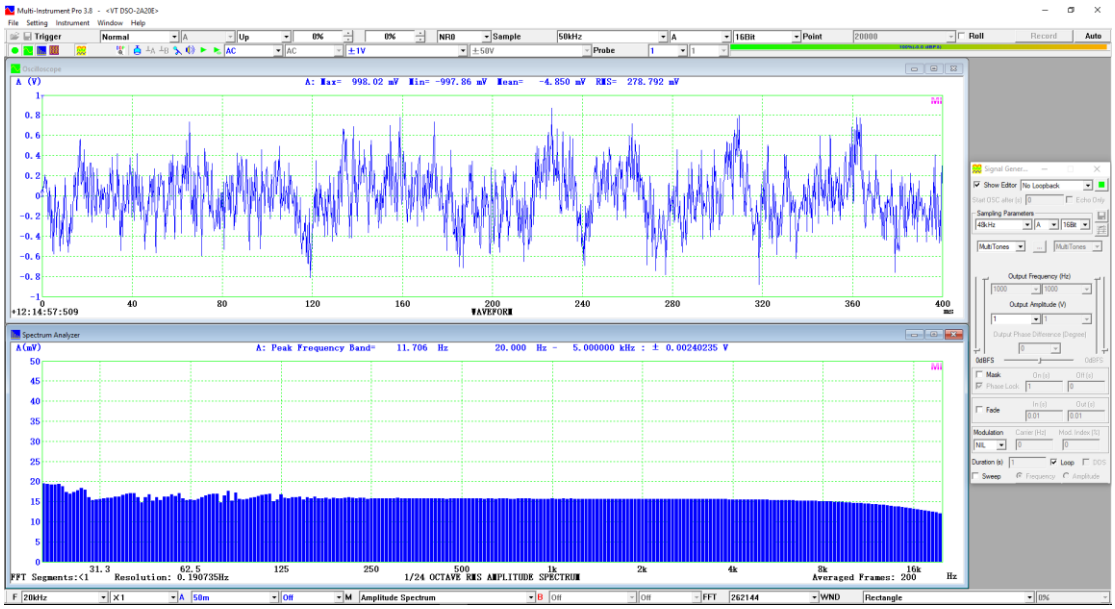
(2) Multitone stimulus aligned to 1/24 Octave bands

Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 400 ms (20000 samples)
Trigger Mode: Normal, Trigger Level: 0%

Spectrum Analyzer:
FFT size: 262144, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: 200-frame Linear Average
X Scale: 1/24 octave, Y Scale: Vrms,
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

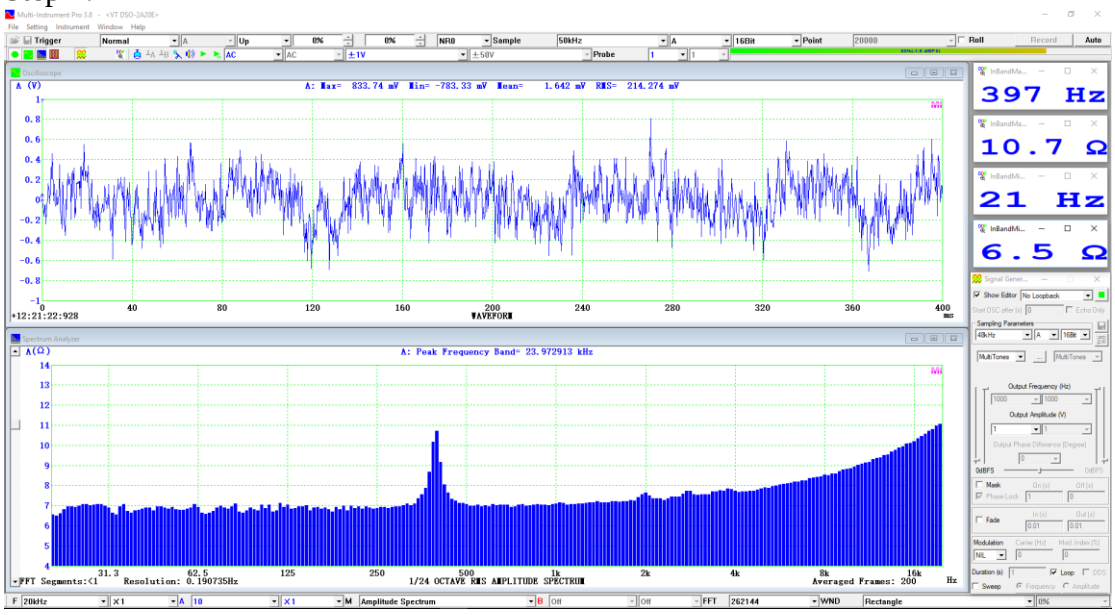
Signal Generator:
DAC Device: Sound Card Headphone Output, Duration: Loop
Waveform: Multitones aligned to 1/24 octave, Frequency Range: 10 Hz ~ 24 kHz
Sampling Frequency: 48 kHz

Step 1:



The above figure actually shows the overall frequency response of the setup. It mainly reflects the fact that VT DSO-2A20E's anti-aliasing filter has a cutoff frequency at 0.443 × [Sampling Frequency] = 0.443 × 50 kHz = 22.15 kHz under its 16-bit mode. The small undulation at the lower frequency end is due to the fact that octave bands have a much higher resolution at lower frequencies. This calls for a longer record length in the oscilloscope. Note that there is no need to compensate the frequency response in this test.

Step 2:



The measured resonant frequency is 397 Hz at the impedance peak 10.7 Ω. The lowest impedance is 6.5 Ω at the lower frequency end.

(3) Frequency logarithmically swept sine wave

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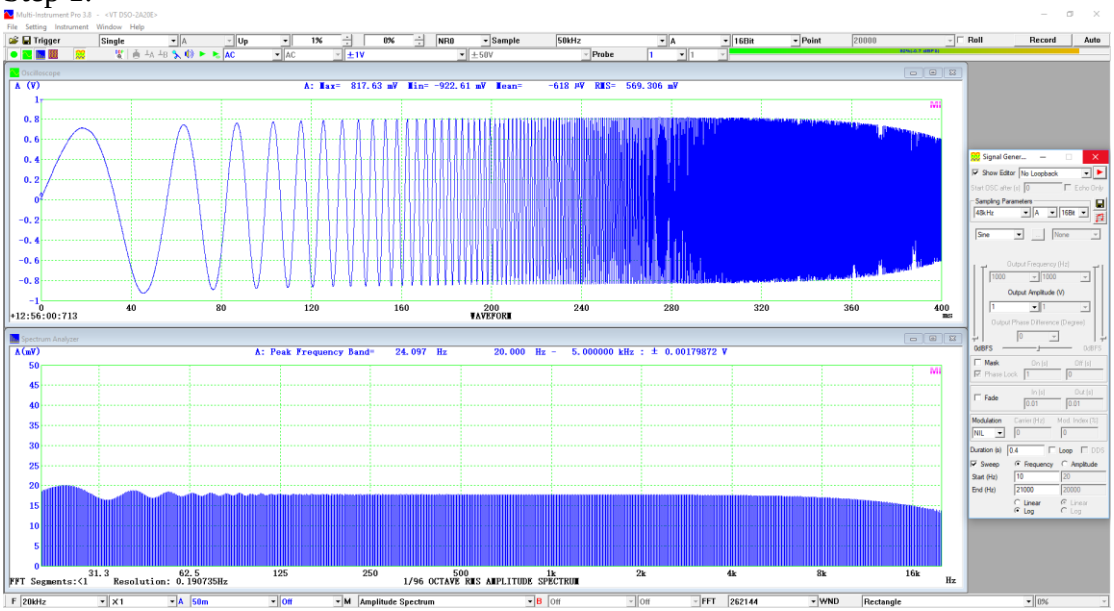
Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 400 ms (20000 samples)
Trigger Mode: Single, Trigger Level: 1%

Spectrum Analyzer:
FFT size: 262144, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: None
X Scale: 1/96 octave, Y Scale: Vrms,
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

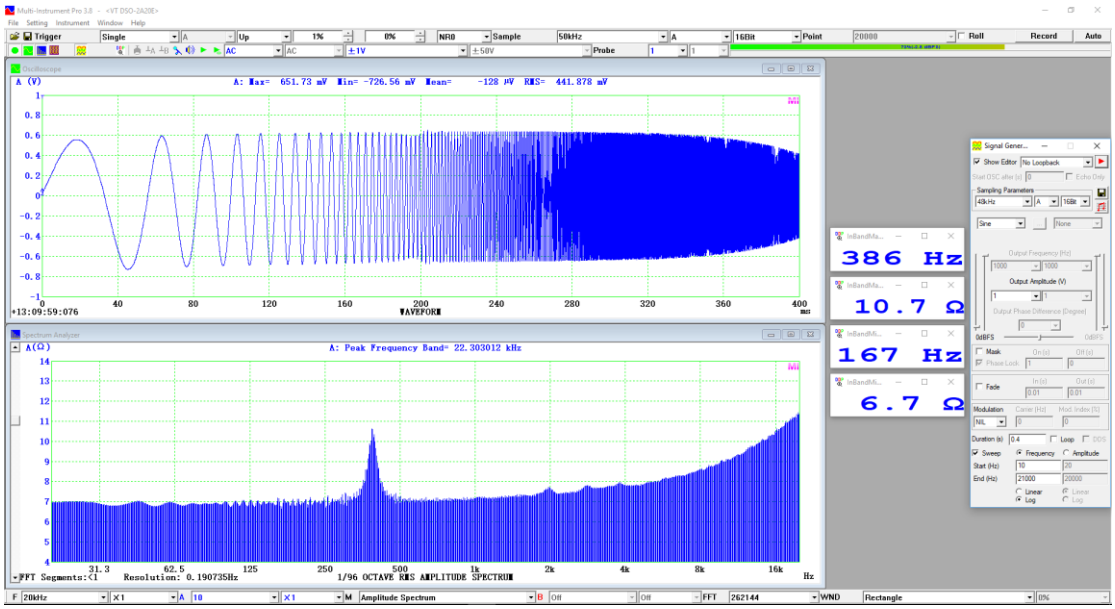
Signal Generator:
DAC Device: Sound Card Headphone Output, Duration: 400 ms
Waveform: Frequency logarithmically swept sine wave from 10 Hz to 21 kHz
Sampling Frequency: 48 kHz

Step 1:



The above figure actually shows the overall frequency response of the setup. It mainly reflects the fact that VT DSO-2A20E's anti-aliasing filter has a cutoff frequency at 0.443 × [Sampling Frequency] = 0.443 × 50 kHz = 22.15 kHz under its 16-bit mode. The small undulation at the lower frequency end is due to the fact that octave bands have a much higher resolution at lower frequencies. This calls for a longer record length in the oscilloscope. Note that there is no need to compensate the frequency response in this test.

Step 2:



The measured resonant frequency is 386 Hz at the impedance peak 10.7 Ω. The lowest impedance is 6.7 Ω at the lower frequency end.

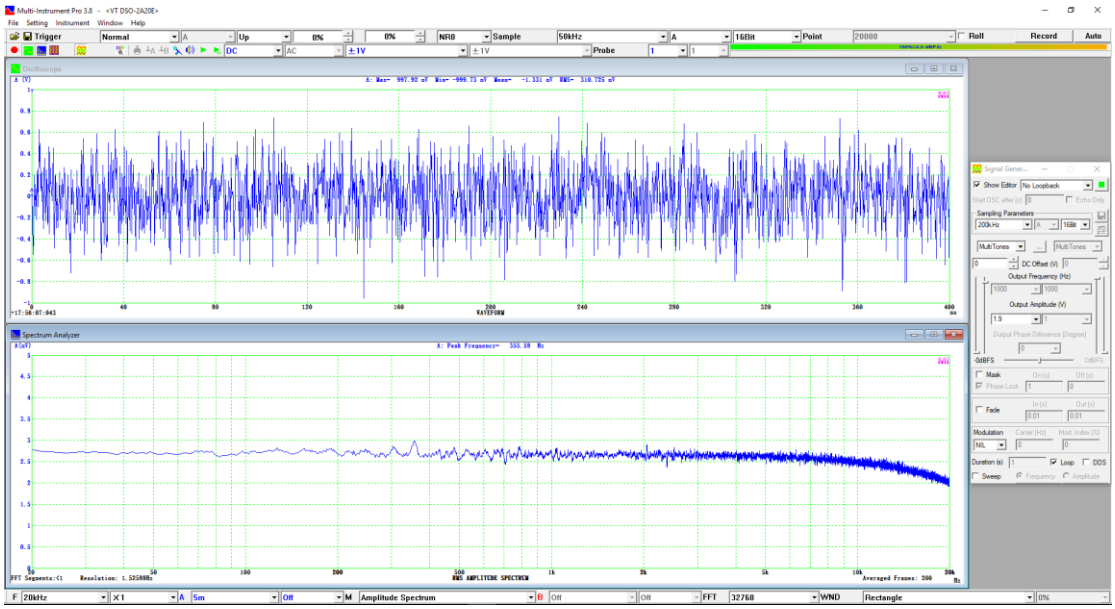
(4) Multitone stimulus aligned to FFT bands (Periodic White Noise)

Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 400 ms (20000 samples)
Trigger Mode: Normal, Trigger Level: 0%

Spectrum Analyzer:
FFT size: 32768, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: 200-frame Linear Average
X Scale: Log, Y Scale: Vrms,
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

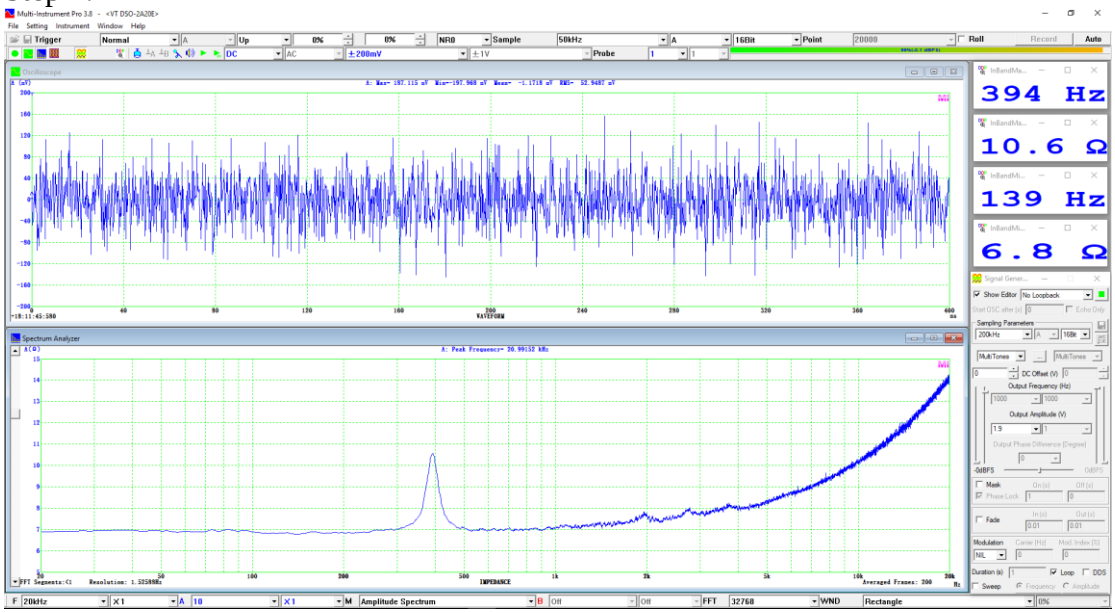
Signal Generator:
DAC Device: VT DSO-2A20E streaming mode, Probe CAL: Disabled
Waveform: Multitones aligned to FFT, Frequency Range: 10 Hz ~ 21 kHz
Sampling Frequency: 200 kHz, Duration: Loop

Step 1:



The above figure actually shows the overall frequency response of the setup. It mainly reflects the fact that VT DSO-2A20E's anti-aliasing filter has a cutoff frequency at 0.443 × [Sampling Frequency] = 0.443 × 50 kHz = 22.15 kHz under its 16-bit mode. Note that there is no need to compensate the frequency response in this test.

Step 2:



The measured resonant frequency is 394 Hz at the impedance peak 10.6 Ω. The lowest impedance is 6.8 Ω at the lower frequency end.

(5) Multitone stimulus aligned to 1/24 Octave bands

Oscilloscope:

ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 400 ms (20000 samples)

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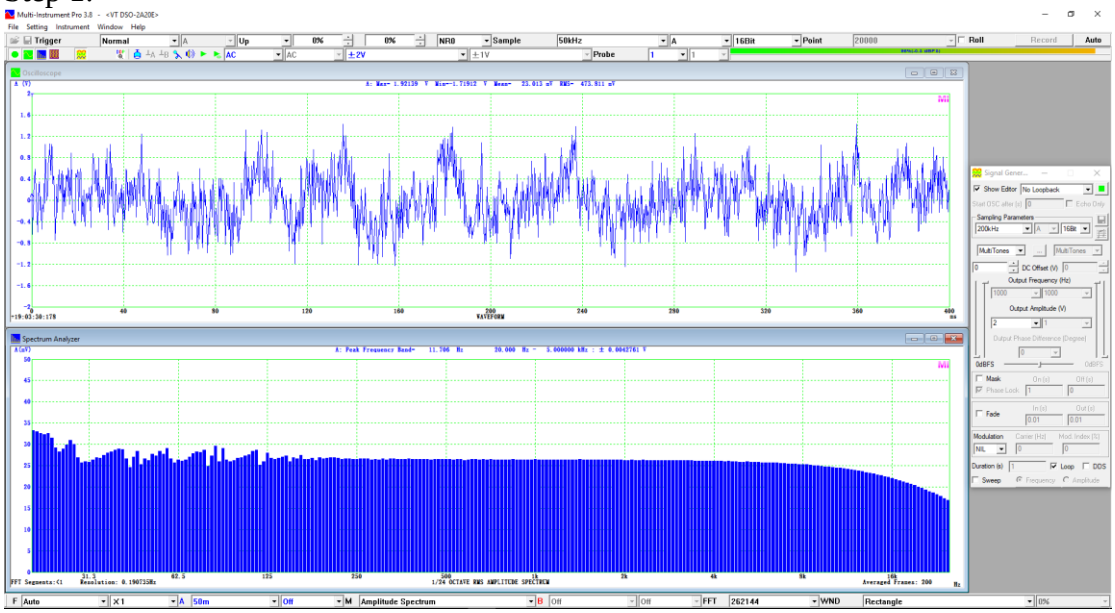
Trigger Mode: Normal, Trigger Level: 0%

Spectrum Analyzer:
FFT size: 262144, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: 200-frame Linear Average
X Scale: 1/24 octave, Y Scale: Vrms,
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

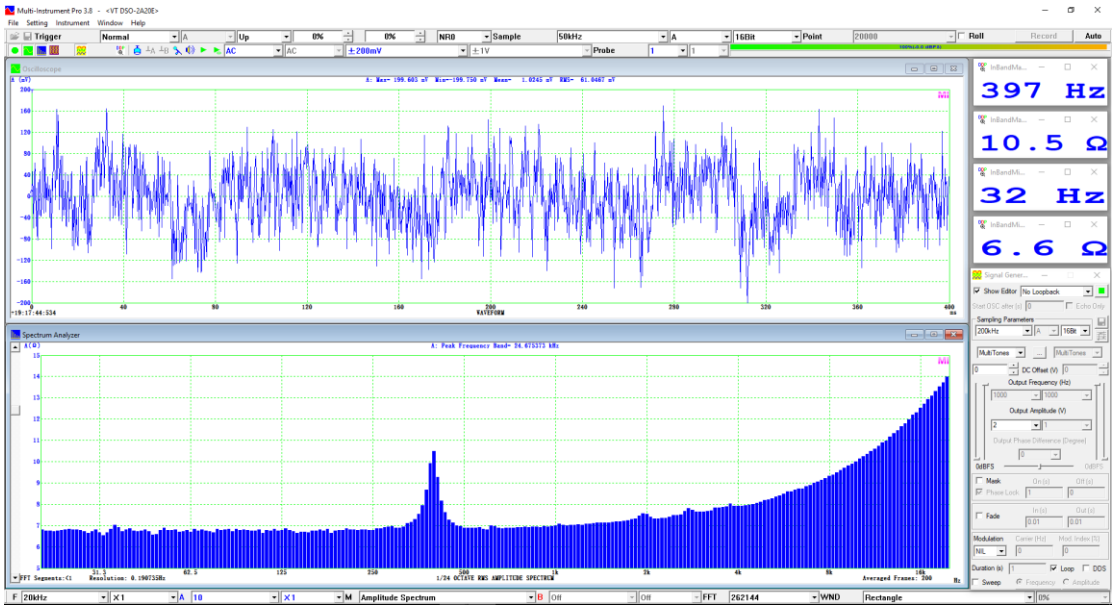
Signal Generator:
DAC Device: VT DSO-2A20E streaming mode, Probe CAL: Disabled
Waveform: Multitones aligned to 1/24 octave, Frequency Range: 10 Hz ~ 24 kHz
Sampling Frequency: 200 kHz, Duration: Loop

Step 1:



The above figure actually shows the overall frequency response of the setup. It mainly reflects the fact that VT DSO-2A20E's anti-aliasing filter has a cutoff frequency at 0.443 × [Sampling Frequency] = 0.443 × 50 kHz = 22.15 kHz under its 16-bit mode. The small undulation at the lower frequency end is due to the fact that octave bands have a much higher resolution at lower frequencies. This calls for a longer record length in the oscilloscope. Note that there is no need to compensate the frequency response in this test.

Step 2:



The measured resonant frequency is 397 Hz at the impedance peak 10.5 Ω. The lowest impedance is 6.6 Ω at the lower frequency end.

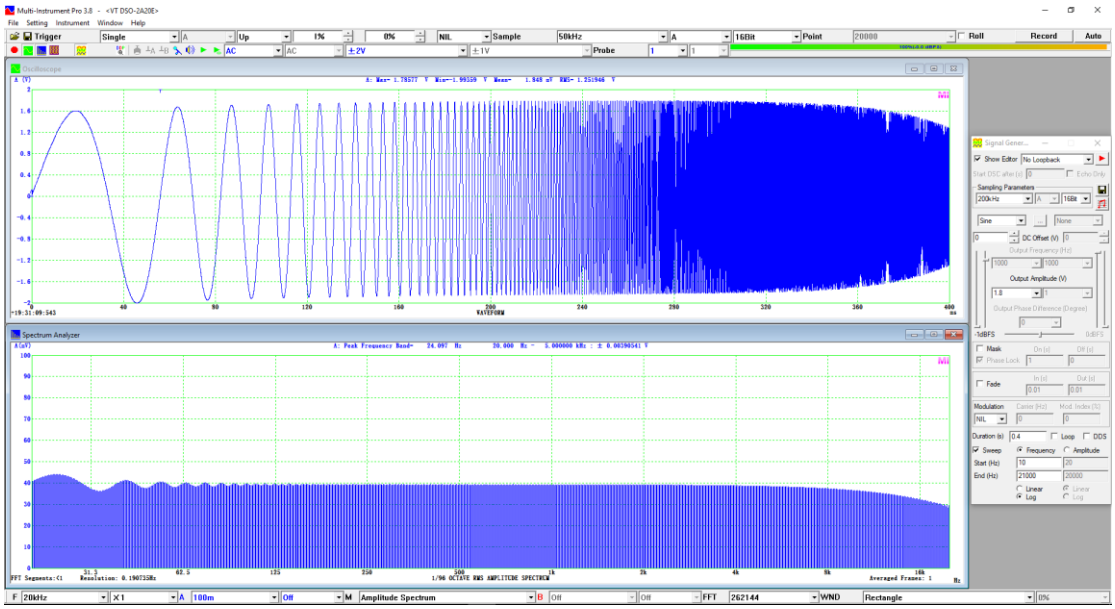
(6) Frequency logarithmically swept sine wave

Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 400 ms (20000 samples)
Trigger Mode: Single, Trigger Level: 1%

Spectrum Analyzer:
FFT size: 262144, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: None
X Scale: 1/96 octave, Y Scale: Vrms,
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

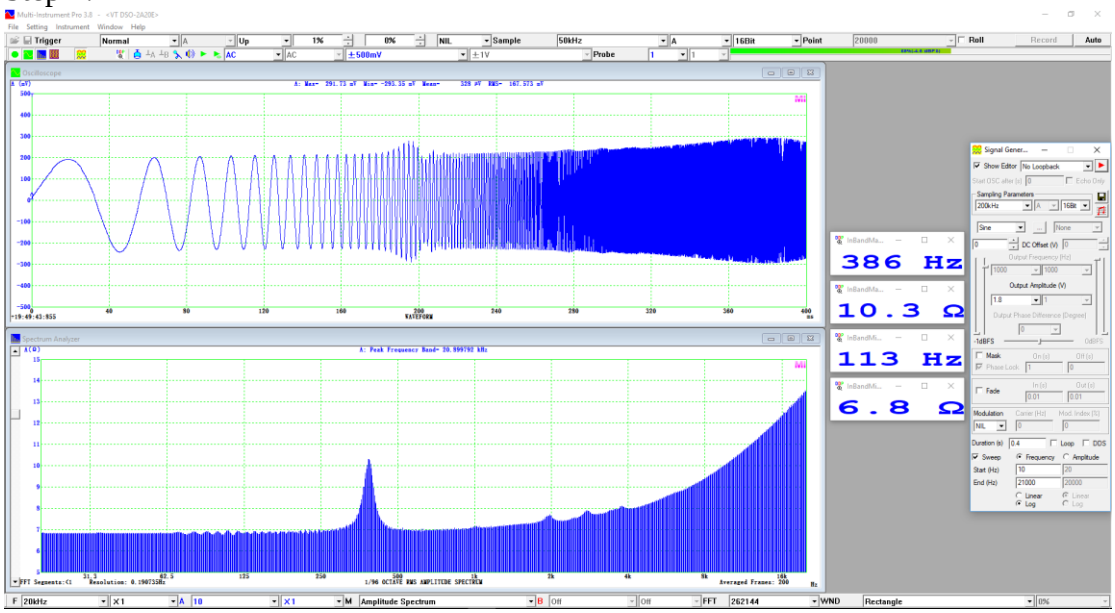
Signal Generator:
DAC Device: VT DSO-2A20E streaming mode, Probe CAL: Disabled
Waveform: Frequency logarithmically swept sine wave from 10 Hz to 21 kHz
Sampling Frequency: 200 kHz, Duration: 400 ms

Step 1:



The above figure actually shows the overall frequency response of the setup. It mainly reflects the fact that VT DSO-2A20E's anti-aliasing filter has a cutoff frequency at 0.443 × [Sampling Frequency] = 0.443 × 50 kHz = 22.15 kHz under its 16-bit mode. The small undulation at the lower frequency end is due to the fact that octave bands have a much higher resolution at lower frequencies. This calls for a longer record length in the oscilloscope. Note that there is no need to compensate the frequency response in this test.

Step 2:



The measured resonant frequency is 386 Hz at the impedance peak 10.3 Ω. The lowest impedance is 6.8 Ω at the lower frequency end.

All the above tests obtain similar results. Among them, the frequency logarithmically swept sine wave is the fastest method. Multitone stimuli should have converged faster if a record length of 1 or multiple seconds had been chosen.

The Panel Setting Files for Step 2 of the above Examples (4), (5) and (6) can be found in the software's installation directory:

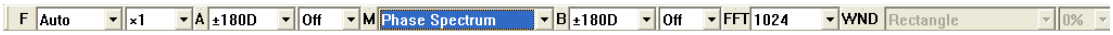
psf\VTDSO\Impedance_MultiTones_FFT32768_2810E.psf,
psf\VTDSO\Impedance_MultiTones_Octave24_2810E.psf,
psf\VTDSO\Impedance_LogSweptSine_Octave96_2810E.psf , respectively.

They can be used directly to measure an impedance in the order of 50Ω using VT DSO-2810E, 2820E, 2A10E, or 2A20E. General reference Vo files have already been provided for these settings and thus Step 1 can be omitted. However, it is recommended to perform Step 1 to re-calibrate the Reference Vo file for higher accuracy. If the reference Rr in [Spectrum Analyzer Y Scale] is changed to 600Ω instead, then the modified settings can be used to measure an impedance in the order of 600Ω using VT DSO-2810, 2820, 2A10, or 2A20. The Signal Generators of these DSOs have an output impedance of about 600Ω.

If a sound card is used as both the ADC and DAC devices for impedance measurement, the following Panel Setting Files corresponding to Step 1 of the above Examples (1), (2) and (3) can be used:

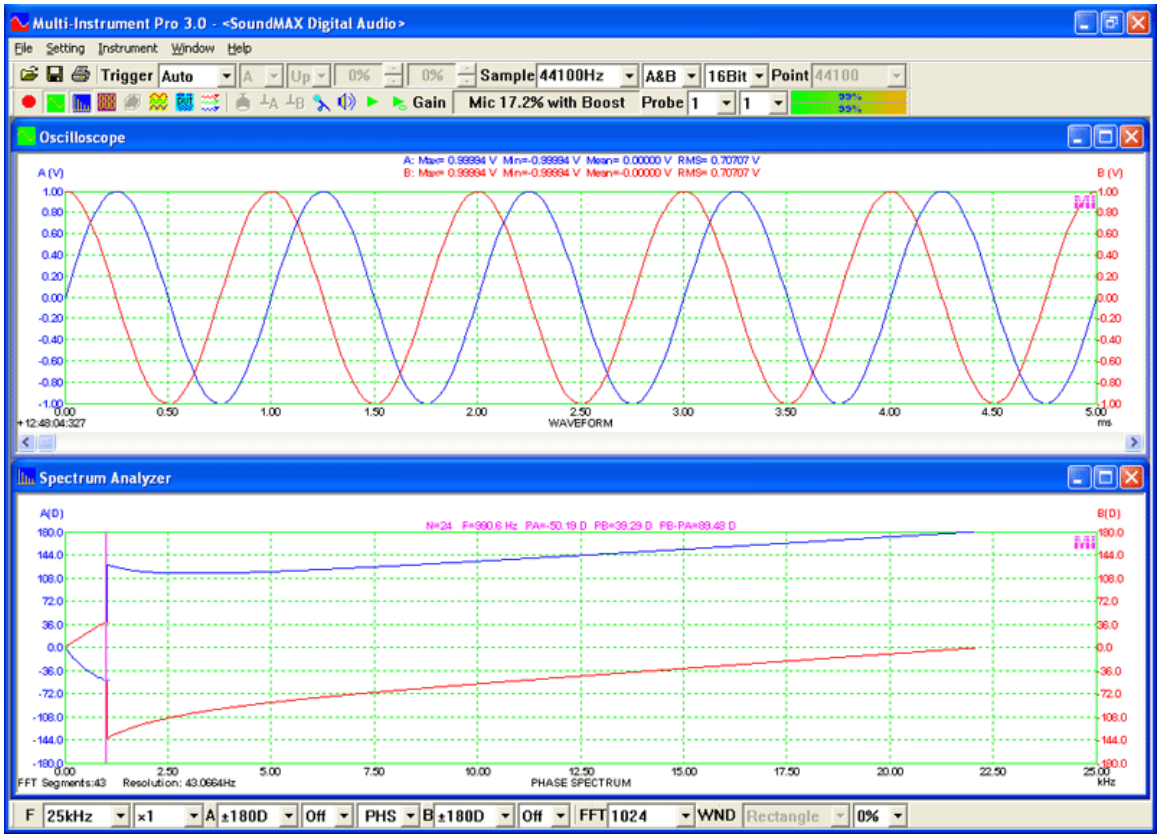
psf\Pro\Impedance_MultiTones_FFT65536_SoundCard.psf,
psf\Pro\Impedance_MultiTones_Octave24_SoundCard.psf,
psf\Pro\Impedance_LogSweptSine_Octave96_SoundCard.psf, respectively.

3.2.2 View Parameters for Phase Spectrum Display



Selecting Phase Spectrum in the above View Type selection box switches the Spectrum Analyzer to phase spectrum display mode which shows the phase vs frequency graph of the measured signal. Phase spectrum is normally used to compare or read out the phase difference between the two channels rather than read out the absolute phase information of each channel.

The following figure illustrates the phase spectrum of two 1 kHz sine waves with a phase difference of 90 degrees. The cursor reader in the phase spectrum indicates a 90-degreee difference around 1 kHz.



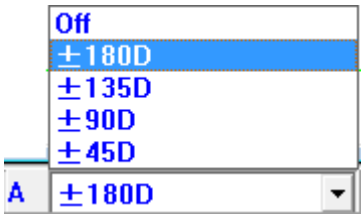
3.2.2.1 Frequency Range (F)

Same as Amplitude Spectrum.

3.2.2.2 Frequency Multiplier

Same as Amplitude Spectrum.

3.2.2.3 Channel A Display Range



Five options available: Off, ± 180D (i.e. ± 180 degree), 135D, 90D, 45D.

When "Off" is selected, the data in Channel A will not be displayed.

3.2.2.4 Channel A Multiplier

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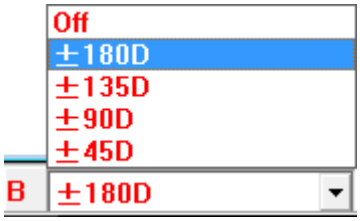
Same as Amplitude Spectrum.

3.2.2.5 Channel B Display Range



Five options available: Off, ± 180D (i.e. ± 180 degree), 135D, 90D, 45D.

When "Off" is selected, the data in Channel B will not be displayed.

3.2.2.6 Channel B Multiplier

Same as Amplitude Spectrum.

3.2.2.7 FFT Size

Same as Amplitude Spectrum.

3.2.2.8 Window Function

Same as Amplitude Spectrum.

3.2.2.9 Window Overlap Percentage

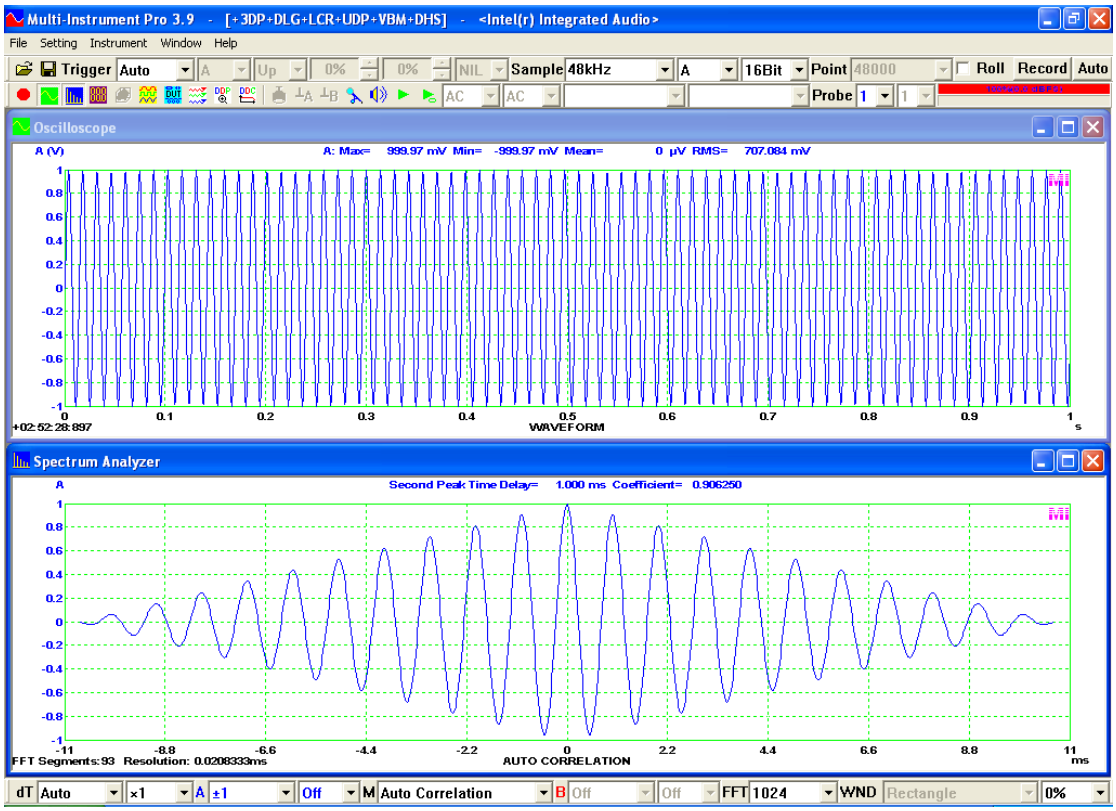
Same as Amplitude Spectrum.

3.2.3 View Parameters for Auto Correlation Function Display

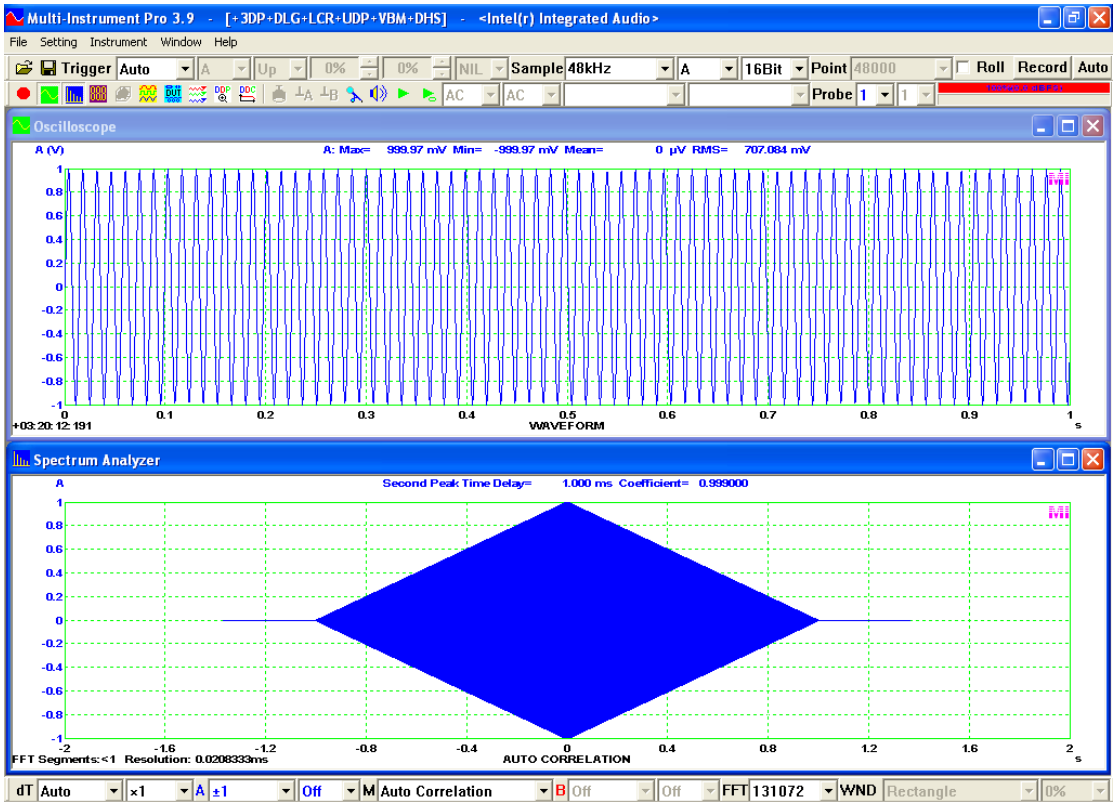
Selecting Auto Correlation in the above View Type selection box switches the Spectrum Analyzer to auto correlation display mode which shows the auto correlation coefficient (i.e. normalized auto correlation function) vs time delay graph of the measured signal. The auto correlation function here is acyclic or linear by default. It is symmetric and always has the highest auto correlation coefficient (=1) at [Time Delay] =0. Therefore, there is no point for the software to display textual information about this peak. The time delay and auto correlation coefficient of the second highest peak are shown instead. They can also be accessed through these Derived Data Points (DDPs): SecondPeakACFTimeDelay_A(s) and SecondPeakACFCoef_A for Channel A, and SecondPeakACFTimeDelay_B(s) and SecondPeakACFCoef_B for Channel B.

The highest auto correlation peak may be surrounded by a few sub-peaks. A deadband (in milliseconds) is used to avoid detecting one of these surrounding sub-peaks, within the deadband of the highest peak, as the second highest peak. The deadband can be adjusted via [Spectrum Analyzer Processing]> "Parameter Measurement" > "Deadband". Its default value is 0.25 ms.

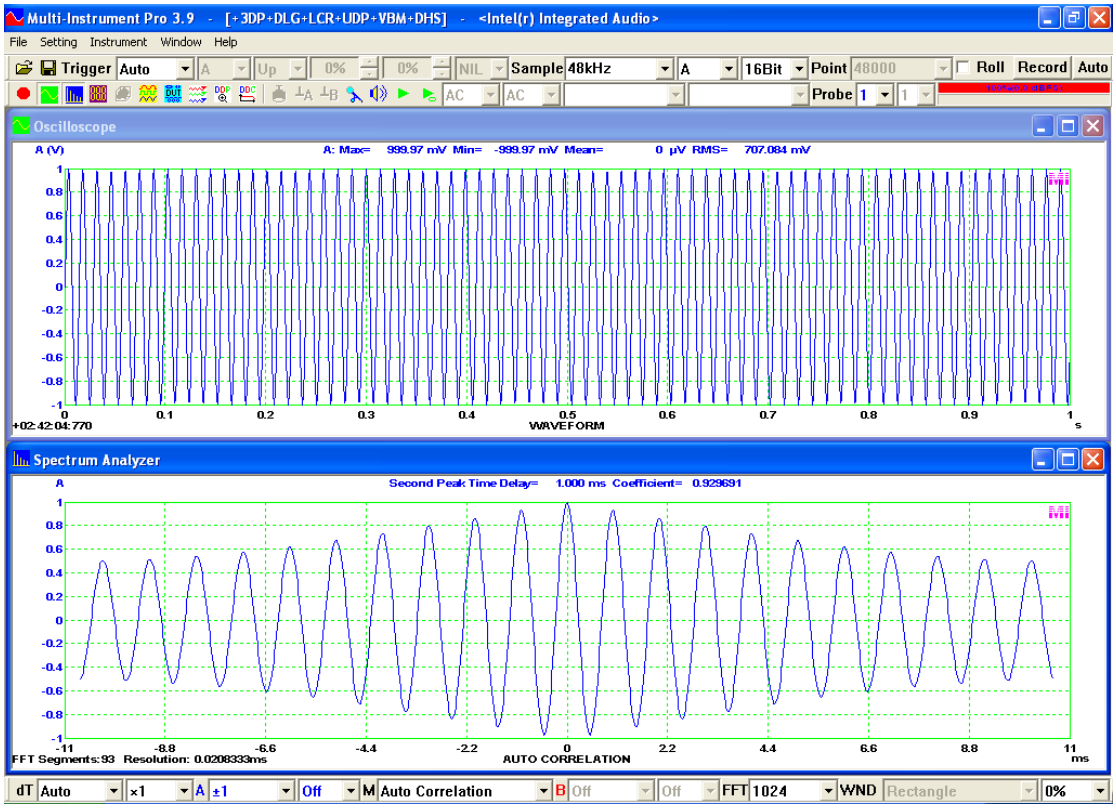
The following figure illustrates the linear auto correlation function of a 1 kHz sine wave with [Sampling Frequency]=48kHz, Record Length=48000, FFT Size=1024. It shows that the time interval between correlation function peaks is 1 ms, which is equal to the period of the signal. The second highest peak is located at [Time Delay]=1 ms with a coefficient of about 0.9.



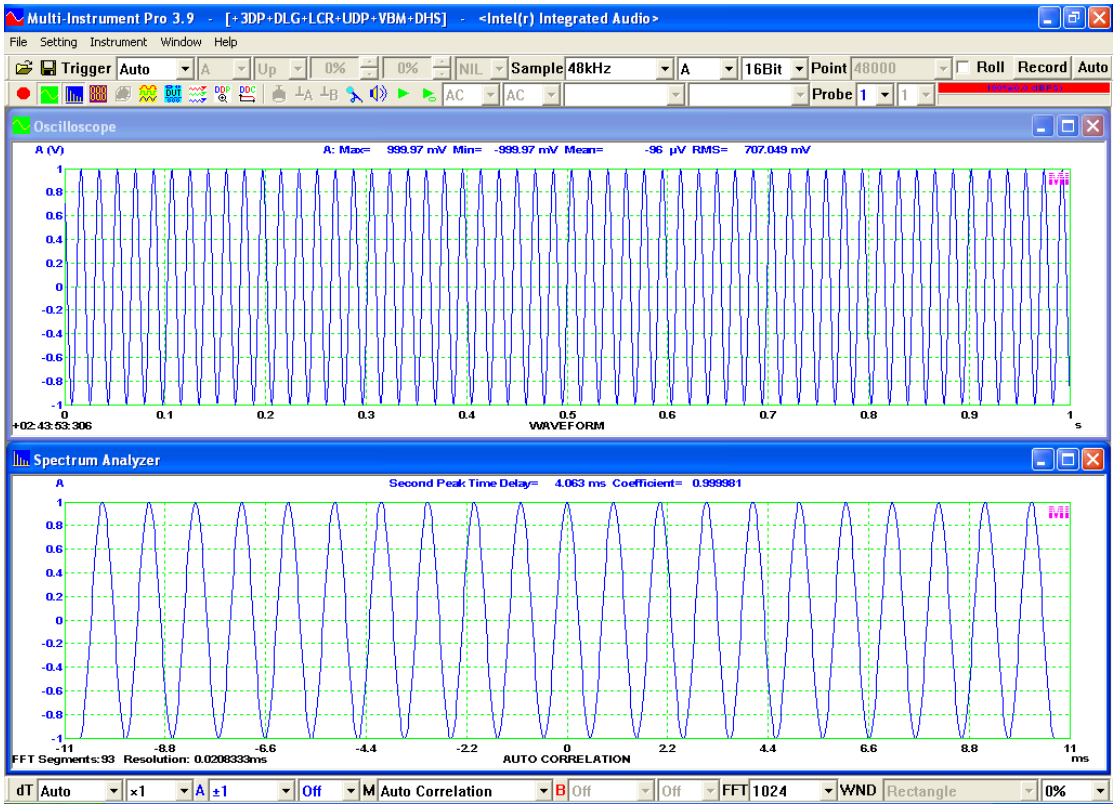
The auto correlation function here is computed indirectly using FFT in order to save time. The time delay range evaluated varies with the FFT size. To do a full-range evaluation, select a FFT size equal to or greater than twice of the Record Length in the Oscilloscope. This will yield 2×[Record Length]-1 auto correlation points centered at [Time delay]=0. The following figure has the same parameters as the above one except that the FFT size is changed to 131072 to evaluate the full-range of the linear auto correlation of the 48000-point data.



Linear auto correlation coefficient always decreases from 1 at [Time Delay]=0 to zero at the maximum time delays (i.e. ± time length of the data). Its tendency to dwindle with the time delay is due to the fact that linear correlation assumes that the data outside the time length are all zeros. Multi-Instrument supports circular auto correlation as well. It can be set via [Spectrum Analyzer Processing]>[Intra-Frame Processing]>"Circular Correlation". Unlike linear correlation, circular correlation assumes that the data outside the time length are just circular replicas of the data inside it. Therefore the circular auto correlation coefficient at the maximum time delays may not be zeros, as illustrated in the following figure. It shows the circular auto correlation function of a 1 kHz sine wave with [Sampling Frequency]=48kHz, [Record Length]=48000, [FFT Size]=1024.



In the above example, if the 1kHz signal frequency is changed to a "No Spectral Leakage" frequency (this concept will be introduced later) such as 984.375 Hz, then the circular auto correlation function becomes periodic with the same period as the original one (see figure below). It should be noted, however, if the FFT size is greater than the Record Length, then the software will pad [FFT Size]-[Record Length] zeros at the end of the Record Length, reducing the effect of the circular arrangement. When the FFT size is equal to or greater than twice of the Record Length, the circular correlation will become the same as the linear correlation.



The auto correlation function of a periodic signal has correlation peaks recurring at the same interval as the signal period. Non-periodic signals, such as noises, songs and music, usually only have one dominant auto correlation peak at zero time delay and very small values elsewhere (zeros for the case of a white-noise like signal). This feature has many applications. For example, auto correlation function can be used to measure the echo delay time embedded in a non-periodic signal. The following figure shows the linear auto correlation function of a 100-second song with an echo delay time of 1.3s. The deadband is set to be 50ms. It can be seen that the echo delay time is correctly detected at the second highest peak of the auto correlation function. Similarly, this method can be used to check whether the same sounds from two speakers arrive at a listening position at the same time.

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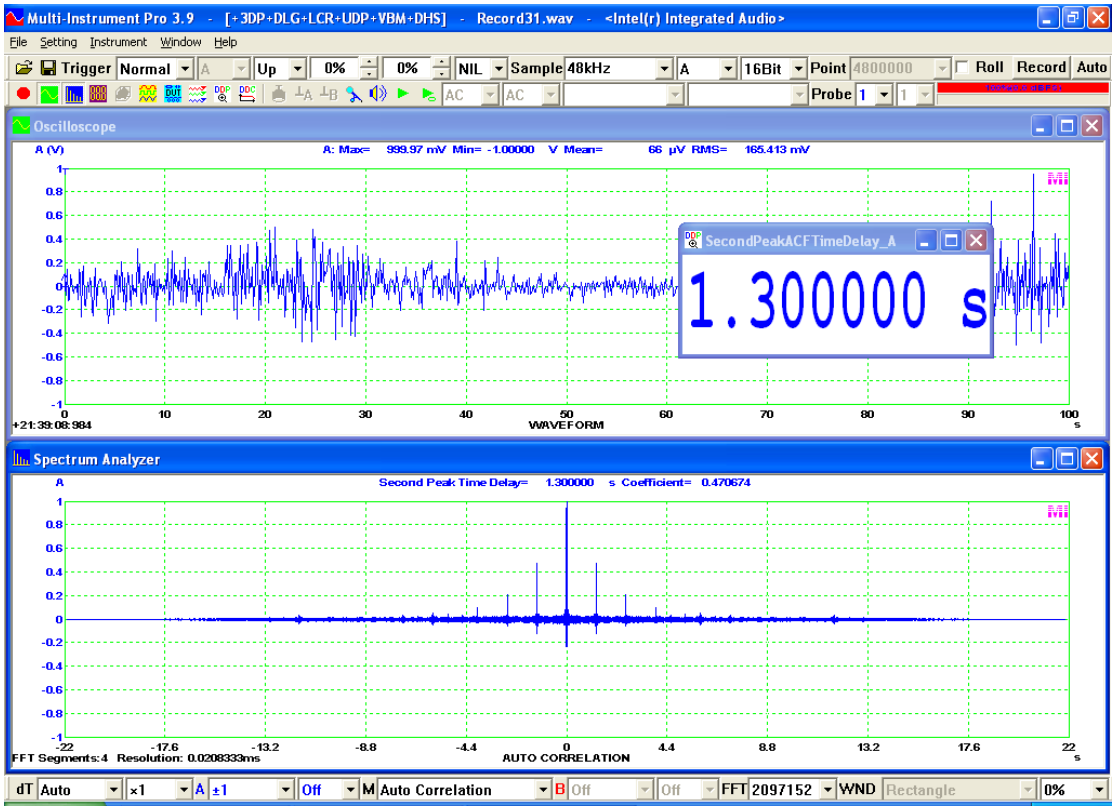
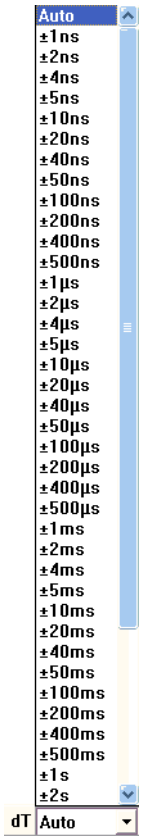


3.2.3.1 Time Delay Range (dT)



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This parameter is applicable to Auto or Cross Correlation Function Display. 49 options are available: Auto, ±1ns, ±2ns, ±4ns, ±5ns, ±10ns, ±20ns, ±40ns, ±50ns, ±100ns, ±200ns, ±400ns, ±500ns, ±1µs, ±2µs, ±4µs, ±5µs, ±10µs, ±20µs, ±40µs, ±50µs, ±100µs, ±200µs, ±400µs, ±500µs, ±1ms, ±2ms, ±4ms, ±5ms, ±10ms, ±20ms, ±40ms, ±50ms, ±100ms, ±200ms, ±400ms, ±500ms, ±1s, ±2s, ±4s, ±5s, ±10s, ±20s, ±40s, ±50s, ±100s, ±200s, ±400s, ±500s.

When “Auto” is selected, the Time Delay Range will be set by the software automatically based on the following formula:

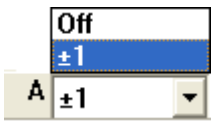
Time Delay Range = the nearest integer that is equal to or greater than the value of 0.5 × [FFT Size] / [Sampling Frequency]

Auto correlation and cross correlation functions are calculated using FFT in order to achieve a faster speed. The computable range of the Time Delay for a frame of data is thus ± 0.5 × [FFT size]/[Sampling Frequency].

3.2.3.2 Time Delay Multiplier

Same as Amplitude Spectrum display.

3.2.3.3 Channel A Display Range



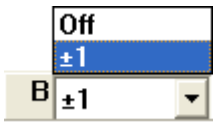
Only two options available: Off, ± 1.

When "Off" is selected, the data in Channel A will not be displayed.

3.2.3.4 Channel A Multiplier

Same as Amplitude Spectrum display.

3.2.3.5 Channel B Display Range



Only two options available: Off, ± 1.

When "Off" is selected, the data in Channel B will not be displayed.

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3.2.3.6 Channel B Multiplier

Same as Amplitude Spectrum display.

3.2.3.7 FFT Size

Same as Amplitude Spectrum display.

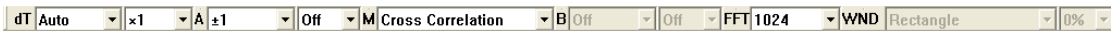
3.2.3.8 Window Function

Disabled.

3.2.3.9 Window Overlap Percentage

Same as Amplitude Spectrum display.

3.2.4 View Parameters for Cross Correlation Function Display



Selecting Cross Correlation in the above View Type selection box switches the Spectrum Analyzer to cross correlation display mode which shows the cross correlation coefficient (i.e. normalized cross correlation function) between Channels A and B vs time delay graph. The cross correlation function here is acyclic or linear by default. The time delay and cross correlation coefficient of the peak are also displayed as text above the graph. They can be accessed through these Derived Data Points (DDPs): PeakCCFTimeDelay_AB(s) and PeakCCFCoef_AB.

The following figure illustrates the linear cross correlation function of two 1 kHz sine wave signals, with the one in Channel A being 90º in phase ahead of the one in Channel B. It shows that the cross correlation peak is at -250 µs, which means that the signal in Channel A is 250 µs (i.e. 1/4 period) ahead of the signal in Channel B.

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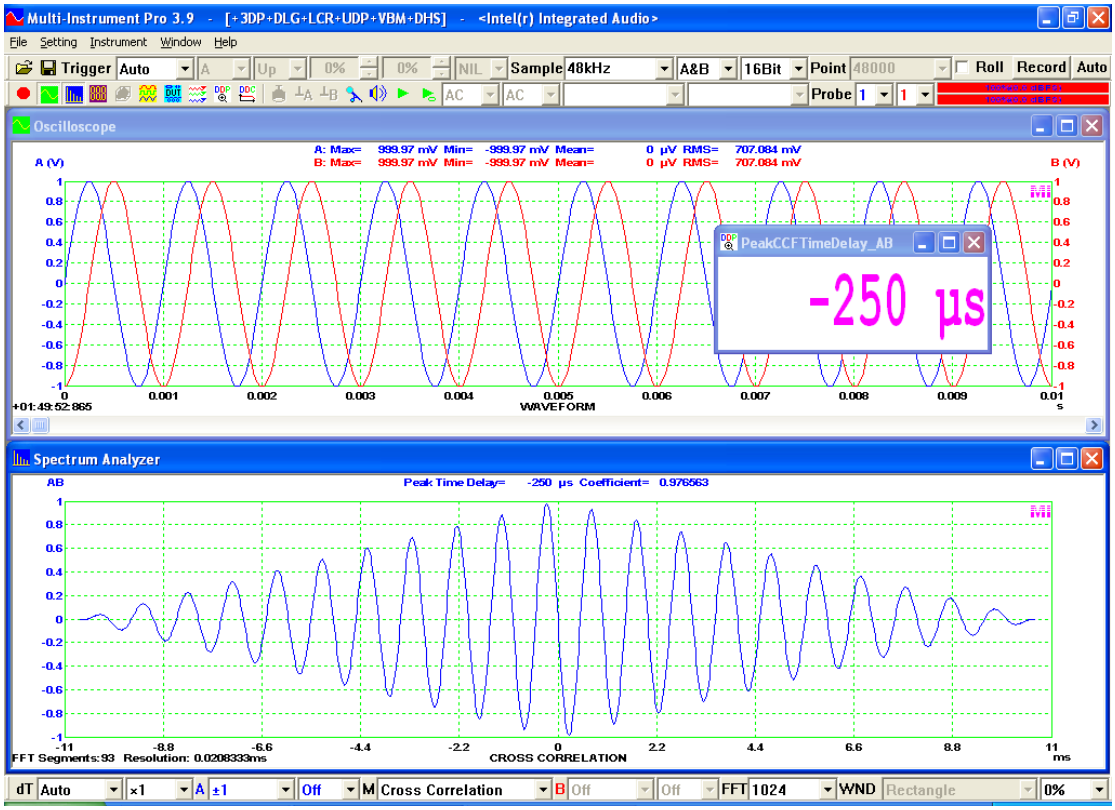
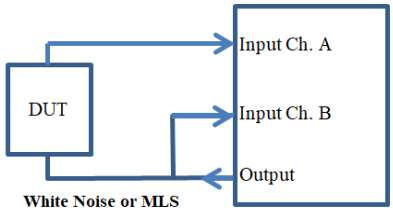


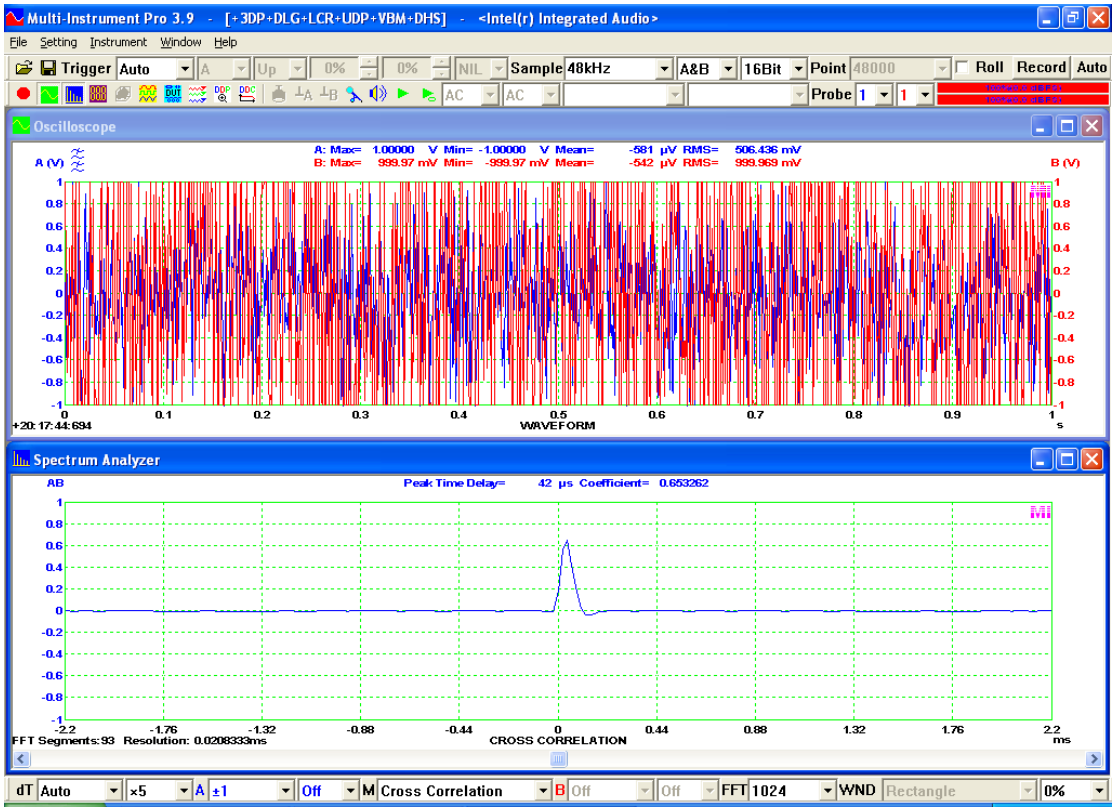
Similar to the case of auto correlation function, the cross correlation function here is also computed indirectly using FFT. The time delay range evaluated varies with the FFT size. To do a full-range evaluation, select a FFT size equal to or greater than twice of the Record Length in the Oscilloscope. This will yield 2×[Record Length]-1 cross correlation points centered at [Time delay]=0. Multi-Instrument supports circular cross correlation as well. It can be set via [Spectrum Analyzer Processing]>[Intra-Frame Processing]>"Circular Correlation". Again, it should be noted that, if the FFT size is greater than the Record Length, then the software will pad [FFT Size]-[Record Length] zeros at the end of the Record Length, reducing the effect of the circular arrangement. When the FFT size is equal to or greater than twice of the Record Length, the circular correlation will become the same as the linear correlation.

Cross correlation can be used to measure signal transmission delay as introduced previously in Section "Time Delay Removal" of Chapter "Oscilloscope". It can also be used to measure the impulse response of a LTI system using the same method (see figure below). To do that, the stimulus fed into the DUT must satisfy one requirement: its auto correlation function must be zero everywhere except at zero time delay. The stimuli that meet this requirement are white noise, MLS and unit impulse, etc. Given a white-noise-like stimulus, circular cross correlation between the output and the input yields the impulse response of the DUT. In Multi-Instrument, Channel B must be fed with the same stimulus sent to the DUT and Channel A must be fed with the response from the DUT. Note that the signal path from the output to the input Ch. B can be either a physically wired path or purely a software path (i.e. "iB=oA" loopback mode of the Signal Generator that will be introduced later). The former is a two-channel method while the latter is usually referred to as a single–channel method (it still requires two channel operations in the software). More generally, both signal paths,

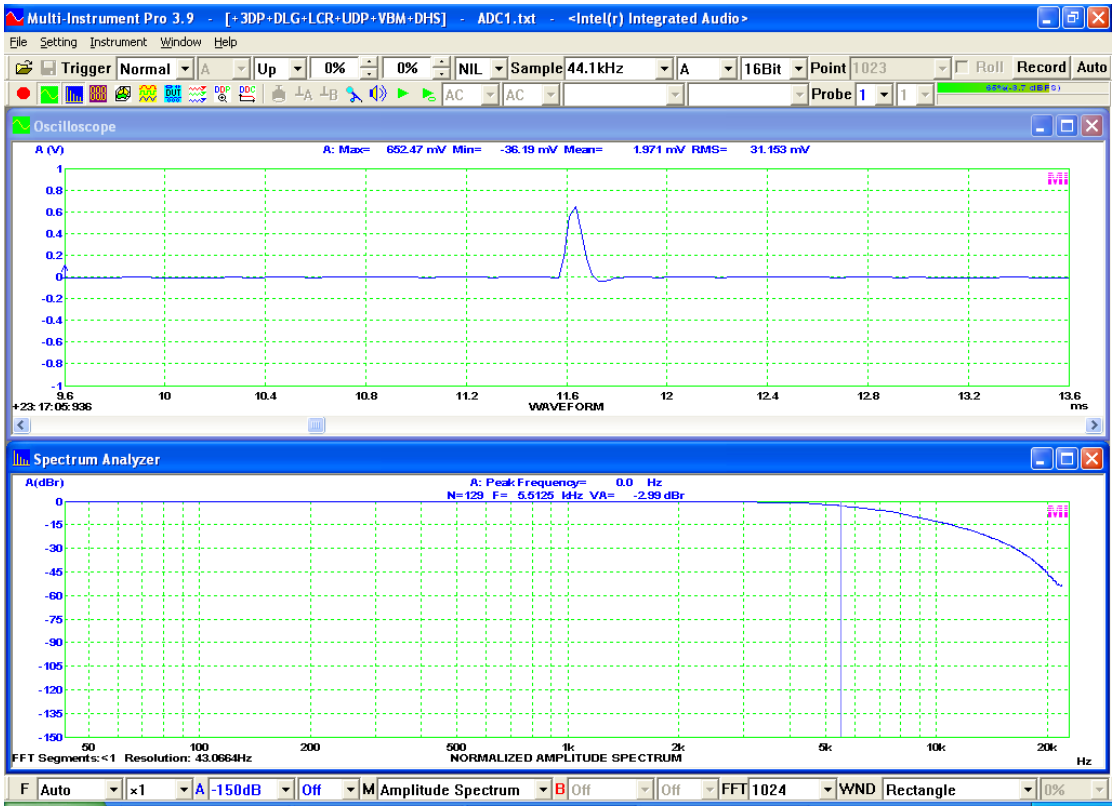
from the single output channel to the two input channels, can be purely software paths (i.e. "iA=oA, iB=oA" loopback mode of the Signal Generator that will be introduced later). This is referred to as a simulation mode. It can be used to measure the impulse response of, for example, a software digital filter, if the filter is applied to Channel A only.



The following figure shows the impulse response of a 5513 Hz second order Butterworth low pass filter, measured using circular cross correlation with a 65535-point MLS stimulus.



The impulse response (or circular cross correlation) data in the above figure can be exported to a TXT file by right clicking anywhere within the Spectrum Analyzer and select [Spectrum Analyzer Export]. It can then be imported via [File]>[Import] into the Oscilloscope in order to show its magnitude frequency response in the Spectrum Analyzer (see figure below). It can be seen from the figure that the -3dB cutoff frequency of the low pass filter is correctly measured to be 5513Hz.



Impulse response contains phase response information as well. However, it is noted that the impulse response loaded in the Oscilloscope does not start from time=0. As a result, its phase response graph will usually look messy due to phase wrapping (i.e. all phase values are constrained to the range -180º~180º). This issue can be resolved by manually removing the time delay in the impulse response. The data from the start of the impulse response to its end can be extracted and saved to a separate WAV file via [File]>[Extract]. Then load that WAV file via [File]>[Open] and switch the Spectrum Analyzer to phase spectrum display to show the phase response (see figure below). It can be seen from the figure that the phase at the cutoff frequency 5513Hz is correctly measured to be about 90º.

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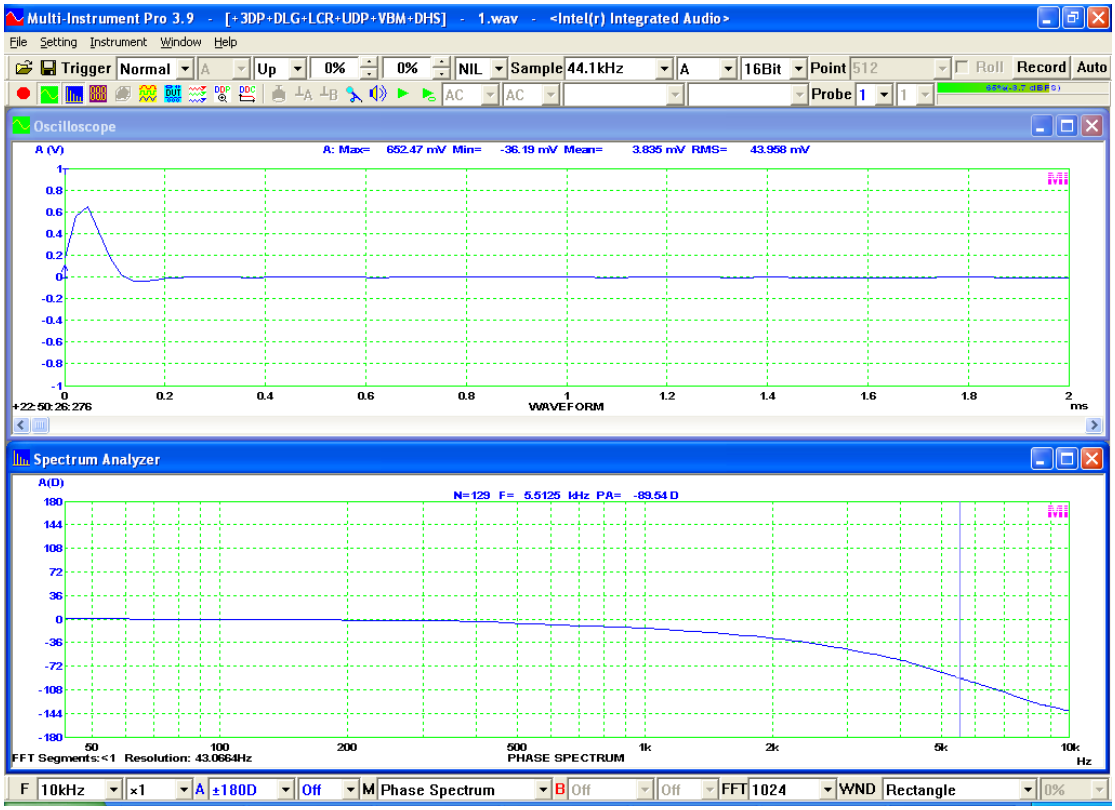
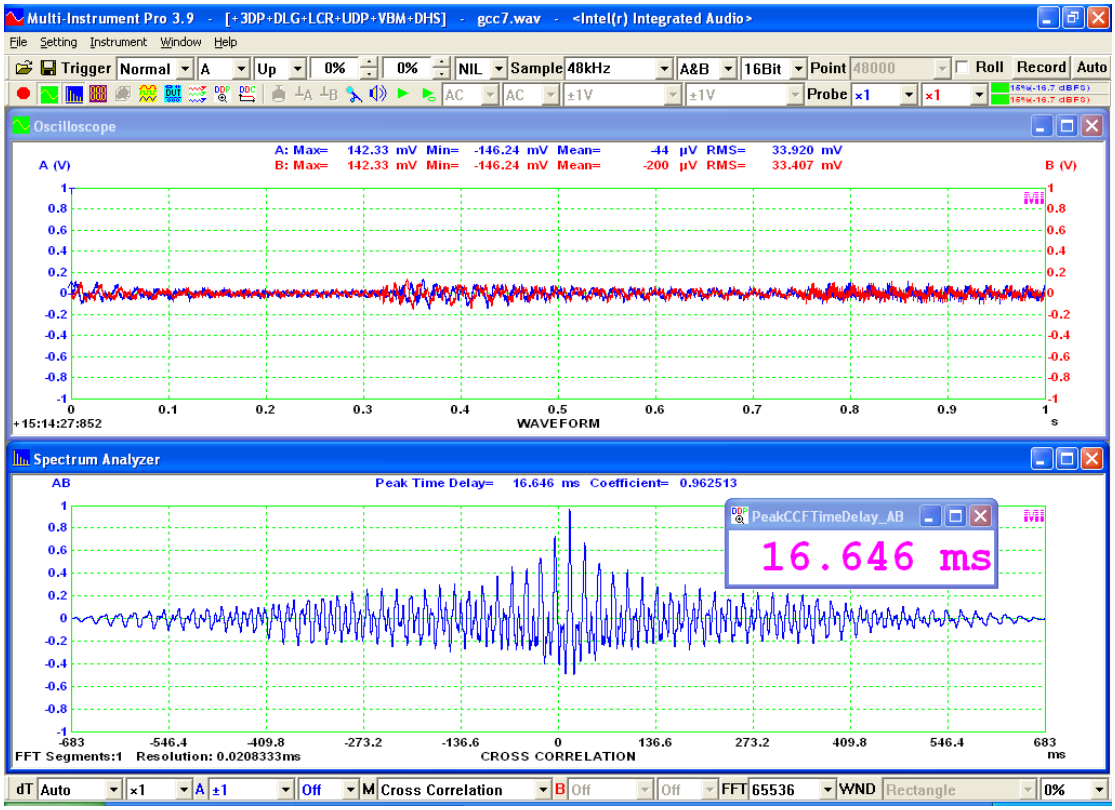
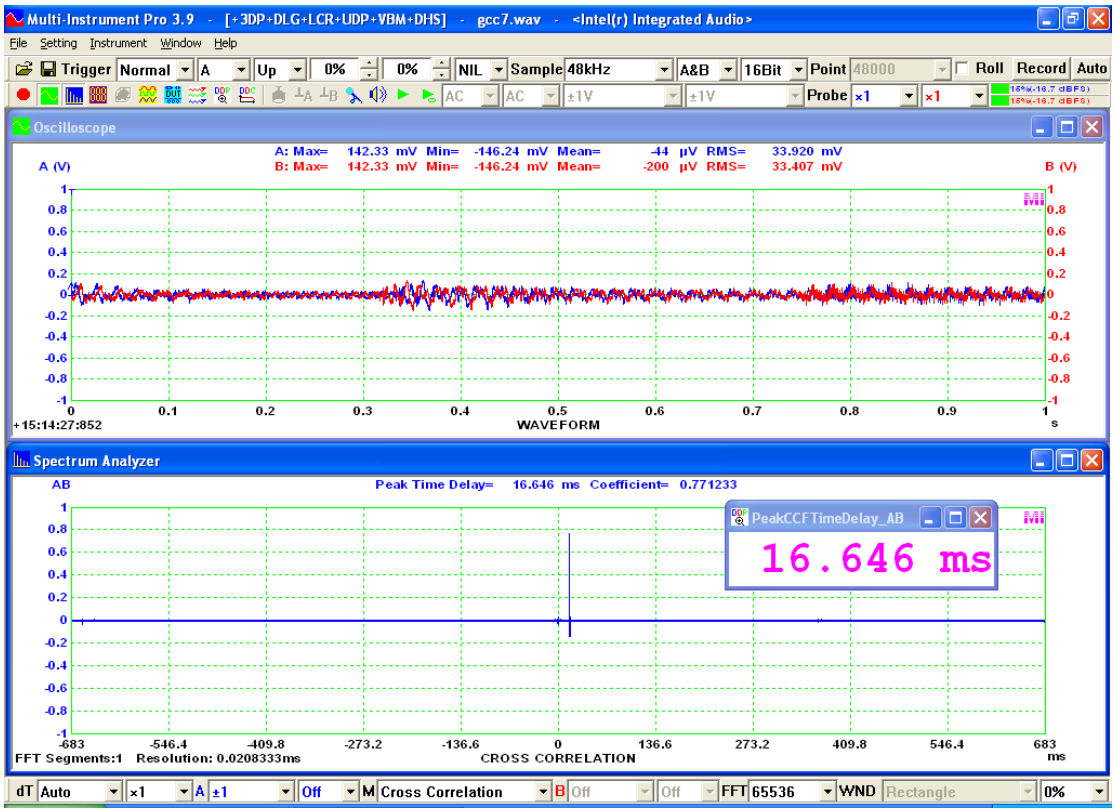


Impulse response can also be used to derive other information such as reverberation time and speech transmission index. This has been introduced previously. Some sample panel setting files for impulse response measurements using cross correlation are provide in the PSF subdirectory of the software.



When cross correlation is used to measure signal transmission delay, the signal needs to be "white" enough to ensure a single prominent correlation peak right at the time delay. This condition may not be satisfied if the signal is not generated intentionally by the testing equipment or when there are strong background noise and reverberation. The above figure shows a music signal (from the same sound source) captured by two microphones. Cross correlation is used to find the Time Difference of Arrival (TDOA) between them. It can be seen that the "true" cross correlation peak is surrounded by many interfering sub-peaks. These sub-peaks are caused by the periodic components of the music signal. They may outweigh the "true" correlation peak under some conditions and lead to a wrong TDOA measurement. Generalized Cross Correlation (GCC) can be used to improve the signal-to-noise ratio in this type of measurements. The most commonly used GCC technique is called GCC Phase Transform (GCC-PHAT). It whitens the cross power spectrum in the frequency domain before the conversion back to the time domain to get the cross correlation. In Multi-Instrument, the GCC flag can be set via [Spectrum Analyzer Processing]>[Intra-Frame Processing]>"Generalized Cross Correlation". The following figure shows the generalized cross correlation of the signal in the above example. A significant improvement on the measurement SNR can be observed. This method is often used with a microphone array to locate a sound source.



Generalized Cross Correlation greatly broadens the range of test signals that can be used for time delay measurement. However, it still cannot measure the time delay of a periodic signal such as a sinewave. In fact, when the test signal contains strong periodic components, such as a slow piano solo, spurious cross correlation peaks often show up and may lead to wrong peak identification. Multi-Instrument features a uniquely improved GCC-PHAT algorithm which is able to identify the correct cross correlation peak even under such challenging conditions.

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3.2.4.1 Time Delay Range (dT)

Same as Auto Correlation Function.

3.2.4.2 Time Delay Multiplier

Same as Auto Correlation Function.

3.2.4.3 Channel A&B Display Range

Same as Channel A Display Range in Auto Correlation Function.

3.2.4.4 Channel A&B Multiplier

Same as Channel A Multiplier in Amplitude Spectrum.

3.2.4.5 Channel B Display Range

Not applicable and disabled.

3.2.4.6 Channel B Multiplier

Not applicable and disabled.

3.2.4.7 FFT Size

Same as Amplitude Spectrum.

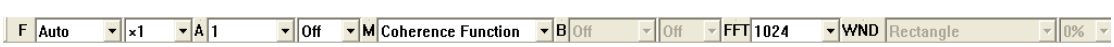
3.2.4.8 Window Function

Disabled.

3.2.4.9 Window Overlap Percentage

Same as Amplitude Spectrum.

3.2.5 View Parameters for Coherence / Non-Coherence Function Display



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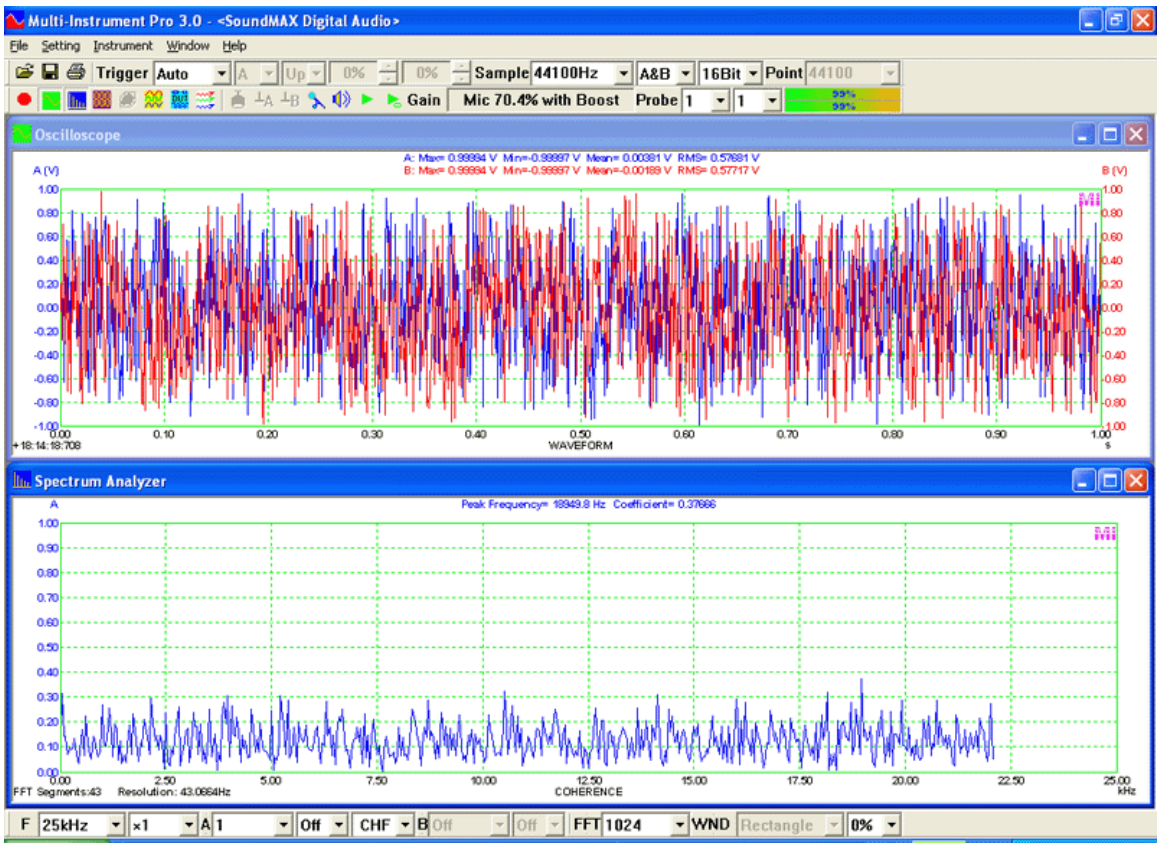
Selecting Coherence Function in the above View Type selection box switches the Spectrum Analyzer to coherence function display mode which shows the coherence coefficient between Channels A and B vs frequency graph. The coherence function is defined as the ratio of the magnitude of the cross-power spectrum between the input and output signals of a system to the square root of the product of their auto-power spectra. It has a value between 0 and 1. If the signals are ergodic and the system is linear, the coherence function can be used to estimate the causality between the input and output signals. Its square value reflects the percentage of the output power that is contributed by the input. Thus the coherence function is often used to examine the quality of a measured transfer function. The peak coherence coefficient and its corresponding frequency will be shown as text above the graph. They can also be accessed through these Derived Data Points (DDPs): PeakCHFCoef_AB and PeakCHFFreq_AB(Hz).

The following figure illustrates the coherence function of two independent white noise signals. It shows that the coherence value at each frequency is well below 1 after 42 times of intra-frame averaging (Number of data points in time domain: 44100, FFT size: 1024, number of FFT segments: 44100/1024 = 43). The values should converge to zero at all frequencies if an infinite number of times of intra-frame averaging is performed. Note that if there is only one FFT segment (i.e. no intra-frame averaging), the coherence values will always be 1 even if the two signals are fully independent and un-correlated. Thus, to get a valid coherence function, at least one time of intra-frame averaging must be performed. In other words, the Record Length of the Oscilloscope must be at least twice the FFT size of the Spectrum Analyzer. The software will show a warning message inside the Spectrum Analyzer if this requirement is not met.



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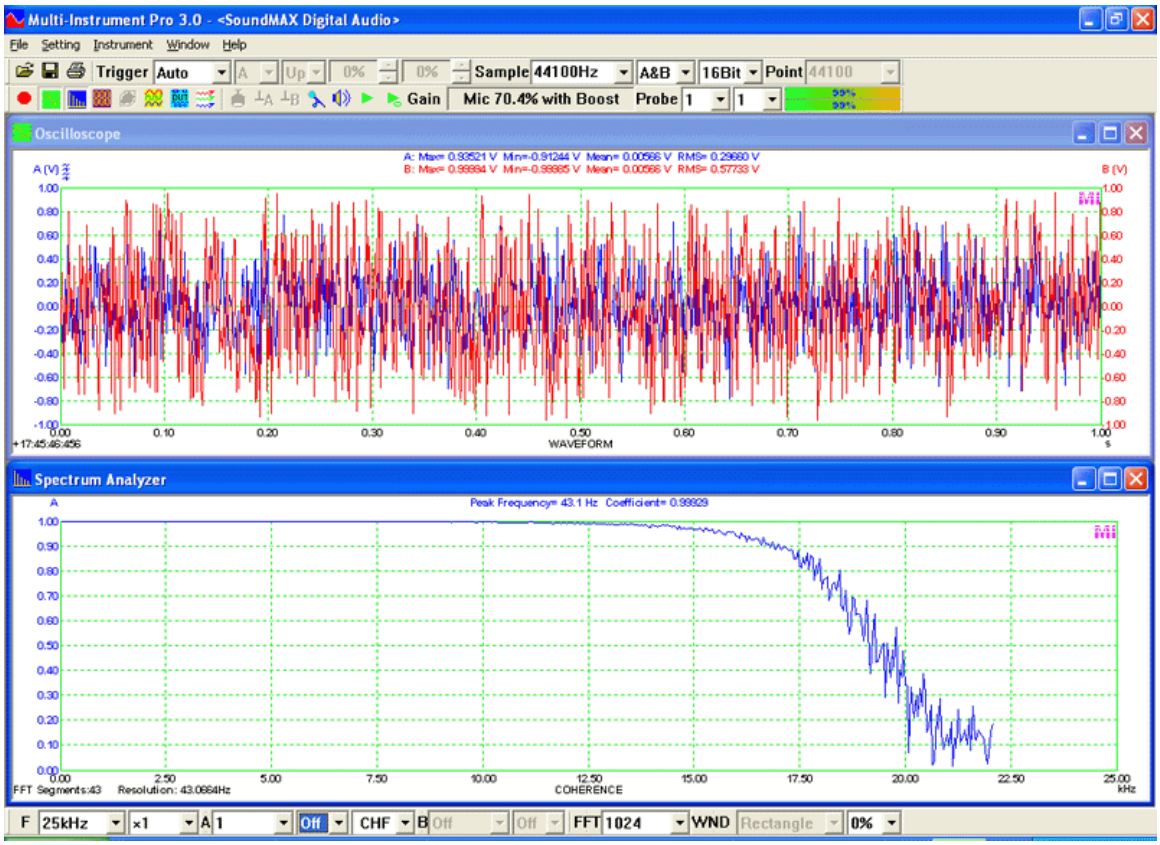
The following figure illustrates the coherence function of a white noise stimulus (in Channel B) and the corresponding response (in Channel A) from a 5513 Hz second order Butterworth low pass filter. The figure shows that the coherence value maintains at nearly 1 from 0 Hz to about 15 kHz and then starts to drop towards zero quickly. The reason why the high frequencies tend to be un-correlated is that their energy in the response signal is greatly suppressed by the low pass filter, resulting in a very poor signal-to-noise ratio at these frequencies.

The coherence function is often used in conjunction with the transfer function as an indication of the quality of the transfer function measurement. It indicates how much of the response energy is correlated to the stimulus energy.



3.2.5.1 Frequency Range (F)

Same as Amplitude Spectrum.

3.2.5.2 Frequency Multiplier

Same as Amplitude Spectrum.

3.2.5.3 Channel A&B Display Range

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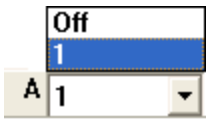


Only two options available: Off, 1.

When "Off" is selected, the data in Channel A will not be displayed.

3.2.5.4 Channel A&B Multiplier

Same as Channel A Multiplier in Amplitude Spectrum.

3.2.5.5 Channel B Display Range

Not applicable and disabled.

3.2.5.6 Channel B Multiplier

Not applicable and disabled.

3.2.5.7 FFT Size

Same as Amplitude Spectrum.

3.2.5.8 Window Function

Same as Amplitude Spectrum.

3.2.5.9 Window Overlap Percentage

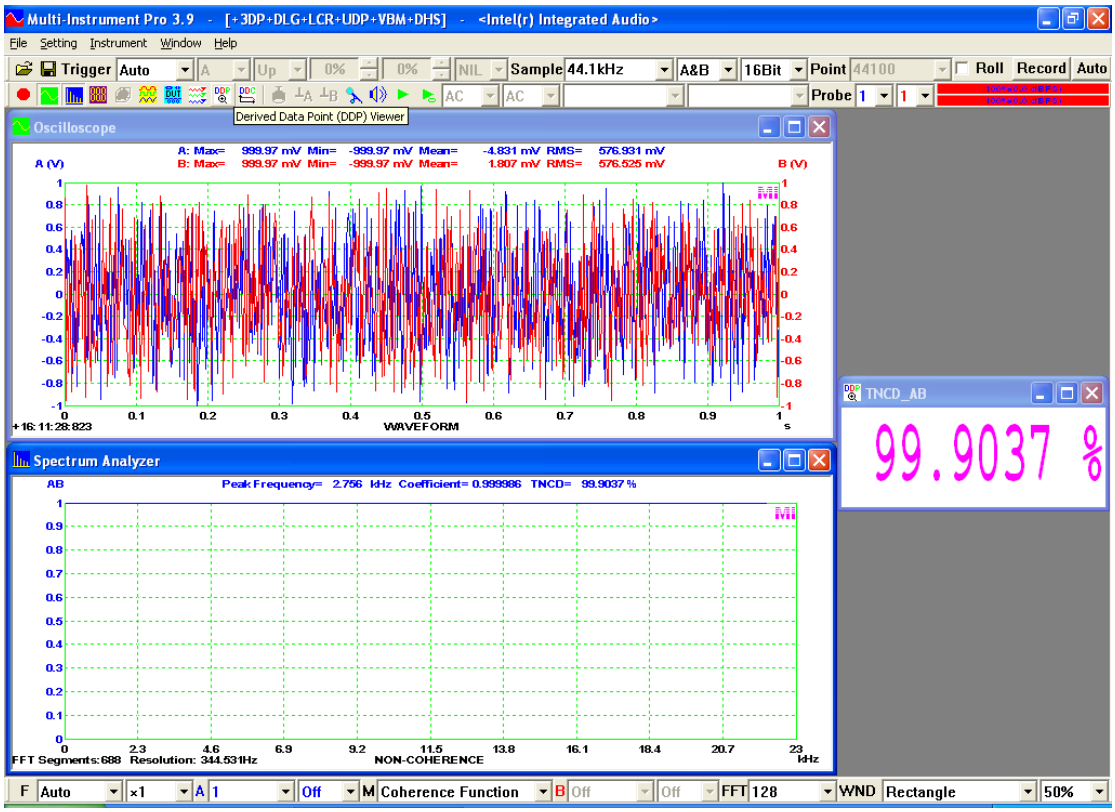
Same as Amplitude Spectrum.

3.2.5.10 Non-Coherence Function and Total Non-Coherent Distortion and Noise

The coherence function can be seen as the square root of the ratio of coherent output power to output power against frequency. Coherent output power is the part of output power that is linearly related to the input. The remainder of the output power is called non-coherent output power, which is caused by all kinds of non-linear distortions (e.g. THD, IMD, etc..) and noise. The non-coherence function is thus defined as the square root of the ratio of non-coherent output power to output power against frequency. It equals to the square root of (1-[Coherent Function]^2). To switch to non-coherence function display mode, right click anywhere within the Spectrum Analyzer window and select [Spectrum Analyzer Processing] in the pop-up menu, then tick “Non Coherence” checkbox.

Non-Coherence function can be used to indicate the variation of the non-coherent distortion and noise with frequency. The non-coherent distortion takes into account all kinds of non-linear distortions. Integrating the non-coherent output power against frequency gives the total non-coherent output power. Total Non-Coherent Distortion and noise (TNCD) is a single value defined as the square root of the ratio of the total non-coherent output power to the total output power. It is displayed in percentage (%). This value will be shown on the top of the non-coherence coefficient vs frequency graph. It can also be accessed through the Derived Data Point (DDP): TNCD_AB(%). It is possible to specify a frequency range in which the TNCD is calculated. The default range is from 20 Hz to 20 kHz.

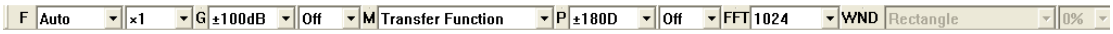
The following figure illustrates the non-coherence function of two independent white noise signals. It shows that the non-coherence coefficient at each frequency approaches 1 after 687 times of intra-frame averaging (Number of data points in time domain: 44100, FFT size: 128, window overlap: 50%, number of FFT segments: 688). The values should converge to 1 at all frequencies and the TNCD should be 100% if an infinite number of times of intra-frame averaging are performed. Note that if there is only one FFT segment (i.e. no intra-frame averaging), the non-coherence coefficients will always be zero even if the two signals are completely independent and un-correlated. Thus, to get a valid non-coherence function, at least one time of intra-frame averaging must be performed. In other words, the Record Length of the Oscilloscope must be at least twice the FFT size of the Spectrum Analyzer. However, if the non-coherence function and TNCD are used to quantify the non-coherent distortion + noise, then a much greater number of times of intra-frame averaging should be performed, in order to minimize its artifact on the measurement accuracy and allow a small value of non-coherent distortion + noise to show up. This requires a longer record length of the Oscilloscope.



Unlike those traditional non-linear distortion measurements, such as THD, IMD, and TD+N, whereby a single tone, dual-tone, or multi-tone stimulus is used as the input signal, non-coherent distortion measurement can use any broad-band stimulus such as white noise, pink noise, multitones, broadband music, or broadband speech.

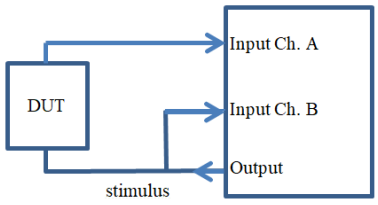
For Coherence and Non-Coherence Function measurement using the software, Channel B must be fed with the same stimulus sent to the Device Under Test (DUT) and Channel A must be fed with the response from the DUT (see figure in the next section).

3.2.6 View Parameters for Transfer Function Display / Impedance Analyzer



Selecting Transfer Function in the above View Type selection box switches the Spectrum Analyzer to transfer function display mode under which the system's gain & phase shift vs frequency graph is shown. Transfer function is a convenient representation of a Linear Time Invariant (LTI) system. It describes the relationship between the system's input and output. Frequency response is a transfer function expressed in frequency domain. It is a measure of magnitude and phase of the output as a function of frequency, in comparison to the input. Frequency response includes magnitude frequency response and phase frequency response. A graph that shows both are called Bode plot or gain and phase plot.

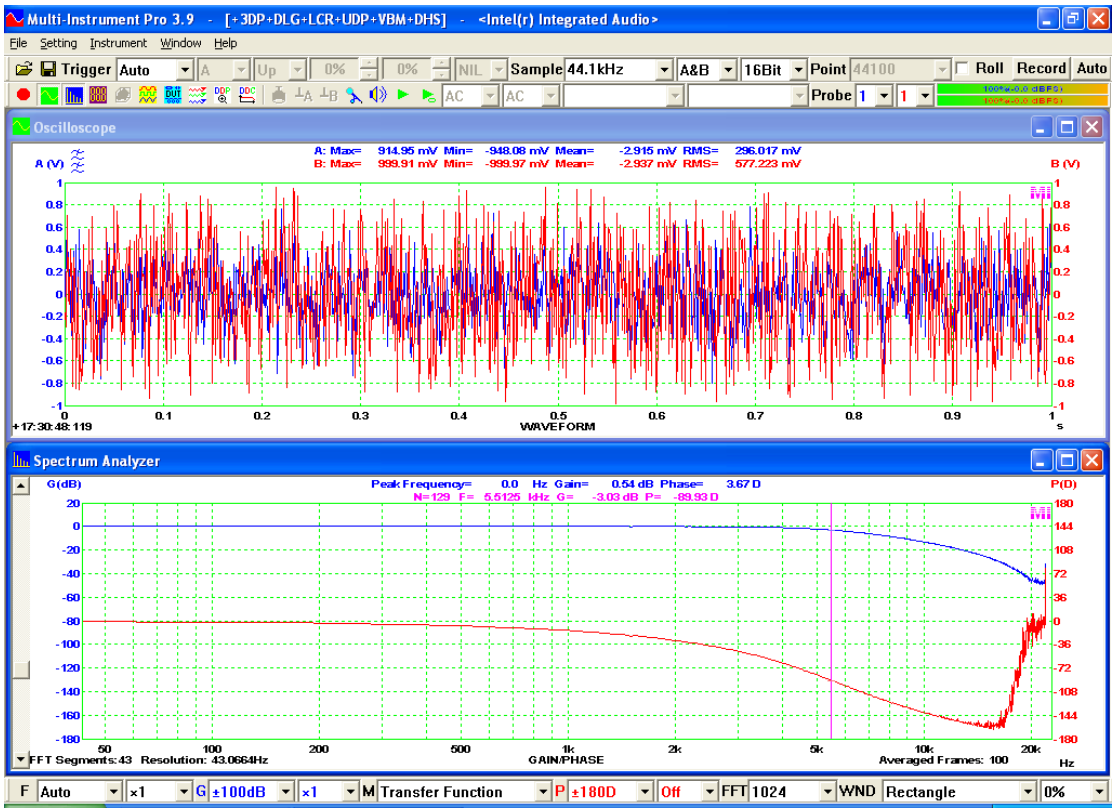
For transfer function measurement using the software, Channel B must be fed with the same stimulus sent to the Device Under Test (DUT) and Channel A must be fed with the response from the DUT (see figure below). To achieve a good measurement accuracy, significant stimulus energy must be present in the frequency range of interest. Commonly used test signals are chirp signal and white noise as well as multitones capable of simultaneously stimulating all FFT frequency bands within the frequency range of interest. Transfer function measurement requires that the two input channels themselves have the same frequency response in the frequency range of interest. This can be verified by feeding the stimulus to both input channels directly and measuring the transfer function. The measured gain and phase shift in the frequency range of interest should be very close to 0 dB and 0°. If not, you can generate a gain and phase compensation file and load it to compensate the disparity between the two input channels. This will be described later in this chapter.



The following figure illustrates the transfer function of the aforementioned 5513 Hz second order Butterworth low pass filter, measured using a white noise stimulus, with the stimulus data stored in Channel B and the response data stored in Channel A. 100 oscilloscope frames were inter-frame averaged and 43 intra-frame averages were

performed in each oscilloscope frame. The figure shows that the gain maintains at nearly 0 dB from 0 Hz to about 5513 Hz, and then start to drop very quickly, meanwhile the phase changes gradually from 0º at 0 Hz towards –180º as the frequency goes to infinity. The cursor reader indicates that at 5512.5 Hz (i.e. around the cutoff frequency), the gain is about –3.03 dB and the phase is about -89.93º.

As indicated in the previous section, the coherence function of this measurement drops very quickly towards zero after 15 kHz, thus the transfer function results beyond 15 kHz may not be trustable because the response and stimulus are no more closely correlated due to the poor signal-to-noise ratio.



The following figure shows a first order RC low pass filter. The resistor R is 200Ω and the capacitor C is 2.2μF. Its cutoff frequency can be calculated as: $1/(2\pi RC)=362Hz$. Its transfer function was measured using a sound card. The stimulus was a white noise. 143 oscilloscope frames were inter-frame averaged and two intra-frame averages were performed in each oscilloscope frame. The measured bode plot is show as follows. The measured cutoff frequency is about 360Hz and the measured phase shift changes from 0° to -90° as frequency increases.

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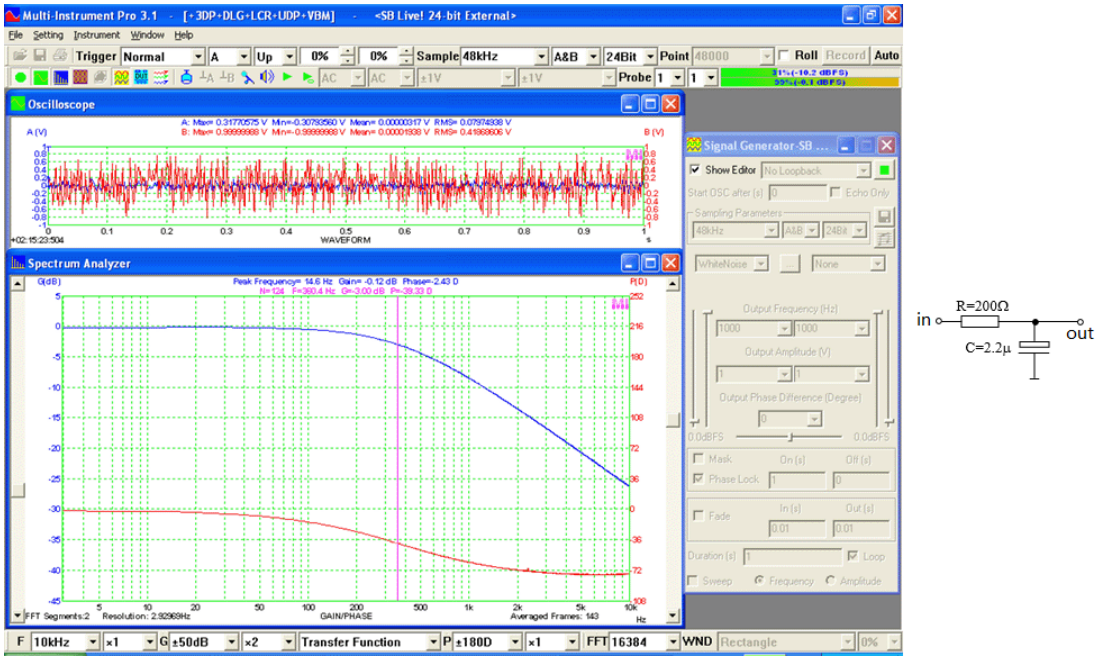


The following DDPs are generated by the transfer function display mode:

(1) PeakGainFreq_AB(Hz): Frequency at the peak gain
(2) PeakGainValue_AB(dB): Peak gain
(3) PeakGainPhase_AB(D): Phase at the peak gain
(4) GainAtGeneratedFreq(dB): Gain at the frequency generated by Channel A of the Signal Generator
(5) PhaseAtGeneratedFreq(D): Phase at the frequency generated by Channel A of the Signal Generator
(6) InBandMinF_A(Hz): Frequency at which the minimum gain is located in the specified frequency band
(7) InBandMinRMS_A(EU): Minimum gain in the specified frequency band, its unit will be updated to dB
(8) InBandMaxF_A(Hz): Frequency at which the maximum gain is located in the specified frequency band
(9) InBandMaxRMS_A(EU): Maximum gain in the specified frequency band, its unit will be updated to dB
(10) InBandMinF_B(Hz): Frequency at which the minimum gain is located in the specified frequency band
(11) InBandMinRMS_B(EU): Phase at the minimum gain in the specified frequency band, its unit will be updated to D
(12) InBandMaxF_B(Hz): Frequency at which the maximum gain is located in the specified frequency band
(13) InBandMaxRMS_B(EU): Phase at the maximum gain in the specified frequency band, its unit will be updated to D

The above (1), (2) and (3) are also displayed in the upper part of the Spectrum Analyzer. (4) and (5) are usually used in transfer function measurements using frequency stepped sine stimulations through Device Test Plan. If the signals acquired in the two input channels are not time aligned, and it is required to remove the time delay between them before the transfer function is calculated, then please refer to the

Time Delay Removal method introduced in the previous chapter. Some sample panel setting files for Bode plot measurements are provided in the PSF subdirectory of the software.

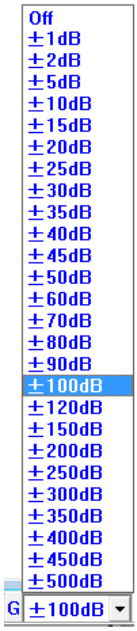
3.2.6.1 Frequency Range (F)

Same as Amplitude Spectrum.

3.2.6.2 Frequency Multiplier

Same as Amplitude Spectrum.

3.2.6.3 Channel A Display Range



27 options available: Off, ±1dB, ±2dB, ±5dB, ±10dB, ±15dB, ±20dB, ±25dB, ±30dB, ±35dB, ±40dB, ±45dB, ±50dB, ±60dB, ±70dB, ±80dB, ±90dB, ±100dB, ±120dB, ±150dB, ±200dB, ±250dB, ±300dB, ±350dB, ±400dB, ±450dB, ±500dB.

When "Off" is selected, the data in Channel A will not be displayed.

3.2.6.4 Channel A Multiplier

Same as Amplitude Spectrum.

3.2.6.5 Channel B Display Range

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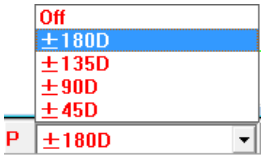

Five options available: Off, ± 180D (i.e. ± 180 degree), 135D, 90D, 45D.

When "Off" is selected, the data in Channel B will not be displayed.

3.2.6.6 Channel B Multiplier

Same as Amplitude Spectrum.

3.2.6.7 FFT Size

Same as Amplitude Spectrum.

3.2.6.8 Window Function

Same as Amplitude Spectrum.

3.2.6.9 Window Overlap Percentage

Same as Amplitude Spectrum.

3.2.6.10 Impedance Analyzer Mode

Under the Transfer Function display mode, to switch to Impedance Analyzer mode, right click anywhere within the Spectrum Analyzer window and select [Spectrum Analyzer Y Scale] in the pop-up menu, then tick “Convert to Impedance” checkbox (see figure below).

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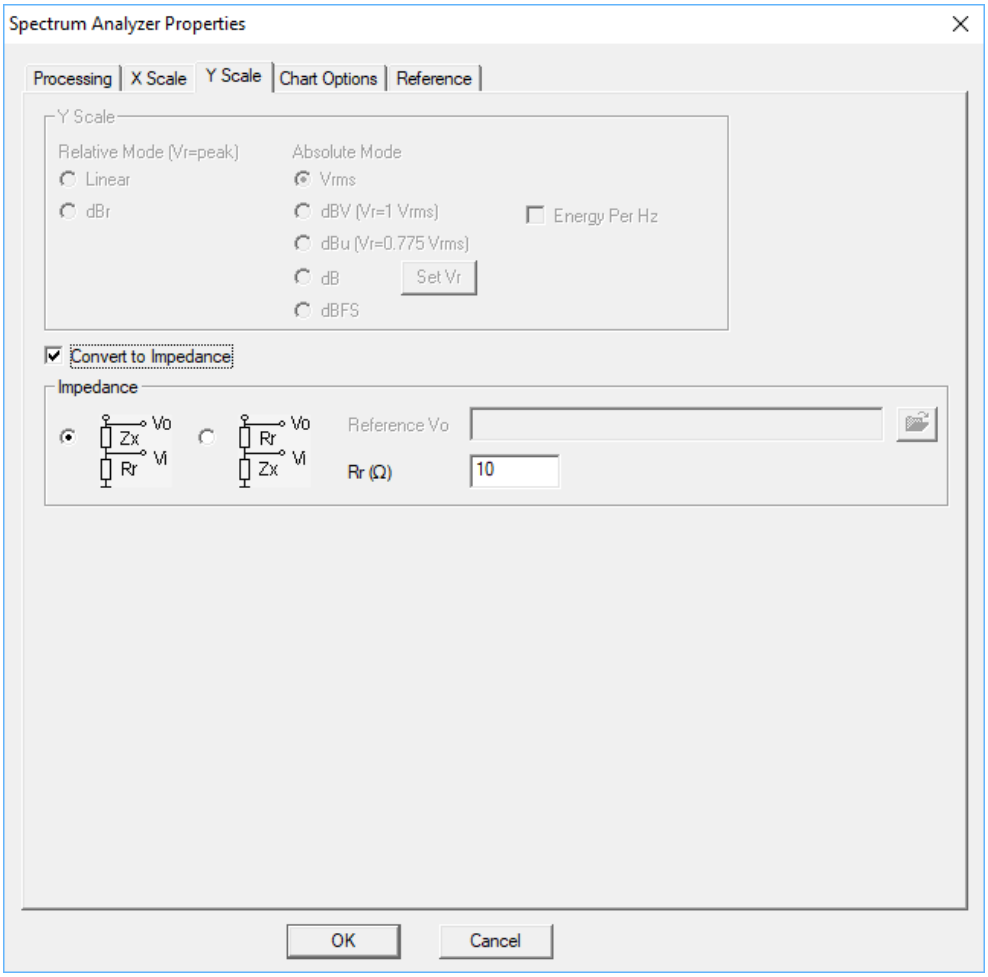
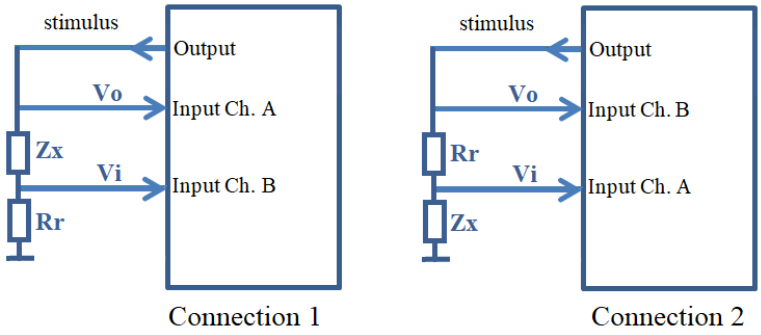


Impedance is a two-dimensional vector quantity consisting of two independent scalars: resistance and reactance. It is represented as a complex quantity Z. In polar form, the impedance is represented by its magnitude and phase angle, both of which are functions of frequency. The impedance magnitude and phase vs frequency plot is thus quite similar to the gain and phase vs frequency plot of a LTI system. So does the measurement method. Two connection methods are supported in the software (see figure below).



where Zx is the impedance to be measured, Rr the reference resistor, Vo the output voltage from the signal generator and Vi the input voltage fed by the voltage divider formed by Zx and Rr. Both Vo and Vi are ground referenced, so measuring devices with either ground referenced inputs or differential inputs can be used. The input

impedance of the measuring device must be much higher than Zx and Rr so that their effect on the measurement accuracy can be ignored. The output impedance and overall frequency response of the measuring device do not affect the measurement accuracy directly thanks to the dual-channel measurement method, but it is still recommended for the measuring device to have a low output impedance and a flat overall frequency response. The dual-channel measurement method requires the two input channels to have minimum gain and phase differences within the frequency range of interest. Unlike the transfer function measurement, these differences cannot be compensated in the software.

Rr is an external reference resistor. You need to find the resistor and make the connection by yourself. It is recommended to have Rr's value comparable to that of Zx. The actual value of Rr should be entered into the Rr(Ω) edit box in [Spectrum Analyzer Y Scale] dialog box. The actual connection used also needs to be selected there. Unlike the single-channel impedance measurement method under Amplitude Spectrum display mode, the dual-channel method does not require a Reference Vo file as Vo is already measured simultaneously with Vi. In Connection 1, Zx = Rr × (Vo-Vi) / Vi while in Connection 2, Zx = Rr × Vi / (Vo-Vi). To achieve a good measurement accuracy, significant stimulus energy must be present in the frequency range of interest. Commonly used signals are chirp signal (frequency swept sine) and white noise as well as multitones capable of simultaneously stimulating all FFT frequency bands in the frequency range of interest. Various average methods can be used to smoothen the result.

Under the Impedance Analyzer mode, Channel A Display Range will be updated to display impedance in Ω. The "Auto" option of the display range is about 5×Rr. The following DDPs are generated by the Impedance Analyzer mode:

(1) PeakGainFreq_AB(Hz): Frequency at the peak impedance
(2) PeakGainValue_AB(dB): Peak impedance value, its unit will be updated to Ω
(3) PeakGainPhase_AB(D): Phase at the peak impedance
(4) GainAtGeneratedFreq(dB): Impedance at the frequency generated by Channel A of the Signal Generator, its unit will be updated to Ω
(5) PhaseAtGeneratedFreq(D): Phase at the frequency generated by Channel A of the Signal Generator
(6) InBandMinF_A(Hz): Frequency at which the minimum impedance is located in the specified frequency band
(7) InBandMinRMS_A(EU): Minimum impedance in the specified frequency band
(8) InBandMaxF_A(Hz): Frequency at which the maximum impedance is located in the specified frequency band
(9) InBandMaxRMS_A(EU): Maximum impedance in the specified frequency band
(10) InBandMinF_B(Hz): Frequency at which the minimum impedance is located in the specified frequency band
(11) InBandMinRMS_B(EU): Phase at the minimum impedance in the specified frequency band, its unit will be updated to D
(12) InBandMaxF_B(Hz): Frequency at which the maximum impedance is located in the specified frequency band
(13) InBandMaxRMS_B(EU): Phase at the maximum impedance in the specified frequency band, its unit will be updated to D

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The above (1), (2) and (3) are also displayed in the upper part of the Spectrum Analyzer. (4) and (5) are usually used in impedance measurements using frequency stepped sine stimulations through Device Test Plan.

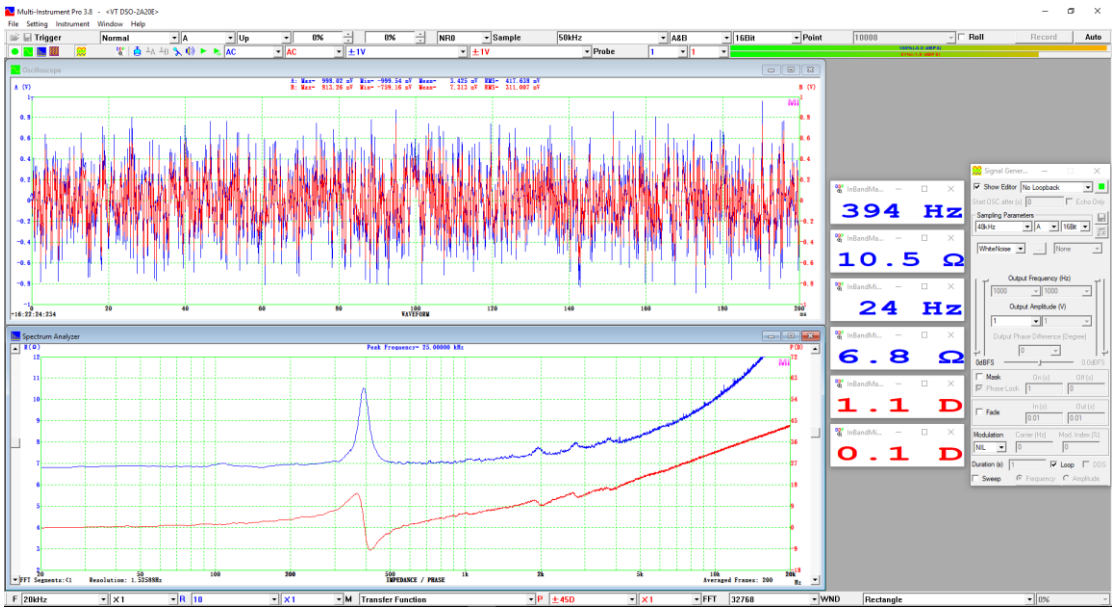
The following examples use different stimuli and configurations to measure a 2.5" 8Ω speaker's impedance with a 25Ω reference resistor.

(1) White Noise

Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 200 ms (10000 samples)
Trigger Mode: Normal, Trigger Level: 0%

Spectrum Analyzer:
FFT size: 32768, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: 200-frame linear average
X Scale: Log, Y Scale: Convert to Impedance
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

Signal Generator:
DAC Device: Sound Card Headphone Output, Duration: Loop
Waveform: White Noise, Sampling Frequency: 48 kHz



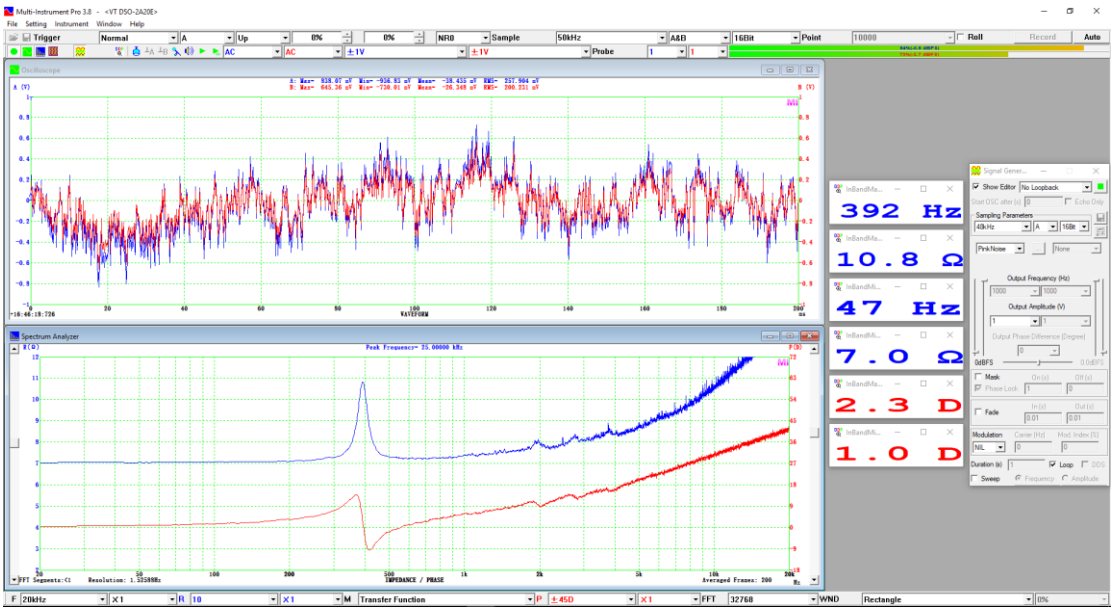
The measured resonant frequency is 394 Hz at the impedance peak 10.5 Ω. The lowest impedance is 6.8 Ω at the lower frequency end. The phases at the resonant frequency and lower frequency end are 1.1 degree and 0.1 degree respectively, which are very close to zero.

(2) Pink Noise

Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 200 ms (10000 samples)
Trigger Mode: Normal, Trigger Level: 0%

Spectrum Analyzer:
FFT size: 32768, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: 200-frame linear average
X Scale: Log, Y Scale: Convert to Impedance
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

Signal Generator:
DAC Device: Sound Card Headphone Output, Duration: Loop
Waveform: Pink Noise, Sampling Frequency: 48 kHz



The measured resonant frequency is 392 Hz with an impedance peak of 10.8 Ω. The lowest impedance is 7.0 Ω at the lower frequency end. The phases at the resonant frequency and lower frequency end are 2.3 degree and 1.0 degree respectively, which are very close to zero.

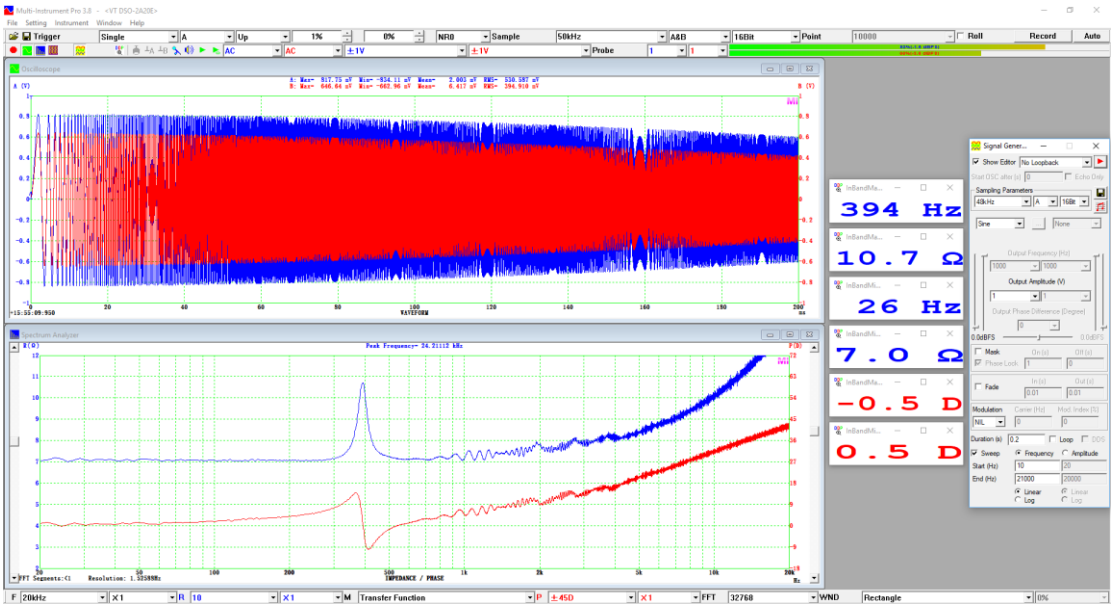
(3) Frequency linearly swept sine wave

Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 200 ms (10000 samples)
Trigger Mode: Single, Trigger Level: 1%

Spectrum Analyzer:
FFT size: 32768, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: None

Inter-Frame Processing: None
X Scale: Log, Y Scale: Convert to Impedance
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

Signal Generator:
DAC Device: Sound Card Headphone Output, Duration: 200 ms
Waveform: Frequency linearly swept sine wave from 10 Hz to 21 kHz
Sampling Frequency: 48 kHz



The measured resonant frequency is 394 Hz with an impedance peak of 10.7 Ω. The lowest impedance is 7.0 Ω at the lower frequency end. The phases at the resonant frequency and lower frequency end are -0.5 degree and 0.5 degree respectively, which are very close to zero.

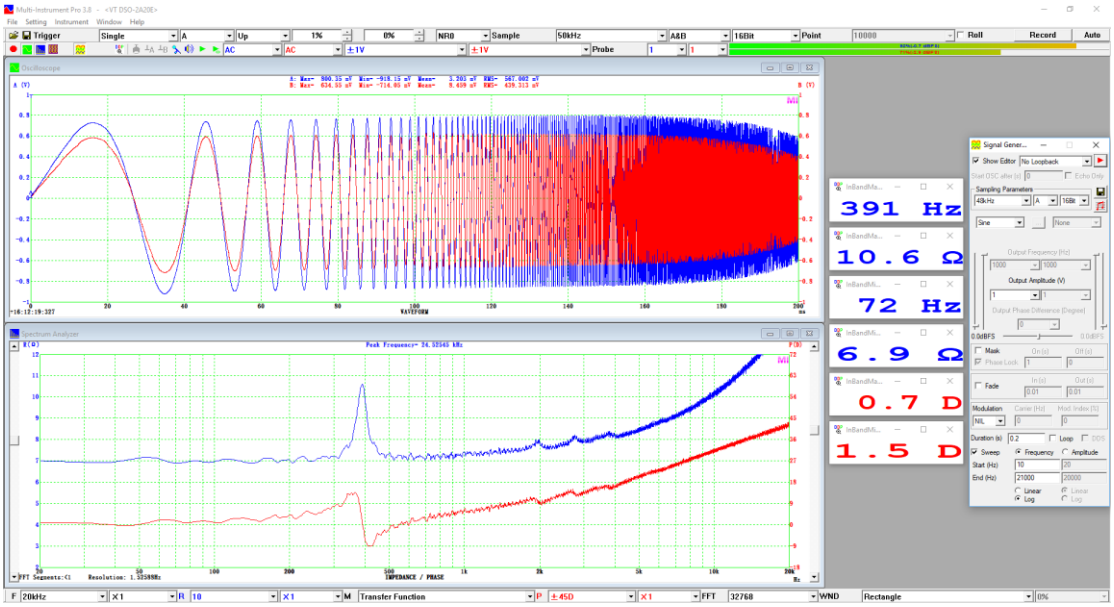
(4) Frequency logarithmically swept sine wave

Oscilloscope:
ADC Device: VT DSO-2A20E, Bit Resolution: 16
Sampling Frequency: 50 kHz, Record Length: 200 ms (10000 samples)
Trigger Mode: Single, Trigger Level: 1%

Spectrum Analyzer:
FFT size: 32768, Window Function: Rectangle
Intra-Frame Processing: Smooth via Moving Average: 10 points
Inter-Frame Processing: None
X Scale: Log, Y Scale: Convert to Impedance
Max. and Min. Impedance Searching Range: 20 Hz ~ 5 kHz
Frequency Range of Interest: 20 Hz ~ 20 kHz

Signal Generator:
DAC Device: Sound Card Headphone Output, Duration: 200 ms

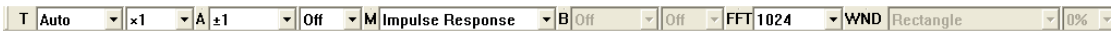
Waveform: Frequency logarithmically swept sine wave from 10 Hz to 21 kHz
Sampling Frequency: 48 kHz



The measured resonant frequency is 391 Hz with an impedance peak of 10.6 Ω. The lowest impedance is 6.9 Ω at the lower frequency end. The phases at the resonant frequency and lower frequency end are 0.7 degree and 1.5 degree respectively, which are very close to zero.

All the above tests obtain similar results. Among them, the frequency swept sine wave, either the linear one or the logarithmic one, is the fastest method. If a longer record length is chosen, the frequency resolution of the results would be finer.

3.2.7 View Parameters for Impulse Response Display



Selecting Impulse Response in the above View Type selection box switches the Spectrum Analyzer to impulse response display mode under which the impulse response graph is shown. The Fourier transform of the impulse response of a LTI system is precisely the frequency response of the system. The impulse response here is computed from the frequency response (i.e. transfer function) measured in the previous section through inverse FFT.

For impulse response measurement using the software, Channel B must be fed with the stimulus sent to the Device Under Test (DUT) and Channel A must be fed with the response from the DUT. To achieve a good measurement accuracy, significant stimulus energy must be present in the frequency range of interest. Commonly used test signals are chirp signal (swept sine) and white noise as well as multitones capable of simultaneously stimulating all FFT frequency bands within the frequency range of interest.

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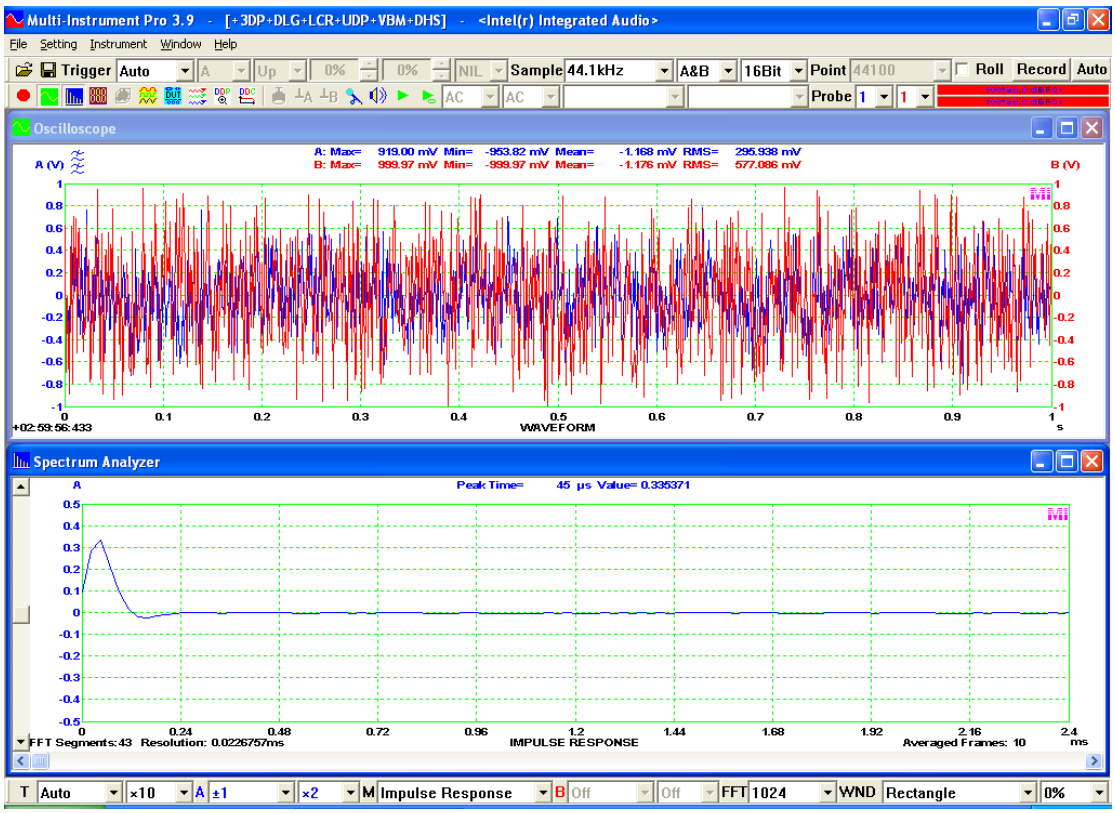
The following figure illustrates the impulse response of the aforementioned 5513 Hz second order Butterworth low pass filter. It can be seen that the measured impulse response is the same as the one measured using circular cross-correlation previously, except that they may be have different vertical scaling factors. Similarly, the measured impulse response here can also be exported to a TXT file and then imported into the Oscilloscope for further analysis (e.g. reverberation and speech intelligibility analysis). Unlike the circular cross correlation method, the impulse response measurement method here does not requires the stimulus to be white-noise like, any broadband signal that convers the frequency band of interest can be used.

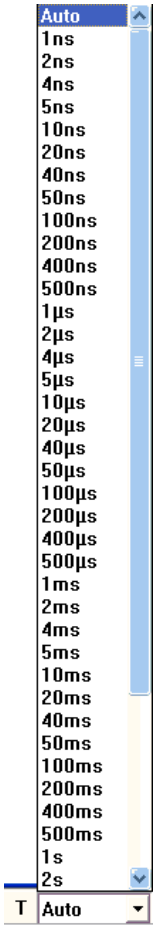


The following DDPs are generated by the impulse response display mode:

(1) PeakIPRTime_AB(s): Time at the peak of the Impulse Response
(2) PeakIPRValue_AB: Peak Impulse Response Value

These values are also displayed in the upper part of the Spectrum Analyzer. Some sample panel setting files for impulse response measurements are provided in the PSF subdirectory of the software.

3.2.7.1 Time Range (T)



49 options are available: Auto, 1ns, 2ns, 4ns, 5ns, 10ns, 20ns, 40ns, 50ns, 100ns, 200ns, 400ns, 500ns, 1µs, 2µs, 4µs, 5µs, 10µs, 20µs, 40µs, 50µs, 100µs, 200µs, 400µs, 500µs, 1ms, 2ms, 4ms, 5ms, 10ms, 20ms, 40ms, 50ms, 100ms, 200ms, 400ms, 500ms, 1s, 2s, 4s, 5s, 10s, 20s, 40s, 50s, 100s, 200s, 400s, 500s.

When “Auto” is selected, the Time Range will be set by the software automatically based on the following formula:

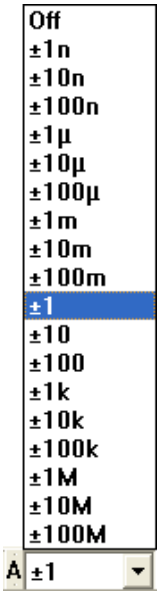
Time Range = the nearest integer that is equal to or greater than the value of [FFT Size] / [Sampling Frequency]

The impulse response here is obtained by converting the frequency response (i.e. Transfer Function) from the frequency domain to the time domain using inverse FFT. The time range is thus [FFT size]/[Sampling Frequency].

3.2.7.2 Time Multiplier

Same as Amplitude Spectrum.

3.2.7.3 Channel A Display Range



19 options available: Off, ±1n, ±10n, ±100n, ±1µ, ±10µ, ±100µ, ±1m, ±10m, ±100m, ±1, ±10, ±100, ±1k, ±10k, ±100k, ±1M, ±10M, ±100M.

When "Off" is selected, the data in Channel A will not be displayed.

3.2.7.4 Channel A Multiplier

Same as Amplitude Spectrum.

3.2.7.5 Channel B Display Range

Not applicable and disabled.

3.2.7.6 Channel B Multiplier

Not applicable and disabled.

3.2.7.7 FFT Size

Same as Amplitude Spectrum.

3.2.7.8 Window Function

Same as Amplitude Spectrum.

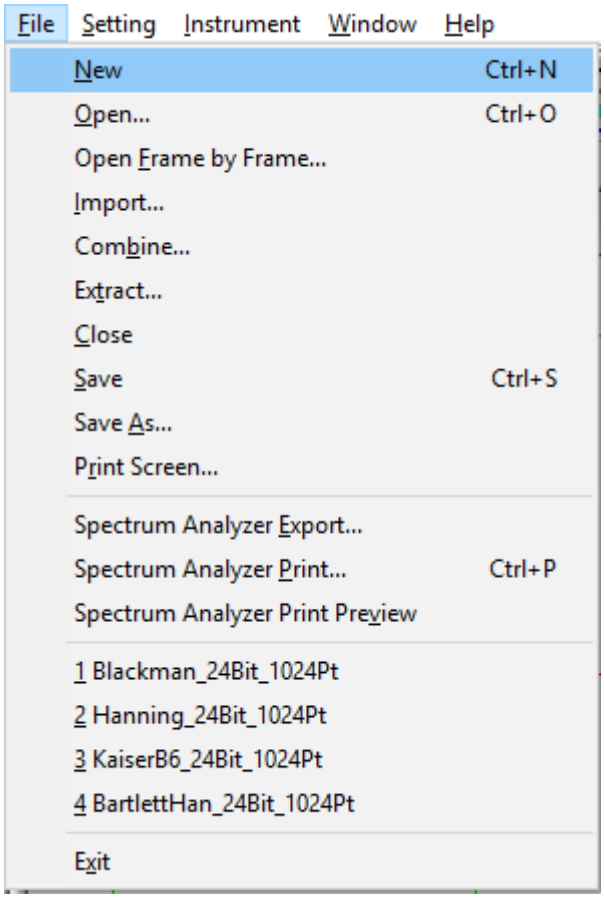
3.2.7.9 Window Overlap Percentage

Same as Amplitude Spectrum.

3.3 Menu

Spectrum Analyzer has its own menu bar and additional functions can be accessed through the menu items in each submenu. Click anywhere within the Spectrum Analyzer window will switch the software's main menu to the Spectrum Analyzer menu.

3.3.1 File SubMenu



This submenu provides access to the file operation and printing functions.

3.3.1.1 New (File SubMenu) (ALT-F-N, CTRL-N)

The command is used to create a new, blank document for measurement, with the system default panel setting fixed in the factory. The new document will be used to hold the latest collected frame of data.

This command is also available through the button in the Instrument Toolbar. When there is no Oscilloscope, Spectrum Analyzer and Multimeter opened, click the

above "Spectrum Analyzer" button will open a new document without changing the current Trigger and Sampling Parameters.

3.3.1.2 Open (File SubMenu) (ALT-F-O, CTRL-O)

Same as Oscilloscope.

3.3.1.3 Open Frame by Frame (File SubMenu) (ALT-F-F)

Same as Oscilloscope.

3.3.1.4 Import (File SubMenu) (ALT-F-I)

Same as Oscilloscope.

3.3.1.5 Combine (File SubMenu) (ALT-F-B)

Same as Oscilloscope.

3.3.1.6 Extract (File SubMenu) (ALT-F-T)

Same as Oscilloscope.

3.3.1.7 Close (File SubMenu) (ALT-F-C)

Same as Oscilloscope.

3.3.1.8 Save (File SubMenu) (ALT-F-S, CTRL-S)

Same as Oscilloscope.

3.3.1.9 Save As (File SubMenu) (ALT-F-A)

Same as Oscilloscope.

3.3.1.10 Print Screen (File SubMenu) (ALT-F-R)

Same as Oscilloscope.

3.3.1.11 Spectrum Analyzer Export (File SubMenu) (ALT-F-E)

This command is used to export either the calculated data (RMS Amplitude Spectrum, Octave Amplitude Spectrum, Phase Spectrum, Auto Correlation Function, Cross

Correlation Function, Coherence Function, Transfer Function, Impulse Response, depending on the current view type) to a TXT file or the currently displayed graph to a BMP file. When clicked, a "Save As" window will pop up. You can specify whether you want to export as a TXT file or a BMP file by selecting "Text File (*.txt)" or "Bitmap File (*.bmp)" in the "Save as type" combo box. The text file can be imported into third party software such as Microsoft Excel for further processing and analysis.

This command is disabled when the document is empty.

3.3.1.12 Spectrum Analyzer Print (File SubMenu) (ALT-F-P, CTRL-P)

Similar to Oscilloscope.

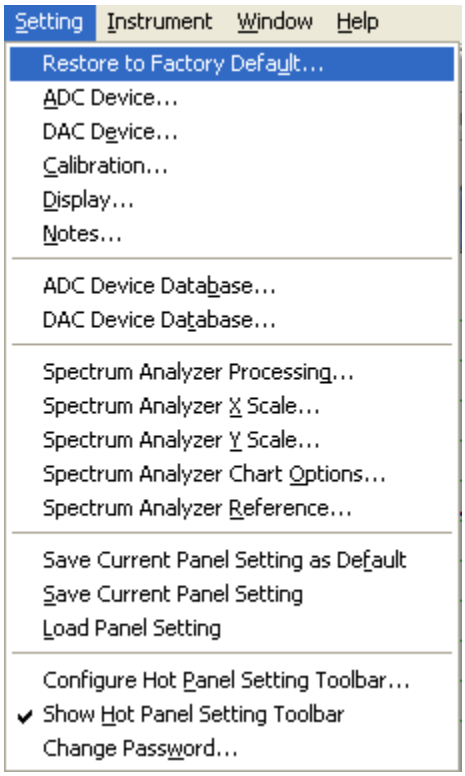
3.3.1.13 Spectrum Analyzer Print Preview (File SubMenu) (ALT-F-V)

Similar to Oscilloscope.

3.3.1.14 Recent File (File SubMenu)

Same as Oscilloscope.

3.3.2 Setting SubMenu



This submenu provides access to various setting functions.

3.3.2.0 Restore to Factory Default (Setting SubMenu) (ALT-S-U)

Same as Oscilloscope.

3.3.2.1 ADC Device (Setting SubMenu) (ALT-S-A)

Same as Oscilloscope.

3.3.2.2 DAC Device (Setting SubMenu) (ALT-S-E)

Same as Oscilloscope.

3.3.2.3 Calibration (Setting SubMenu) (ALT-S-C)

Same as Oscilloscope.

3.3.2.4 Display (Setting SubMenu) (ALT-S-D)

Same as Oscilloscope.

3.3.2.5 Note (Setting SubMenu) (ALT-S-N)

Same as Oscilloscope.

3.3.2.6 ADC Device Database (Setting SubMenu) (ALT-S-B)

Same as Oscilloscope.

3.3.2.7 DAC Device Database (Setting SubMenu) (ALT-S-T)

Same as Oscilloscope.

3.3.2.8 Spectrum Analyzer Processing (Setting SubMenu) (ALT-S-G)

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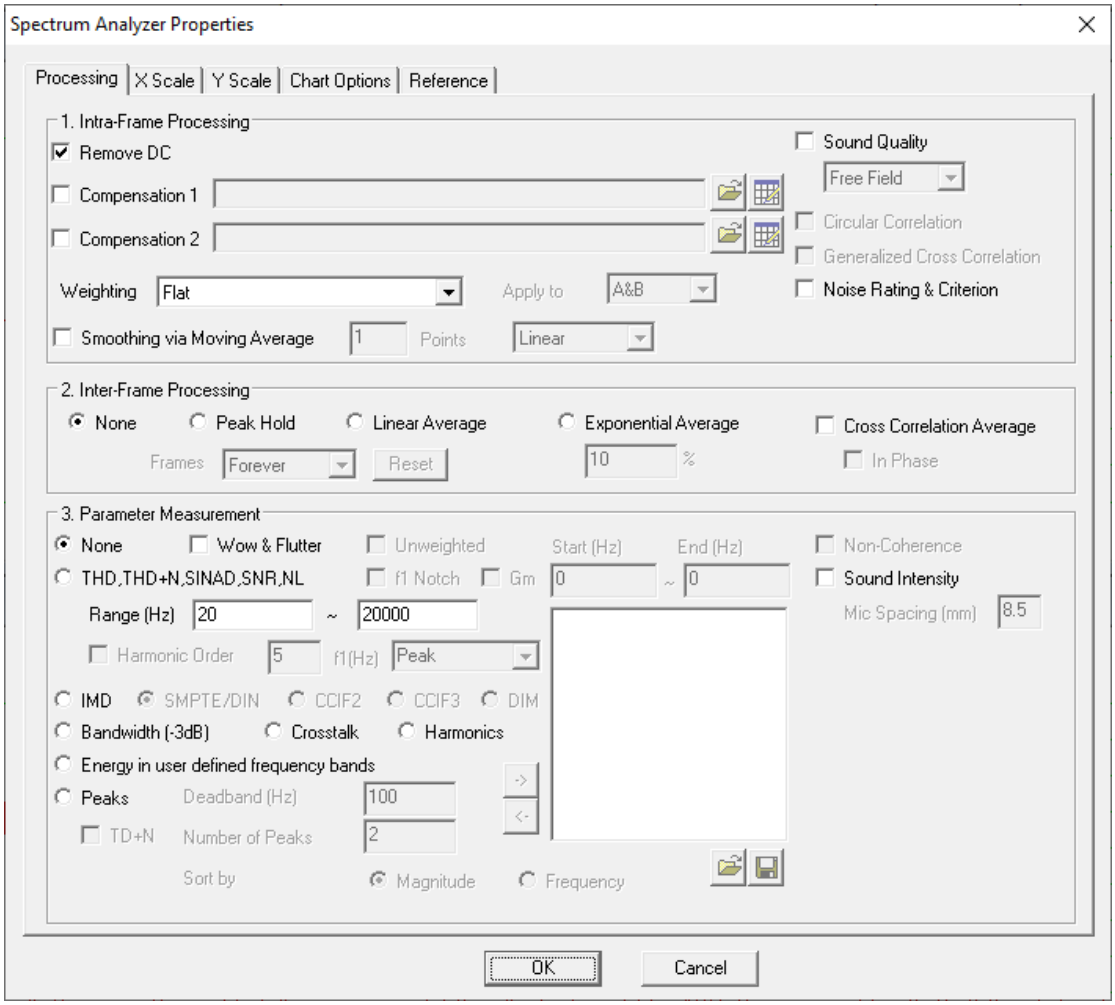

After being processed in time domain in the Oscilloscope, the signal will undergo the following processes in frequency domain sequentially:

(1) Intra-frame Processing
(2) Inter-frame Processing
(3) Parameter Measurement

The data frame here is the same as that in the Oscilloscope. A data frame can be further divided into one or multiple FFT segments depending on the ratio of Record Length in the Oscilloscope and FFT Size in the Spectrum Analyzer.

3.3.2.8.1 Intra-Frame Processing

The Intra-Frame Processing, as its name suggests, is performed within a data frame in frequency domain based on FFT segments. There are four types of Intra-Frame Processing: Remove DC, Frequency Compensation, Frequency Weighting, Smoothing via Moving Average. They are performed in sequence as above.

3.3.2.8.1.1 Remove DC

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If this checkbox is ticked, the mean value of the current frame of data will be subtracted from the data before FFT computation.

Some of the sound cards may introduce a small background DC offset even though the sound card is AC coupled. This DC offset, if not compensated in time domain, will result in one or a few frequency peaks around 0 Hz and sometimes these peaks can be higher than those frequency peaks of interest. Leaving this checkbox ticked will remove the DC component present in the input signal during FFT. This checkbox is ticked by factory default.

It should be noted that the mean value of the current frame of data is calculated without applying any non-uniform window function (i.e. only Rectangle Window is applied). If a window function other than Rectangle is used in the Spectrum Analyzer, some DC residual may still exist after DC removal and under some conditions, the DC residual may be even greater than the case if this option is not chosen.

3.3.2.8.1.2 Compensation

There are two types of compensation: frequency compensation under amplitude spectrum mode, and gain and phase compensation under transfer function mode. "-C-" will be displayed at the bottom left of the Spectrum Analyzer window if a frequency compensation is applied.

3.3.2.8.1.2.1 Frequency Compensation

Under amplitude spectrum mode, if you check the "Compensation1" or "Compensation2" checkbox, you are allowed to load a Frequency Compensation File (*.fcf) to compensate the amplitude spectrum. You can specify up to two compensation files. For example, you can use one compensation file to compensate the magnitude frequency response of the measuring circuit, and use another compensation file to compensate the magnitude frequency response of the microphone.

A Frequency Compensation File is a Comma Separated Value (CSV) TXT file that defines the magnitude compensation value (in dB) at each frequency. A mono frequency compensation file has the following format (which follows the same format as the Frequency Response File (*.frf) defined previously):

Example:

1,20,-50.5
2,25,-44.7
3,31.5,-39.4
4,40,-34.6
……

Each line contains three comma separated values. The first variable is the sequential number. The second one is the frequency value in Hz. And the third one is the corresponding gain value in dB. Note that 0 dB represents the unit gain. Any frequency that falls outside the defined range will be given a gain value of –1000 dB.

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A stereo frequency compensation file has a fourth variable in each line, specifying the gain value in dB for the second channel. If the sampled data are stereo but the frequency compensation file is mono, then the same frequency compensation will be applied to both channels.

When the magnitude frequency response of a measuring device itself is obtained in the Spectrum Analyzer, the corresponding frequency compensation file can be generated by right clicking anywhere within the Spectrum Analyzer window and selecting "Generate Frequency Compensation File (Flat)". The generated compensation file can be used to eliminate the effect of the non-flat magnitude frequency response of the measuring device itself on the measurement.

Some sample files are provided in the FCF directory of the software and can be used as templates. Frequency compensation is only applicable to the Amplitude Spectrum display.

It is also possible to load an "Original Frequency Response File (*.txt)" here. This type of TXT files is normally provided by device manufacturers. It contains a few lines of textual information in the header, followed by many lines of data (see example below). Each header line starts with a non-numeric character while each data line starts with a numeric one. Each data line contains two (frequency in Hz and gain in dB) or three (frequency in Hz, Ch.A gain in dB, Ch.B gain in dB) space-separated variables. The products, whose original frequency response files are compatible with the above format, include Dayton EMM-6 and UMM-6 measurement microphones, miniDSP UMIK-1 & UMIK-2 USB measurement microphones, Earthworks measurement microphones, etc.. It is important to note that signs of the gain values in the frequency response file will be automatically reversed in the software in order to compensate for them.

Example:

```
*1000Hz        -40.6

20.00   -2.4
20.55   -2.1
…
```

3.3.2.8.1.2.2 Gain and Phase Compensation

Under transfer function mode, if you check the "Compensation1" or "Compensation2" checkbox, you are allowed to load a Gain and Phase Compensation file (*.gpc) to compensate the gain and phase plot. You can specify up to two compensation files.

A Gain and Phase Compensation file is a Comma Separated Value (CSV) TXT file that defines the gain compensation value (in dB) and phase compensation value (in degree) at each frequency. It has the following format:

Example:

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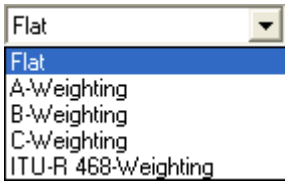
1,20,-50.5,0.49
2,25,-44.7,2.85
3,31.5,-39.4,4.54
4,40,-34.6,-0.38
……

Each line contains four comma separated values. The first variable is the sequential number. The second one is the frequency value in Hz. The third one is the corresponding gain compensation value in dB, and the fourth one the corresponding phase compensation value in degree. Note that 0 dB represents the unit gain. Any frequency that falls outside the defined range will be given a gain compensation value of 0 dB and a phase compensation value of 0 º.

When the transfer function of a measuring device itself is obtained (usually by directly injecting the same test signal into both input channels) in the Spectrum Analyzer, the corresponding gain and phase compensation file can be generated by right clicking anywhere within the Spectrum Analyzer window and selecting "Generate Frequency Compensation File (Flat)". The generated compensation file can be used to eliminate the effect of the gain and phase difference between the two input channels of the measuring device itself on the measurement.

Gain and phase compensation is only applicable to the transfer function display. Note that under the impedance measurement submode of the transfer function mode, these compensation files, if any, will be ignored.

3.3.2.8.1.3 Frequency Weighting



Four commonly used weighting profiles are supported: A, B, C and ITU-R 486. The following figure illustrates the four weighting curves obtained by analyzing white noise with the corresponding weighting profile. All curves cross at 1 kHz with a gain value of 0 dB.

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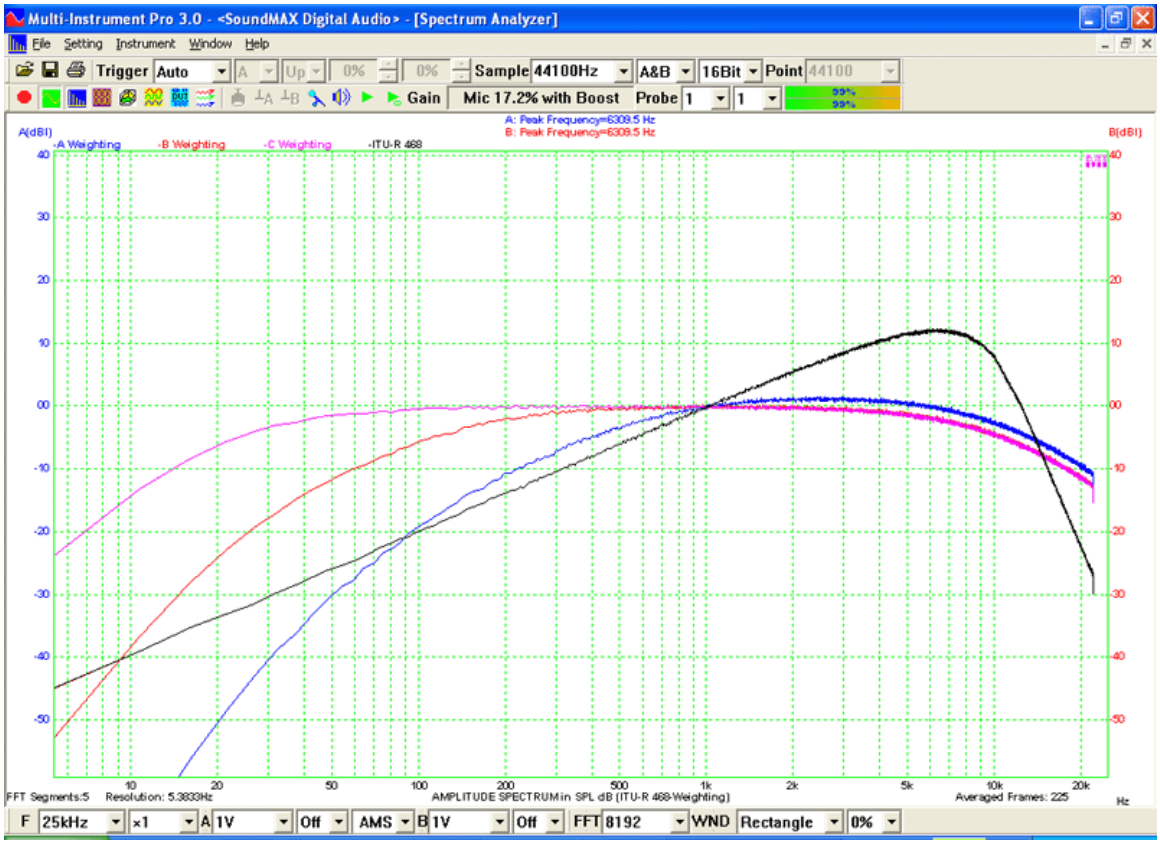


In addition, the following frequency weighting schemes are supported via their respective frequency compensation files:

(1) ITU_R_ARM
(2) Recording RIAA
(3) Reproducing RIAA
(4) 10Hz-20kHz Z_Weighting
(5) 20Hz-20kHz Band Width Limiting
(6) AES6-2008-W&F (Wow & Flutter)
(7) ISO-2631-1-Wc (horizontal whole-body vibration x-axis)
(8) ISO-2631-1-Wd (horizontal whole-body vibration x-axis or y-axis)
(9) ISO-2631-1-We (rotating whole-body vibration)
(10) ISO-2631-1-Wf (low-frequency vertical whole-body vibration z-axis)
(11) ISO-2631-1-Wj (vertical head vibration)
(12) ISO-2631-1-Wk (vertical whole-body vibration z-axis)
(13) ISO-2631-2-Wm (whole-body vibration on non-specific posture)
(14) ISO-2631-4-Wb (vertical whole-body vibration z-axis)
(15) ISO-5349-1-Wh (hand-arm vibration)

Frequency weighting is only applicable to the Amplitude Spectrum display. It can be applied to Channel A, Channel B, or both of them. Different weighting options generate different DDPs. A, C weighting generates RMSDBA_A(dBA) and RMSDBC_A(dBC) respectively for Channel A, and RMSDBA_B(dBA) and RMSDBC_B(dBC) respectively for Channel B. These DDPs can be displayed in the respective display mode in the Multimeter.

For more information on frequency weightings for sound and vibration, please refer to *Frequency Weightings for Sound and Vibration Perceived by Humans using Multi-Instrument*.

Download link:
https://www.virtins.com/doc/Frequency-Weightings-for-Sound-and-Vibration-Perceived-by-Humans-using-Multi-Instrument.pdf

3.3.2.8.1.4 Smoothing via Moving Average

If this option is ticked, moving average will be applied to the data in frequency domain to make the displayed curves smoother. The size of the moving average window is adjustable. The bigger the moving average window, the smoother the displayed curves, the coarser the frequency resolution.

Along the frequency axis, the size of the moving average window can be a fixed one which is the number of "Points" under the "linear" mode, or a variable one according 1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96 octave bands. The latter is referred to as octave smoothing. Its moving average window size increases exponentially with frequency. This is especially useful when the X axis is scaled logarithmically. Under the octave smoothing mode, the number of "Points" specified will still play a part. It acts as the minimum moving average window size in case the fractional octave bandwidth is too small in the low frequency region.

The moving average is performed in terms of power under the amplitude spectrum display mode while under other display modes, it is just an arithmetic average.

3.3.2.8.1.5 Parameter Measurements before Inter-Frame Processing

Some parameters are measured before inter-frame processing. Therefore, their values will reflect only the current frame of data in the Oscilloscope and will not be affected by the inter-frame processing. These parameters include dBA, dBZ and dBC in the Multimeter, and sound quality parameters such as loudness, loudness level and sharpness.

3.3.2.8.1.5.1 Sound Quality

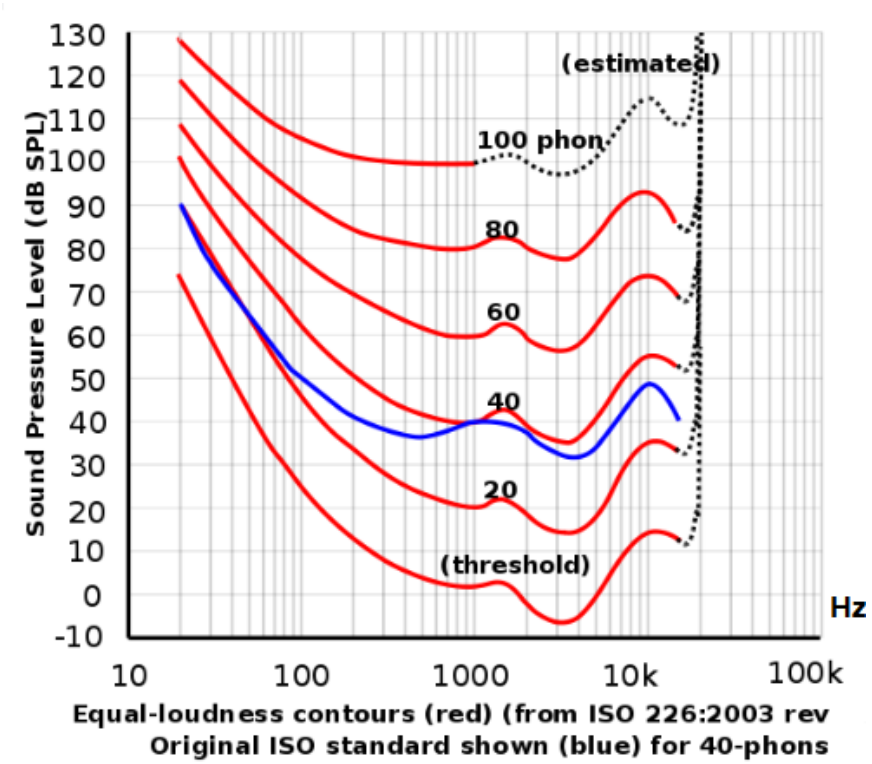
It has long been known that the conventional acoustic metrics, such as A-weighted sound pressure level, don't correlate well with perceptions of sound quality by humans. The term sound quality here refers to the overall experience of the information in the sound that leads to a person's liking it or not, or that leads to a perception of the non-acoustical qualities of the device emitting the sound (that is, engine power, robust construction, etc.). There are a large number of metrics, some of which are well defined and others which are not. Very few have been standardized and the usefulness of a particular metric is dependent on the nature of the sound being tested. The following sections describe the sound quality parameters such as loudness, loudness level, sharpness, articulation index and open articulation index, which can be measured by the software.

3.3.2.8.1.5.1.1 Loudness and Loudness Level

Sound loudness is a subjective term describing the strength of human ear's perception of a sound. Different standards exist for loudness calculation. ISO 532B (DIN45631) is used here. It is the most widely accepted standard for loudness calculation of a stationary sound. The calculation is based on 1/3 octave band levels of a sound. Options are provided to allow the user to specify the type of sound fields: Free Field or Diffuse Field.

The unit of loudness is sone. The sone scale is linear. Doubling the perceived loudness doubles the sone value. A loudness of 1 sone is equivalent to the loudness of a signal at 40 phons, the loudness level of a 1 kHz tone at 40 dB SPL.

The unit of loudness level is phon. The phon scale is logarithmic. The number of phons of a sound is the dB SPL of a sound at a frequency of 1 kHz that sounds just as loud. The following figure shows the equal-loudness contours.



Loudness and loudness level are interconvertible. When the loudness is equal to or greater than 1 sone, [loudness level] = 40 + 10×$\log_2$(loudness). When the loudness is less than 1 sone, [loudness level] = 40×(loudness+0.0005)$^{0.35}$. In the software, their values are accessible through the Derived Data Points (DDPs): Loudness_A(SONE) and LoudnessLevel_A(PHON) for Channel A, and Loudness_B(SONE) and LoudnessLevel_B(PHON) for Channel B. It should be noted that the loudness calculation is performed after "Remove DC" and Frequency Compensation but before Frequency Weighting. Therefore, their values will not be affected by the frequency weighting option.

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For loudness calculation, FFT size should be set greater than 16384 (required by 1/3 octave analysis at a sampling frequency of 48 kHz) and normally Rectangle window function should be used for wide band noises.

3.3.2.8.1.5.1.2 Sharpness

Sharpness is a hearing sensation related to frequency and independent of loudness. Sharpness corresponds to the sensation of a sharp, painful, high-frequency sound and is the comparison of the amount of high frequency energy to the total energy. Sharpness delineates human sensation in a linear manner as well. In general, sharpness is increased by adding higher frequency content, and decreased by adding lower frequency content. The unit of sharpness is acum. Sharpness of one acum is produced by a narrow-band noise at 1 kHz with a bandwidth smaller than 150 Hz and a level of 60 dB.

Different standards exist for sharpness calculation but there is no international standard yet. DIN45692 is used here. It is calculated together with the loudness and loudness level introduced previously.

In the software, sharpness values are accessible through the Derived Data Points (DDPs): Sharpness_A(ACUM) for Channel A, and Sharpness_B(ACUM) for Channel B.

3.3.2.8.1.5.1.3 Articulation Index and Open Articulation Index

Articulation index (AI), as defined in ANSI S3.5-1969 and ANSI S3.5-1997, is used to indicate how much background sound levels can interfere with human speech. It was used originally in applications like measuring speech privacy and the effectiveness of communication systems. Nowadays, it is also used to rate vehicle interior noise, the quietness of white goods and more. It has a value between 0% (no speech understood) to 100% (all speech understood).

There can be slight variations in the calculation method of articulation index. It is calculated here based on the A-weighted 1/3 octave spectrum of the measured sound, as shown in the following table.

1/3 Octave Center Frequency (Hz)	Low Limit (dBA)	High Limit (dBA)	Weighting Factor (%)
200	23.1	53.1	1
250	30.4	60.4	2
315	34.4	64.4	3.25
400	38.2	68.2	4.25
500	41.8	71.8	4.5
630	43.1	73.1	5.25
800	44.2	74.2	6.5
1000	44.0	74.0	7.25
1250	42.6	72.6	8.5
1600	41.0	71.0	11.5
2000	38.2	68.2	11.0
2500	36.3	66.3	9.5
3150	34.2	64.2	9.0
4000	31.0	61.0	7.75

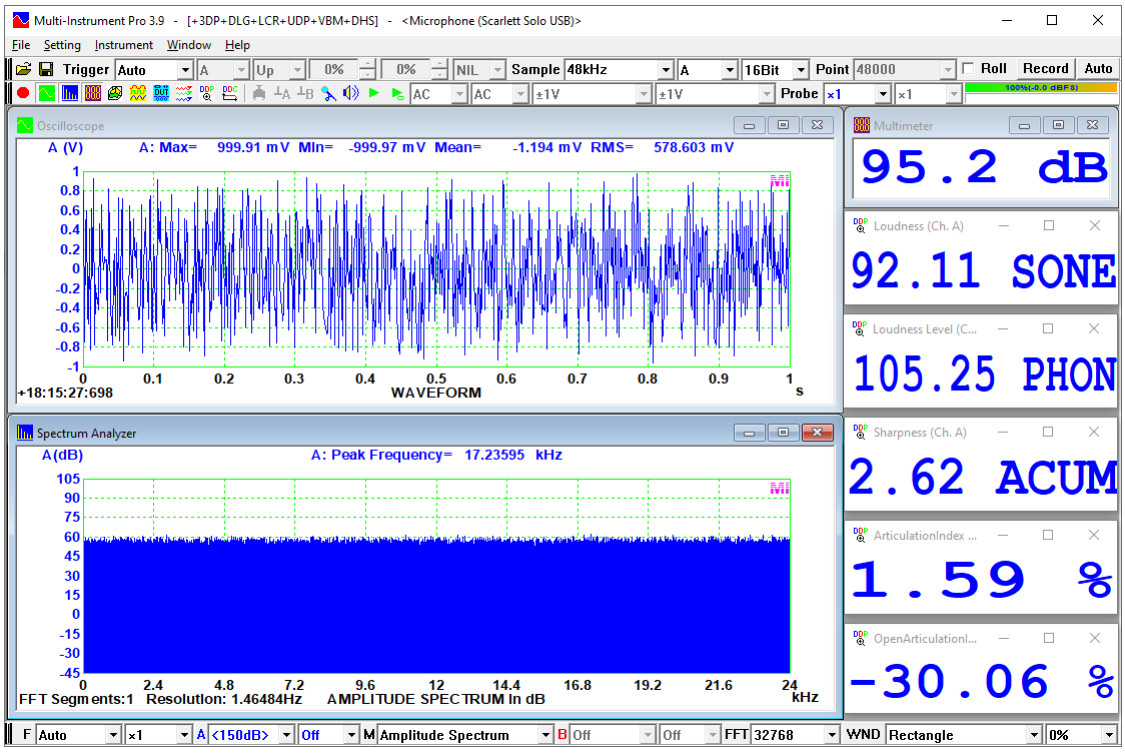
5000	26.5	56.5	6.25
6300	20.9	50.9	2.5

The difference between the High Limit and Low Limit in the above table is fixed at 30 dB. When the measured value in a 1/3 octave band is greater than the respective High Limit, the result in that octave band is 0. When that value is lower than the respective Low Limit, the result is 1. When that value is in between, the result is given by ([High Limit]- [Measured Value])/30. All results from 200Hz to 6300Hz are weighted and averaged to yield a single value of articulation index.

Open articulation index has a wider range than 0% ~ 100%. Its calculation method is similar to that of the articulation index. But the measured value is not limited by the above octave band specific High Limit and Low Limit during calculation. Instead, it is limited by a fixed 80dB range, from 20dBA to 100dBA. As a result, the open articulation index has a value between -108.56% and 158.11%.

In the software, Articulation index and open articulation index are accessible through the Derived Data Points (DDPs): ArticulationIndex_A(%) and OpenArticulationIndex_A(%) for Channel A, and ArticulationIndex_B(%) and OpenArticulationIndex_B(%) for Channel B.

The following figure shows the measured dBSPL, loudness, loudness level, sharpness, articulation index and open articulation index of a white noise in a free field.



Two preconfigured panel setting files for loudness, loudness level, sharpness and articulation index measurement can be found in the PSF subdirectory of the software:
\pro\Loudness.psf
\pro\Loudness_RTAFCF.psf

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3.3.2.8.1.5.2 Noise Rating and Criterion

Noise Rating (NR) and Noise Criterion (NC) are similar methods used to evaluate and quantify the noise levels in indoor environments, often for noise related to mechanical sources such as air conditioning systems. The Noise Criterion method is used primarily in the United States, while the Noise Rating method is more common in Europe and much of the rest of the world. Both methods take into account the frequency content of the noise and the response of the human ear, providing a single value based on the 1/1 octave band spectrum of the noise. To derive the NR/NC value, the 1/1 octave band measurements are compared against the standard NR/NC curves (see tables below) to find the lowest curve where the noise does not exceed the NR/NC level in any octave band.

NR Curve	1/1 Octave Band Sound Pressure Level (dB)								
	31.5Hz	63Hz	125Hz	250Hz	500Hz	1000Hz	2000Hz	4000Hz	8000Hz
NR 0	55	36	22	12	5	0	-4	-6	-8
NR 10	62	43	31	21	15	10	7	4	2
NR 20	69	51	39	31	24	20	17	14	13
NR 30	76	59	48	40	34	30	27	25	23
NR 40	83	67	57	49	44	40	37	35	33
NR 50	89	75	66	59	54	50	47	45	44
NR 60	96	83	74	68	63	60	57	55	54
NR 70	103	91	83	77	73	70	68	66	64
NR 80	110	99	92	86	83	80	78	76	74
NR 90	117	107	100	96	93	90	88	86	85
NR 100	124	115	109	105	102	100	98	96	95
NR 110	130	122	118	114	112	110	108	107	105
NR 120	137	130	126	124	122	120	118	117	116
NR 130	144	138	135	133	131	130	128	127	126

NC Curve	1/1 Octave Band Sound Pressure Level (dB)							
	63Hz	125Hz	250Hz	500Hz	1000Hz	2000Hz	4000Hz	8000Hz
NC 15	47	36	29	22	17	14	12	11
NC 20	51	40	33	26	22	19	17	16
NC 25	54	44	37	31	27	24	22	21
NC 30	57	48	41	35	31	29	28	27
NC 35	60	52	45	40	36	34	33	32
NC 40	64	56	50	45	41	39	38	37
NC 45	67	60	54	49	46	44	43	42
NC 50	71	64	58	54	51	49	48	47
NC 55	74	67	62	58	56	54	53	52
NC 60	77	71	67	63	61	59	58	57
NC 65	80	75	71	68	66	64	63	62
NC 70	83	79	75	72	71	70	69	68

NR/NC values can be accessed through the Derived Data Points (DDPs): NoiseRating_A and NoiseCriterion_A for Channel A, and NoiseRating_B and NoiseCriterion_B for Channel B.

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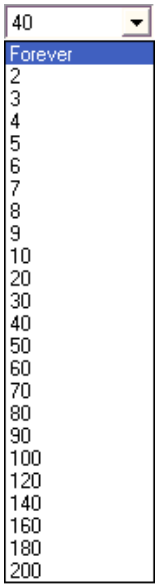
3.3.2.8.2 Inter-Frame Processing

There are four options: None, Peak Hold, Linear Average, Exponential Average. You can specify the number of consecutive frames to be processed. The available options are: 2, 3, 4, 5, 6, 7, 8, 9, 10, 20, 30, 40, 50, 60, 70, 80, 90, 100, 120, 140, 160, 180, 200 and forever. Note that when the Spectrum 3D Plot is running, the number of frames here must be less than or equal to the number of frames specified in Spectrum 3D Plot. When "Forever" is selected, you can reset the process using the Reset button when necessary. The number of frame processed is displayed at the bottom right corner of the Spectrum Analyzer window. It can be accessed through DDP: NumberOfFrames.



3.3.2.8.2.1 Peak Hold

If Peak Hold is selected, the Spectrum Analyzer will keep track of each data frame acquired and only display the peak value (in terms of absolute value) at each frequency for the specified number of frames acquired most recently. The number of frames processed will be displayed at the bottom-right corner of the Spectrum Analyzer view.

You can utilize the peak hold function together with a swept sine signal generated via the Signal Generator to obtain the magnitude frequency response of a Device Under Test (DUT). This method requires the Record Length, FFT frequency resolution and sweep duration to be carefully chosen. The finer the frequency resolution, the longer the sweep duration. The following figure illustrates the magnitude frequency response of a laptop built-in sound card measured using an external loopback cable. The sinusoidal signal was swept linearly from 20 Hz to 22050 Hz for 300 seconds. The frame size of the Oscilloscope was 200 ms with a sampling rate of 44100 Hz. The Spectrum Analyzer was in the Peak Hold mode with 1024 FFT points. The figure shows that the magnitude frequency response was finally obtained by peak holding

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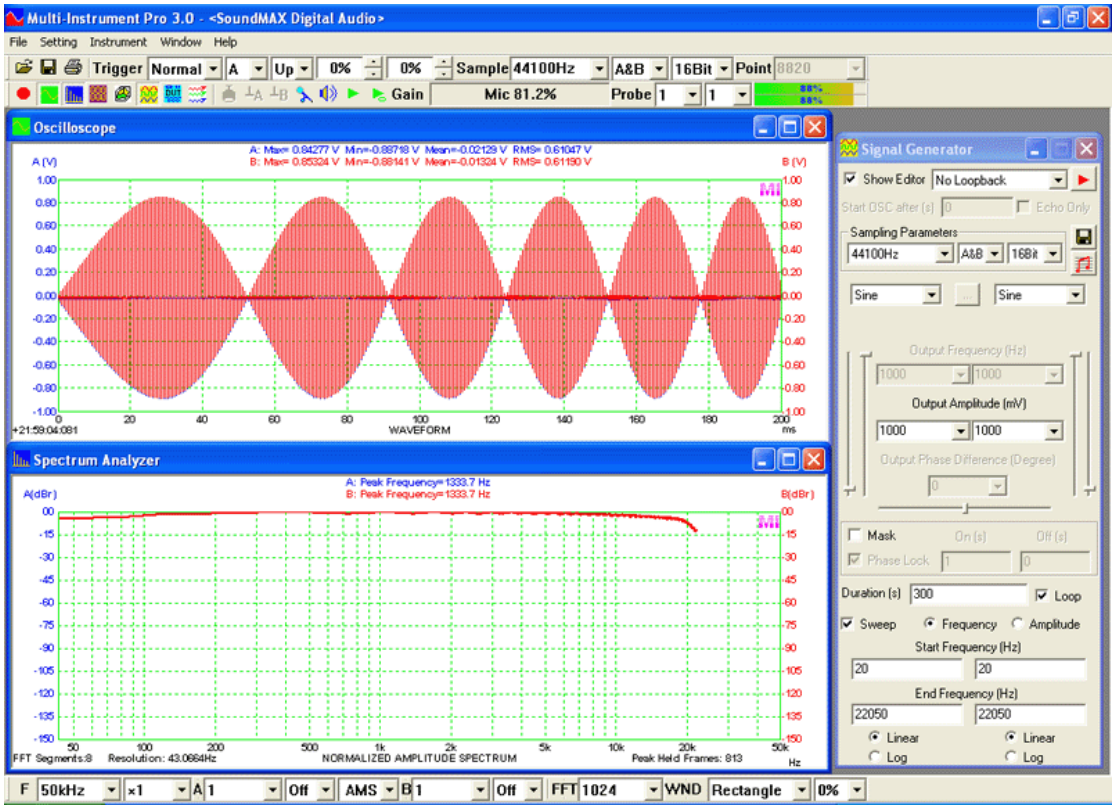
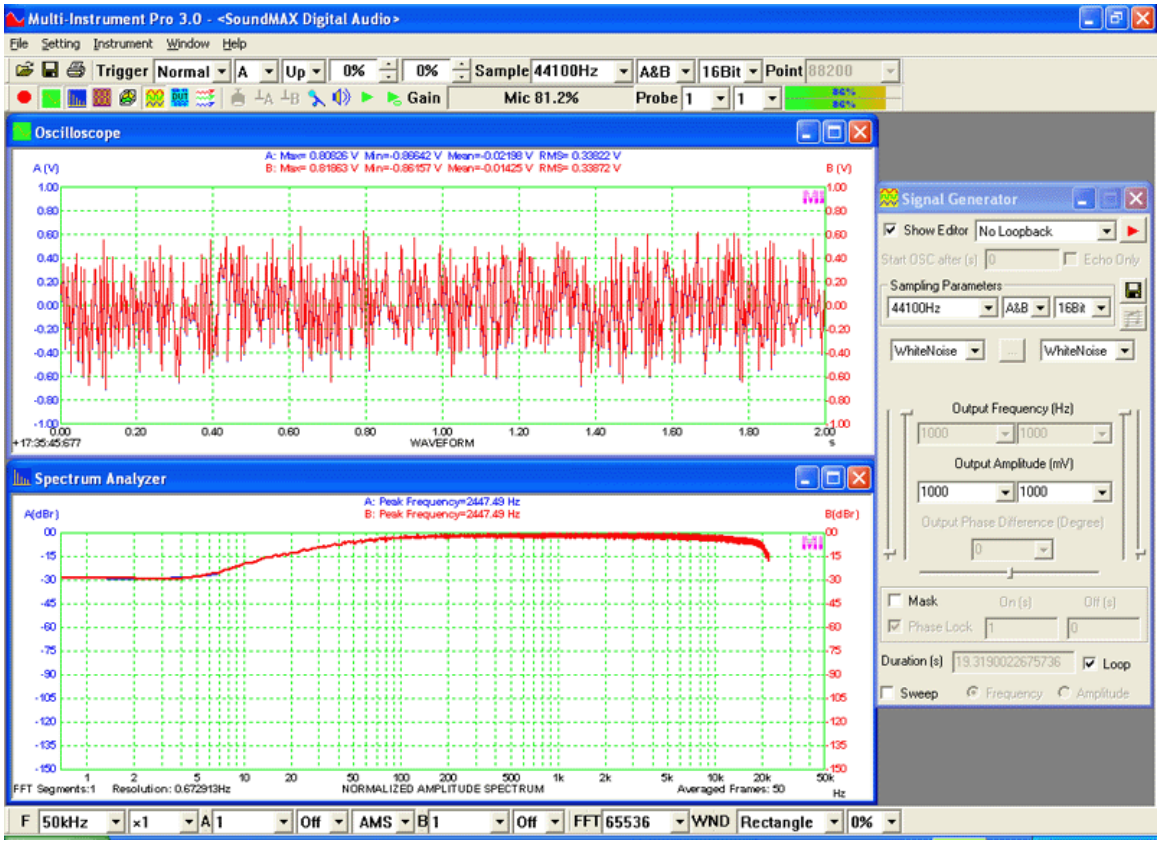
813 frames of data. This panel setting can be found in the Panel Setting File (PSF) directory of the software.



3.3.2.8.2.2 Linear Average

If Linear Average is selected, the Spectrum Analyzer will keep track of each data frame acquired and only display the averaged value at each frequency for the specified number of frames acquired most recently. The number of frames processed will be displayed at the bottom-right corner of the Spectrum Analyzer view.

The following figure illustrates the magnitude frequency response of a laptop built-in sound card measured using an external loopback cable. White noise was used as the test signal. The frame size of the Oscilloscope was 2 s with a sampling rate of 44100 Hz. The Spectrum Analyzer was in the averaging mode with 65536 FFT points. The result was obtained by averaging 50 frames of white noise. The more the averaged frames, the smoother the curve. This panel setting can be found in the Panel Setting File (PSF) directory of the software.



3.3.2.8.2.3 Exponential Average

Unlike linear average where all data frames used for average are given equal weights, in exponential average, the weighting factor for each data frame decreases with time exponentially, giving much more importance to recent observations while still not discarding older observations entirely. The degree of weighting decrease is expressed as a constant α in percentage. The greater the α, the faster the decrease. Alternatively, α may be expressed in terms of N, where $\alpha = 2/(N+1)$, and $N \times$ [time interval between the start times of two adjacent data frames] is called the time constant.

3.3.2.8.2.4 Cross Correlation Average

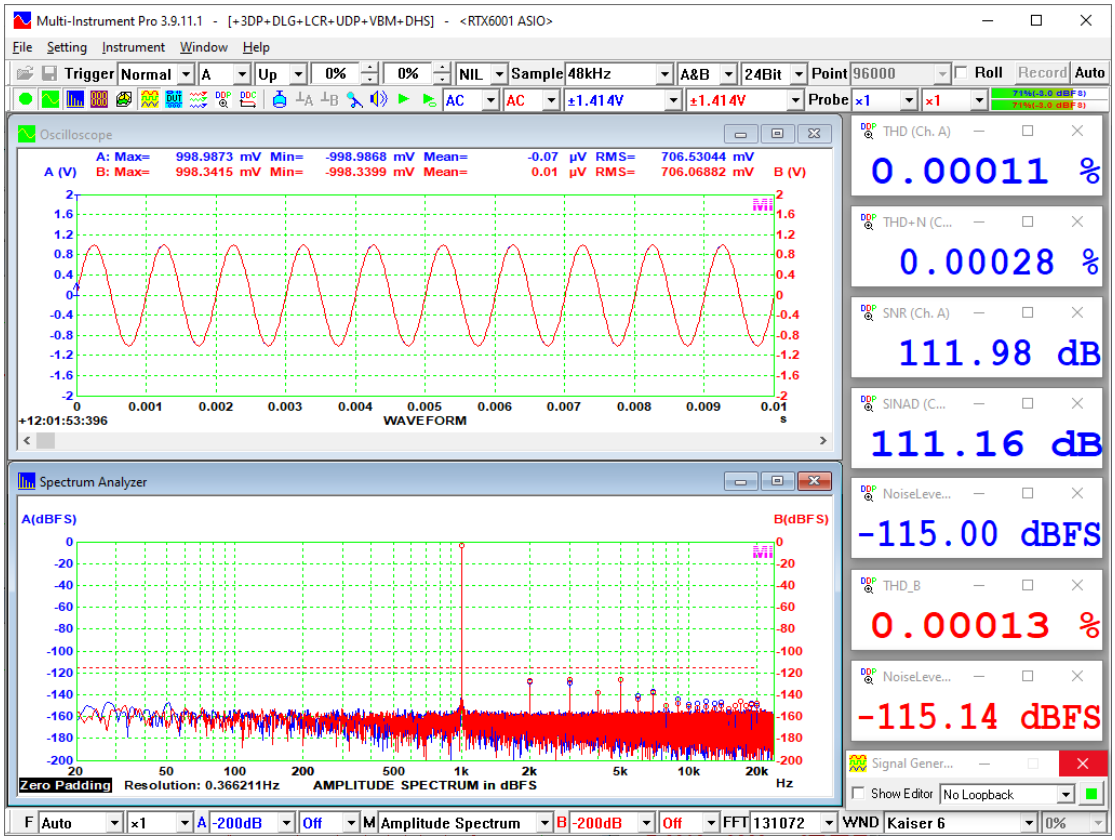
This option will only be enabled for amplitude spectrum display mode when the inter-frame processing is set to None, Linear Average, or Exponential Average and Spectrum 3D Plot is not running.

As mentioned previously in Section 3.2.1.7, when the input channels work independently, auto power averaging is used for both intra-frame and inter-frame averaging. This type of averaging reduces the fluctuation of the spectral data but does not reduce the absolute noise level in the amplitude spectrum, which includes the contribution from the measurement system itself. To suppress this adverse contribution, the same signal under test can be fed into both input channels of the measurement system simultaneously, and cross power vector averaging can be used to reduce the uncorrelated noise in the two input channels. This "Cross Correlation Average" technique is often used when the signal under test is very weak or even

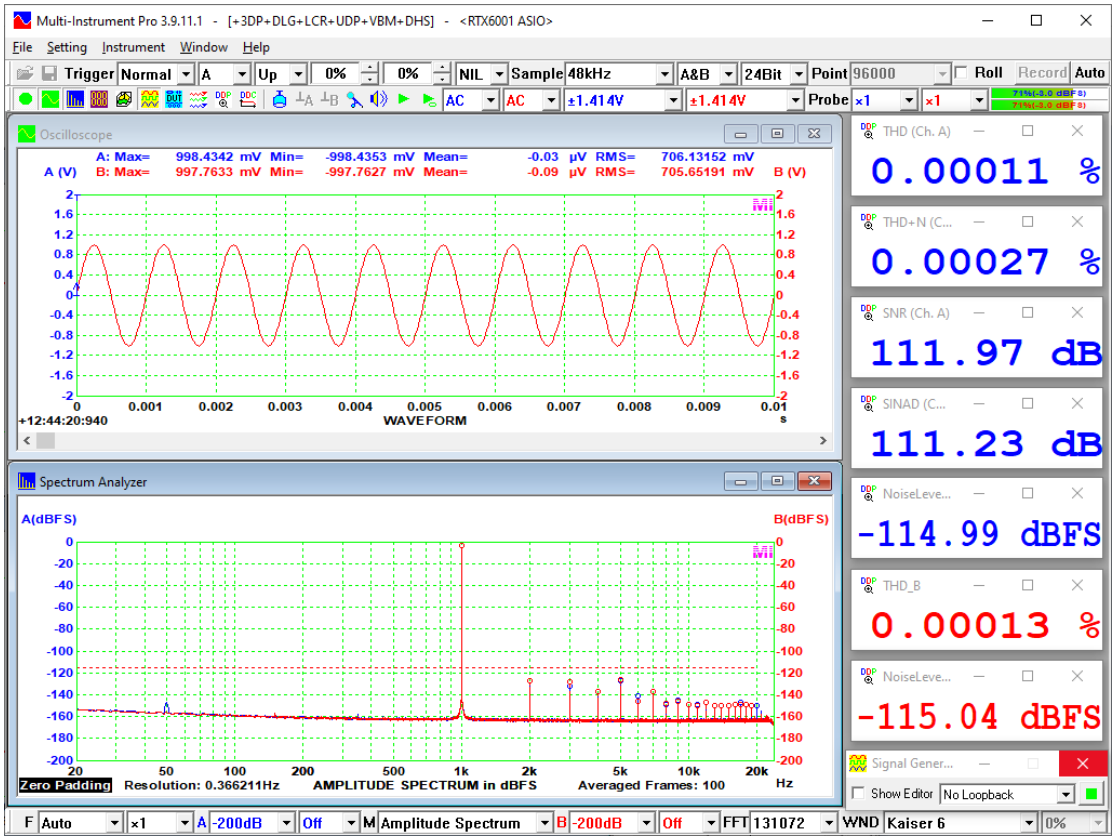
buried under the noise floor of the measurement system. The uncorrelated noise will theoretically be reduced by $5\log_{10}$(Number of Averaging) dB. For example, with 100 averages, the noise reduction would be 10dB.

Please note that if this option is selected, the "Cross Correlation Average" technique will be used for both intra-frame and inter-frame averaging. Additionally, this technique does not require the signals in the two input channels to be in phase, unless the "In Phase" checkbox is ticked. This is advantageous, for example, when measuring extremely low sound pressure levels using a microphone pair, where the signals in the two channels may not be in-phase. If the signals in the two channels are strictly in phase or strictly with inverted polarities, selecting the "In Phase" option will improve the noise reduction performance by $5\log_{10}(2)$=1.5dB. Unlike the amplitude spectrum, which will be identical for both channels after "Cross Correlation Average", the phase information will still be calculated individually and is available in the respective DDPs.

The following screenshot shows a THD loopback test of an RTX6001 audio analyzer without any averaging. The THD and noise level values are 0.00011%, -115.00 dBFS for Channel A, and 0.00013%, -115.14 dBFS for Channel B. It can be observed from the Spectrum Analyzer that the variation in the noise floor is quite significant.



The following screenshot shows the same test with 100 inter-frame linear averages. There are nearly no changes in the measured THD and noise levels, except that the noise floor becomes much cleaner.



The same test was repeated using the "Cross Correlation Average" method (without selecting "In Phase") with 100 averages. The results are shown below. The measured THD is 0.00012%, which is about the average of the two independent channels. The noise level was measured to be -123.78 dB, a reduction of about 8.8 dB. Other noise-related parameters, including THD+N, SNR, SINAD, all show pronounced improvement, except that the noise floor is still not as clean as in the previous screenshot.

For more information on noise reduction techniques, please refer to *Software Methods for Non-Linear Distortion Compensation and Noise Reduction in Measurements*.

Download link:
https://www.virtins.com/doc/Software-Methods-for-Non-Linear-Distortion-Compensation-And-Noise-Reduction-In-Measurements.pdf

3.3.2.8.3 Parameter Measurement after Inter-frame Processing

Parameter Measurement is performed after the Intra-Frame and Inter-Frame processing. It is done only in the Amplitude Spectrum display. The following parameters can be measured.

3.3.2.8.3.1 THD, THD+N, SINAD, SNR, NL

If the option is ticked, THD, THD+N, SINAD, SNR, and NL values will be displayed in the upper portion of the Spectrum Analyzer view. These parameters are measured with reference to the single test tone frequency, which is detected automatically with a sub-FFT-bin-size accuracy, in other words, its accuracy is much higher than the FFT frequency resolution. These parameters can also be measured with frequency weighting such as A-weighting.

THD (Total Harmonic Distortion) is defined here as the square root of the ratio of the sum of the powers of all harmonic frequencies above the fundamental frequency to the power of the fundamental. It is displayed in both percentage (%) and dB. It can be

accessed through the Derived Data Points (DDPs): THD_A(%) and THDDB_A(dB) for Channel A, and THD_B(%) and THDDB_B(dB) for Channel B.

THD+N (Total Harmonic Distortion plus Noise) is defined here as the square root of the ratio of the sum of the powers of all harmonic frequencies above the fundamental frequency plus noise to the total power. In other words, it equals to the square root of the ratio of the total power less the power of the fundamental frequency to the total power. It is displayed in both percentage (%) and dB. It can be accessed through the Derived Data Points (DDPs): THDN_A(%) and THDNDB_A(dB) for Channel A, and THDN_B(%) and THDNDB_B(dB) for Channel B.

SINAD (Signal in Noise and Distortion) is defined here as the ratio of the total power to the total power less the power of the fundamental frequency, converted to dB. It can be accessed through the Derived Data Points (DDPs): SINAD_A(dB) for Channel A and SINAD_B(dB) for Channel B.

SNR (Signal-to-Noise Ratio) is defined here as the ratio of the power of the fundamental frequency to the power of the noise, converted to dB. It can be accessed through the Derived Data Points (DDPs): SNR_A(dB) for Channel A and SNR_B(dB) for Channel B.

NL (Noise Level) is defined here as the square root of the noise power. The unit of the Noise Level depends on the Y scale chosen. Apart from the value display, the noise level will also be displayed as a dotted line in the Amplitude Spectrum. Note that the noise level will always appear higher than the "apparent" noise level because it represents the total power of all noises, not the noise power at a particular frequency bin. The "apparent" noise level varies with the FFT size. The bigger the FFT size, the smaller each FFT frequency bin, the less the noise power contained in each FFT frequency bin, thus the lower the "apparent" noise level. However, the real noise level will not go down because the total noise power will not change with the FFT size. The noise level can be accessed through the Derived Data Points (DDPs): NoiseLevel_A(EU) for Channel A and NoiseLevel_B(EU) for Channel B.

ENOB (Effective Number of Bits) is defined here as: ENOB = (SINAD-1.76dB) / 6.02. It can be accessed through the Derived Data Points (DDPs): ENOB_A(Bit) for Channel A and ENOB_B(Bit) for Channel B.

The above parameters are automatically computed based on the peak frequency detected in the amplitude spectrum. In order to achieve a high measurement accuracy, test signal frequency, sampling frequency, Record Length and FFT size must be carefully chosen such that a FFT segment contains exactly an integer number of cycles of the test signal, in order to avoid any artificial noise introduced due to the spectral leakage inherent in FFT algorithm. It is recommended to use the following formula to derive the test signal frequency: N × [Sampling Frequency]/[FFT Size], where N is an integer. For example, when the sampling frequency is 44100 Hz and the FFT size is 16384, and you want a test signal of about 1 kHz, then the recommended test frequency would be 1001.293945 Hz (i.e. N=372) and the Record Length should be set to a value equal to or greater than the FFT size to avoid zero padding. The following table lists the recommended test frequencies for different

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sampling frequencies and FFT sizes. No window function is necessary if the recommended test frequency is used.

FFT Size \ Sampling Frequency	44100 Hz	48000 Hz	96000 Hz	192000 Hz
1024	990.5273438	984.375	1031.25	937.5
2048	990.5273438	1007.8125	984.375	1031.25
4096	1001.293945	996.09375	1007.8125	984.375
8192	1001.293945	1001.953125	996.09375	1007.8125
16384	1001.293945	999.0234375	1001.953125	996.09375
32768	999.9481201	1000.488281	999.0234375	1001.953125
65536	999.9481201	999.7558594	1000.488281	999.0234375
131072	999.9481201	1000.12207	999.7558594	1000.488281
262144	999.9481201	999.9389648	1000.12207	999.7558594

In many cases, however, it may not be practical to ensure a FFT segment contains exactly an integer number of cycles of the test signal (e.g. due to the round off error of the test frequency, discrepancy in ADC and DAC clock frequency…), thus a proper window function must be used to suppress the spectrum leakage, otherwise very different and inaccurate results can be obtained. The software uses Kaiser 6 as the default window function for the above parameter measurement. Other window functions that can be used here includes: Kaiser 7 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261.

Generally, if you use a sound card to generate the test signal and use the same sound card (i.e. share the same ADC and DAC clock) to measure the above parameters, then you should use the above recommended test frequencies, FFT size and sampling frequency. The Record Length should be set to a value equal to or greater than the FFT size (i.e. no zero padding) and Rectangle window function should be chosen in the Spectrum Analyzer. If the ADC and DAC do not share the same clock, Kaiser 6 window and the like is then recommended, and zero padding is allowed in this case.

To measure the above parameters, you must define a frequency band within which the total power of the signal (exclusive of the DC component), the power of the fundamental frequency, the total power of all the harmonic frequencies, and the total power of the noises are calculated. By default, the frequency band is set to 20 Hz ~20 kHz. Note that the upper limit of the frequency band must be equal to or lower than ½ of the sampling frequency. Otherwise, ½ of the sampling frequency will be used as the upper limit of the frequency band internally by the software. If you tick the harmonic order checkbox, you can further specify the maximum order of harmonics used for the calculation of the total power of the harmonic frequencies. That is, the power of a certain order of harmonic frequency will be taken into account only if it is within the frequency band specified, and it is equal to or lower than the harmonic order specified.

The “f1(Hz)” combo box specifies how the fundamental frequency f1 is determined:

(1) Peak (default)

The fundamental frequency is the peak frequency detected in each channel.

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(2) Max_A(EU), Min_A(EU), Mean_A(EU), RMS_A(EU), f1Freq_A(Hz) ~ f7Freq_A(Hz), Freq_A, RPM_A(rpm)

In Channel A, the fundamental frequency is the peak frequency detected in Channel A.

In Channel B, the fundamental frequency is the value of the DDP selected. Essentially it tracks the variation of the DDP in Channel A. Its applications include order tracking. Max_A(EU), Min_A(EU), Mean_A(EU), RMS_A(EU) are usually voltages that represent frequencies such as analog outputs from some tachometers. They can be calibrated to Hz or RPM through [Setting]>[Calibration]> "Sensor"> "Sensitivity" and "Unit".

(3) Freq_B(Hz), RPM_B(rpm)

In Channel A, the fundamental frequency is the DDP selected. Essentially it tracks the variation of the DDP in Channel B. Its applications include order tracking.

In Channel B, the fundamental frequency is the peak frequency detected in Channel B.

The options here are not symmetric with those in (2) for software simplicity.

(4) A positive value (in Hz)

The fundamental frequency in each channel is the one specified by the user. It can be used in the cases where certain harmonics have higher amplitudes than the fundamental.

The "f1 Notch" option should generally not be selected. If enabled, the power of the fundamental frequency used in THD, THD+N, SINAD, SNR, and NL calculations will be measured within a one-octave span around the fundamental frequency, in accordance with the AES17-2015 standard notch filter setting. This frequency span is significantly wider than the one typically used for these calculations, resulting in the inclusion of noise in the fundamental power measurement while reducing the measured total noise power. This effect becomes more pronounced as the fundamental frequency increases. To prevent excessive deviation, the software automatically disables this option internally when the measured fundamental frequency exceeds 1100Hz. This option is useful only when comparing results with the AES17-2015 standard notch filter method or when the spectral skirt of the fundamental frequency is very wide due to factors such as ADC and DAC using different sampling clocks, jitter, or spectral leakage.

Some recommended panel settings for the above parameter measurement are provided in the Panel Setting File (PSF) directory of the software.

The following figure shows the THD related parameters of a laptop built-in sound card measured using an external loopback cable.

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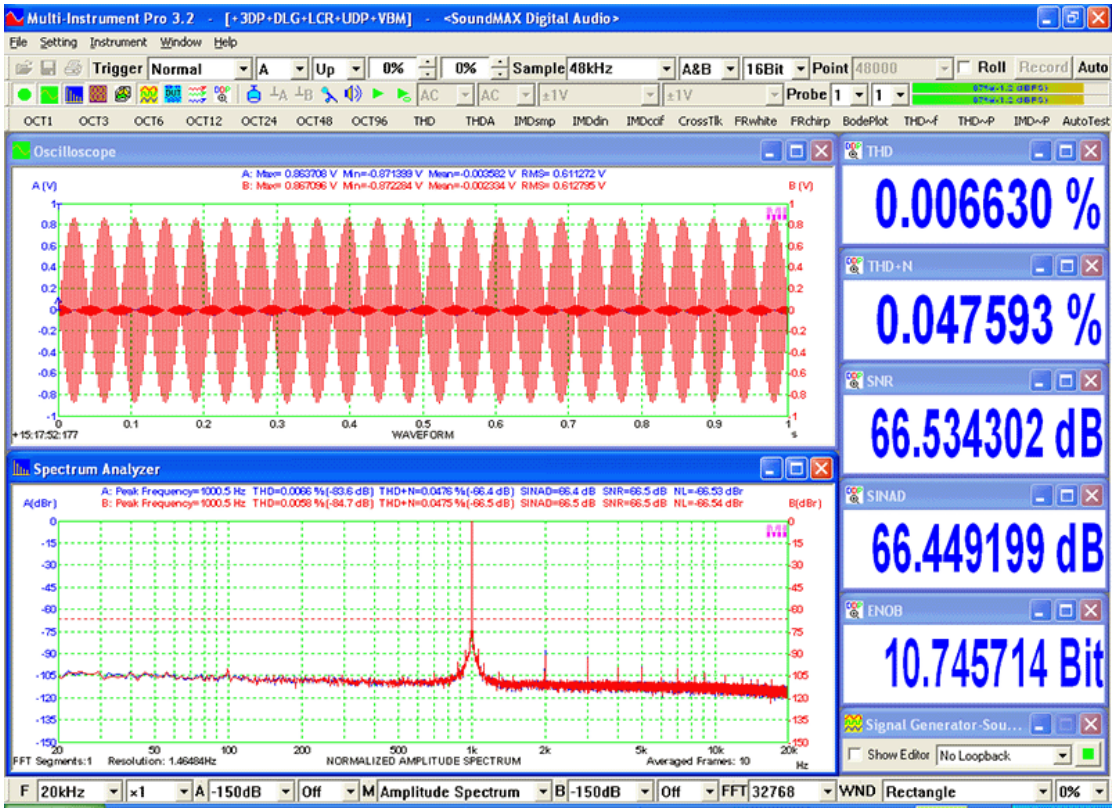
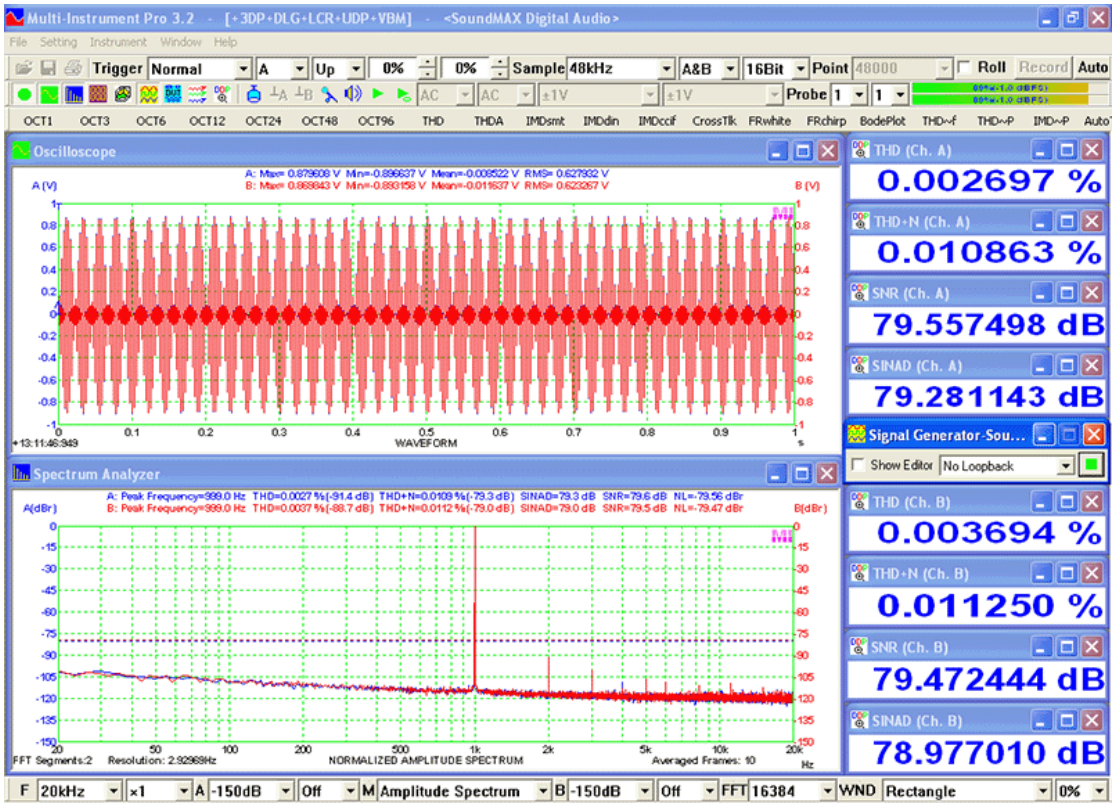
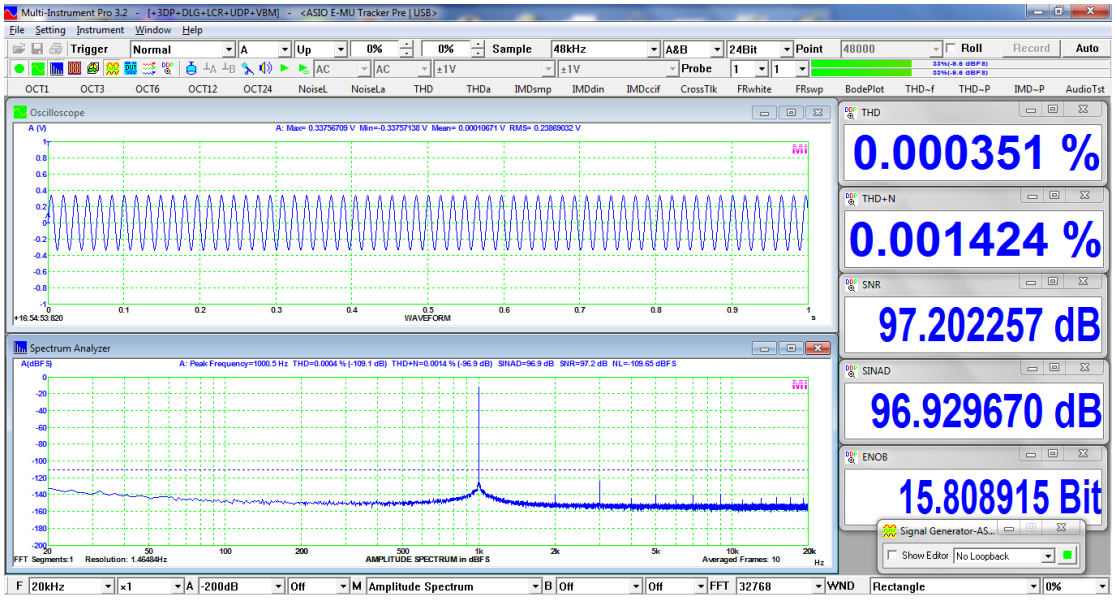


The following figures shows the THD related parameters of the same laptop built-in sound card measured via a loopback at the sound card's mixer level. As you can see that all the parameters are substantially better than the above case, due to the fact that the current measurement excluded the adverse contribution from the sound card's power amplifier (output) and mic amplifier (input) as well as the possible external noises induced via the external loopback cable.



Some professional sound cards in the market have excellent performances in audio parameter measurements. The following figure shows the loopback measurement result of an EMU Tracker Pre USB sound card. The measured THD is only 0.000351% and THD+N is only 0.001424%. Sound Card ASIO driver was used here to avoid the possible Sampling Rate Conversion (SRC) performed automatically by Sound Card MME driver which may introduce some additional distortions and noises.



A professional sound card together with the software is a very cost effective audio analyzer solution. When measuring the output of an audio power amplifier, the signal may need to be attenuated first using a linear attenuation circuit such as a simple voltage divider formed by two resistors. It should also be noted that the loopback

performance of the measuring device should be better than the performance of the Device Under Test to ensure measurement quality. The loopback results of some sound cards can be found at:

- *EMU Tracker Pre Test Report using Multi-Instrument*
  Download link:
  https://www.virtins.com/doc/D1004/EMU_Tracker_Pre_Report_D1004.pdf

- *EMU 0204 Test Report using Multi-Instrument*
  Download link:
  https://www.virtins.com/doc/D1007/EMU_0204_Report.pdf

- *Focusrite Scarlett Solo Test Report using Multi-Instrument*
  Download link:
  https://www.virtins.com/doc/Focusrite-Scarlett-Solo-Test-Report-using-Multi-Instrument.pdf

For more information on the measurement of THD and its related parameters, please refer to *Measurement of Total Harmonic Distortion and Its Related Parameters using Multi-Instrument*, download link:
https://www.virtins.com/doc/Measurement-of-Total-Harmonic-Distortion-and-Its-Related-Parameters-using-Multi-Instrument.pdf

and *Software Methods for Non-Linear Distortion Compensation and Noise Reduction in Measurements*, download link:
https://www.virtins.com/doc/Software-Methods-for-Non-Linear-Distortion-Compensation-And-Noise-Reduction-In-Measurements.pdf

3.3.2.8.3.2 IMD

Intermodulation is the result of two or more signals of different frequencies being mixed together, forming additional signals at frequencies that are not in general, at harmonic frequencies (integer multiples) of either. Three types of Intermodulation Distortion (IMD) can be measured: SMPTE/DIN IMD, CCIF2 IMD, CCIF3 IMD. Note that the fourth option, DIM (Dynamic Intermodulation distortion), will be introduced separately later.

SMPTE/DIN IMD is the most common IMD measurement. SMPTE standard RP120-1983 and DIN standard 45403 are similar. Both specify a two-sinewave test signal consisting of a large amplitude low-frequency tone linearly mixed with a high-frequency tone at ¼ the amplitude of the low frequency tone. SMPTE specifies 60 Hz and 7 kHz mixed at 4:1. The DIN specification allows several choices in both frequencies, with 250 Hz and 8 kHz being the most common. The IMD under this category is defined here as the square root of the ratio of the power of the sidebands to the power of the upper frequency. It is displayed in both percentage (%) and dB. The sidebands used for this type of IMD calculation are f_H-f_L, f_H+f_L, f_H-2f_L, f_H+2f_L, where f_H and f_L are the high frequency and low frequency of the test signal respectively. IMD measured in this way is also called Modulation IMD (MOD).

CCIF IMD, also called Twin-Tone IMD, or Difference-Tone IMD, is another common IMD measurement. The test specifies two equal-amplitude closely spaced high frequency signals. The IMD under this category is defined here as the square

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root of the ratio of the power of the intermodulation distortion products to the square of the RMS amplitude sum of the two test frequencies. It is displayed in both percentage (%) and dB. It has two sub-types: CCIF2 IMD and CCIF3 IMD. IMD measured in this way is also called Difference Frequency Distortion (DFD). It is described in IEC60118 and IEC60268.

For CCIF2 IMD, the commonly used frequencies are: 19 kHz and 20 kHz. The intermodulation distortion products used for this type of IMD calculation is: f_H-f_L, i.e. only the low-frequency second-order product is used.

For CCIF3 IMD, the commonly used frequencies are: 13 kHz and 14 kHz, 14 kHz and 15 kHz, or 15 kHz and 16 kHz. The intermodulation distortion products used for this type of IMD calculation is: f_H-f_L, $2f_L$- f_H, $2f_H$-f_L, i.e. up to the third-order products are used.

To conduct the above IMD tests, you can use the MultiTone generation function of the Signal Generator to generate the test signal. Some sample test tones for IMD measurements are provided under the Tone Configuration File (TCF) directory of the software. It is also possible to use other signal generation source such as a CD, as the software is able to detect the two test frequencies automatically.

Similar to the THD measurement, the IMD measurement is also very sensitive to spectral leakage. According to the specified two test frequencies and their intermodulation terms, it is almost impossible to ensure that each FFT segment contains integer numbers of cycles of all these individual test frequencies. Therefore, a window function must be used. The software uses the Kaiser 6 window by default for the IMD tests. Other window functions that can be used includes: Kaiser 7 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261. Some recommended panel settings for different IMD measurements are provided in the Panel Setting File (PSF) directory of the software.

The following figure illustrates the SMPTE IMD of a laptop built-in sound card measured using an external loopback cable.

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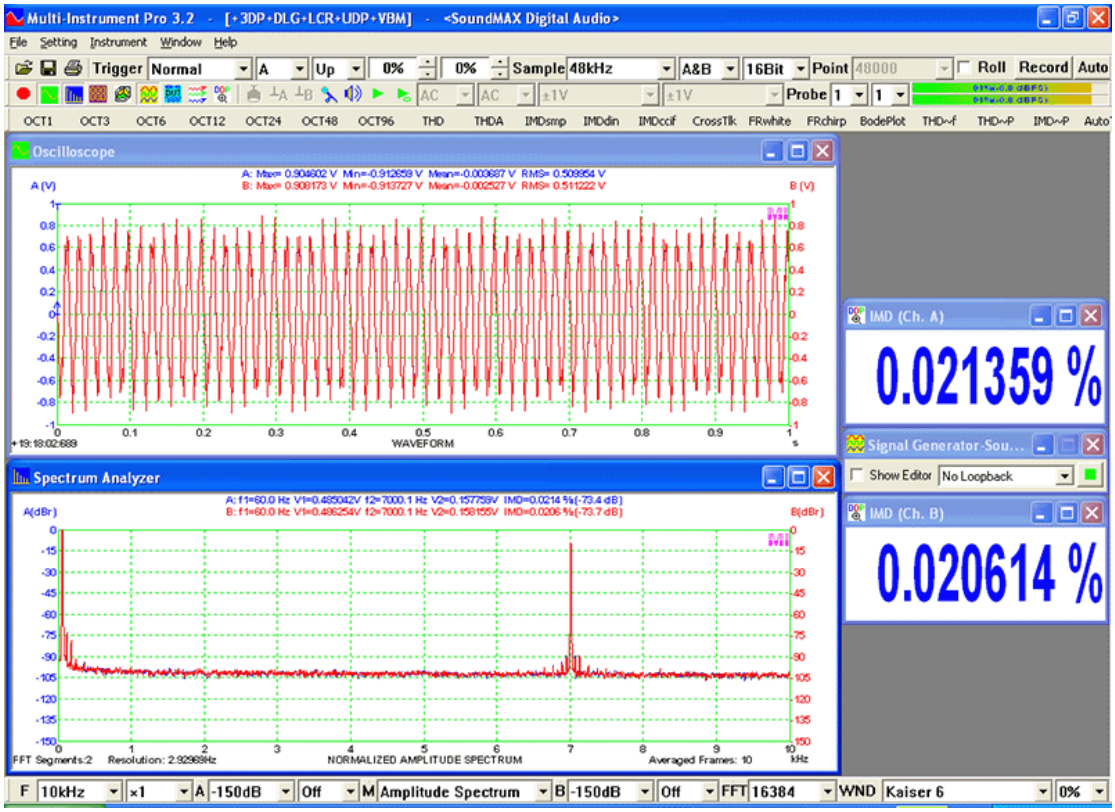
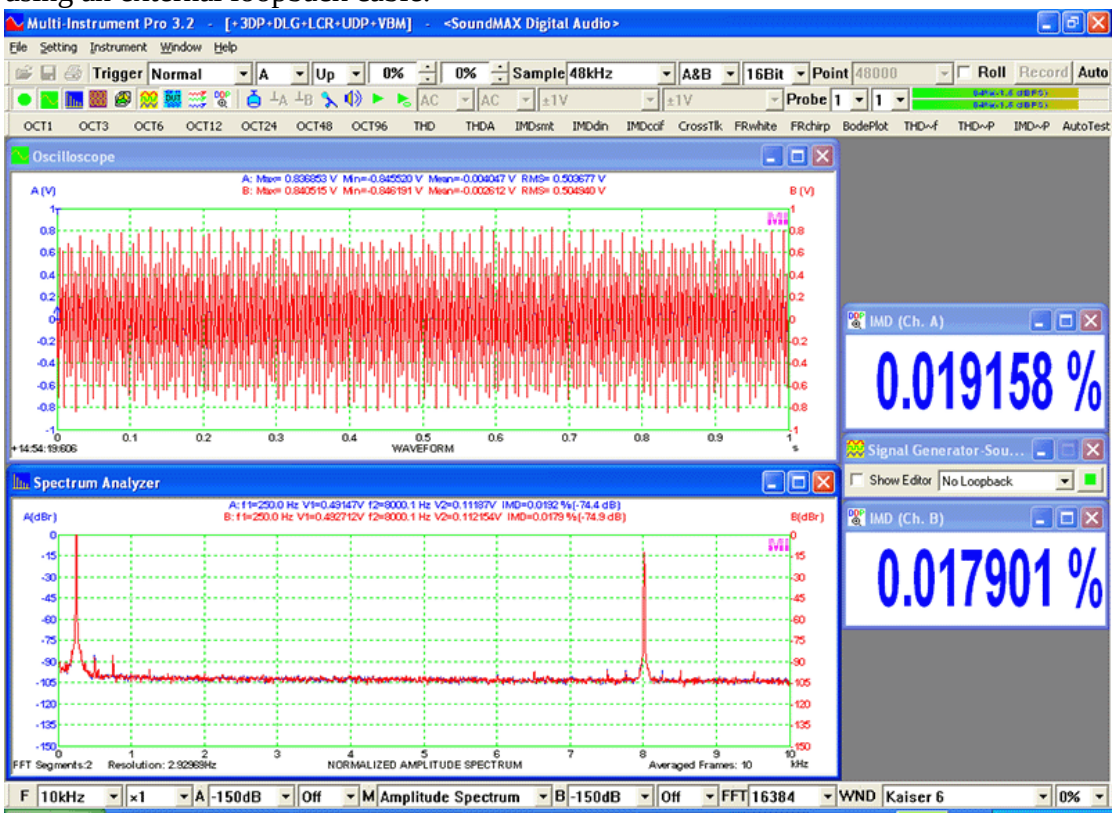


The following figure illustrates the DIN IMD of a laptop built-in sound card measured using an external loopback cable.



The following figure illustrates the CCIF2 IMD of a laptop built-in sound card measured using an external loopback cable.

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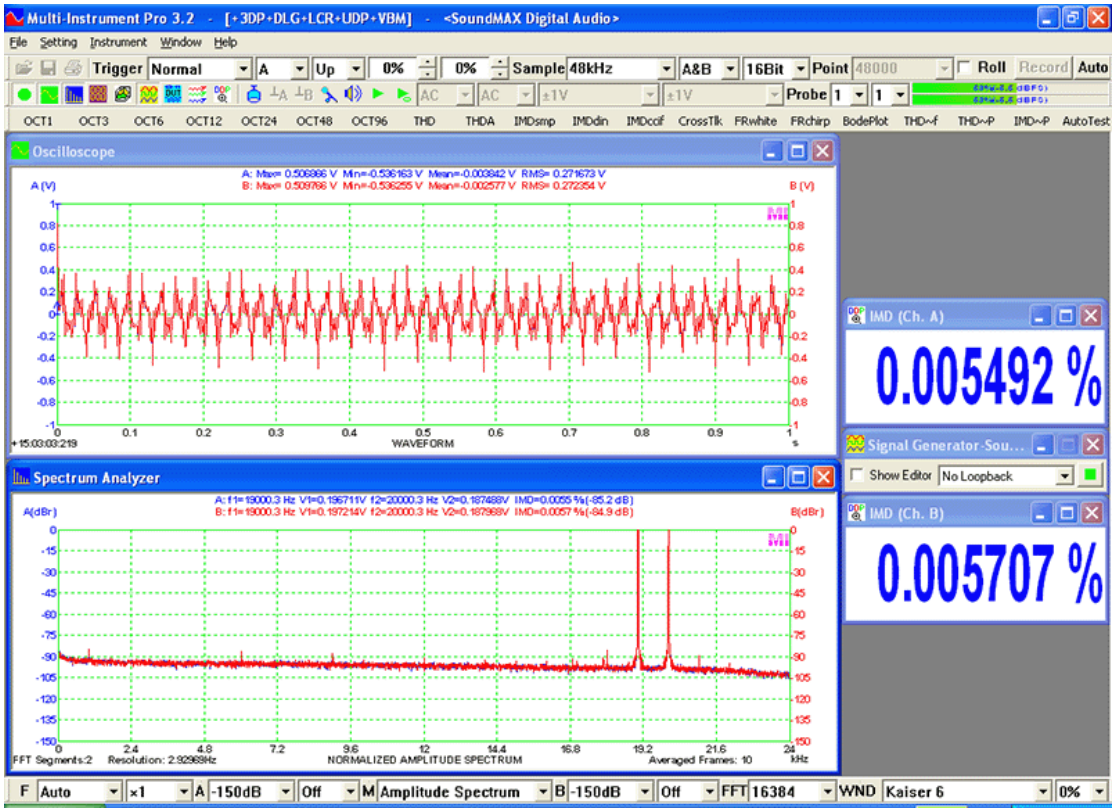
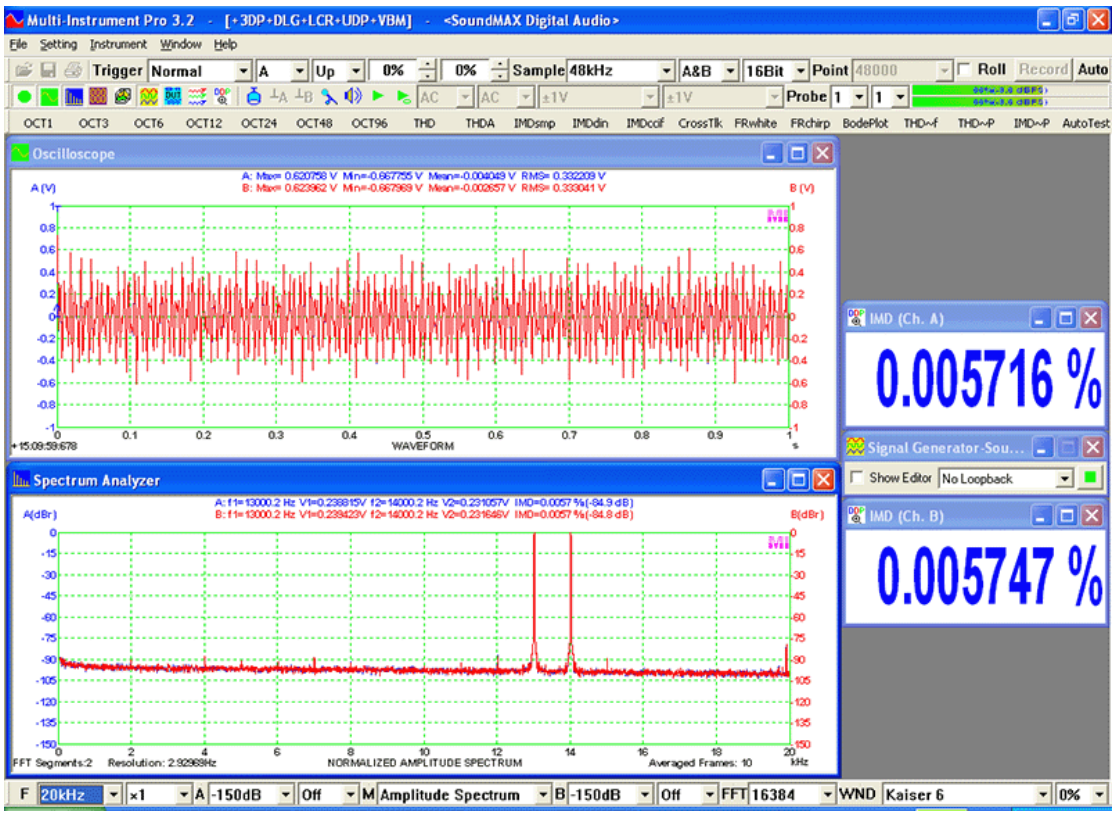
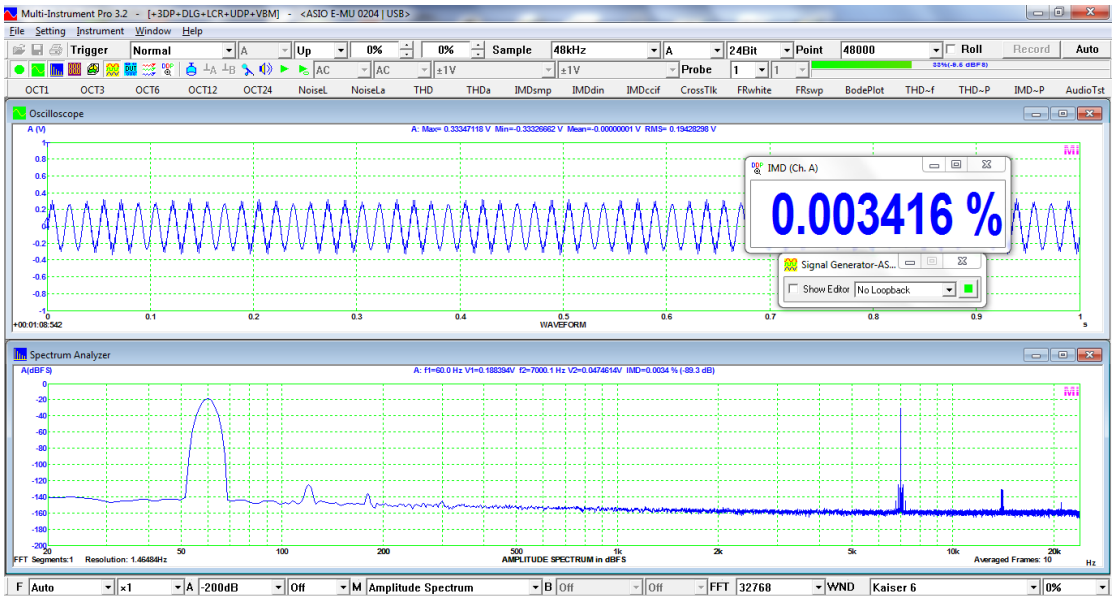


The following figure illustrates the CCIF3 IMD of a laptop built-in sound card measured using an external loopback cable.



Some professional sound cards in the market have excellent performances in audio parameter measurements. The following figure illustrates the SMPTE IMD of an EMU Tracker Pre USB sound card measured using an external loopback cable.



In addition to the IMD values, the two test tone frequencies and their RMS values measured will also be displayed. The test tone frequencies here are measured with a sub-FFT-bin-size accuracy. IMD can be accessed through the DDPs: IMD_A(%) and IMD_A(dB) for Channel A and IMD_B(%) and IMD_B(dB) for Channel B.

For more information on IMD measurement, please refer to *Measurements of Various Intermodulation Distortions (IMD, TD+N, DIM) using Multi-Instrument*.

Download link:
https://www.virtins.com/doc/Measurements-of-Various-Intermodulation-Distortions-IMD-TD+N-DIM-using-Multi-Instrument.pdf

3.3.2.8.3.3 Bandwidth

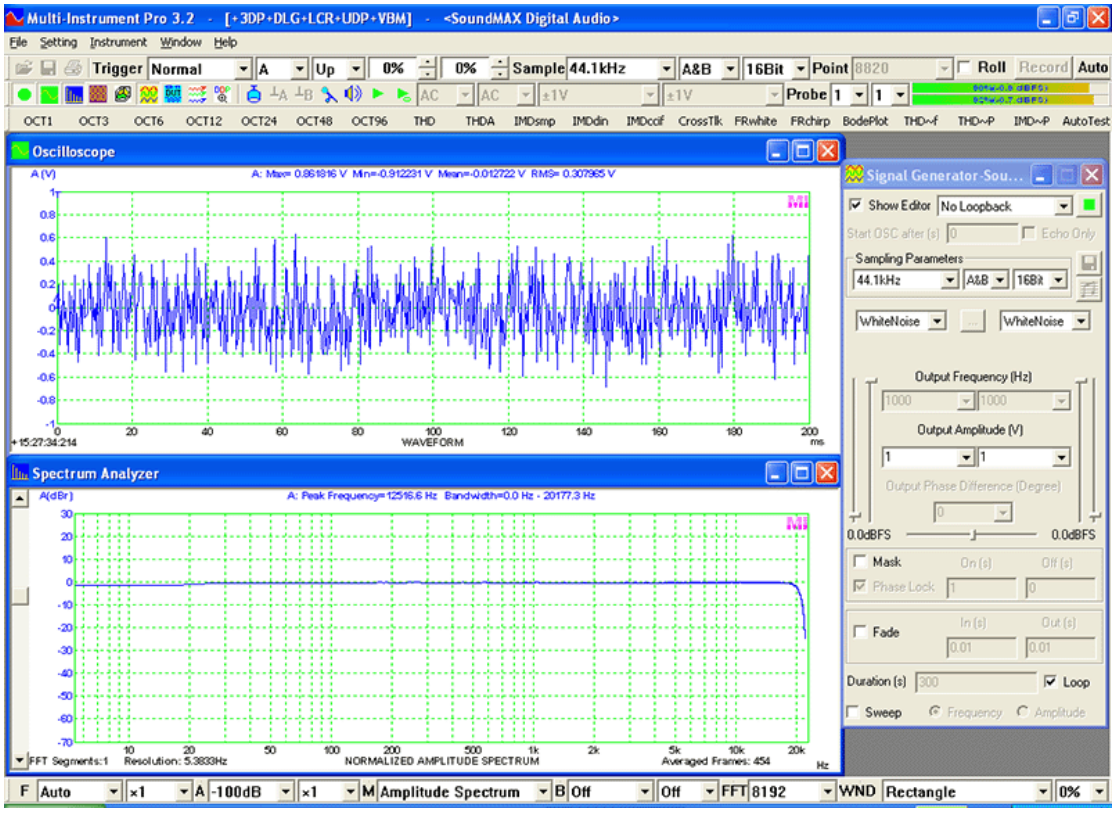
Bandwidth (-3dB) is defined here as the difference between the upper and lower cutoff frequencies where the magnitude frequency response drops by –3 dB compared with the highest peak. To obtain accurate bandwidth measurement, the magnitude frequency response must be measured accurately with little fluctuation along the curve. You can use the aforementioned “Smoothing via Moving Average” function to make the curve smoother. Alternatively, you can always use the cursor reader to obtain the bandwidth manually.

In most of the cases, the bandwidth and flatness of the magnitude frequency response of a system is of more interest than the absolute gain of the output over the input at different frequencies (i.e. the gain graph in Bode Plot). The former is easier to measure as it measures only the output signal and does not need to measure the input signal. There are quite a few ways to do it:

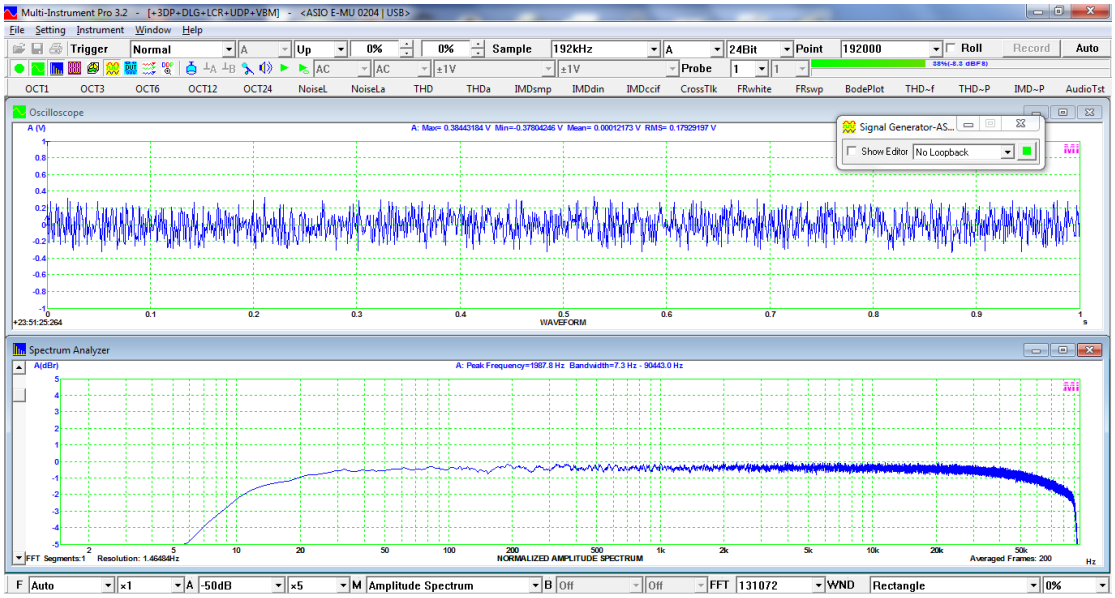
(1) White noise stimulation + inter-frame average in Spectrum Analyzer
(2) Linearly swept sine stimulation + inter-frame peak hold in Spectrum Analyzer
(3) Linearly swept sine stimulation + it is captured and analyzed in one frame in Spectrum Analyzer
(4) Pink noise stimulation + octave analysis and inter-frame average in Spectrum Analyzer
(5) Multi-tones stimulation with each tone with an equal amplitude at the center frequency of each octave band + octave analysis in Spectrum Analyzer
(6) Logarithmically swept sine stimulation + octave analysis in Spectrum Analyzer
(7) Impulse stimulation + Spectrum Analyzer
(8) MLS stimulation + inter-frame average in Spectrum Analyzer
(9) Periodic white noise stimulation + Spectrum Analyzer
(10) Periodic pink noise stimulation + octave analysis in Spectrum Analyzer
…

Some recommended panel settings for the above parameter measurement are provided in the Panel Setting File (PSF) directory of the software.

The following figure illustrates the Bandwidth of a laptop built-in sound card measured using the mixer-level loopback. White noise was used as the stimulus and 454 frames were averaged.



The following figure illustrates the bandwidth of an EMU Tracker Pre USB sound card measured using an external loopback cable. The measured bandwidth is about 7.3Hz ~ 90443Hz when the sampling rate is 192 kHz. White noise was used as the stimulus and 200 frames were averaged.



Bandwidth can be accessed through these DDPs: BandWidthLowLimit_A(Hz) and BandWidthHighLimit_A(Hz) for Channel A, and BandWidthLowLimit_B(Hz) and BandWidthHighLimit_B(Hz) for Channel B. They can also be used to derive Quality Factor and Damping Ratio.

To describe the degree of flatness of the magnitude frequency response within a specified frequency band, four DDPs are provided per channel:

InBandMinRMS_A(EU): the minimum in the specified frequency band in Ch. A.
InBandMinF_A(Hz): the frequency at which the minimum is located in Ch. A.
InBandMaxRMS_A(EU): the maximum in the specified frequency band in Ch. A.
InBandMaxF_A(Hz): the frequency at which the maximum is located in Ch. A.

Similarly, InBandMinRMS_B(EU), InBandMinF_B(Hz), InBandMaxRMS_B(EU) and InBandMaxF_B(Hz) are used for Ch. B.

3.3.2.8.3.4 Crosstalk

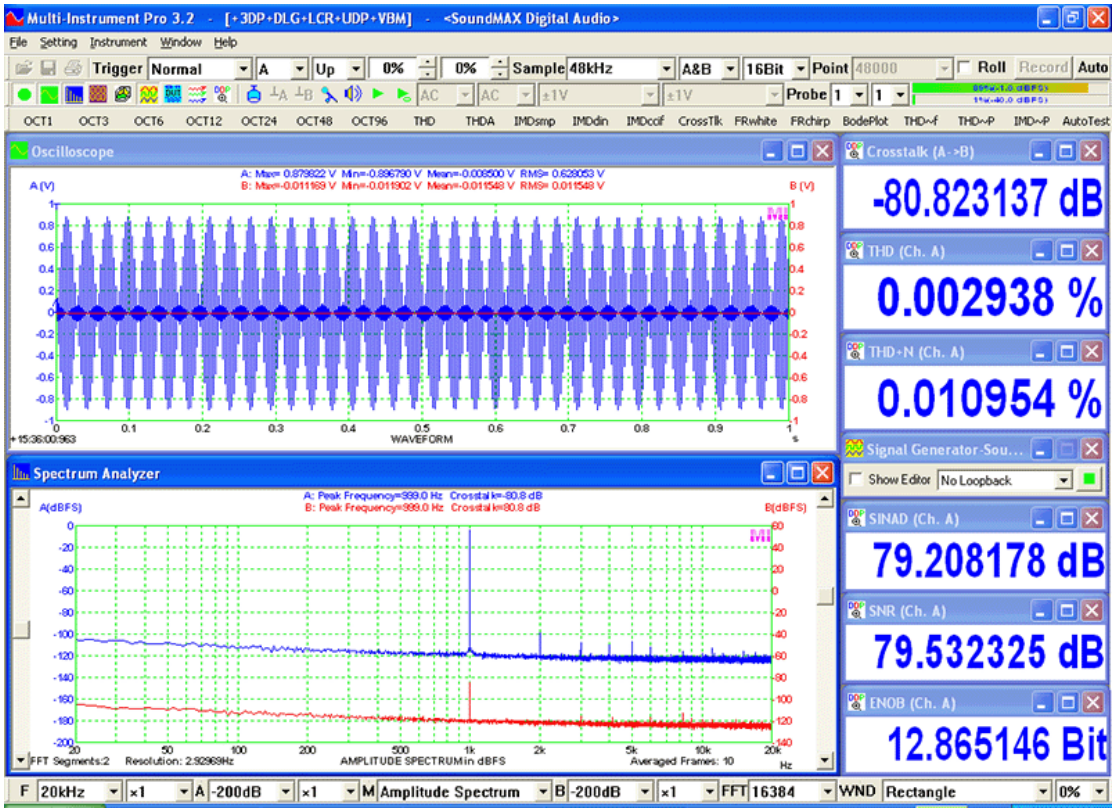
Crosstalk occurs when signal from one channel leaks into another channel. Crosstalk is defined here as the ratio of the power of the fundamental frequency between the channel without the test signal and the channel with the test signal. It is expressed in dB. The input of the channel without the test signal must be terminated to the signal ground to prevent external noises from entering. The test signal used should be a single-frequency signal and the crosstalk value measured is for that frequency only. The test tone frequency is measured with a sub-FFT-bin-size accuracy.

In the Spectrum Analyzer, two crosstalk values will be displayed, one is from Channel A to Channel B, the other is from Channel B to Channel A. To measure the former one, the test signal should be injected into Channel A, and to measure the latter one, the test signal should be injected into Channel B. The test signal is normally a sine wave. It is possible to inject two test signals with different frequencies

(can be very close with each other but not the same) into the two channels respectively so as to measure the mutual crosstalks simultaneously.

To minimize the effect of spectral leakage on the crosstalk measurement, the software uses the Kaiser 6 window by default. Other window functions that can be used includes: Kaiser 7 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261.

The following figure illustrates the Crosstalk at 1 kHz from Channel A to Channel B in a laptop built-in sound card measured using the mixer-level loopback.



The following figure illustrates the crosstalk at 1 kHz from Channel A to Channel B of an EMU Tracker Pre USB sound card measured using an external loopback cable. The crosstalk is about -109dB.

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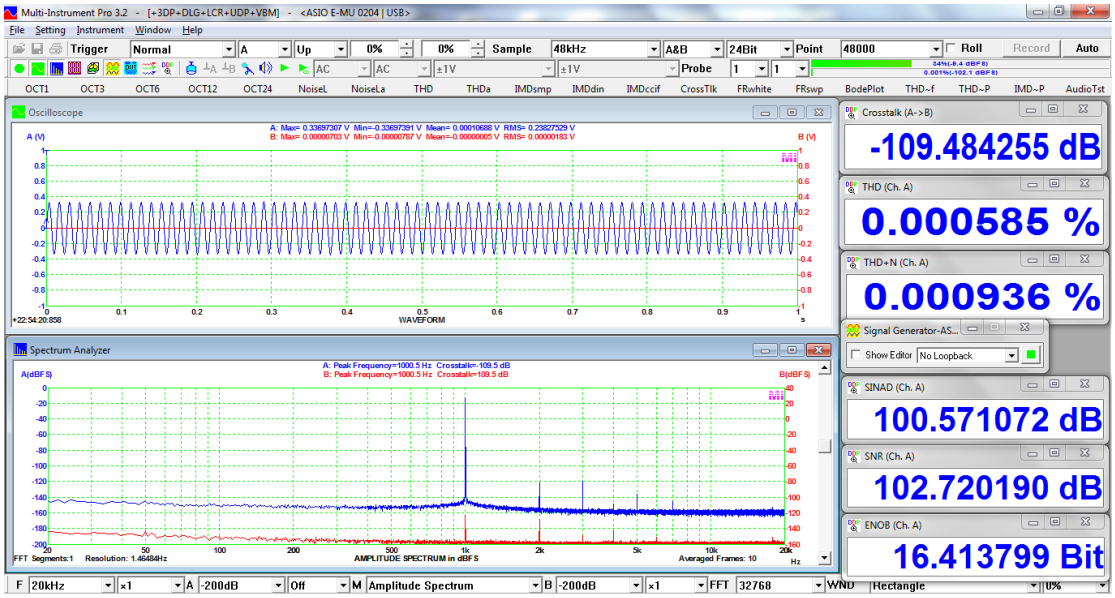


Crosstalk can be accessed through these DDPs: CrossTalkAB(dB) for crosstalk from Channel A to Channel B and CrossTalkBA(dB) for crosstalk from Channel B to Channel A. Some recommended panel settings for crosstalk measurement are provided in the Panel Setting File (PSF) directory of the software.

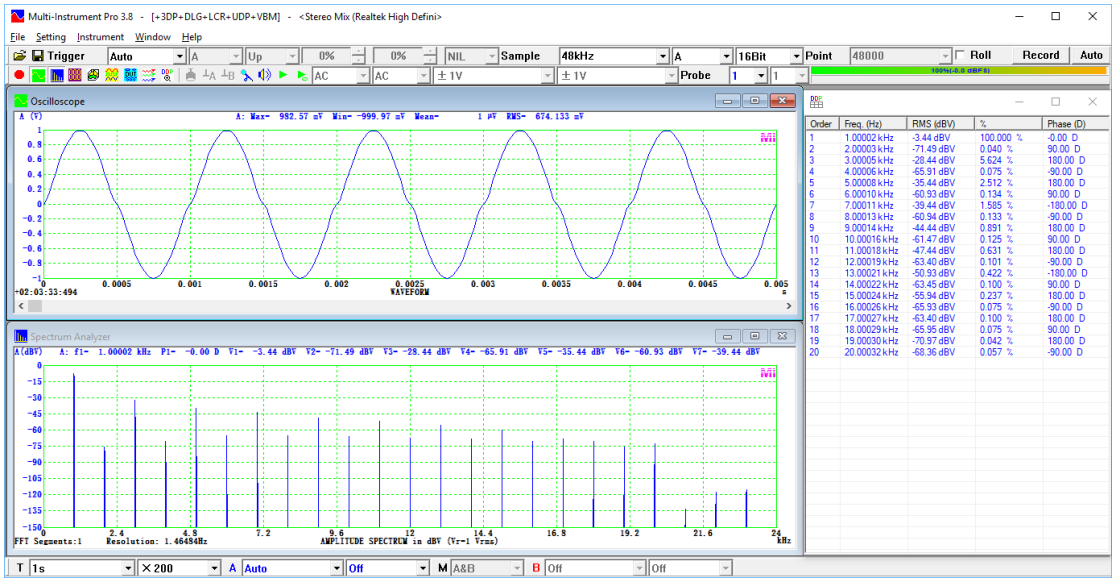
3.3.2.8.3.5 Harmonics

When this option is selected, the software will analyze up to the 100th harmonic and display the fundamental frequency and its RMS value as well as the RMS values of its 2nd ~ 7th harmonics on the top of the spectrum graph. The fundamental frequency is the peak frequency detected (with a sub-FFT-bin-size accuracy) if "f1(Hz)" is set to "Peak". Please refer to the previous THD section for detailed description of all options of "f1(Hz)". The fundamental frequency and its harmonic frequencies can be accessed through the DDPs: f1Freq_A(Hz)~f7Freq_A(Hz) for Channel A, and f1Freq_B(Hz)~f7Freq_B(Hz) for Channel B. Their respective RMS values can be accessed through the DDPs: f1RMS_A(EU) ~ f7RMS_A(EU) for Channel A, and f1RMS_B(EU) ~ f7RMS_B(EU) for Channel B. It should be noted that the fundamental frequency and its RMS value can always be accessed in Amplitude Spectrum Display regardless of the parameter measurement option selected. To minimize the effect of spectral leakage on the fundamental and harmonic RMS value measurement, the software uses the Kaiser 6 window by default. Other window functions that can be used includes: Kaiser 7 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261.

The phase angle of the fundamental frequency will also be displayed. It can be accessed through these DDPs: f1Phase_A(D) for Channel A and f1Phase_B(D) for Channel B. This option can be used to perform quantitative phase difference measurement between the two channels, in contrast to the qualitative phase difference measurement through Lissajous pattern. Note that the phase angle is of a sine function.

The fundamental and harmonic frequencies, RMS values and phases can also be accessed through the following DDP Arrays: fnFreq_A(Hz)_Array[?],

fnRMS_A(EU)_Array[?] and fnPhase_A(D)_Array[?] for Channel A, and fnFreq_B(Hz)_Array[?], fnRMS_B(EU)_Array[?] and fnPhase_B(D)_Array[?] for Channel B. The "?" should be replaced by the actual order of harmonic-1. That is, "0" for fundament, "1" for the 2nd harmonic and so on. The maximum index number allowed is "99". Multi-Instrument features a specially designed algorithm which is able to measure the phase of a harmonic accurately even when its amplitude is only 0.01% of that of the fundamental or the signal itself (when sampling bit resolution =16). The following figure shows an example of harmonic analysis. DDP array viewer is used to display the frequency, RMS value and phase of each harmonic. DDP array viewer will be introduced later.



Two panel setting examples can be found at:

Pro\Harmonic_FFT32768_SR48000_SingleChannel.psf
Pro\Harmonic_FFT32768_SR48000.psf

3.3.2.8.3.6 Energy in User Defined Frequency Bands

When this option is selected, the software can analyze up to 100 user defined frequency bands and display the RMS values of up to 8 of them on the top of the spectrum graph. These values can be accessed through these DDPs: fBand1RMS_A(EU)~ fBand8RMS_A(EU) for Channel A, and fBand1RMS_B(EU)~ fBand8RMS_B(EU) for Channel B. Note that the maximum number of bands that can be defined here is actually 100. Their RMS values can be accessed through the DDP arrays: fnRMS_A(EU)_Array[?] for Channel A and fnRMS_B(EU)_Array[?] for Channel B. The"?" should be replaced by the actual frequency band number starting from 0. The maximum allowed number is "99".

The start and end frequencies of each frequency band can be specified and then the specified frequency band can be added into the list box. Please note that the calculation here is based on narrow band FFT and thus wide band octave analysis should not be used. The finer the FFT frequency resolution, the more accurate the

results. It is possible to save the frequency band configuration as a CSV text file so that it can be loaded later for convenience.

If the X scale is set to an octave scale, this option will be disabled. Under octave analysis, it is possible to obtain the RMS values of a particular octave band through the DDP arrays: oBandRMS_A(EU)_Array[?] for Channel A and oBandRMS_B(EU)_Array[?] for Channel B. The"?" should be replaced by the actual octave band number starting from 0. Please note that when using a cursor reader to read the RMS value of an octave band from the screen, the octave band number "N" displayed starts from 1 instead of 0. Therefore, if you want to access the octave band number N=31 in Channel A on the screen, for example, oBandRMS_A(EU)_Array[30] should be used.

The concept of DDP array greatly expands the number of DDPs without defining additional DDPs. It is possible to access multiple elements of the array simultaneously. One efficient way to view the DDP array is to use the DDP array viewer which will be introduced later.

3.3.2.8.3.7 Peaks

When this option is selected, the software can detect up to 100 peaks and display up to 7 peak frequencies and their RMS values on the top of the spectrum graph. They can be accessed through these DDPs: f1Freq_A(Hz)~f7Freq_A(Hz) and f1RMS_A(EU)~f7RMS_A(EU) for Channel A, and f1Freq_B(Hz)~f7Freq_B(Hz) and f1RMS_B(EU)~f7RMS_B(EU) for Channel B. You can specify the number of peaks to be displayed and the deadband between adjacent peaks. A peak spectral line in frequency domain is normally surrounded by a few sub-peaks. The deadband is used to avoid detecting these surrounding sub-peaks as peaks. Within the dead band specified, only one peak can be detected. In other words, two adjacent peaks must be at least one deadband apart. If the deadband specified is narrower than the main lobe of the window function used, then the width of the main lobe, which is usually in the range of 15~30 FFT frequency resolution, will be used for this purpose instead by the software. The detected peaks can be sorted by either magnitude or frequency. It should be noted that the maximum number of peaks that can be detected here is actually 100. The frequencies, RMS values and phases of the peaks can be accessed through these DDP arrays: fnFreq_A(Hz)_Array[?], fnRMS_A(EU)_Array[?] and fnPhase_A(D)_Array[?] for Channel A, and fnFreq_B(Hz)_Array[?], fnRMS_B(EU)_Array[?] and fnPhase_B(D)_Array[?] for Channel B. The "?" should be replaced by the actual peak number starting from 0. The maximum allowed number is "99". One efficient way to view the DDP array is to use the DDP array viewer which will be introduced later.

To minimize the effect of spectral leakage on the peak RMS value measurement, the software uses the Kaiser 6 window by default. Other window functions that can be used includes: Kaiser 7 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261. All the peak frequencies here are measured with a sub-FFT-bin-size accuracy.

SFDR (Spurious Free Dynamic Range) is also calculated here. It is defined here as the ratio of the power of the highest peak to that of the second highest peak, converted to dB. It can be accessed through the Derived Data Points (DDPs): SFDR_A(dB) for Channel A, and SFDR_B(dB) for Channel B.

3.3.2.8.3.7.1 Total Distortion + Noise (TD+N)

Nonlinear distortion measurements depend heavily on the spectral content and level of the stimulus. Single-tone THD and Dual-Tone IMD measurements are often criticized for being far too simple to represent a real-world audio signal such as music or speech. A multitone signal contains more than two frequencies and is a closer resemblance of a real-world audio signal. It is usually used as a test signal for fast frequency response measurements (a.k.a linear distortion measurements). Meanwhile, it can also be used as a test signal for nonlinear distortion measurements. It excites both harmonic and intermodulation distortions in a DUT simultaneously.

Total Distortion + Noise (TD+N) is defined here as the square root of the ratio of the power of the total distortions plus noise to the power of all the fundamentals, in other words, the square root of the ratio of the total power less the power of all the fundamentals to the power of all the fundamentals. It is expressed in both percentage (%) and dB. It can be accessed through the Derived Data Points (DDPs): TDN_A(%) and TDN_A(dB) for Channel A, and TDN_B(%) and TDN_B(dB) for Channel B. The frequency range of TD+N calculation can be set via [Spectrum Analyzer Processing]>"Parameter Measurement">"Range (Hz)". The default range is 20Hz~20kHz. Multi-Instrument detects the fundamentals in the multitone response automatically using its peak detection function. Therefore it is crucial that the "Number of Peaks" and "Deadband" are set correctly according to the stimulus's multitone configuration.

The fundamental frequencies configured in the multitone signal should be carefully chosen such that they do not coincide with their harmonics and intermodulation products. The number of harmonic and intermodulation tones grows extremely fast with the number of stimulus tones. Therefore, the number of stimulus tones is usually kept below 32. Their phases should be selected to minimize the crest factor of the overall signal. This can usually be achieved by randomizing or collectively optimizing their initial phases.

The following figure shows the TD+N measurement results of an RTX6001 audio analyzer. The multitone stimulus used consists of 250Hz, 1938Hz, 3626Hz, 5314Hz, and 7002Hz tones.

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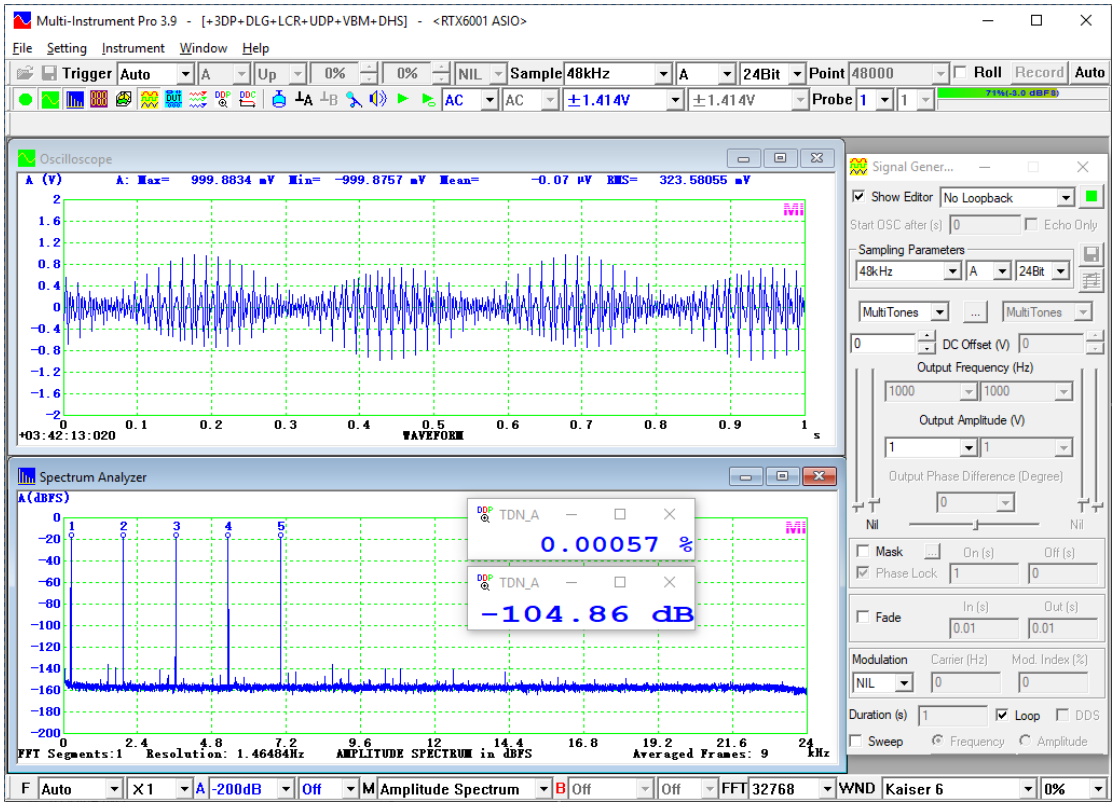


For more information on TD+N measurement, please refer to *Measurements of Various Intermodulation Distortions (IMD, TD+N, DIM) using Multi-Instrument*.

Download link:
https://www.virtins.com/doc/Measurements-of-Various-Intermodulation-Distortions-IMD-TD+N-DIM-using-Multi-Instrument.pdf

3.3.2.8.3.8 Wow and Flutter

Flutter, wow, drift, and "scrape flutter" are all forms of distortion in analog recording and reproducing systems that use a moving medium. These are caused by undesired frequency modulation introduced into the signal by an irregular motion of the recording medium during the recording, duplicating, and reproducing processes. The measurement of wow and flutter quantifies the amount of 'frequency wobble' (caused by speed fluctuations) present in subjectively valid terms. Different standards exist for wow and flutter measurement. AES6-2008 (r2012) is used here. It is compatible with IEC 60386, IEEE Std-193, CCIR 409-2, and DIN 45507. In particular, "Two Sigma" method is used.

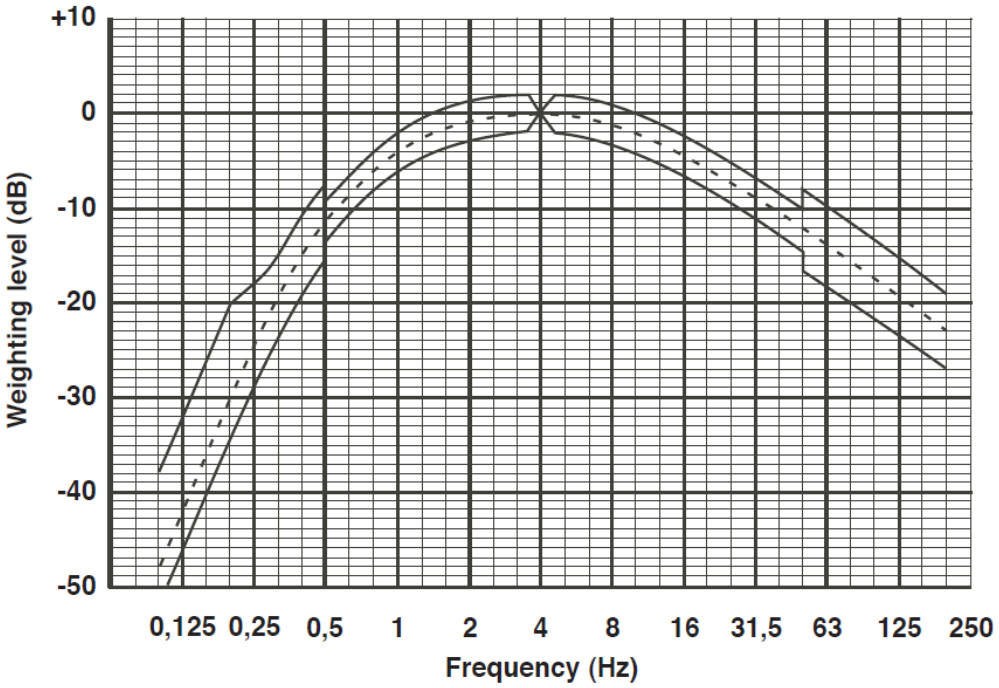
Drift refers to frequency modulation of the signal in the range below approximately 0.5 Hz resulting in distortion which may be perceived as a slow changing of the average pitch.

Wow refers to frequency modulation of the signal in the range of approximately 0.5 Hz to 6 Hz resulting in distortion which may be perceived as a fluctuation of pitch of a tone or program.

Flutter refers to frequency modulation of the signal in the range of approximately 6 Hz to 100 Hz resulting in distortion which may be perceived as a roughening of the sound quality of a tone or program.

"Scrape Flutter" refers to frequency modulation of the signal in the range above approximately 100 Hz, which is caused by stick-slip motion (stiction) of the tape. It results in distortion which may be perceived as a noise added to the signal – that is, a noise not present in the absence of a signal. It is also called "friction noise" or "stiction noise".

During wow and flutter measurement, a test frequency, usually 3150Hz, is used. The measured signal is demodulated first to obtain the instantaneous frequency deviation from that test frequency. The variation of the instantaneous frequency deviation is then weighted according to the subjective perception of human ears. The weighting curve is shown as follows. It can be seen that the "Drift" and "Scrape Flutter" are greatly suppressed by the weighting factor, leaving only the "Wow and Flutter" prominent. An unweighted option is also provided in the software.



Then "Two Sigma" statistical analysis method is used to find the weighted peak flutter (%). The cumulative time for which the instantaneous speed deviations (%) exceeds the weighted peak flutter (%) in the positive or negative direction is equal to 5% of the given time interval. The peak wow and flutter value can be accessed through the Derived Data Points (DDPs): WowAndFlutter_A(%) for Channel A, and WowAndFlutter_B(%) for Channel B. In addition, RMS wow and flutter, RMS wow, RMS flutter and RMS drift are also measured. They can be accessed through WowAndFlutterRMS_A(%), WowRMS_A(%), FlutterRMS_A(%), DiftRMS_A(%) for Channel A and a similar set of DDPs for Channel B. Their frequency ranges are as follows.

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1) Peak and RMS Wow and Flutter, 0*~400Hz (weighted), 0*~0.4×[Test Frequency] Hz (unweighted)
2) RMS Wow, 0.5~6Hz (weighted and unweighted)
3) RMS Flutter, 6~400Hz (weighted), 6~0.4×[Test Frequency] Hz (unweighted)
4) RMS Drift, 0.05*~0.5 Hz (unweighted always)

*The lowest frequency is equal to 1/[Sampling Duration]. For example, if the [Sampling Duration] is 10 seconds, then the lowest frequency measurable is 0.1 Hz.

It should be noted that for this measurement, the Record Length of the Oscilloscope must be greater than 5 s (corresponding to an achievable frequency resolution of 1/5 = 0.2 Hz) in order to achieve a sufficient accuracy. The test frequency does not have to be 3150 Hz as the software can detect the average signal frequency automatically, based on either the frequency peak detected in the Spectrum Analyzer or the frequency counted by the frequency counter of the Multimeter. The latter will take precedence if the Multimeter is opened and set to frequency counter mode. A FM signal spans a frequency range and the peak frequency in that range may not be located at the center. This measurement error in the Spectrum Analyzer usually increases with the degree of wow and flutter. Typically, for 1% wow and flutter, the error is less than 1%. Thus, there is generally no need to correct this error. On the other hand, the frequency counted by the frequency counter of the Multimeter reflects the true average frequency. Pre-configured panel setting files are available for both methods. When the wow & flutter is lower than 1%, there is little difference between these two methods.

Other than the average frequency detection, which may be affected by the settings in the Spectrum Analyzer or frequency counter of the Multimeter, the rest of wow and flutter measurement algorithm is independent and self-contained.

The following figure shows a 30s-long 3150Hz carrier frequency modulated by 20Hz with a maximum frequency deviation of 3.15Hz. The theoretical unweighted peak wow and flutter value is thus $3.15 / 3150 \times \sin(0.95 \times 90°) = 0.0997\%$. The theoretical weighted peak value is then $0.508 \times 0.0997\% = 0.0506\%$. This value is accurately measured by the software as shown below.

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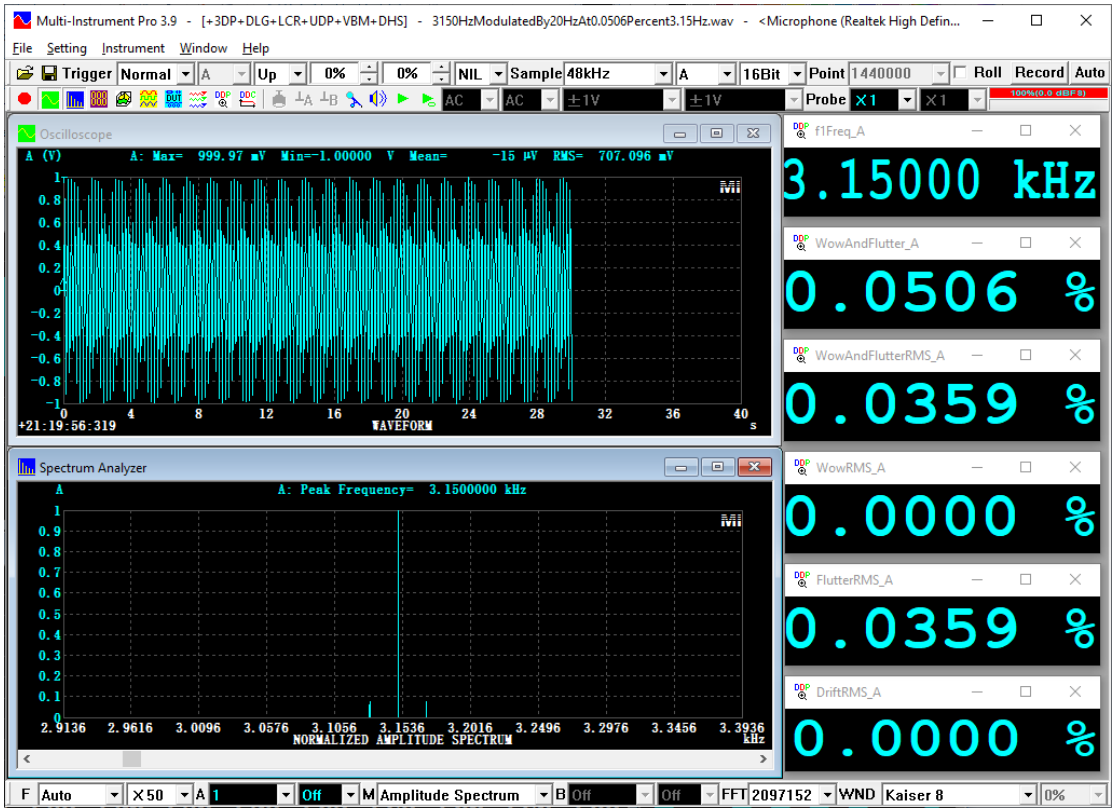


Four panel setting examples can be found at:
psf\Pro\WeightedPeakWowAndFlutter_10s.psf
psf\Pro\UnweightedPeakWowAndFlutter_10s.psf
psf\Pro\WeightedPeakWowAndFlutterFrequenceCounter_10s.psf
psf\Pro\UnweightedPeakWowAndFlutterFrequenceCounter_10s.psf

Please refer to the article *Wow and Flutter Measurement using Multi-Instrument* at https://www.virtins.com/doc/Wow-and-Flutter-Measurement-using-Multi-Instrument.pdf for more information.

It should be noted that the wow and flutter measurement here requires the carrier signal to be sinusoidal. If it is not, modulation domain analysis for the measurement of jitter statistics described later in the Multimeter chapter should be used.

3.3.2.8.3.9 Non-Coherence

This option is used to switch between the coherence and non-coherence function display modes when the View Type of the Spectrum Analyzer is set to Coherence Function. The Total Non-Coherent Distortion and Noise (TNCD) is calculated only under the non-coherence function display mode. Please refer to the section for Coherence / Non-Coherence Function for details.

3.3.2.8.3.10 GedLee Metric

It has long been known that the conventional nonlinearity metrics, such as THD and IMD, don't correlate well with human auditory perception. A number of new nonlinearity metrics have been proposed by many researchers in the past, attempting

to take into account the subjective assessment in addition to the objective one. Among them, a notable one is GedLee Metric, proposed by Dr. E.R.Geddes and Dr. L.W.Lee in their paper "*Auditory Perception of Nonlinear Distortion - Theory*," presented at the 115th Convention of the Audio Engineering Society, October, 2003.

GedLee metric (Gm) is a perceptually weighted quantitative measure of the nonlinearity of an ideal memory-less (static) nonlinear system, which has no frequency dependence or memory in time and thus has an infinite bandwidth. In such a system, the instantaneous output y is determined only by the instantaneous input x through the nonlinear transfer function T(x), i.e. y = T(x). Gm is defined by:

$$G_m = \sqrt{\int_{-1}^{1} \left(\cos\left(\frac{x\pi}{2}\right)\right)^2 \left(\frac{d^2}{dx^2}T(x)\right)^2 dx}$$

The second derivative of T(x) corresponds to the curvature of the T(x) curve. If the system is linear, then the curve becomes a straight line and thus the curvature is zero. As a result, Gm becomes zero. Squaring the second derivative of T(x) alleviates the sign problem and makes upward and downward concave, both of which deviate from a straight line, equally weighted. By taking the second derivative of the nonlinear transfer function T(x), the metric gives more weight to higher order distortions as the second derivative increases in value according to n×(n-1) where n is the order of non-linearity in T(x)'s polynomial expansion. This attempts to account for the masking effect of the human ear which tends to make higher order nonlinearities more audible than lower order ones. Finally, the offset and linear gain terms in T(x) are completely removed by taking the second derivative, leaving only the nonlinear distortion in Gm calculation.

The squared cosine term in the above formula attempts to addresses another masking effect which tends to cause nonlinearities that occur at low signal levels to be more audible than those that occur at higher signal levels. It is unity at small values of the signal and zero at the largest ones.

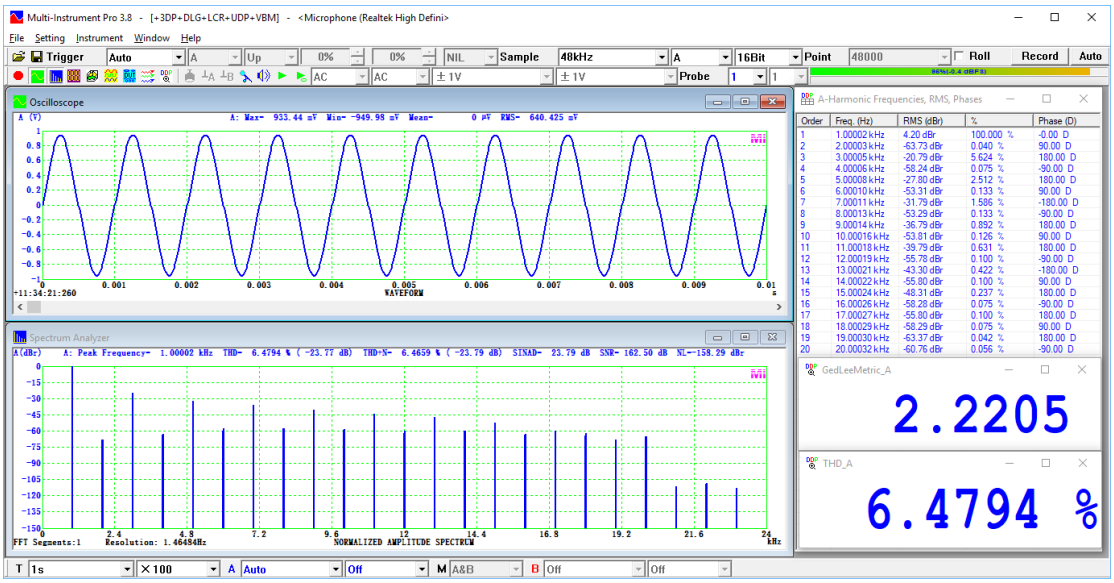
The product of the above two terms is then integrated along x from -1 to +1 (the normalized range of the output signal) after which the square root is taken to yield Gm.

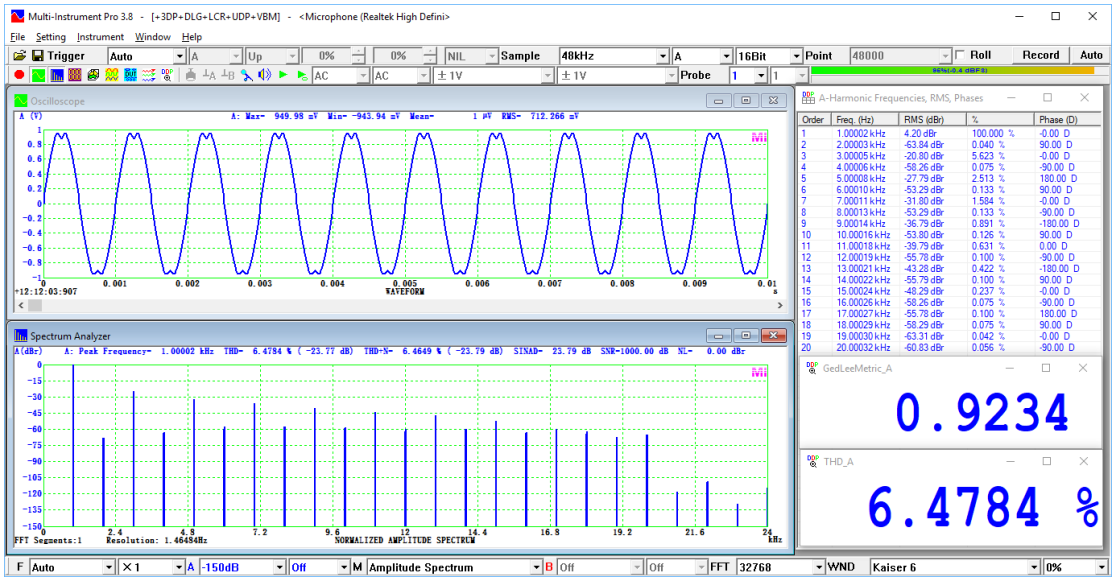
In the real world, it is difficult to find an ideal memory-less system. Thus, T(x) is frequency dependent in a real system, i.e. T(x,f). As a result, Gm(f) is also frequency dependent. However, there is no ambiguity in performing calculation if Gm is measured at a particular frequency. This is completely analogous to the case of THD. Therefore, in Multi-Instrument, Gm is measured using a sine wave as the stimulus at a particular frequency. It can be measured together with THD, THD+N, SNR, etc. Please refer to the previous THD measurement section for the selection of a proper testing frequency and window function in order to avoid or suppress the spectral leakage in FFT and obtain correct measurement results. Unlike THD, Gm takes into account the relative phase information of harmonics as well and thus it requires the software to be able to measure the relative phase of each harmonic accurately. In Multi-Instrument, to measure the phase of each harmonic, the FFT size must be set to a value greater than ½ of the Record Length, but less than or equal to 1/1 of it. Multi-

Instrument features a specially designed algorithm which is able to measure the phase angle of a harmonic accurately even when its amplitude is only 0.01% of that of the fundamental or the signal itself. Same as the THD measurement, it is possible to specify the frequency range and the highest order of harmonics used to calculate Gm. The highest order of harmonics allowed in Gm calculation is 100.

According to the aforementioned researches by Dr. Geddes and Dr. Lee, Gm < 1.0 can be expected to yield subjective ratings of "imperceptible" and Gm < 3.0 can be expected to yield subjective ratings of "barely perceptible but not annoying".

The following two examples (see figures below) are quoted from Keith Howard's article "Weighting Up". The stimulus is a 1 kHz sine wave. In these two examples, the measured harmonics have the same amplitudes but different relative phases. In particular, the second example has its 3rd, 7th, 11th, 15th and 19th harmonics' polarities inverted (i.e. a phase difference of 180 degree) as compared to those of the first example. The phase differences result in different waveforms: the first example shows obvious cross-over distortions while the second one has obvious distortions at its peaks and troughs. In both cases, the measured THD values are the same (about 6.48%), implying that THD is not able to differentiate these two cases. On the other hand, the measured Gm values are very different: 2.22 and 0.92 respectively, illustrating that Gm places greater emphasis on the nonlinear distortion perceived at low signal levels than high ones, which is in accordance with the psychoacoustic findings as well as the fact that the signal amplitude of a music is close to the zero crossing for much of the time, with only infrequent excursions to high amplitudes.





It should be noted that the phase angles displayed in the DDP array viewer (will be introduced later) in the above two examples reference to sine functions. Internally in the software, the nonlinear transfer function T(x) is expressed as a series of harmonically related cosine terms. The measured phase angles of harmonics are normalized with respect to that of the fundamental in terms of cosine functions, and the measured amplitudes of the harmonics are normalized by that of the fundamental. Gm is calculated according to its definition after these normalizations. No inter-frame average in the Spectrum Analyzer is allowed in Gm calculation. Averaging, if needed, can be done through DDP viewer.

Gm can be accessed through the Derived Data Points (DDPs): GedLeeMetric_A for Channel A, and GedLeeMetric_B for Channel B. Two Panel Setting File examples can be found at:

\psf\Pro\GedLeeMetric_FFT32768_SR48000_SingleChannel.psf
\psf\Pro\GedLeeMetric_FFT32768_SR48000.psf

3.3.2.8.3.11 Dynamic Intermodulation Distortion (DIM)

Nonlinear distortions, particularly in power amplifiers, can be categorized as either static or dynamic. Those that result from basic device nonlinearity are called static distortions. They are not really frequency dependent although their magnitude may change with the change in feedback factor with frequency. Thus, the distortions vary with frequency in direct proportion to the change in feedback with frequency. This distinguishes such distortions from ones which inherently change magnitude with frequency - often called “dynamic” distortions. Since dynamic distortions change with frequency, even without a change in feedback with frequency, the addition of frequency dependent feedback creates a high order dependence of distortion on frequency. The term dynamic distortion has commonly referred to mechanisms which get worse with fast or high frequency signals. In other words, dynamic distortion is very much slew rate related, and thus it is sometimes called Slewing Induced Distortion (SID) or Slope Induced Distortion (SID), Transient Intermodulation

Distortion (TIM). It is excited by the signal rate of change and becomes worse when slew rate limiting occurs.

Numerous techniques have been proposed for measuring static and dynamic distortions. Measuring THD as a function of frequency and level yields a complete performance profile of a DUT. For those severely bandlimited DUTs, other techniques must be employed. If static distortions are of special interest, SMPTE/DIN IMD can be used. If dynamic distortions are of concern, CCIF should be measured. If the DUT's bandwidth allows, measuring DIM is an effective and efficient way to characterize both the static and dynamic distortions. It is designed to be particularly sensitive to distortions produced during transient conditions. According to IEC 60268-3, the DIM test signal consists of a bandlimited square wave component at a frequency of 3.15 kHz and a sine wave component at a frequency of 15 kHz with an amplitude of ¼ of that of the square wave component. Two DIM tests are common: DIM30 and DIM100. DIM30 uses a single-pole low-pass filter with a cutoff frequency of 30 kHz while in DIM100 the cutoff frequency is 100 kHz. In both tests, nine intermodulation products are measured. They are listed in the following table.

Intermodulation (f_q=3.15 kHz, f_s=15 kHz)	Frequency (kHz)	Symbol
$5f_q - f_s$	0.75	U1
$f_s - 4f_q$	2.40	U2
$6f_q - f_s$	3.90	U3
$f_s - 3f_q$	5.55	U4
$7f_q - f_s$	7.05	U5
$f_s - 2f_q$	8.70	U6
$8f_q - f_s$	10.20	U7
$f_s - f_q$	11.85	U8
$9f_q - f_s$	13.35	U9

DIM is calculated as the square root of the ratio of the power of the above nine intermodulation products to the power of the 15 kHz sine wave. It is expressed as a percentage or in dB. The following DDPs are used to store the related parameters.

(1) IMD_A(%): DIM in % for Channel A
(2) IMD_A(dB): DIM in dB for Channel A
(3) F1Freq_A(Hz)~ F7Freq_A(Hz): 3.15 kHz, 15 kHz, frequencies of U1~U5 for Channel A
(4) f1RMS_A(EU): 3.15 kHz RMS value, 15 kHz RMS value, RMS values of U1~U5 for Channel A
(5) fnFreq_A(Hz)_Array[0]~fnFreq_A(Hz)_Array[10]: 3.15 kHz, 15 kHz, frequencies of U1~U9 for Channel A
(6) fnRMS_A(EU)_Array[0]~fnRMS_A(EU)_Array[10]: 3.15 kHz RMS value, 15 kHz RMS value, RMS values of U1~U9 for Channel A

There are a similar set of DDPs for Channel B.

The DIM test signal can be generated using the Multitone function of the Signal Generator of the software. Four DIM test tone configurations can be found in the software's TCF subdirectory:

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(1) DIM30_SharplyBandLimited
The square wave component is sharply bandlimited by a 30 kHz brick-wall (ideal) low-pass filter. An output sampling rate of at least 96 kHz is recommended for signal generation.

(2) DIM30_SinglePole_LowpassFiltered
The square wave component is bandlimited by a 30 kHz single-pole (first-order) low-pass filter. An output sampling rate of at least 192 kHz is recommended.

(3) DIM100_SharplyBandLimited
The square wave component is sharply bandlimited by a 100 kHz brick-wall (ideal) low-pass filter. An output sampling rate of at least 192 kHz is recommended for signal generation.

(4) DIM100_SinglePole_LowpassFiltered
The square wave component is bandlimited by a 100 kHz single-pole (first-order) low-pass filter. An output sampling rate of at least 384 kHz is recommended for signal generation.

Two panel setting examples for DIM30 and DIM100 measurements respectively can be found at:
\psf\pro\DIM30_FFT131072_SR192000.psf
\psf\pro\DIM100_FFT262144_SR384000.psf

The square wave components are single-pole low pass filtered in the above two settings. The default window function for DIM measurement is the Kaiser 6 window. Other window functions that can be used includes: Kaiser 7 ~ Kaiser 20, Blackman Harris 7, Cosine Sum 220, Cosine Sum 233, Cosine Sum 246, Cosine Sum 261. The following figure shows the measured DIM30 of an RTX6001 audio analyzer.

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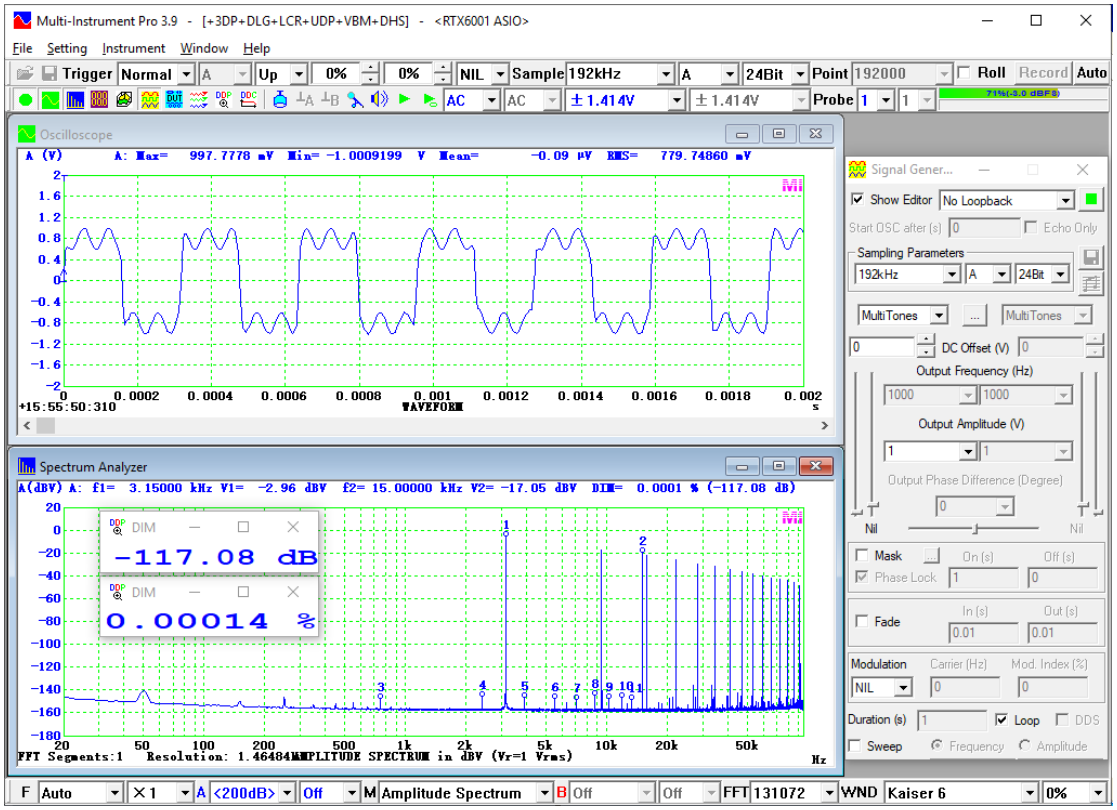


For more information on DIM measurement, please refer to Measurements of Various Intermodulation Distortions (IMD, TD+N, DIM) using Multi-Instrument.

Download link:
https://www.virtins.com/doc/Measurements-of-Various-Intermodulation-Distortions-IMD-TD+N-DIM-using-Multi-Instrument.pdf

3.3.2.8.3.12 Sound Intensity

Sound intensity (sometimes referred to as sound power flux density) is the time-averaged rate at which sound energy is transmitted through a unit area perpendicular to a specified direction at a given point in space. It is equal to the time averaged product of sound pressure (p) and acoustic particle velocity (u) in the specified direction. The SI unit of sound intensity is W/m^2. In contrast to sound pressure, which is a scalar quantity and has only magnitude, sound intensity is a vector quantity, possessing both magnitude and direction. It is usually measured in the direction of sound propagation. Sound Intensity Level (SIL) is a logarithmic measure of the sound intensity relative to a reference value, that is:

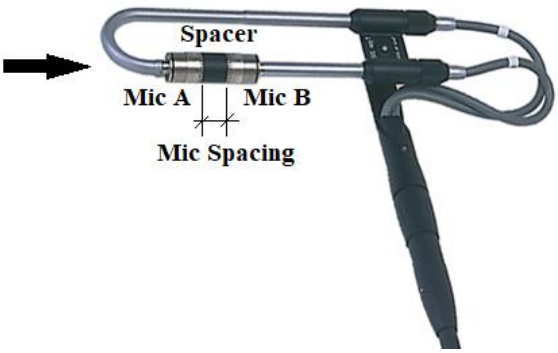
$$SIL\ (dB) = 10\lg(I/I_0)$$

where I is the magnitude of the sound intensity being measured and I_0 the reference sound intensity. The commonly used reference sound intensity is $1\times10^{-12}\ W/m^2$.

Sound intensity measurement is useful for determination of sound power, identification and ranking of sound sources, visualization of sound fields, measurement of transmission loss, identification of transmission paths, etc. The

sound power of a source can be determined from sound pressure measurement, but only if the source is placed in a carefully controlled environment, such as an anechoic or reverberant chamber, where special assumptions about the sound field can be made. In contrast, sound intensity measurement allows for the in-situ determination of sound power in any sound field, even in the presence of strong steady background noise from other sources.

There are generally two types of probes for sound intensity measurement: p-u probes and p-p probes. A p-u probe contains a pressure sensor and velocity sensor, allowing it to measure both sound pressure and particle velocity directly. The Oscilloscope view type "A×B" in Multi-Instrument can be used for deriving sound intensity from these measurements. A p-p probe, on the other hand, consists of two matched pressure sensing microphones separated by a known distance. The sound pressure is determined by averaging the pressure sensed by the two microphones, while the particle velocity is derived from the pressure difference between them, based on a finite difference approximation of the pressure gradient. Please note that the sound intensity measurement option here is only for the p-p approach.



A typical p-p sound intensity probe is shown in the figure above. The spacer between the two microphones determines the frequency range over which sound intensity can be accurately measured. Generally, the wavelength being measured must be at least 6 times the microphone spacing to ensure sufficient accuracy in the finite difference approximation of the pressure gradient. This sets the upper frequency limit. This limit can be doubled if the microphone spacing equals the diameter of the microphones in a face-to-face configuration. This is due to the resonance in the small cavity between the spacer and the membrane of the microphone, which causes a pressure increase for sound incidence along the probe axis and to some extent compensates for the finite difference error. At low frequencies, there will be only a small phase difference between the signals from the two microphones, making the measurements more sensitive to self-generated noise and phase mismatch errors in the measurement system. This problem can be mitigated by using a longer spacer, though this will reduce the upper frequency limit.

For sound intensity measurement in Multi-Instrument, Microphones A&B should be connected to Input Channels A&B, respectively. Their sensitivities in V/Pa should be set via [Setting]>[Calibration]>"Sensor">"Sensitivity" and "Unit". Sound pressure level calibration can be done there individually if necessary. The "0dB Reference Vr" should be configured with the standard reference value, e.g. 0.00002 μPa for sound pressure level measurement in air. These settings are required to enable sound

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intensity calculation. Multi-Instrument uses the cross power spectrum to derive sound intensity as a function of frequency as follows:

$$I(f) = -\frac{I_m(G_{AB}(f))}{2\pi f \rho \Delta r}$$

where the numerator is the imaginary part of the cross power spectrum, ρ the air density, and Δr the mic spacing which requires user input. Air density varies with temperature, pressure and humidity. It is set to 1.2048 kg/m^3 in the software. If the actual value differs significantly, it can be corrected using the entered value of Δr according the formula above.

Under the sound intensity measurement mode, the spectrum in Ch.A will display the measured sound intensity and that in Ch. B will display the measured sound pressure. The level difference of these two is called Pressure Intensity Index, which is a good guide to the quality of the measurement. When exactly the same acoustic signal is fed into the two microphones, the measured sound intensity should ideally be zero. However, in practice, it will not, primarily due to phase mismatch between the two channels. The level difference between the measured sound pressure and the measured residual intensity is referred to as Pressure-Residual Intensity Index. It is often tested to verify the quality of the measurement system. When the Y scale of the Spectrum Analyzer is set to "Pa", Ch.A actually displays the square root of the magnitude of the measured sound intensity in W$^{1/2}$/m, while Ch.B still displays the measured sound pressure in Pa. When the Y scale is set to "dB", both channels will display values in dB, with 0 dB referenced to 10^{-12} W/m^2 (or 10^{-6} W$^{1/2}$/m) for Ch.A and the "0dB Reference Vr" configured in [Setting]>[Calibration] for Ch.B. Unlike sound pressure, sound intensity can be negative, indicating that sound power is flowing in the opposite direction, i.e. from Mic B to Mic A. The spectrum graph will indicate the direction of sound intensity only in octave analysis mode and when the bar chart is used. The color of Ch. EXT, configured via [Setting]>[Display], indicates the negative direction, while the color of Ch. A indicates the positive direction. All exported sound intensity data, as well as the sound intensity data displayed in the DDP viewers and DDP Array viewer, will indicate the direction by the signs of the values, except for data in dB, where "dB+" and "dB-" are used to represent positive and negative directions respectively. This is because the sign of a value in dB is already used to indicate whether its magnitude is above or below the 0dB reference.

Multi-Instrument calculates sound intensity spectrum from 0 Hz to half the sampling frequency. However, only the data within the valid frequency range of the sound intensity probe can be trusted. [Spectrum Analyzer Processing]>"Energy in User Defined Frequency Bands" can be used to extract the sound intensity values within specified frequency bands. The measured sound intensity magnitudes are stored in the DDPs: fBand1RMS_A(EU) ~ fBand8RMS_A(EU) and the DDP array: fnRMS_A(EU)_Array[?]. Their directions are stored in the DDP array: fnPhase_A(D)_Array[?], where the values 1 and -1 represent positive and negative directions, respectively. Under the octave analysis mode, the measured sound intensity magnitudes are stored in the DDP array: oBandRMS_A(EU)_Array[?].

The following screenshot shows an example of sound intensity and pressure measurement of a 1kHz, 1Pa (peak) acoustic signal, with Ch.A lagging Ch.B by 0.562895º. The length of the spacer is 8.5mm. The measured sound intensity and

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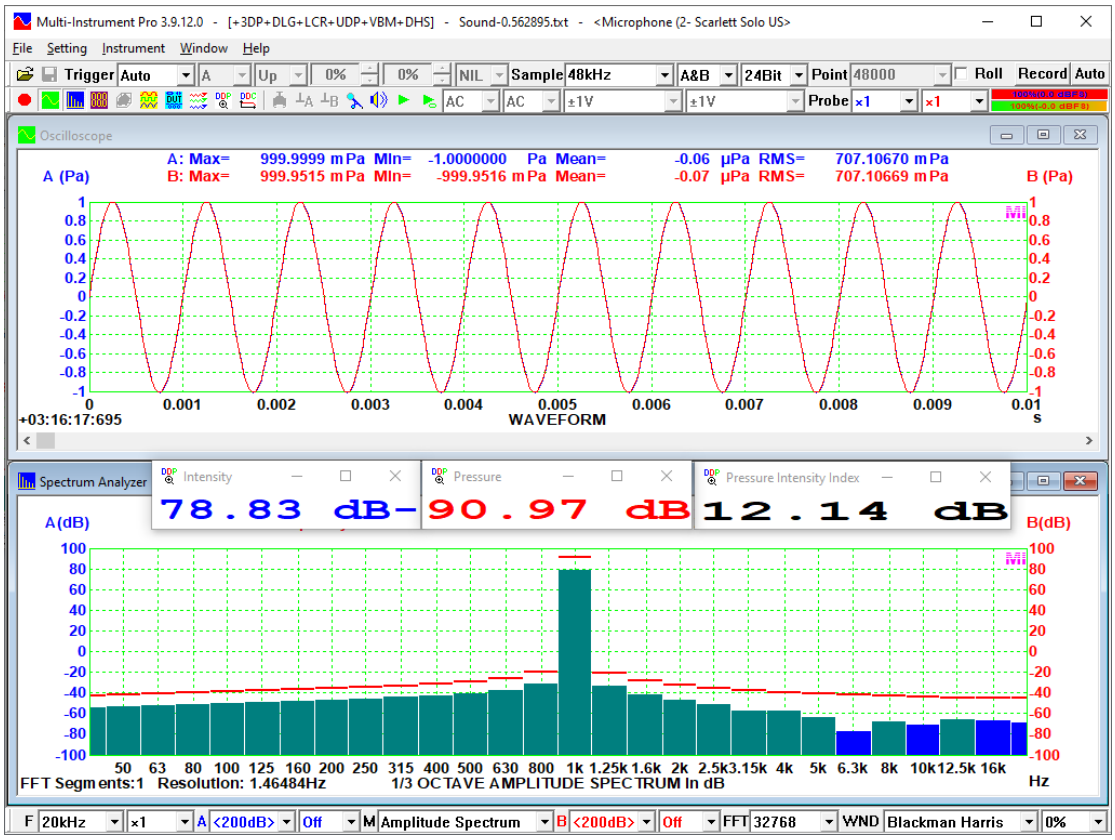
pressure are 78.83 dB and 90.97 dB, respectively. Theoretically, the magnitude of the sound intensity here can be calculated as follows:

$$10 \times \lg\left(\frac{0.707 \times 0.707 \times \sin(0.562895^{\circ})}{2\pi \times 1000 \times 1.2048 \times 0.0085 \times 10^{-12}}\right) = 78.83 \text{ dB}$$



The phase mismatch between the two channels, including the microphone pair, is the primary cause of measurement inaccuracy. It can be calibrated by applying identical acoustic white noise to the microphone pair and measuring sound intensity or transfer function using Multi-Instrument. Once the measurement is completed, right click the Spectrum Analyzer and select "Spectrum Analyzer Generate Frequency Compensation File (Flat)". A phase compensation file (*.pcf) or a gain and phase compensation file (*.gpc) will be generated. This file can later be loaded as a frequency compensation file via [Spectrum Analyzer Processing]> "Compensation1" or "Compensation2" to improve measurement accuracy.

A preconfigured panel setting file for sound intensity measurement with 8.5mm mic spacing can be found in the PSF subdirectory of the software:
\pro\SoundIntensity8.5mm.psf

3.3.2.9 Spectrum Analyzer X Scale (Setting SubMenu) (ALT-S-X)

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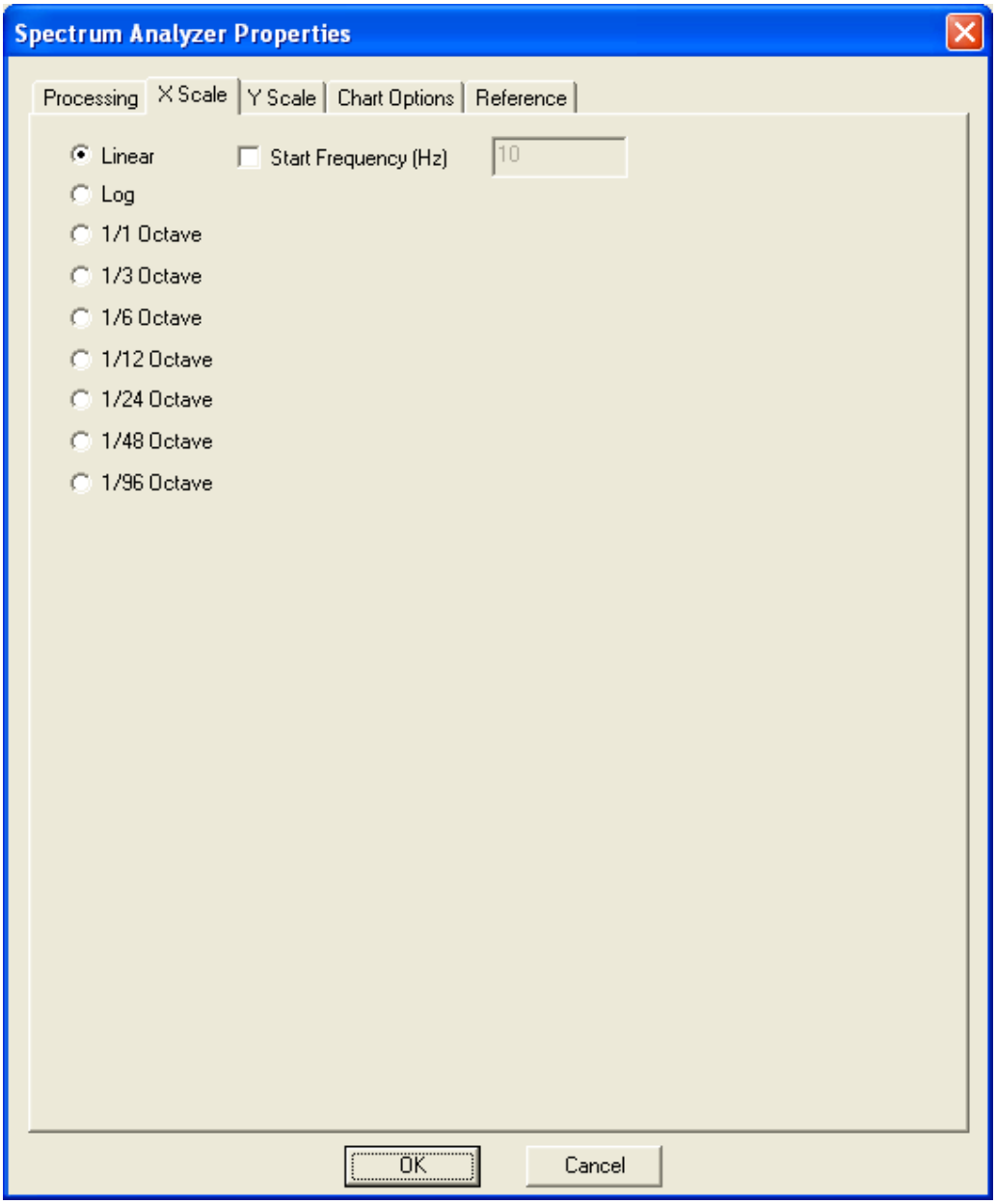

Selection of X Scale is only available in Amplitude Spectrum, Phase Spectrum, Coherence Function, and Transfer Function displays. The horizontal axis (X) can be set to Linear, Logarithmic, Octave (1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96). The X Scale dialog can also be accessed by double clicking on the X axis.

Octave scales are only available in Amplitude Spectrum display. When an octave scale is selected, the software will check the corresponding minimum FFT size required and change the current FFT size automatically if necessary. The larger the FFT size, the more accurate the octave analysis. The window function will be set to "Rectangle" by default under octave analysis.

The X axis in Amplitude Spectrum, Phase Spectrum, Coherence Function, and Transfer Function displays is a frequency axis. By default, the X axis starts from 0 Hz in linear scale, a frequency equal to the FFT frequency resolution in logarithmic scale,

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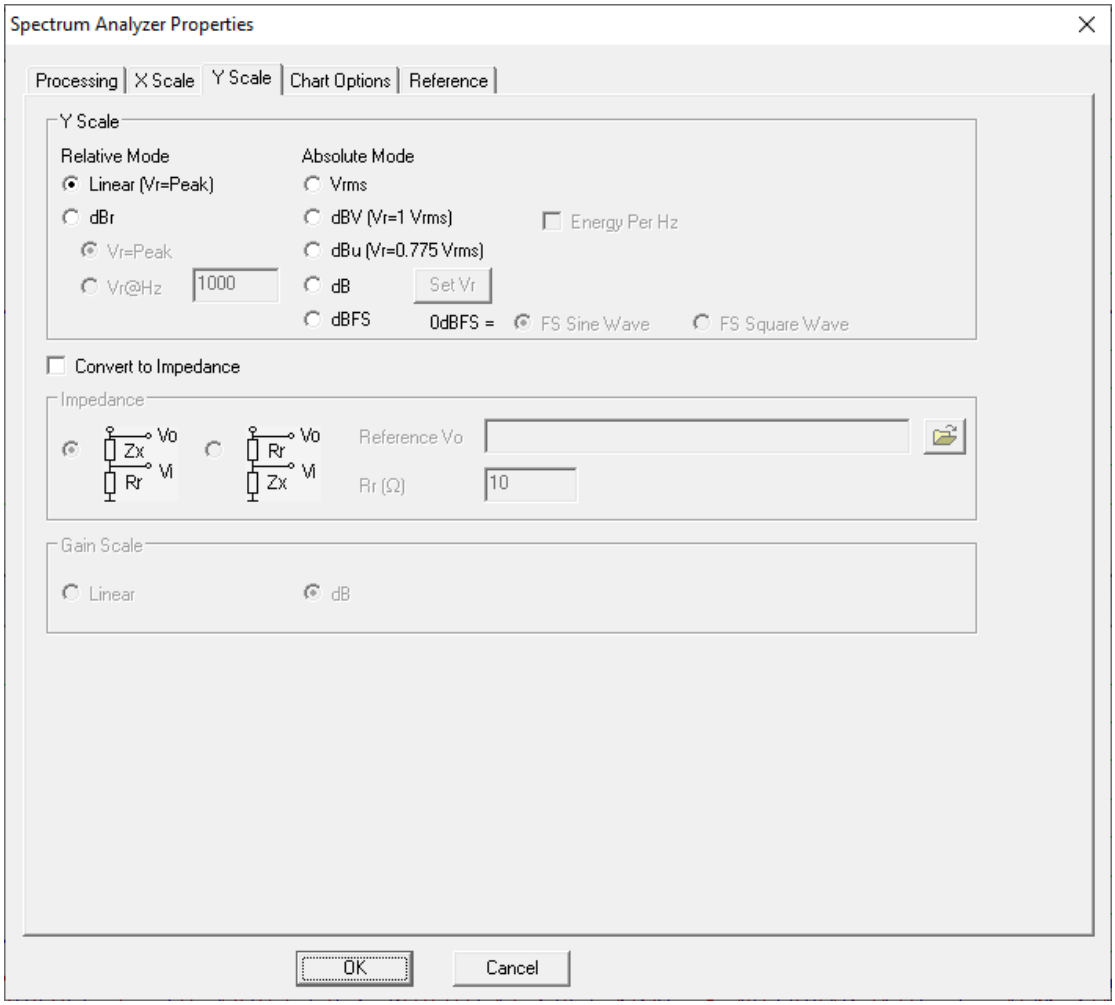
and the lowest octave band in octave scale. If the above Start Frequency checkbox is ticked, you can customize the start frequency of the X axis.

3.3.2.10 Spectrum Analyzer Y Scale (Setting SubMenu) (ALT-S-Y)



Selection of Y Scale is only available in Amplitude Spectrum as well as Transfer Function. It can also be accessed by double clicking on the Y axis.

3.3.2.10.1 Under Amplitude Spectrum Mode

There are two display modes for the vertical axis (Y): Absolute Display Mode and Relative Display Mode.

Under Relative Display Mode, the vertical axis is scaled relatively with reference to the maximum value in the current measurement (i.e. Vr=peak), in either linear or logarithmic (dBr) scale. When “dBr” is selected, it is also possible to set its reference level at a particular frequency instead of using the maximum value.

Under Absolute Display Mode, all spectral data points are plotted based on their absolute values, in Vrms, dBV, dBu, dB, or dBFS. Note that by definition, the reference voltages for dBV and dBu are 1 Vrms and 0.775 Vrms respectively. 1 dBu

in amplitude spectrum is equivalent to 1 dBm in power spectrum when the load is 600 ohms. The reference level (Vr) for dB is user definable. To set the Vr, press the "Set Vr" button, the Calibration Setting dialog box will pop up. Enter the values of 0dB Reference Vr for both channels at the upper right corner. (Please refer to the section for "0dB Reference Vr" introduced previously). The reference voltage for dBFS is the input full-scale voltage (1/2 Vpp). When dBFS is used in the context of peak values, there is no ambiguity. However, when it is used in the context of RMS values such as the ones in the case of spectra, there are two conventions: 0 dBFS = Full Scale Sine Wave or 0 dBFS = Full Scale Square Wave. The difference between the two is 3.01dB. Note that, if the Sensor Engineering Unit is not "V", then the above options will be changed accordingly but the concept remains the same. Please refer to Section "Channel A Display Range" described previously for details.

If the "Per Hz" option is ticked, the Spectrum Analyzer will display the amplitude spectral density function for Vrms and the respective power spectral density function for dBV, dBu, dB and dBFS.

"Convert to Impedance" option is enabled when the Spectrum Analyzer is in Amplitude Spectrum mode or Transfer Function / Impedance Analyzer mode. The former allows using the single-channel method to measure the impedance vs frequency plot, while the latter allows using the dual-channel method to measure the impedance and phase vs frequency plot. These two methods have already been described previously.

3.3.2.10.2 Under Transfer Function Mode

The "Gain Scale" option is enabled under Transfer Function mode. It can be set on either linear or dB scale.

3.3.2.11 Spectrum Analyzer Chart Options (Setting SubMenu) (ALT-S-O)

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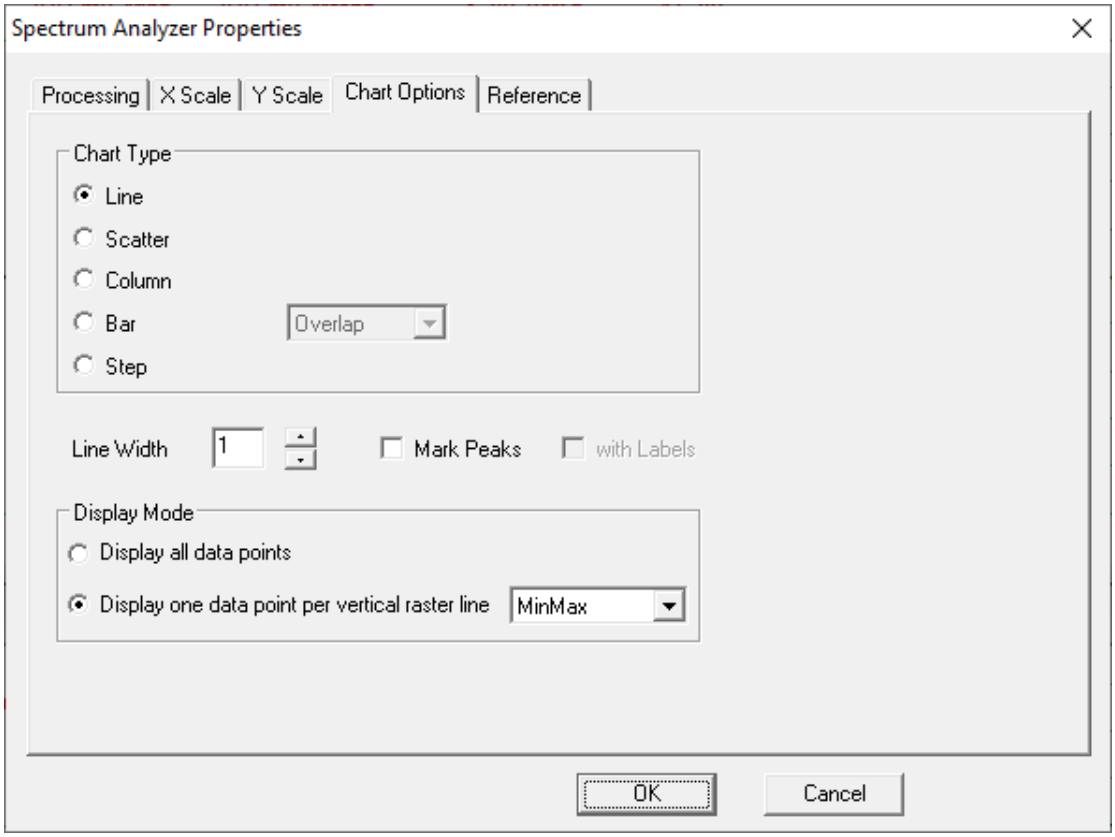
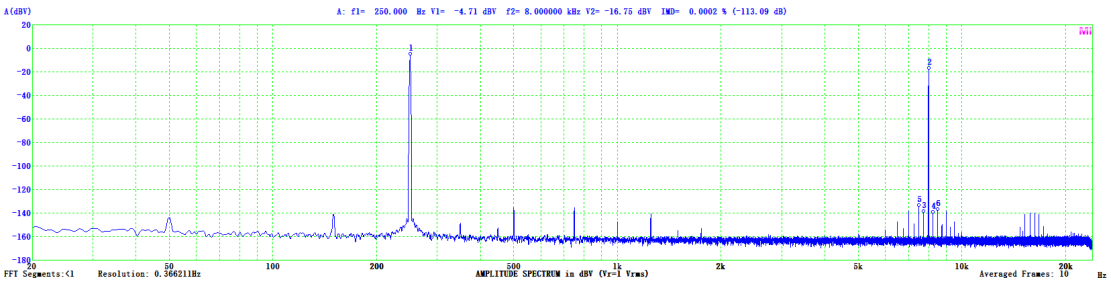


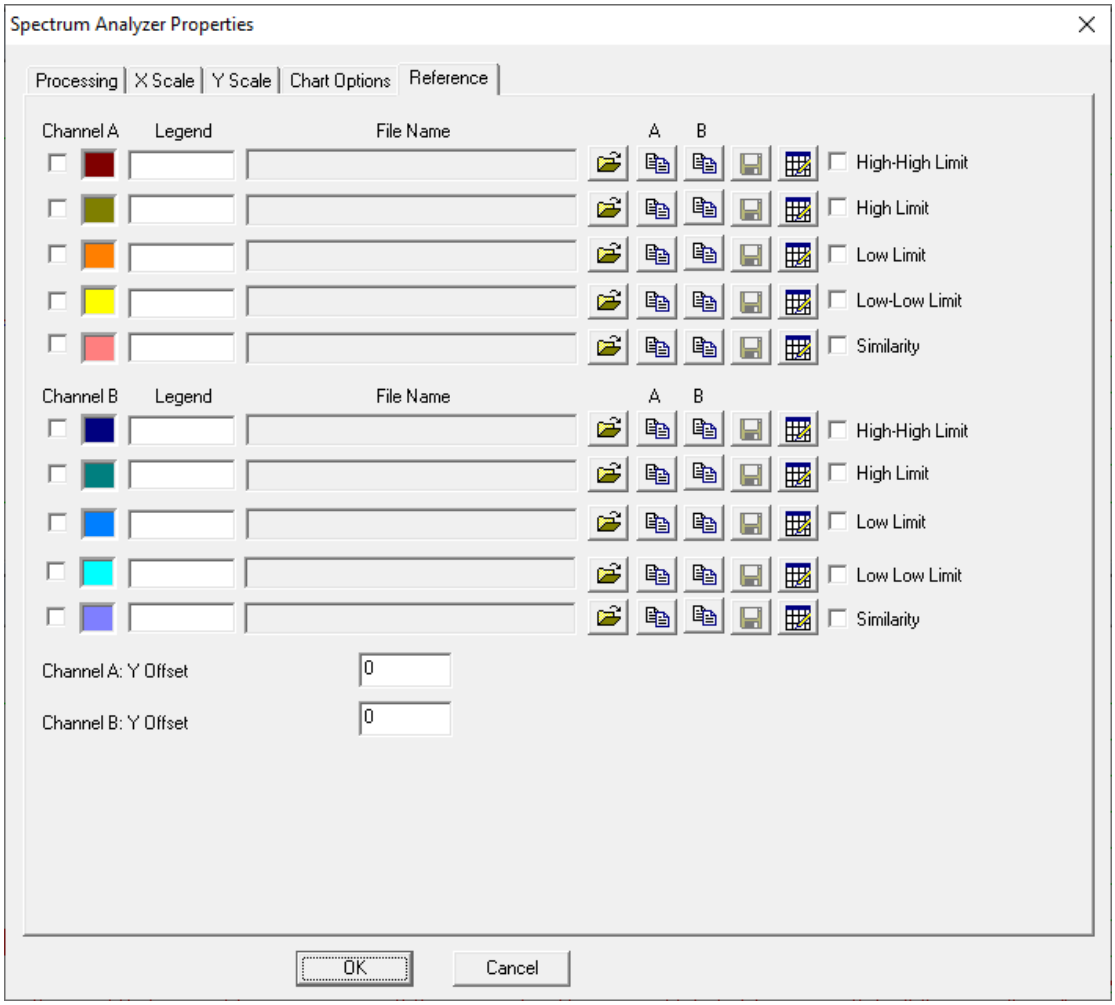
Same as Oscilloscope except the Mark Peaks option.

Mark Peaks option will be enabled for selection under the following parameter measurements: None, THD, IMD, Crosstalk, Harmonics, Peaks. When this option is ticked, the relevant peaks in the spectrum will be marked at their accurately calculated no-spectral-leakage heights and accurately interpolated frequencies which may not be located at the centers of FFT bins, with or without labels. The following figure shows an IMD DIN measurement with the relevant peaks marked with labels.



In single-channel 1/1 or 1/3 octave analysis, if "Mark Peaks" is ticked and [Spectrum Analyzer Y Scale] is in dBV, dBu, dB or dBFS, the Y value of each octave band will be displayed as text on the spectrum graph. This would be convenient during EQ processes.

3.3.2.12 Spectrum Analyzer Reference (Setting SubMenu) (ALT-S-R)



Similar to Oscilloscope. The related six DDPs are: SpectrumComparisonH_A, SpectrumComparisonL_A, SpectrumSimilarity_A, SpectrumComparisonH_B, SpectrumComparisonL_B, SpectrumSimilarity_B.

3.3.2.13 Save Current Panel Setting as Default (Setting SubMenu) (ALT-S-F)

Same as Oscilloscope.

3.3.2.14 Save Current Panel Setting (Setting SubMenu) (ALT-S-S)

Same as Oscilloscope.

3.3.2.15 Load Panel Setting (Setting SubMenu) (ALT-S-L)

Same as Oscilloscope.

3.3.2.16 Configure Hot Panel Setting Toolbar (Setting SubMenu) (ALT-S-P)

Same as Oscilloscope.

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3.3.2.17 Show Hot Panel Setting Toolbar (Setting SubMenu) (ALT-S-H)

Same as Oscilloscope.

3.3.2.18 Change Password (Setting SubMenu) (ALT-S-W)

Same as Oscilloscope.

3.3.3 Instrument SubMenu

Same as Oscilloscope.

3.3.4 Window SubMenu

Same as Oscilloscope.

3.3.5 Help SubMenu

Same as Oscilloscope.

3.4 Cursor Reader and Markers

Similar to Oscilloscope.

3.5 Peak Values

Spectrum Analyzer will display the peak frequency in Amplitude Spectrum display, the second peak time delay and the corresponding coefficient in Auto Correlation Function display, the peak time delay and the corresponding coefficient in Cross Correlation Function display, the peak frequency and the corresponding coefficient in Coherence Function display, the peak frequency and the corresponding gain and phase in Gain and Phase display, the peak time in Impulse Response display.

3.6 Context Menu

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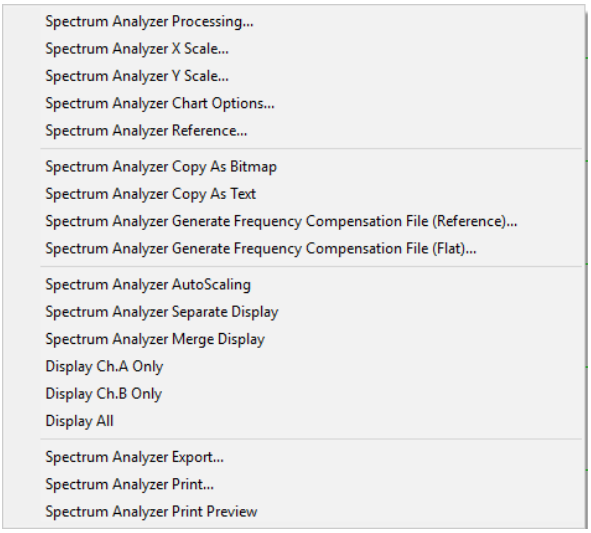

The above context menu will be shown when right clicking anywhere within the Spectrum Analyzer view. It provides additional convenience to you. All menu items in the context menu can also be found in the spectrum analyzer main menu except the following items:

- Copy As Bitmap

  Same as Oscilloscope

- Copy As Text

  Same as Oscilloscope

- Generate Frequency Compensation File (Reference)

  This command is used to generate a Frequency Compensation File, which compensates the currently measured magnitude frequency response in Channel A to that in Channel B with the compensation factor at 1 kHz being 0 dB. Channel A is the one to be compensated and Channel B is the reference. For example, you can connect the microphone to be compensated in Channel A and the reference microphone in Channel B, and generate a Frequency Compensation File for the microphone in Channel A using pink noise or white noise as the stimulus. The generated Frequency Compensation File can be loaded later for frequency compensation.

- Generate Frequency Compensation File (Flat)

  Under amplitude spectrum mode, without selecting the “Sound Intensity” option, this command is used to generate a Frequency Compensation File, which compensates the currently measured magnitude frequency response to a flat one with the compensation factor at 1 kHz being 0 dB. For example, you can use this command to generate a Frequency Compensation File which compensates the frequency response of a sound card to a flat frequency response using white noise and a loopback cable from the sound card’s output to its input. The generated Frequency Compensation File can be loaded later for frequency compensation.

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Under amplitude spectrum mode, with the “Sound Intensity” option selected, this command is used to generate a Phase Compensation File, which compensates the phase mismatch between the two channels to zero. The generated Phase Compensation File can be loaded later to improve the accuracy of sound intensity measurement.

Under transfer function mode (and not in its impedance measurement mode), this command is used to generate a Gain and Phase Compensation file, which can be used to compensates the currently measured gain and phase to 0 dB and 0 degree respectively. For example, you can use this command to generate a Gain and Phase Compensation file which eliminates the small gain and phase difference between the two input channels of the measuring device itself.

- AutoScaling

  It adjusts the vertical display scales based on the peak values in the sampled data frame.

- Separate Display

  It separates the display of Channel A from that of Channel B vertically.

- Merge Display

  It merges the displays of Channel A and Channel B vertically.

- Display Ch. A Only

  It will hide other channels and display Ch. A only in all views.

- Display Ch. B Only

  It will hide other channels and display Ch. B only in all views.

- Display All

  It will display all channels in all views.

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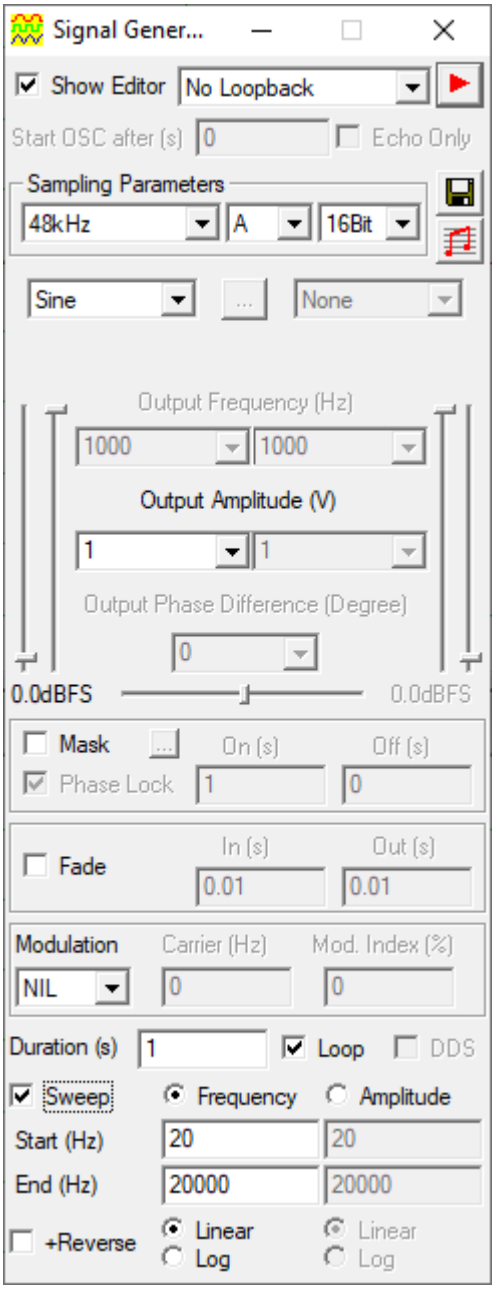
4 Signal Generator

4.1 Overview



This is a dual channel Signal Generator (Sweep / Arbitrary/ Function/ Burst/ Noise/ Modulation/ Playback Generator), providing the following types of waveforms for output:

- Sine
- Rectangle (with adjustable duty cycle)
- Triangle
- Saw Tooth

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- White Noise
- Pink Noise
- Multi-tones
- Arbitrary via user-configurable waveform library (WFLibrary)
- Maximum Length Sequence (MLS), with adjustable length (127~16777215)
- Dual Tone Multi-Frequency (DTMF)
- Unit Impulse
- Unit Step
- Musical Scale
- Wave File

at either a fixed frequency, or a frequency that sweeps linearly or logarithmically within a specified frequency range and time duration. The output signal can be looped back partially (i.e. only one channel while the other channel can be used for field input) or fully (i.e. both channels), via the software itself, to the input of the Oscilloscope for display and analysis in real time. Synchronized operation between the Signal Generator and the Oscilloscope with a timing accuracy in the same order of the sampling frequency is allowed. Amplitude sweeping, burst signal generation, output signal fade in/out and modulation (AM, FM, PM) are also supported. In addition to the streaming mode, the software supports DDS (Direct Digital Synthesis) mode when the hardware used supports DDS.

For a sound card based system, the accuracy of the output signal is dependent on the sound card. Almost all sound cards are "AC coupled" and thus will filter out DC and near-DC components. As a result, signal components below approximately 10 Hz may be distorted. The flatness of pulses and rectangle waves may be affected and ramp signals may become non-linear.

The Signal Generator panel is intentionally made compact so that it can be displayed and operated simultaneously with other instruments within the confinement of the screen. You can un-tick the “Show Editor” checkbox to make the panel even smaller after you have set your desired parameters.

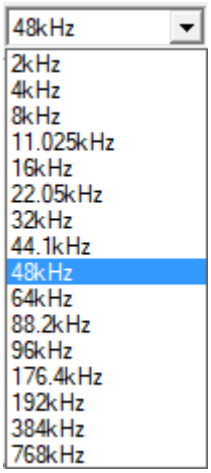
The toggle button in the Instrument Toolbar is used to open or close the Signal Generator. You can also close it by clicking the "Close" button at the upper right corner of the Signal Generator panel.

4.2 Output Sampling Parameters

The sampling capability is fully dependent on the DAC device. Once the sampling parameters are specified and the Run button is clicked, it will attempt to start outputting using the specified sampling parameters. An error message will pop up if the specified sampling parameters are not supported by the DAC device.

Note that some sound cards may not generate error messages even if the sampling frequency specified exceeds their limits. Please check your sound card manual before you can use a sampling frequency greater than 44100 Hz, otherwise measurement error may be introduced.

4.2.1 Sampling Frequency



For a sound card based system, the following sampling frequencies can be selected: 2kHz, 4kHz, 8kHz, 11.025kHz, 16kHz, 22.05kHz, 32kHz, 44.1kHz, 48kHz, 64kHz, 88.2kHz, 96kHz, 176.4kHz, 192kHz, 384kHz, 768kHz.

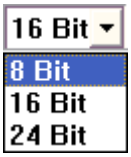
4.2.2 Sampling Channels

Two options are available:

- A
  Only Channel A is sampled. For stereo sound cards, this will output the same signal in both channels.

- A&B
  Both Channel A and Channel B are sampled. Therefore, different signals can be generated in different channels.

4.2.3 Sampling Resolution



Three options are available: 8 Bit, 16 Bit, 24 Bit.

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4.3 Output Signal Parameters

Each channel has its own output signal parameters.

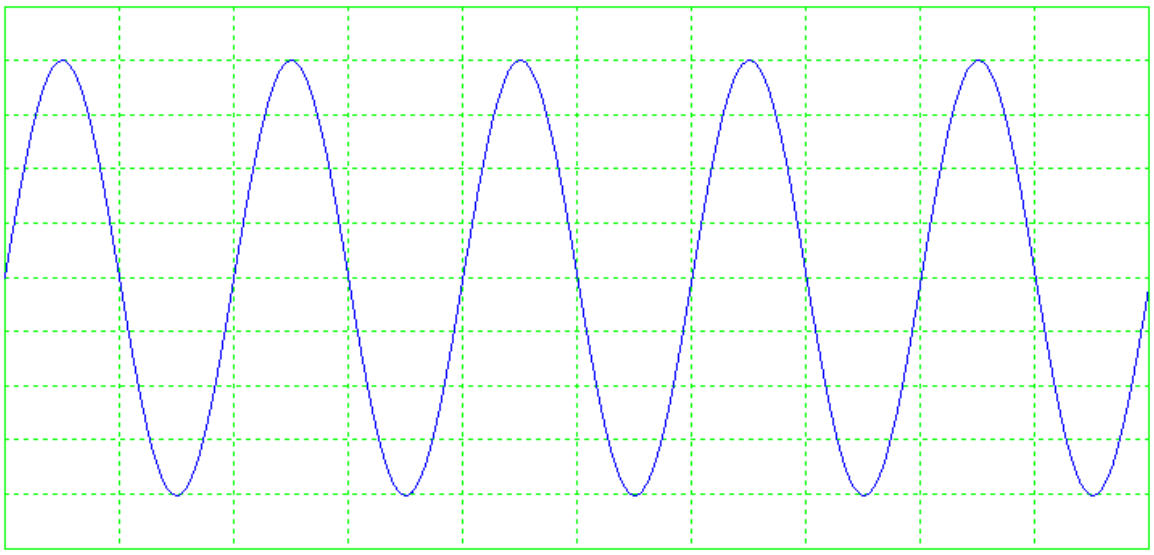
4.3.1 Output Waveform

There are 14 options available.

4.3.1.1 None

No signal will be generated, i.e. the output will be zero. No output frequency and output amplitude need to be specified.

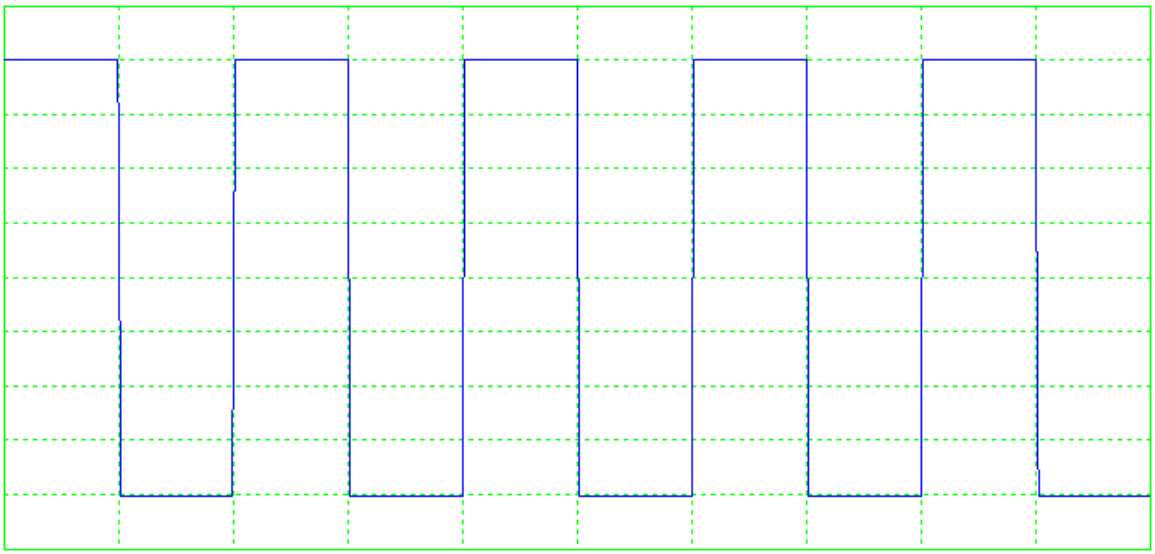
4.3.1.2 Sine



A sinusoidal waveform will be generated. Note that the above figure also illustrates the initial phase of the generated signal. If you tick the "No Spectral Leakage" option, the frequency of the sinusoidal wave you specify will be fine tuned to the nearest

integer multiple of [Sampling Frequency of the Oscilloscope] / [FFT Size of the Spectrum Analyzer], once the "No Spectral Leakage" checkbox is ticked or the Run button of the Signal Generator is pressed. For example, if the sinusoidal wave frequency you specify is 1 kHz, the Sampling Frequency of the Oscilloscope is 48 kHz, and the FFT Size of the Spectrum Analyzer is 16384, then the sinusoidal wave frequency will be changed to 999.0234375 Hz. This option is used to ensure that an FFT segment in the Spectrum Analyzer contains exactly an integer number of cycles of the signal so as to avoid the spectral leakage issue, if the generated signal will be sampled by the Oscilloscope. Rectangle window function should be used in the Spectrum Analyzer when no spectral leakage occurs.

4.3.1.3 Rectangle



A rectangle waveform will be generated. Note that the above figure also illustrates the initial phase of the generated signal. The duty cycle of the rectangle wave can be specified (refer to the figure below).

4.3.1.4 Triangle

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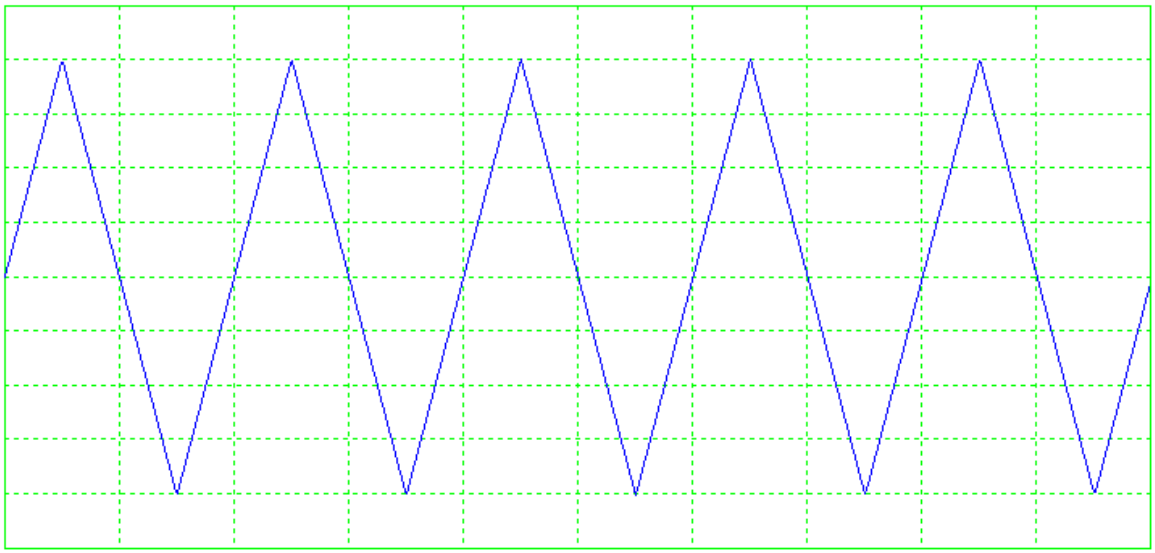
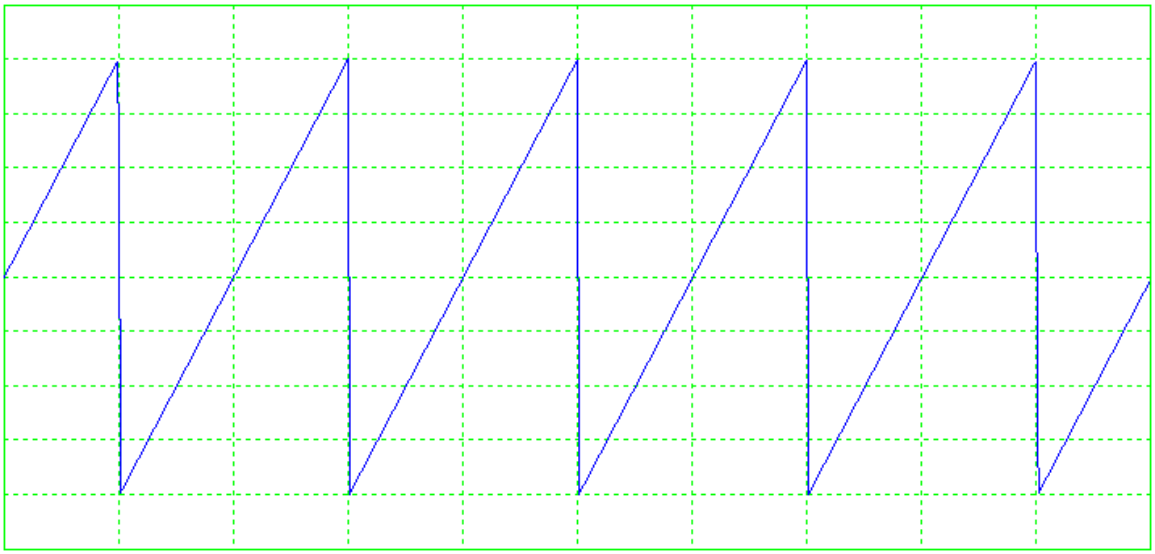


A triangle waveform will be generated. Note that the above figure also illustrates the initial phase of the generated signal.

4.3.1.5 Saw Tooth



A saw tooth waveform will be generated. Note that the above figure also illustrates the initial phase of the generated signal.

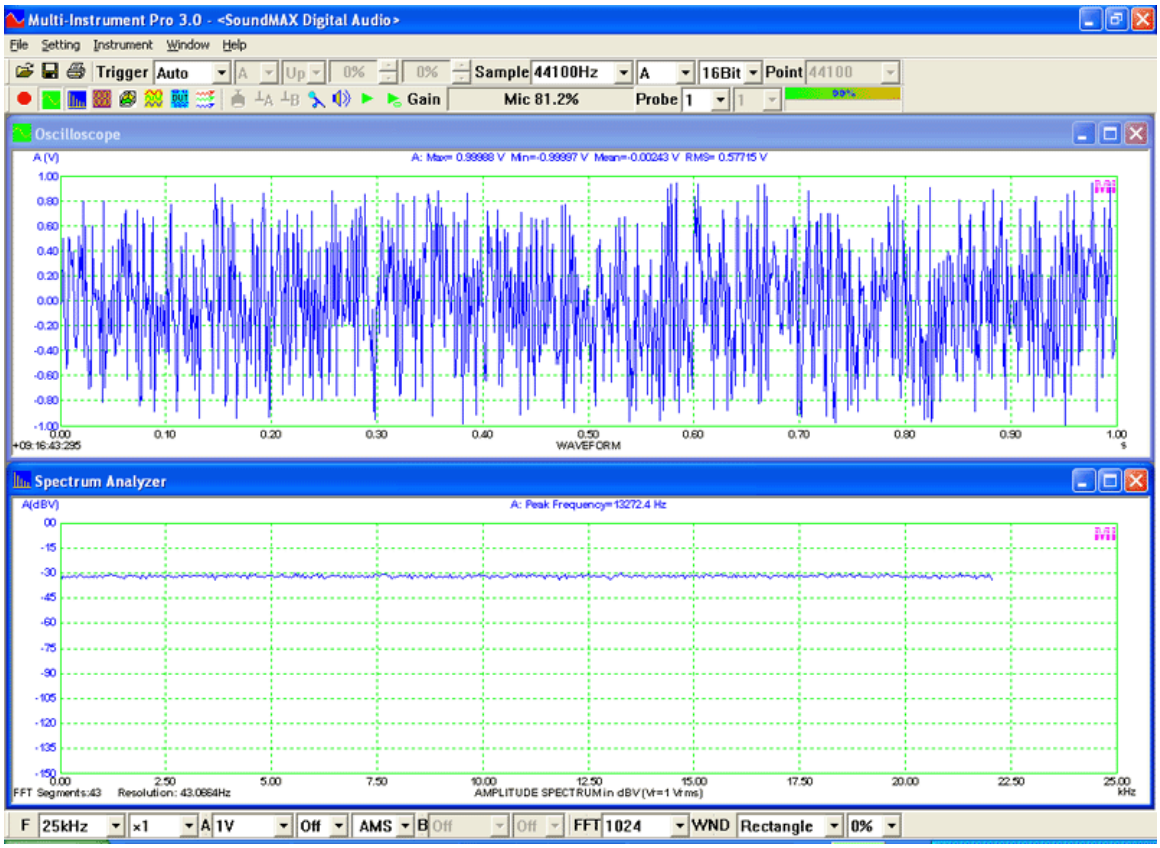
4.3.1.6 White Noise

White Noise will be generated. White noise has an equal amount of energy per Hz of bandwidth. No output frequency needs to be specified.

Unlike most of the digital white noise generators in the market, which eventually repeat the same segment of white noise, the white noise generated here is "true" white noise without any form of repetition. This is very useful when you want to use

averaging method to obtain the frequency response of a Device Under Test (DUT) with a continuous white noise excitation. The white noise generated here has a statistically flat spectrum from 0Hz to ½ sampling frequency. Multi-Instrument is also capable of generating a periodic or pseudo random white noise in a specified frequency range with an ideally flat spectrum even without averaging under restricted conditions. This will be introduced later in the Multitones section.

The following figure illustrates the white noise generated by the software. It has a very flat amplitude spectrum (in dBV) in the range of 0 Hz ~1/2 sampling frequency.



The following figure is the 1/12 octave amplitude spectrum (in dBV) of the same white noise. It shows an incremental of 3dB per octave.

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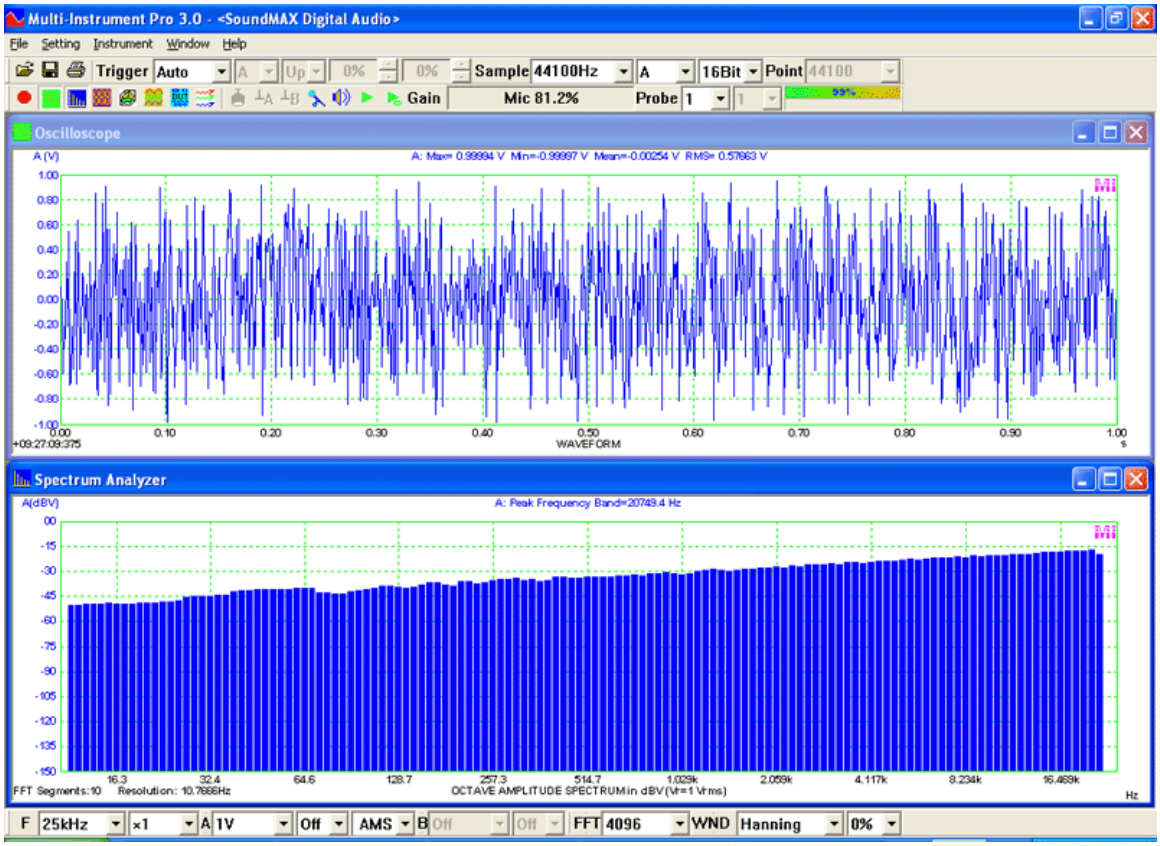
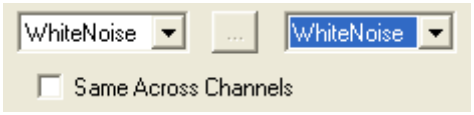


By default, the generated white noises in different channels are not correlated. To generate identical white noises in both channels, tick "Same Across Channels" checkbox in the following figure.

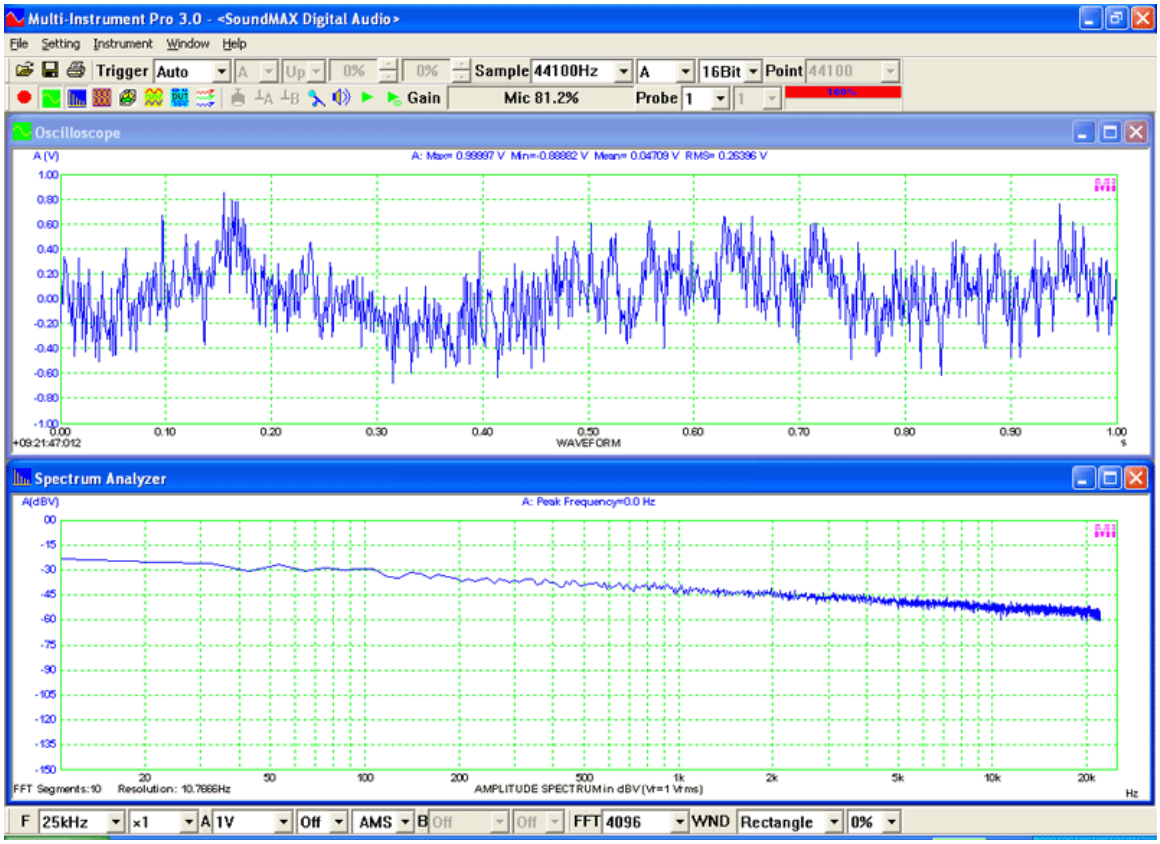


4.3.1.7 Pink Noise

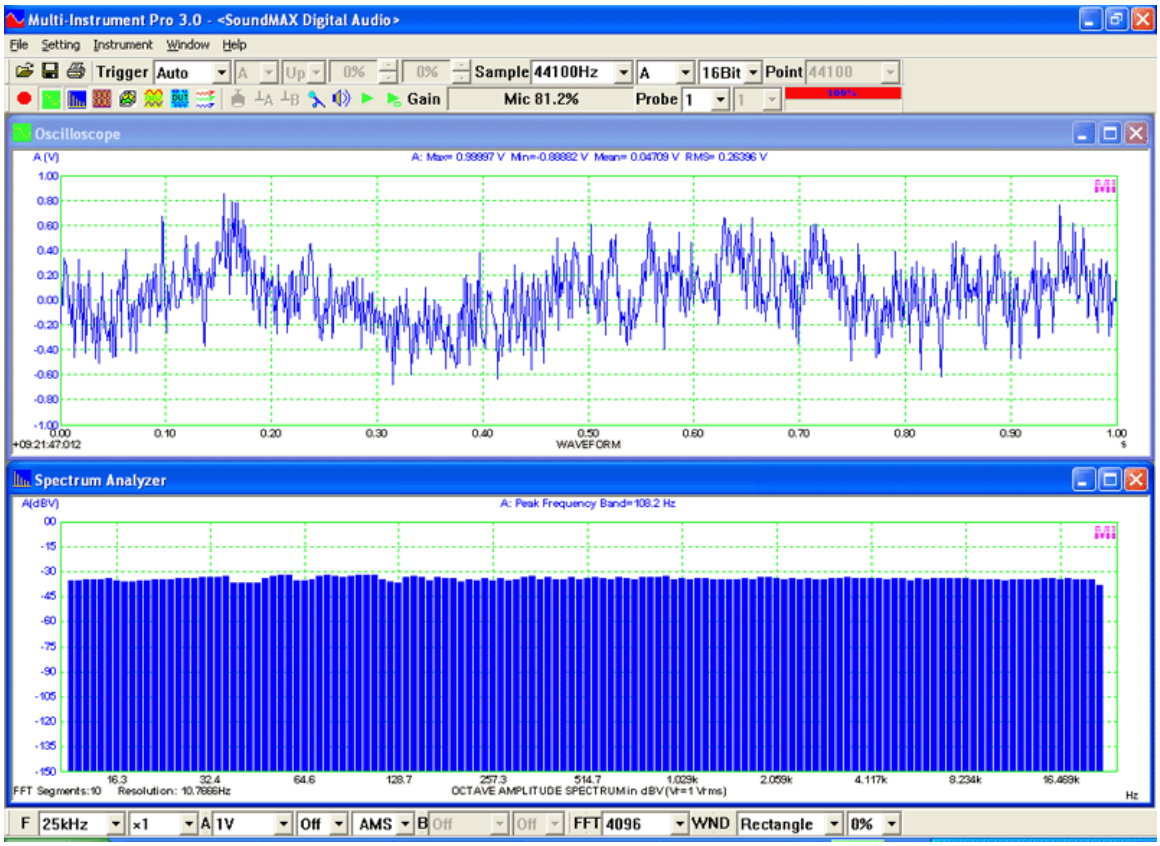
Pink Noise will be generated. Pink noise has an equal amount of energy per octave of bandwidth. No output frequency needs to be specified.

Unlike most of the digital pink noise generators in the market, which eventually repeat the same segment of pink noise, the pink noise generated here is "true" pink noise without any form of repetition. This is very useful when you want to use averaging method to obtain the magnitude frequency response of a Device Under Test (DUT) with a continuous pink noise excitation. The pink noise generated here has a statistically equal amount of energy per octave of bandwidth from 0Hz to ½ sampling frequency. Multi-Instrument is also capable of generating periodic or pseudo random pink noise in a specified frequency range with an ideally equal amount of energy per octave of bandwidth even without averaging under restricted conditions. This will be introduced later in the Multitones section.

The following figure illustrates the pink noise generated by the software. Its amplitude spectrum (in dBV) shows a slope of -3dB/octave.



The following figure is the 1/12 octave amplitude spectrum (in dBV) of the same pink noise. It shows that the spectrum is rather flat in octave scale.



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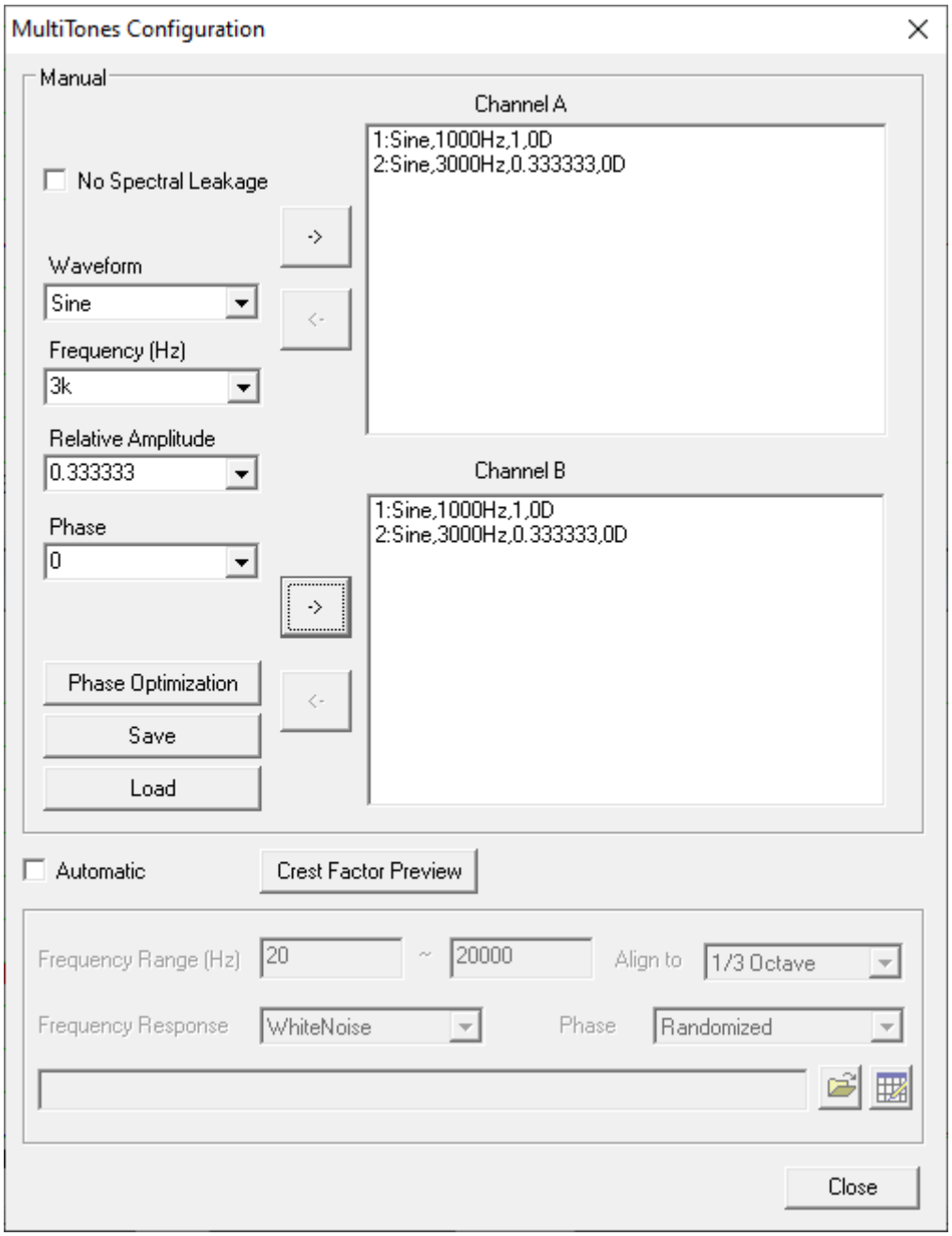
By default, the generated pink noises in different channels are not correlated. To generate identical pink noises in both channels, tick "Same Across Channels" checkbox in the following figure.

4.3.1.8 MultiTones



If MultiTones is selected, the above MultiTones Configuration dialog window will pop up. There are two configuration methods: Manual or Automatic.

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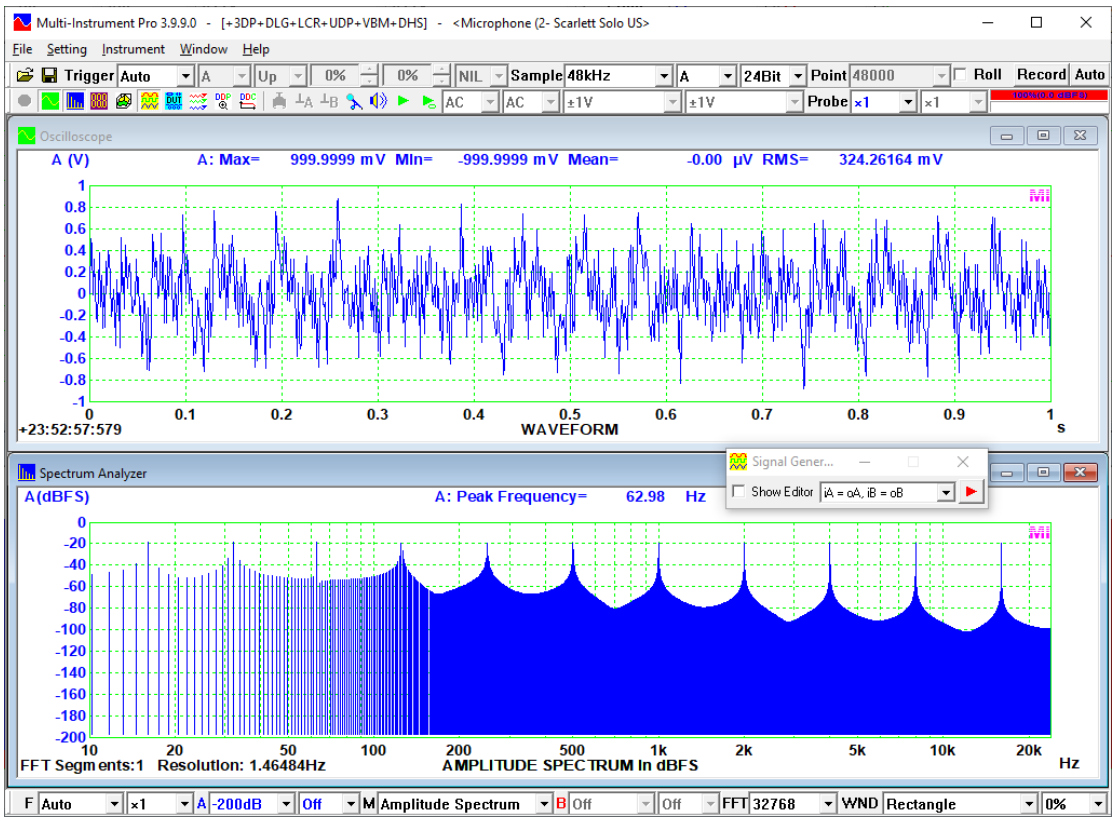
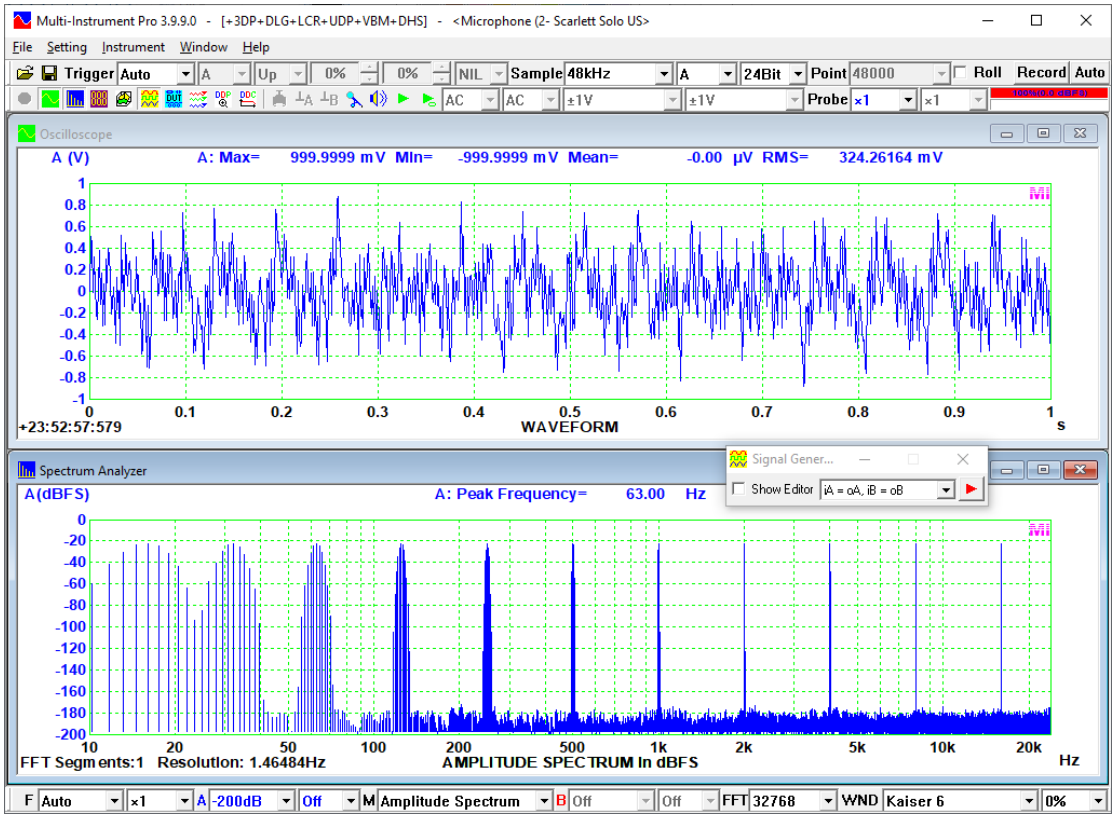
4.3.1.8.1 Manual Configuration

Manual configuration mode allows you to manually add signals with different waveforms, frequencies, amplitudes and initial phases together. You can add as much as 200 tones for each channel. If the "No Spectral Leakage" checkbox is unticked, the available waveform options are: Sine, Square, Triangle, Saw Tooth, White Noise and Pink Noise. Frequencies entered will be rounded to integers under this situation.

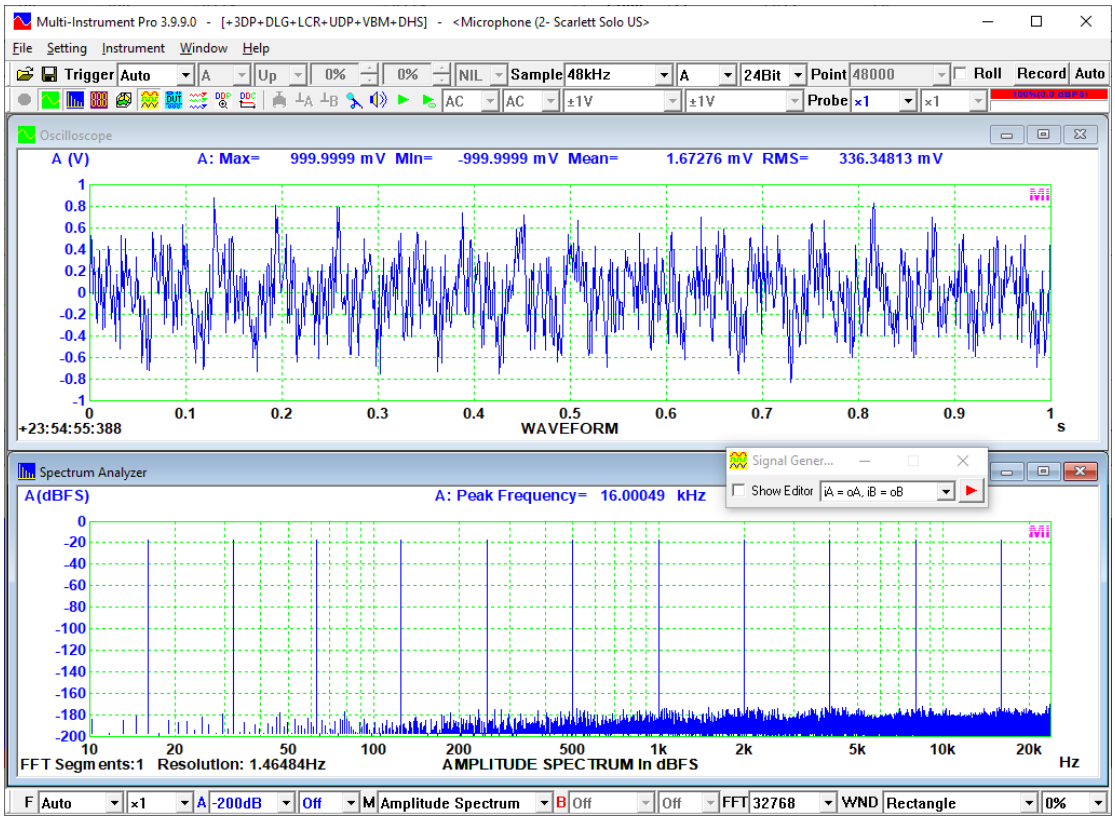
If the "No Spectrum Leakage" checkbox is ticked, only sinewave is allowed. The frequencies specified will be aligned dynamically and silently to the nearest FFT bin centerline frequencies during signal generation so as to avoid spectral leakage. Non-integer frequencies are allowed under this option so that any FFT bin can be targeted at. **For this option to work properly, the sampling rates of the Oscilloscope and Signal Generator must be kept the same.** The signals in the following three screenshots were generated with the same multi-tone configuration: 11 tones corresponding to 1/1 octave centerline frequencies from 16Hz to 16kHz. The comparison shows that this option can be used to excite individual FFT bins without causing spectral leakage.



Generated without "No Spectral Leakage" and Analyzed with Rectangle window



Generated without “No Spectral Leakage” and Analyzed with Kaiser 6 window



Generated with “No Spectral Leakage” and Analyzed with Rectangle window

The available options for Frequency are: 50 MHz, 20 MHz, 10 MHz, 5 MHz, 2 MHz, 1 MHz, 500 kHz, 200 kHz, 100 kHz, 50 kHz, 20 kHz, 10 kHz, 5 kHz, 2 kHz, 1kHz, 500 Hz, 200 Hz, 100 Hz, 50 Hz. Alternatively, you can enter any value that is less than or equal to half of the output sampling frequency specified in the Signal Generator panel. An error message will pop up when this is not satisfied. As the output frequency for each tone has already been specified in this configuration, no output frequency needs to be specified in the Signal Generator panel.

Relative Amplitude, ranging from 0 to 1, is used as the amplitude weighting factor when different waveforms are added together, in order to define their relative strength. However, it has no impact on the absolute output amplitude. The absolute output amplitude is specified in the Signal Generator panel. It refers to the maximum value of the resultant waveform of the MultiTones.

You can add a tone into the MultiTones configuration for each channel by clicking "->" button and you can remove a tone from the MultiTones configuration by clicking "<-" button. The MultiTones configuration can be saved and reloaded. The default extension for a MultiTones configuration file is .tcf.

If all the configured tones are sine waves, the “Phase Optimization” button and the “Crest Factor Preview” button will be enabled. If the “Phase Optimization” button is clicked, the software will try to minimize the crest factor of the synthesized waveform by adjusting the phase of each tone and in the end prompt you to confirm if you want to replace the current phases with the optimized ones. The process may take some time. The optimization algorithm usually works well when all tones are harmonically related.

You can change or review the MultiTones configuration by clicking on the following button in the Signal Generator panel . It will bring up the MultiTones Configuration dialog window.

When the Sweep checkbox in the Signal Generator panel is ticked, a MultiTone waveform can be generated at a frequency (or amplitude) that sweeps linearly or logarithmically within specified frequency (or amplitude) range and time duration. Under frequency swept mode, the frequencies specified in the Multitones Configuration will be ignored and the Start Frequency and End Frequency specified in the Signal Generator panel will be used instead.

4.3.1.8.2 Automatic Configuration

Automatic configuration mode is activated by selecting the “Automatic” checkbox in the MultiTones Configuration dialog window. This mode is used to automatically configure a multitone signal consisting of multiple frequencies within the specified frequency range. These frequencies can be aligned one by one to either the FFT bands or the fractional octave bands (1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96) that are set in the Spectrum Analyzer, in order to achieve a designated spectrum if the generated signal is to be sampled by the Oscilloscope and analyzed by the Spectrum Analyzer. The designated spectrum can be white, pink or arbitrary. The initial phase of each tone is randomized to avoid any prominent time-domain pattern (e.g. much-higher-

than-RMS peaks, steep jumps or falls, etc.) in the aggregate signal and make it noise-like. However, unlike white noise and pink noise, this noise-like multitone signal is deterministic and periodic. Thus it is only pseudo random. One distinct advantage of using this type of periodic noises is that it converges much faster in frequency response tests and thus requires much less number of averages or even no average. The frequency response tests here include magnitude frequency response, transfer function, coherence function, impulse response, impedance & phase tests, etc.. Another distinct advantage is that the test signal's frequency range and the amplitude-frequency profile can be easily controlled. For example, a bandlimited periodic pink noise for speaker tests can be readily generated using this method.

Three phase options are provided: Randomized, Optimized 1 and Optimized 2, with increasing chances of finding a smaller crest factor while taking more time. The phase optimization algorithm here works well under almost all conditions. However, the signals generated with "Optimized 1" and "Optimized 2" are less noise-like than the "Randomized" ones. The "Crest Factor Preview" button can be used to preview the crest factor. Meanwhile, it pre-generates the waveform and save some initial time when outputting the real signals.

4.3.1.8.2.1 Align To FFT Bands

If the multitones are aligned to FFT bands, each FFT band in the specified frequency range will be filled with a tone with a relative level defined by the "Frequency Response". Three options are available in the "Frequency Response" selection box: white noise, pink noise, and arbitrary. These tones are spaced at an interval equal to the FFT resolution. The period of the resulting signal equals to [FFT Size] / [Sampling Frequency].

It should be noted that the sampling rates of the Oscilloscope and Signal Generator must be kept the same in order for this option to function properly.

4.3.1.8.2.1.1 Periodic White Noise

If the "Frequency Response" is set to "White Noise", a periodic white noise will be generated within the specified frequency range. If this signal is used for magnitude frequency response measurement, the following restrictions should be noted:

1) Do not change FFT size and sampling rate after the signal is already generated.
2) Rectangle window should be used
3) [Record Length] >= [FFT Size]
4) Narrowband FFT rather than broadband octave analysis in the Spectrum Analyzer should be used

However, if this signal is employed in transfer function or impulse response measurement which involves division operations in frequency domain, some prominent artefacts may show up if the stimulus is ideally band limited and cannot cover the entire frequency range from 0 to ½ sampling rate. In that case, a window function other than the Rectangle window should be applied.

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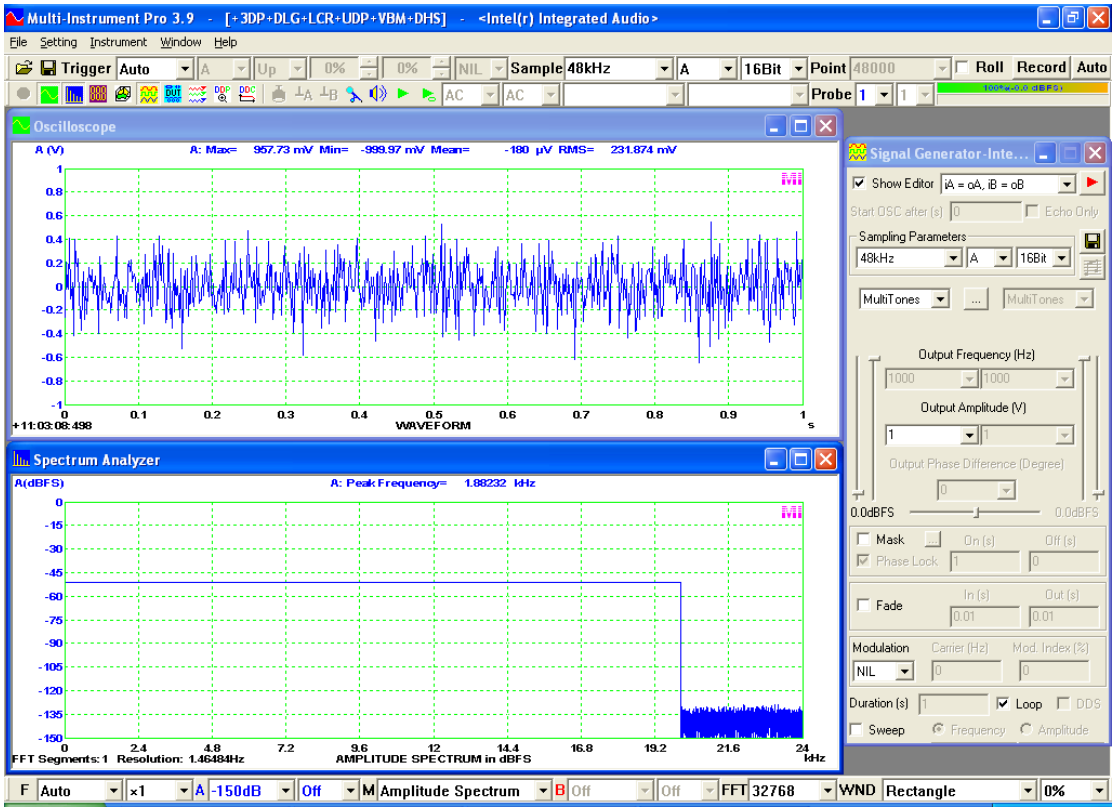
The following figure shows a multitone signal aligned to 32768-point FFT with a white noise frequency response in the range of 20 Hz ~ 20 kHz, and analyzed using narrowband FFT. It can be seen from the figure that the spectrum is ideally flat even without averaging.



4.3.1.8.2.1.2 Periodic Pink Noise

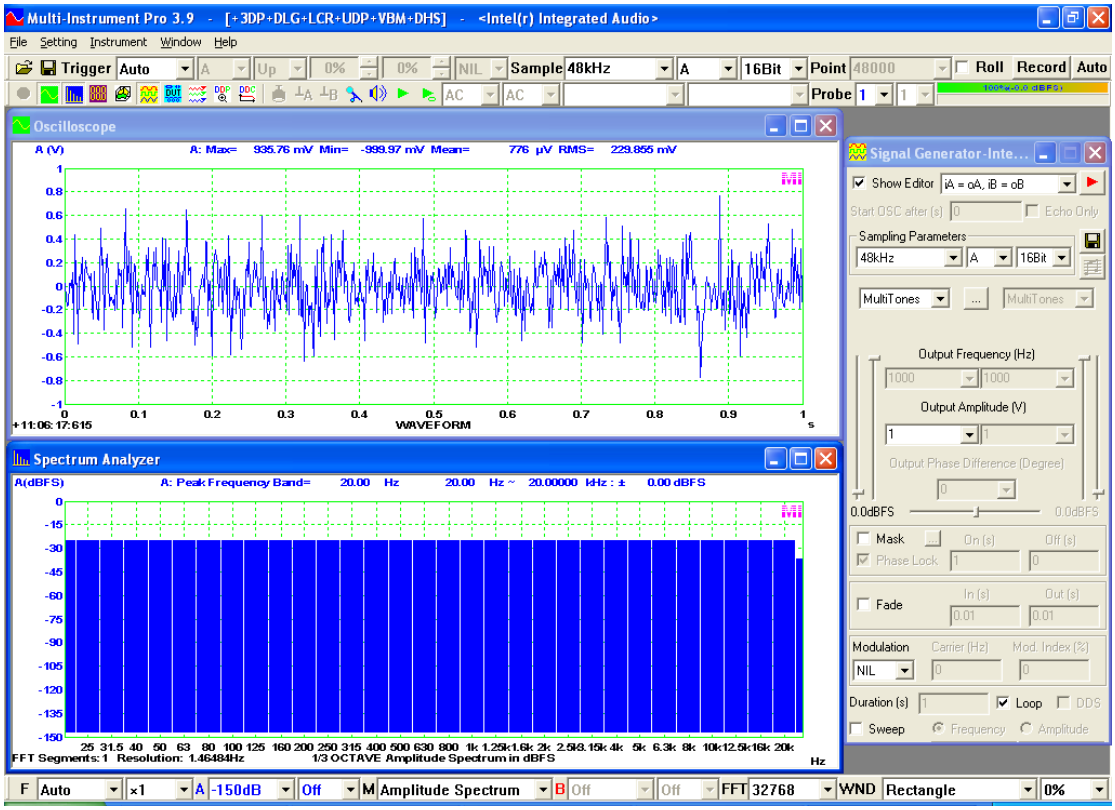
If the "Frequency Response" is set to "Pink Noise", a periodic pink noise will be generated within the specified frequency range. If this signal is used for magnitude frequency response measurement, the following restrictions should be noted:

1) Do not change FFT size and sampling rate after the signal is already generated. The larger the FFT size, the more accurate the octave analysis
2) Rectangle window should be used
3) [Record Length] >= [FFT Size]
4) Broadband octave analysis rather than narrowband FFT in the Spectrum Analyzer should be used

However, if this signal is employed in transfer function or impulse response measurement which involves division operations in frequency domain, some prominent artefacts may show up if the stimulus is ideally band limited and cannot cover the entire frequency range from 0 to ½ sampling rate. In that case, a window function other than the Rectangle window should be applied.

The following figure shows a multitone signal aligned to 32768-point FFT with a pink noise frequency response in the range of 20 Hz ~ 23 kHz, and analyzed using 1/3

octave band analysis. It can be seen from the figure that the spectrum is ideally flat even without averaging.



4.3.1.8.2.1.3 Signal with an Arbitrary Spectrum

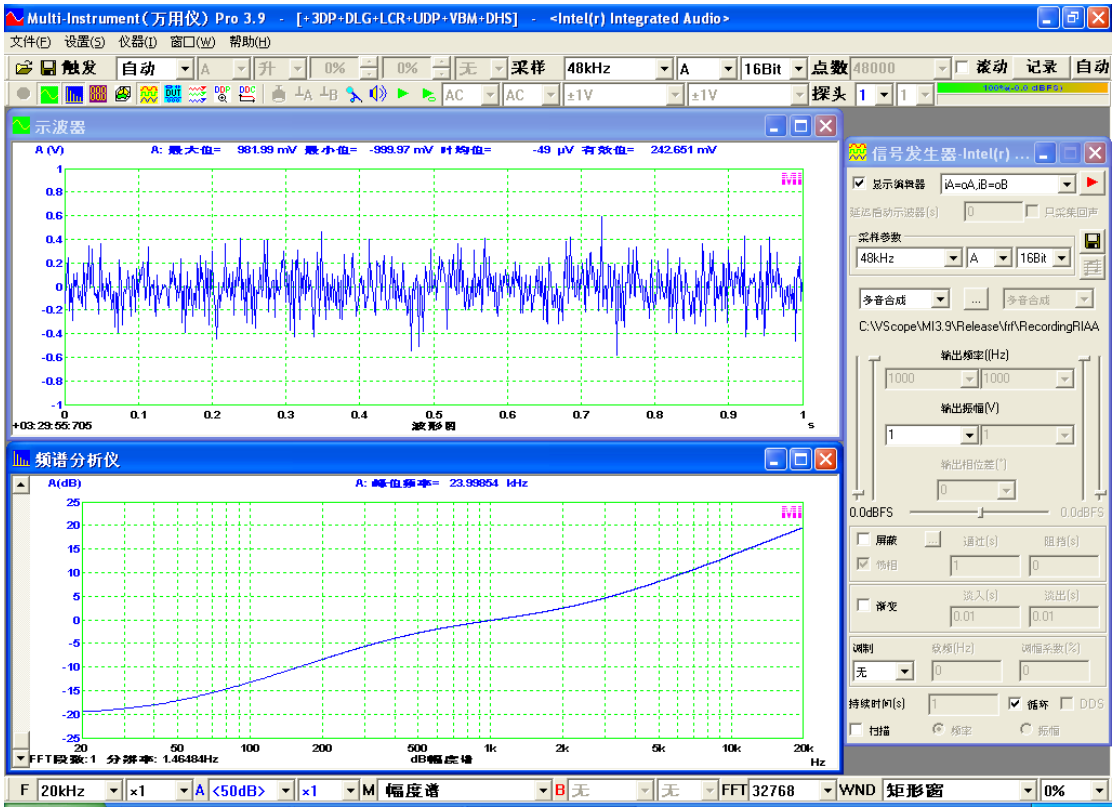
If the "Frequency Response" is set to "Arbitrary", a frequency response file needs to be selected. Please refer to the section regarding arbitrary digital filter in the Oscilloscope described previously for the format of the frequency response file (*.frf). If the generated signal is to be sampled by the Oscilloscope, to achieve the designated spectrum in the Spectrum Analyzer, the following restrictions should be noted:

1) Do not change FFT size and sampling rate after the signal is already generated.
2) Rectangle window should be used
3) [Record Length] >= [FFT Size]

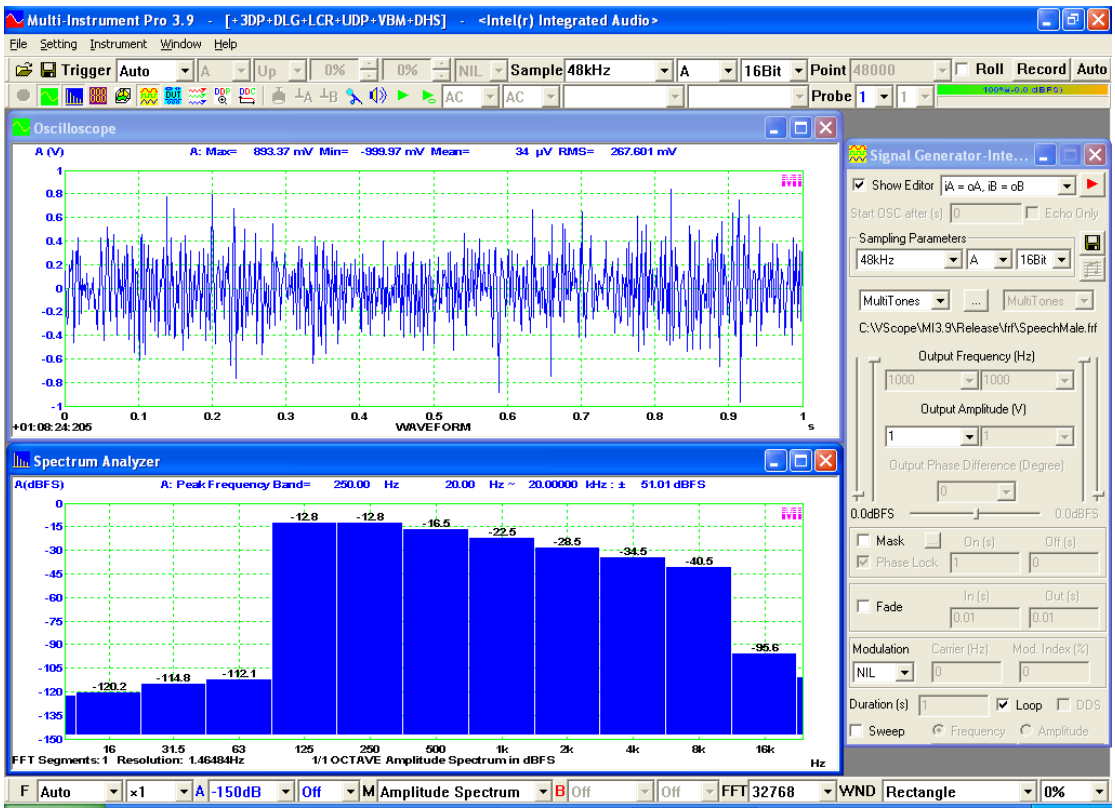
In the FRF subdirectory of the software, some samples frequency response files are provided, including:

(1) RecordingRIAA.frf
(2) ReproducingRIAA.frf
(3) SpeechMale.frf
(4) SpeechFemale.frf

The following figure shows a periodic noise signal generated using RecordingRIAA.frf. It can be used as a frequency response test signal for phono pre-amplifiers.



The following figure shows a periodic noise signal generated using SpeechMale.frf. It can be used as the male Speech Transmission Index test signal according to IEC 60268-16.



The gender-specific spectra of STI test signals are defined in IEC 60268-16 as follows.

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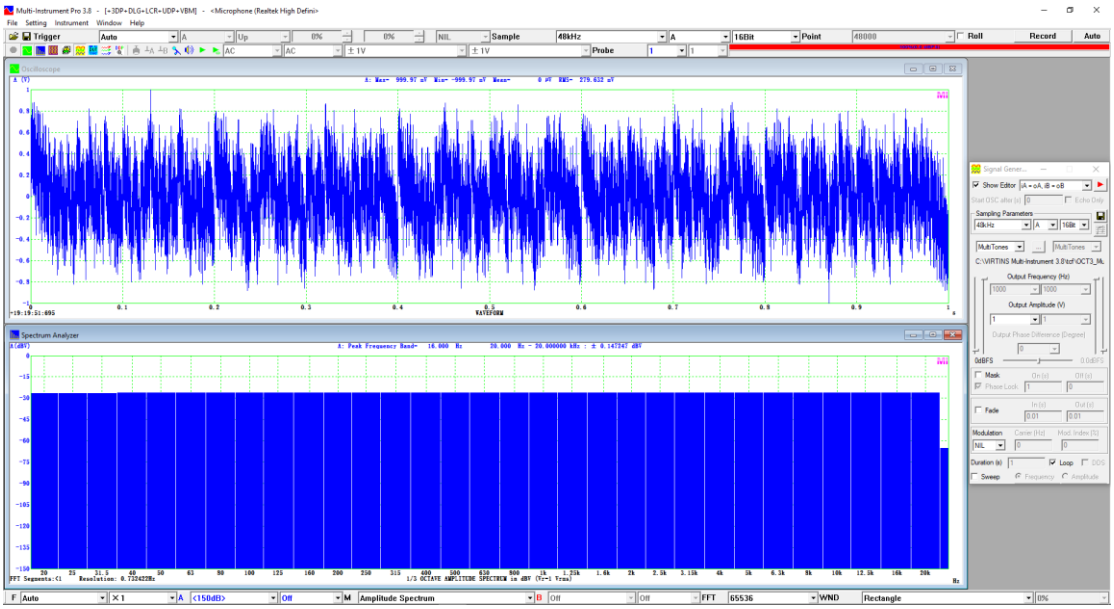
Octave band Hz	125	250	500	1 000	2 000	4 000	8 000	A-weighted
Males dB	2,9	2,9	–0,8	–6,8	–12,8	–18,8	–24,8	0,0
Females dB	–	5,3	–1,9	–9,1	–15,8	–16,7	–18,0	0,0
NOTE： The STIPA test signal is based on the male speech spectrum and only this signal should be generally employed.								

4.3.1.8.2.2 Align To 1/1~1/96 Octave Bands

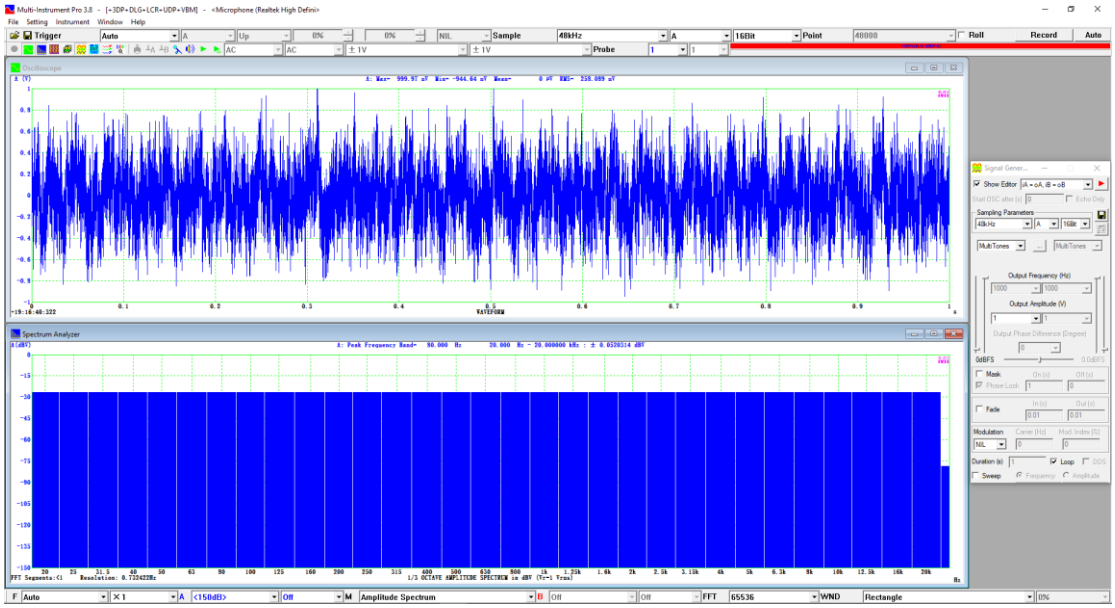
If the multitones are aligned to 1/1~1/96 octave bands, each octave or fractional octave band in the specified frequency range will be filled with a tone with an equal level. No frequency response option is available. These tones are an octave or a fractional octave band apart and thus the spectrum of the multitones follows that of a pink noise. If this signal is used for magnitude frequency response measurement, the following restrictions should be noted:

1) Broadband octave analysis rather than narrowband FFT in the Spectrum Analyzer should be used. Do not change to a finer octave band analysis than the one specified by “Align to”. The larger the FFT size, the more accurate the octave analysis.
2) Rectangle window is recommended for most of the cases.

In the TCF subdirectory of the software, OCT1_MultiTone_11.tcf, OCT3_MultiTone_32.tcf, OCT6_MultiTone_63.tcf and OCT12_MultiTone_125.tcf are manually configured Tone Configuration Files (*.tcf) for 1/1, 1/3, 1/6, 1/12 octave band stimulation in the frequency range of 20 Hz ~ 20 kHz, respectively. Each frequency in these configurations has a fixed initial phase of 0 degree. The following figure shows the 1/3 octave amplitude spectrum (in dBV) of a multitone signal generated using OCT3_MultiTone_32.tcf.

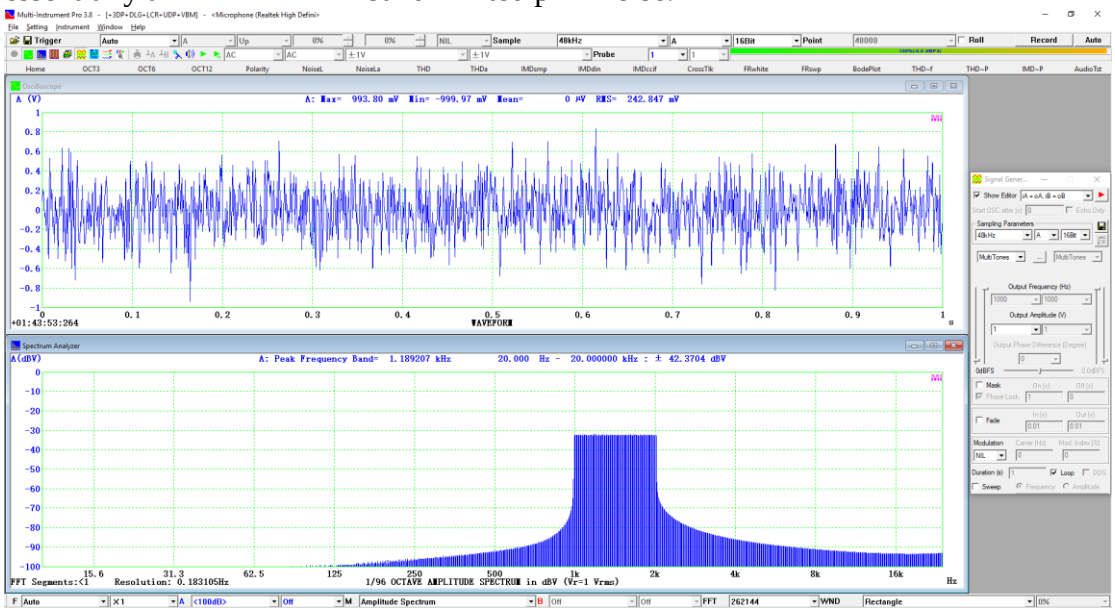


The following figure shows the 1/3 octave amplitude spectrum (in dBV) of a multitone signal aligned to 1/3 octave bands in the range of 20 Hz ~ 20 kHz, generated using the automatic configuration method instead.



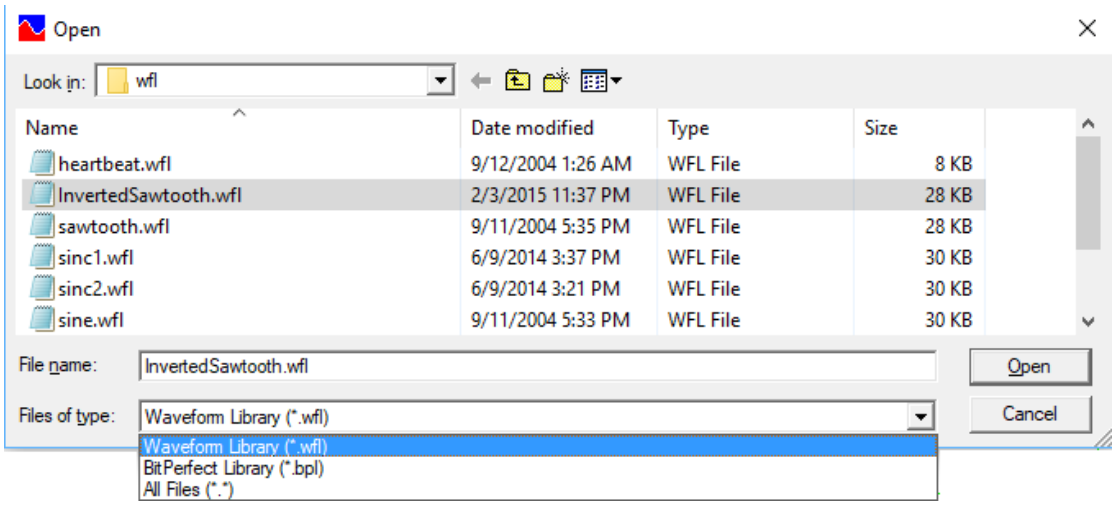
Both of the above two figures have a perfectly flat spectrum in the range of 20 Hz ~ 20 kHz under 1/3 octave analysis with a rectangle window function. However, the former shows some patterns (unwanted sometimes) in time domain while the latter does not.

The following figure shows the 1/96 octave amplitude spectrum (in dBV) of a multitone signal aligned to 1/96 octave bands in the range of 1 kHz ~ 2 kHz. It is essentially a 1 kHz ~ 2 kHz band limited pink noise.



4.3.1.9 Waveform Library (WFLibrary)

If WFLibrary is selected, a dialog window will popup, asking you to select a Waveform Library file (*.wfl), a BitPerfect Library file (*.bpl), or any file (*.*) that conforms to the format of a Waveform Library file (*.wfl).



Both types of waveform libraries *.wfl and *.bpl define an arbitrary wave shape of one cycle of the signal to be generated using a number of discrete samples (points). When a Waveform Library file (*.wfl) is used, the output frequency (i.e. the repetition rate of the defined wave shape) and amplitude can be specified explicitly. This method works in a similar way as the DDS (Direct Digital Synthesis) method. Scaling, resampling and interpolation are involved in order to generate the signal with the specified frequency and amplitude. The resultant digital samples (before DAC) may not have exactly the same values as the ones defined in the library. In this regard, it is not a bit perfect method. In contrast, when a BitPerfect Library file is used, the samples inside in the library file will be output one by one by the output sampling clock without any modification of their values. The output frequency and amplitude are not specified explicitly. They can only be derived, if required, indirectly from the samples in the library. Only file name with an extension ".bpl" will be treated as a BitPerfect library file.

4.3.1.9.1 Waveform Library File (.wfl)*

A Waveform Library file defines an arbitrary wave shape for one cycle of the signal to be generated. It is a Comma Separated Value (CSV) text file with the following format:

Dual-Channel Data with Sequential Numbers or Sampling Time Information

0,0,0
1,0.006135885,0.006135885
2,0.012271538,0.012271538
……
1022,-0.012271645,-0.012271645
1023,-0.006135992,-0.006135992
……

Each row contains the coordinates of a point of the waveform with three variables: the sequential number, the relative amplitude value for Channel A, the relative amplitude value for Channel B. Minimum two points must be specified per waveform and you can define as many points as you want. There is no restriction on the range of the relative amplitude value as relative values are used inside the software. However, for easy understanding purpose, it is recommended to use values between –1 and 1. It should be noted that if there is DC component in the waveform, it will be filtered out if the signal is output via the sound card hardware. The following formats are also supported:

Single-Channel Data with Sequential Numbers or Sampling Time Information

```
0,0
1,0.006135885
2,0.012271538
……
1022,-0.012271645
1023,-0.006135992
……
```

Bare Single-Channel Data

```
0
0.006135885
0.012271538
……
-0.012271645
-0.006135992
……
```

Bare Dual-Channel Data

```
0,0
0.006135885,0.006135885
0.012271538,0.012271538
……
-0.012271645,-0.012271645
-0.006135992,-0.006135992
……
```

You can use Microsoft EXCEL, Notepad or other third party software to composite a Waveform Library file. One advantage of using Microsoft EXCEL is that you can make use of its mathematical function to construct the wave shape and then use its charting function to have a preview on it. There is no restriction on the file extension to be used either. However, it is recommended to use .wfl as the file extension.

Waveform libraries provide great flexibility to the Signal Generator. Any arbitrary wave shape can thus be generated by constructing a corresponding waveform library.

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An arbitrary waveform can be generated at either a fixed frequency (or amplitude) or a frequency (or amplitude) that sweeps linearly or logarithmically within the specified frequency (amplitude) range and time duration.

You can change the waveform library by clicking on the following button in the Signal Generator panel: . It will bring up the file "Open" Window.

Some sample WFL files are provided in the WFL subdirectory of the software.

4.3.1.9.2 BitPerfect Library File (.bpl)*

A BitPerfect Library file defines an arbitrary wave shape for one cycle of the signal to be generated. It is a Comma Separated Value (CSV) text file with the following format:

Dual-Channel Data with Sequential Numbers or Sampling Time Information

0,12582912,12582912
1,12582912,12582912
2,4194304,4194304
3,4194304,4194304

Each row contains the coordinates of a sample of the waveform with three variables: the sequential number, the sample (a positive integer value) for Channel A, the sample (a positive integer value) for Channel B. At least one sample must be specified per waveform and you can define as many samples as you want. The positive integer values must be within the range allowed by the output sampling bits: 0~255 for an 8-bit DAC (unsigned, +FS = 255, -FS = 0), 0~65535 for a 16-bit DAC (signed, +FS = 32767, -FS = 32768), and 0~16777215 for a 24-bit DAC (signed, +FS = 8388607, -FS = 8388608). This is in accordance with the WAV file format. The signed integers are represented by two's complement. Each output sampling clock pulse will push one sample out until all the samples in the library are done before repeating from Sample 0 again. Thus the repetition rate of the defined waveform equals to [Sampling Frequency] / [Number of Samples Defined in the BitPerfect library]. The following formats are also supported:

Single-Channel Data with Sequential Numbers or Sampling Time Information

0,12582912
1,12582912
2,4194304
3,4194304

Bare Single-Channel Data

12582912
12582912
4194304
4194304

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Bare Dual-Channel Data

12582912,12582912
12582912,12582912
4194304,4194304
4194304,4194304

You can change the waveform library by clicking on the following button in the main window of the Signal Generator: . It will bring up the file "Open" Window.

Some sample BPL files including those for jitter tests are provided in the WFL subdirectory of the software.

4.3.1.9.2.1 J-Test Signal for Jitter Tests

The J-Test signal developed by Julian Dunn is widely used to stimulate worse-case jitter over an AES3 digital audio connection, such as balanced and unbalanced pro formats, as well as the coaxial and optical SPDIF consumer variants. It contains an un-dithered “square wave” at 1/4 of the sample rate (e.g. 12kHz for a 48 kHz sample rate), with an amplitude of a half of the full scale (Note: when converted to an analog signal, the "square wave" would be bandlimited at the half of the sampling rate and thus becomes a sine wave with an amplitude of 0.707 of the full scale), to which is added an un-dithered low-frequency (usually at 1/192 of the sampling rate, e.g. 250Hz for a 48kHz sampling rate) square wave with an amplitude of 1 LSB (Least Significant Bit) toggling between 0 to -1 LSB. The spectrum of a 24-bit J-Test signal sampled at 48kHz is shown in the following figure. The high-frequency sine wave has a single frequency component at 12kHz while the low-frequency square wave has a fundamental frequency at 250Hz and all its odd harmonics. It can be seen that the 12kHz signal exceeds its neighboring side bands by more than 170dB and the 250Hz by more than 140 dB under no-jitter condition. This leaves a good room for jitter measurement as jitter will show up as side bands around the signal frequency due to its phase modulation nature.

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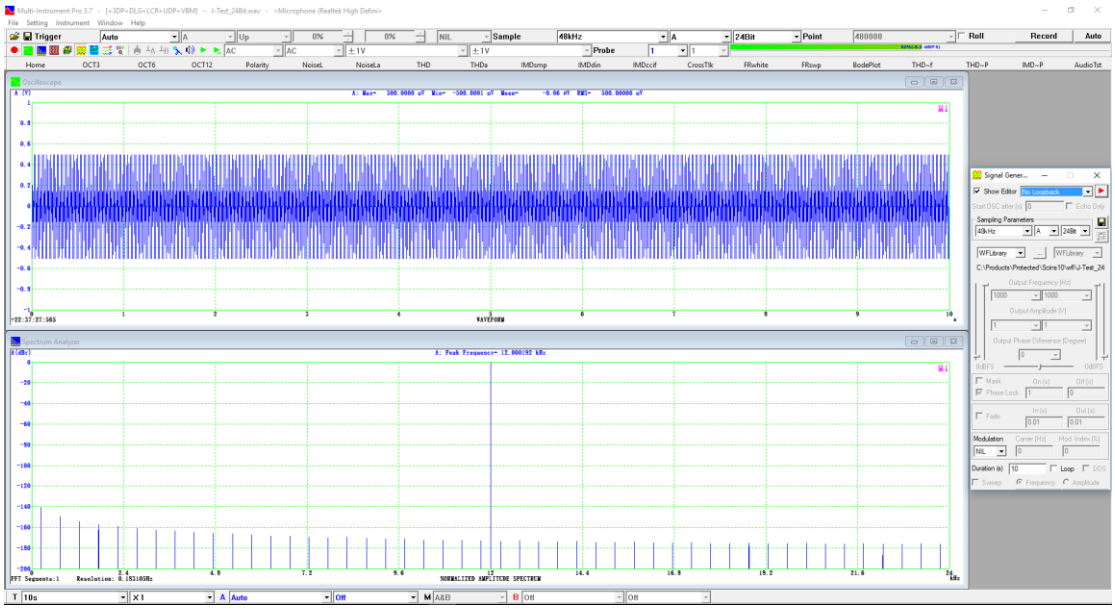

The above 24-bit J-Test signal is generated using the BitPerfect library: J-Test_24Bit.bpl in the software's WFL subdirectory. It consists of 192 samples in HEX format:

C00000,C00000,400000,400000 (× 24)
BFFFFF,BFFFFF,3FFFFF,3FFFFF (× 24)

Or in Decimal format (only this format should be used in the BitPerfect library):

12582912,12582912,4194304,4194304 (× 24)
12582911,12582911,4194303,4194303 (× 24)

The above BPL file must be used with a sampling bit resolution of 24. It automatically adapts to any sampling rate selected. For example, if the sampling rate is 48kHz, the repetition rate of the waveform would be 48000/192=250 Hz. If the sampling rate is 44.1kHz, the repetition rate would be 44100/192=229.6875 Hz. The 16-bit counterpart of this file is J-Test_16Bit.bpl in the same subdirectory. SineAtOneFourthFs_24Bit.bpl and SineAtOneFourthFs_16Bit.bpl in the same subdirectory can be used to generate a clean sine wave signal at ¼ of the sampling rate.

In a J-Test, periodic jitter would show up as sideband pairs around the signal frequency while random jitter would increase the noise level. The jitter frequencies can be determined by the distances from the sidebands to the signal frequency, while the jitter amplitudes can be inferred from the amplitude ratio of the sidebands to the signal frequency. The RMS jitter level in (picoseconds) can be determined by: $R=20\times\log_{10}(J_{rms}\times f)-227dB$, where R is the amplitude ratio (in dB) of one sideband of the jitter frequency to the signal frequency, J_{rms} the RMS jitter level in picoseconds, and f the signal frequency in Hz. A jitter frequency will create a pair of sidebands around the signal frequency. If R is computed based on the combined energy from the double sidebands, then the above formula becomes $R'=20\times\log_{10}(J_{rms}\times f)-224dB$.

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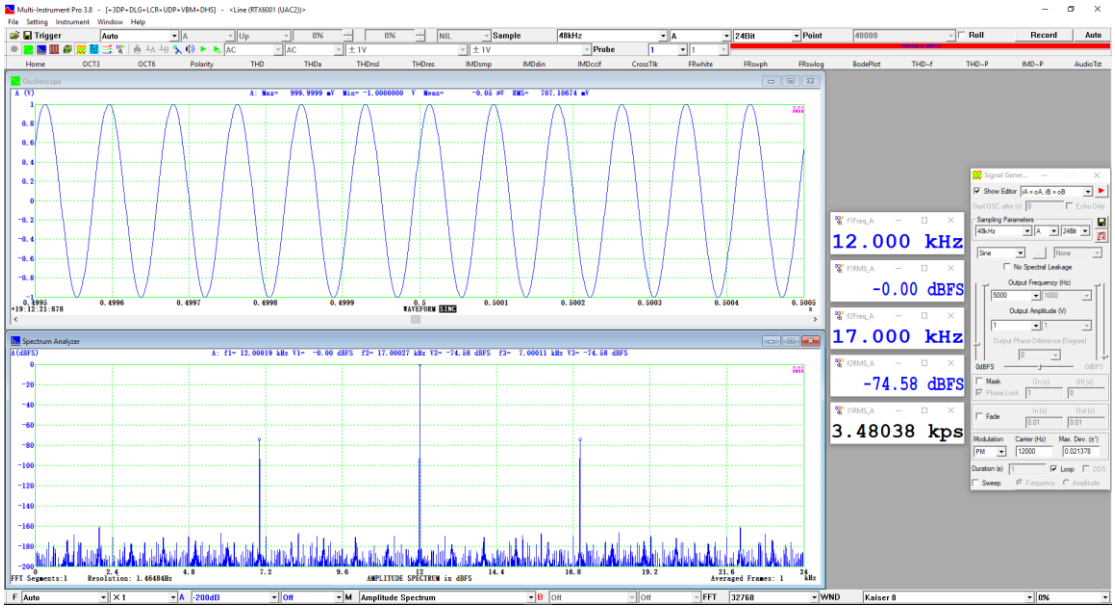
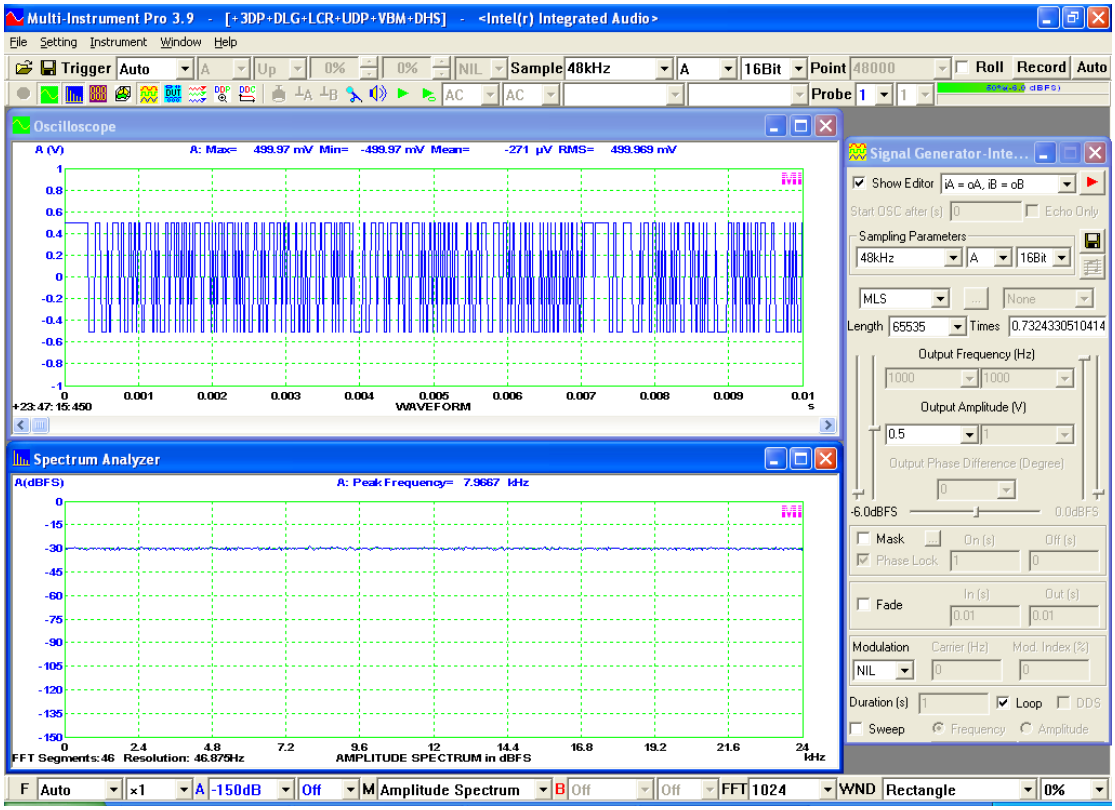
Two preconfigured panel setting files: J-Test_24Bit.psf and J-Test_16Bit.psf are provided in the Panel Setting File (PSF) directory of the software. They can be used to measure the total RMS jitter level in time by assuming all the sidebands and noises are caused by jitter only. The RMS jitter level in time for the second highest peak in the spectrum is also calculated. It is assumed that the second highest peak is one of the sideband of periodic jitter around the first highest peak. These two panel setting files can also be used to measure jitter when the test frequency is a pure sinewave instead of the J-Test signal. The following figure shows a 12kHz tone with a 3.5ns RMS 5kHz sinewave jitter.

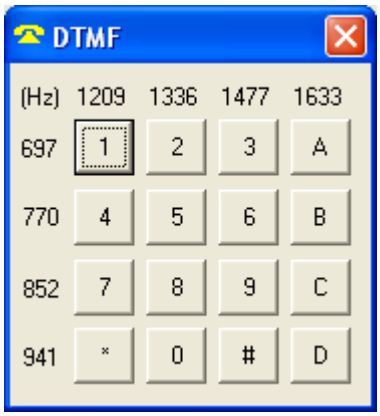


4.3.1.10 MLS

Maximum Length Sequence (MLS) is a pseudo random sequence of pulses consisting of the values 1 and –1. It is actually periodic with the period equal to the length of the sequence, which can be chosen from: 127, 255, 511, 1023, 2047, 4095, 8191, 16383, 32767, 65535, 131071, 262143, 524287, 1048575, 2097151, 4194303, 8388607, and 16777215 points (refer to the figure below). You can also explicitly specify how many times the sequence will be repeated, and the signal "Duration" will be updated automatically. If you opt to specify the signal "Duration" instead, then the "Times" will be updated automatically. MLS is often used as stimulus in frequency response or impulse response measurements. It has a white spectrum from 0 Hz to ½ sampling rate, if its length is equal to or greater than the Record Length, so as to avoid any repetition of the sequence in the sampled data. The figure below shows an example of a 65535-point MLS signal. The Record length is 48000 and the FFT size is 1024. Therefore the number of intra-frame average is 48000/1024=46. It can be seen from the figure that the resulting spectrum is very flat in the range of 0 Hz ~ 1/2 sampling rate.



4.3.1.11 DTMF

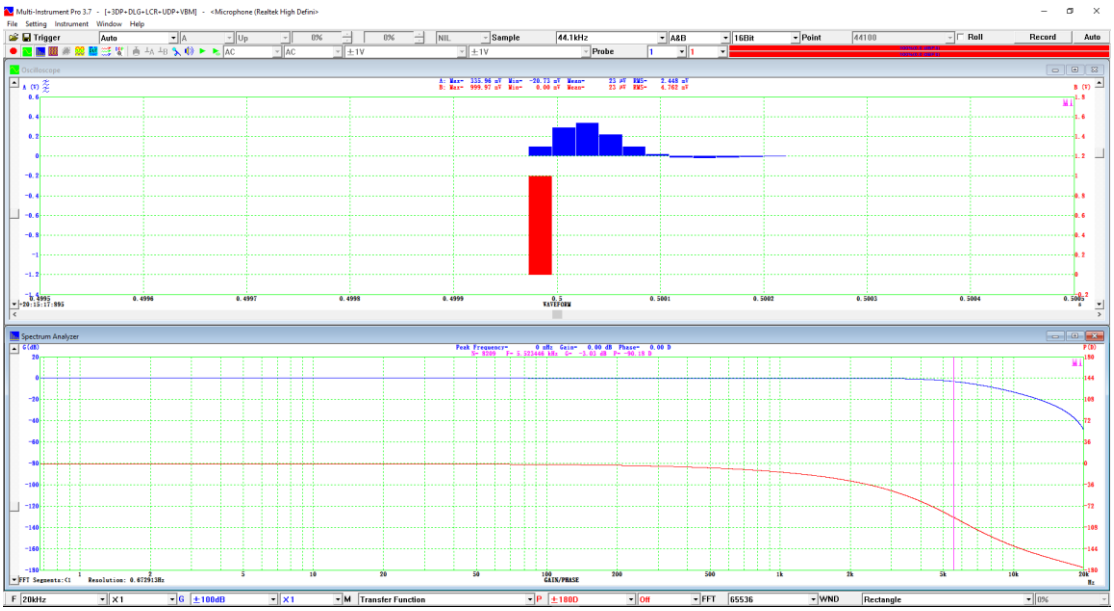


A Dual-Tone Multi-Frequency (DTMF) signal is used for telephone signaling over the line in the voice-frequency band to the call switching center. The above panel (same as the standard telephone panel) will pop up when DTMF is selected. A short signal (0.1s by default) consists of two equal-amplitude frequencies will be generated when a button on the panel is pressed. For example, if the button labelled "8" is pressed, a signal contains 852 Hz and 1336 Hz with an amplitude ratio 1:1 will be generated.

4.3.1.12 Unit Impulse

A Unit Impulse is a signal with zero values at all places except one sample which has a unit value. It can be used to test a system's impulse response. Its amplitude spectrum is a horizontal straight line.

The Unit Impulse has some particular use. The following figure illustrates the transfer function of a digital 5513 Hz second order Butterworth low pass filter, measured using the Unit Impulse, with the stimulus data stored in Channel B and the response data stored in Channel A. It shows that the gain maintains at nearly 0 dB from 0 Hz to about 5513 Hz, and then start to drop down very quickly, meanwhile the phase changes gradually from 0 degree at 0 Hz towards –180 degree as the frequency goes to infinity. The cursor reader indicates that at 5523.45 Hz (i.e. around the cutoff frequency), the gain is about –3.03 dB and the phase is about -90.18 degree. Note that the waveform of Channel A in the Oscilloscope represents the unit impulse response of the filter directly.



The following figure illustrates the unit impulse response of a digital 5000 Hz FIR low pass filter (1022-order Kaiser 6 windowed), measured using the Unit Impulse, with the stimulus data stored in Channel B and the response data stored in Channel A. It shows that the impulse response is delayed by [FIR Order]/2/[Sampling Rate]=1022/2/44100=0.0115873 (s) compared with the stimulus. The impulse response is symmetric and thus the filter has a linear phase response.

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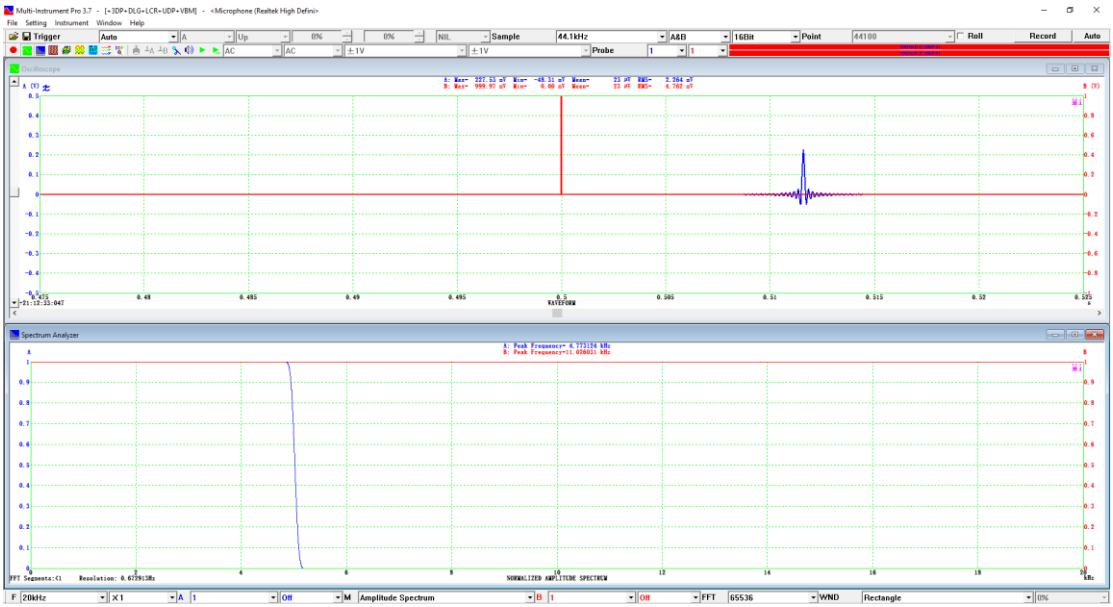
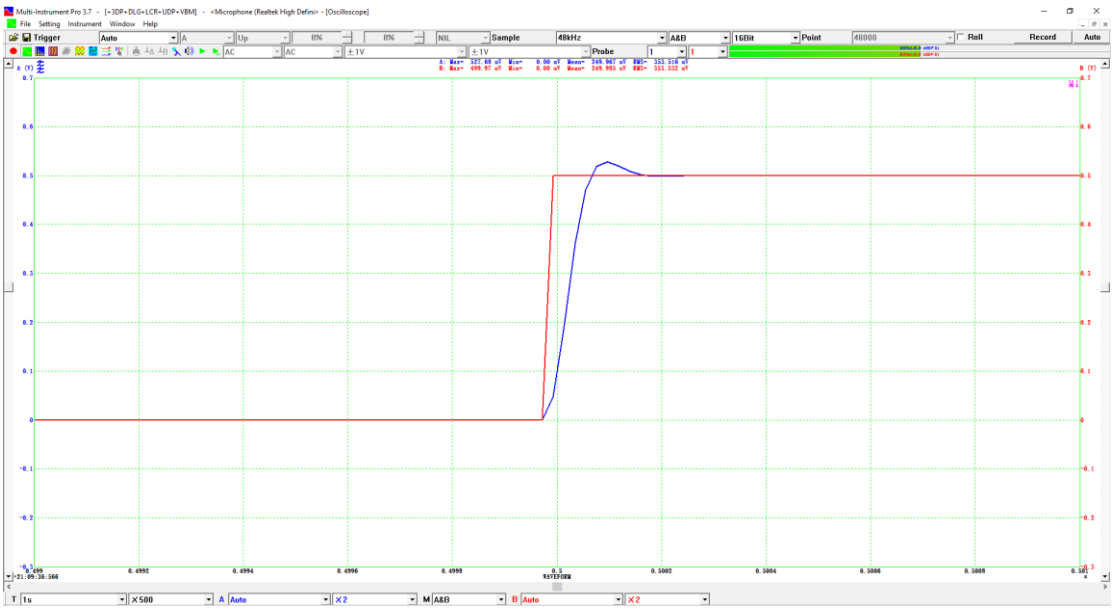


4.3.1.13 Unit Step

A Unit Step is a signal that maintains at zero until a sudden jump from zero to a unit value and then maintains at that level afterwards. It can be used to test a system's step response.

The following figure illustrates the step response of a digital 5513 Hz second order Butterworth low pass filter, measured using the Unit Step signal, with the stimulus data stored in Channel B and the response data stored in Channel A. To avoid the clipping of the digital filter, the stimulus is generated at half of the output full-scale voltage. The figure shows that the overshoot for this case is about 5.5%.



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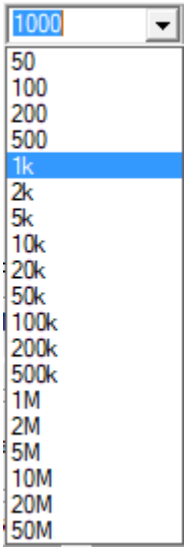
4.3.1.14 Wave File

An 8-bit, 16-bit, 24-bit, or 32-bit standard or extensible wave file in PCM format can be replayed through the Signal Generator. The sampling frequency and bit resolution of the wave file do not have to be the same as those of the Signal Generator as the software is able to convert them automatically. If the wave file contains two channels but the signal generator has only one channel, then the user will be prompted to select which channel of the wave file he/she wants to output. The wave file replaying function is only supported in streaming mode and not supported in DDS mode. Whether the output data are continuous or not depends on the system throughput, sampling frequency, bit resolution, number of sampling channels, etc. It is generally possible to output signals continuously without any interruption using sound cards. In case there is a possible discontinuity in the output signal, a warning cross mark will be shown as follows. If this happens, you will have to offload the computer (e.g. stop the Oscilloscope), or choose a lower sampling frequency, etc.

It is possible to use the Record function in the Oscilloscope to record a wave file and later replay it through the Signal Generator.

4.3.2 Output Frequency



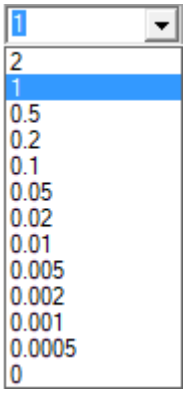
The available options are: 50MHz, 20MHz, 10MHz, 5MHz, 2MHz, 1MHz, 500kHz, 200kHz, 100kHz, 50kHz, 20kHz, 10kHz, 5kHz, 2kHz, 1kHz, 500Hz, 200Hz, 100Hz, 50Hz.

If you cannot find the output frequency you want from the above options, you can enter the value you want directly. It can be a non-integer value. You can also use the left most vertical scroll bar (for Channel A) or the right most vertical scroll bar (for Channel B) to adjust the value of the output frequency.

It should be noted that the software will not allow the fundamental frequency of an output signal to be greater than half of the sampling frequency selected.

The Output Frequency is not applicable and is disabled when the waveform is "None", "White Noise", "Pink Noise" or "MultiTone", "MLS", "DTMF", "Unit Impulse", "Unit Step", or the Signal Generator is under frequency sweep mode.

4.3.3 Output Amplitude



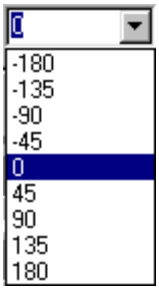
The available options are: 2, 1, 0.5, 0.2, 0.1, 0.05, 0.02, 0.01, 0.005, 0.002, 0.001, 0.0005, 0. Note that the engineering unit of the Output Amplitude can be either V (voltage) or A (ampere) depending on the DAC device used.

If you cannot find the output amplitude you want from the above options, you can enter the value you want directly. You can also use the vertical scroll bar to adjust the value of the output amplitude. The output amplitude will also be displayed in dBFS, dBu or dBV, just below the vertical scroll bar. Click on the text to switch the display.

It should be noted that the software will not allow you to specify an output amplitude greater than the output full-scale voltage (1/2 Vpp) configured.

The Output Amplitude is disabled when the waveform is "None" or the Signal Generator is under amplitude sweep mode.

4.3.4 Output Phase Difference



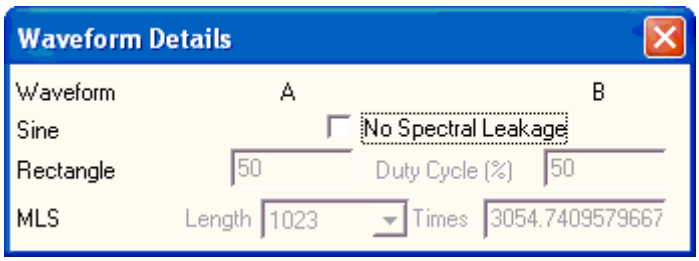
You can specify the Output Phase Difference between the output signals in the two channels when their output frequencies are the same. This is only applicable to

waveforms "Sine", "Rectangle", "Triangle", "Saw Tooth", in non-sweep mode. For different waveforms, the phase difference is calculated with respect to the initial phase described in the waveform sections.

4.3.5 Output DC Offset

For DAC devices that support output DC offset, the above spin boxes will become visible just below the waveform selection combo boxes in the Signal Generator panel. You can use the spin boxes to adjust the output DC offset.

4.3.6 Waveform Details



Some types of waveforms have additional parameters to be specified, such as the duty cycle of a rectangle wave. These parameters are normally displayed just below the waveform selection combo boxes. However, if that place is taken up by the output DC offset parameters, then you can click the button in the Signal Generator panel to call up the Waveform Details dialog.

4.4 Sweep Parameters

When “Sweep” checkbox is ticked and “Frequency” radio button is selected, the waveform can be output at a frequency that sweeps linearly or logarithmically within the specified frequency range and time duration. This is called frequency sweep.

When “Sweep” checkbox is ticked and “Amplitude” radio button is selected, the waveform can be output at an amplitude that sweeps linearly or logarithmically within the specified amplitude range and time duration. This is called amplitude sweep.

4.4.1 Frequency Sweep

4.4.1.1 Start Frequency

Each channel has its own Start Frequency. You can enter a value between 0 Hz and half of the Output Sampling Frequency specified. You will be notified if the value is out of range.

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4.4.1.2 End Frequency

Each channel has its own End Frequency. You can enter a value between 0 Hz and half of the Output Sampling Frequency specified. You will be notified if the value is out of range.

4.4.2 Amplitude Sweep

4.4.2.1 Start Amplitude

Each channel has its own Start Amplitude. You can enter a value between 0 and the output full-scale voltage (1/2 Vpp). You will be notified if the value is out of range.

4.4.2.2 End Amplitude

Each channel has its own End Amplitude. You can enter a value between 0 and the output full-scale voltage (1/2 Vpp). You will be notified if the value is out of range.

4.4.3 Sweep Mode

You can specify either linear or logarithmic sweep mode.

A frequency linearly swept sine wave has an equal amount of energy per Hz of bandwidth within the swept frequency range while a frequency logarithmically swept sine wave has an equal amount of energy per octave of bandwidth within the swept frequency range. The spectrum of the former is white-noise-like while that of the latter is pink-noise-like.

For magnitude frequency response tests, a frequency linearly swept sine wave should be analyzed using narrow band FFT analysis while a frequency logarithmically swept sine wave should be analyzed using octave band analysis. The longer the sweep duration, the better the frequency resolution as well as the accuracy at the lower frequency end. Linear frequency sweep has an equal frequency resolution everywhere along the frequency axis. In contrast, logarithmic frequency sweep has finer frequency resolutions at lower frequencies and coarser frequency resolutions at higher frequencies. The oscilloscope frame width must be set to be equal to or greater than the frequency sweep duration, and the trigger parameters should be set properly to ensure that the entire frequency sweep signal is captured in one oscilloscope frame, from the start frequency to the end frequency. The FFT size must be equal to or greater than the Record Length of the Oscilloscope to avoid the intra-frame average. The “loop” checkbox in the Signal Generator panel must be unticked so that the sweep can be fired individually. This is to prevent the Oscilloscope from capturing something that spans two continuous sweeps.

The following figure shows the spectrum of a 1-second 10 Hz ~ 21 kHz frequency linearly swept sine wave analyzed using 65536-point FFT with a Rectangle window, at the sampling rate of 48 kHz. It can be seen that the frequency response is perfectly

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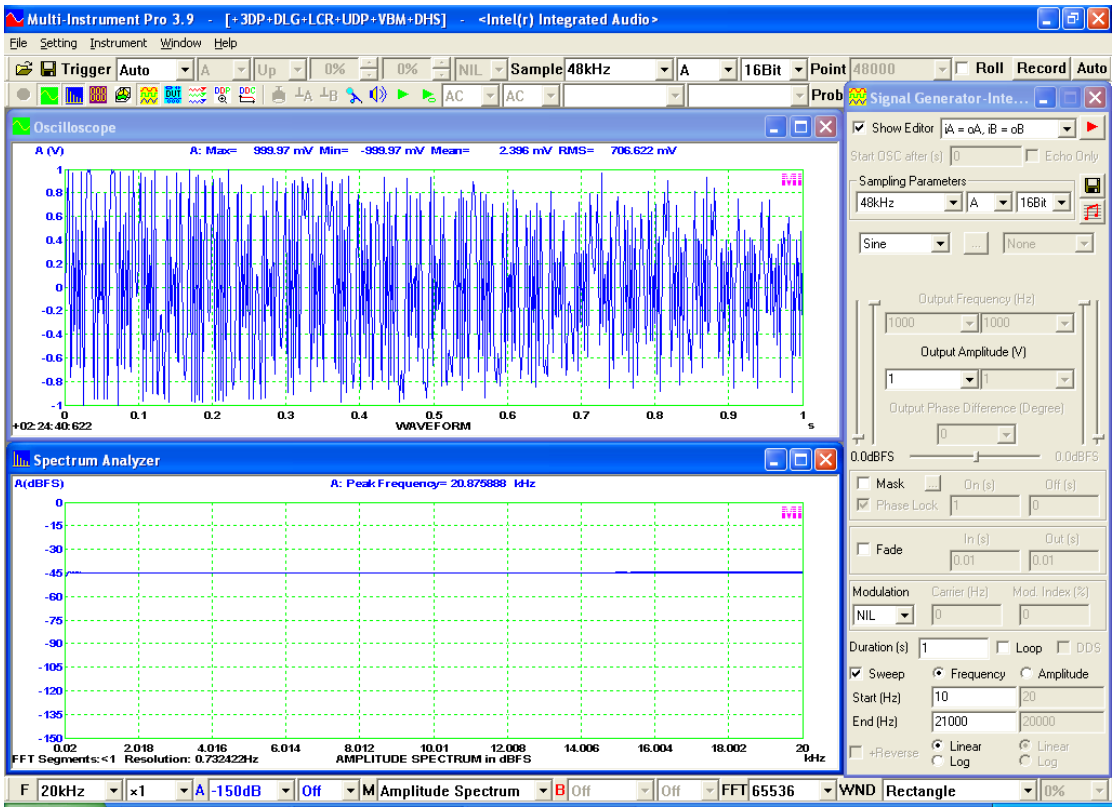
flat within the range of 20 ~ 20 kHz. Note that the frequency sweep range is a little wider than the frequency range of interest in order to minimize any potential boundary effects.



The following figure shows the spectrum of a 1-second 10 ~ 24 kHz frequency logarithmically swept sine wave analyzed using 1/3 octave analysis based on 65536-point FFT with a Rectangle window, at the sampling rate of 48 kHz. It can be seen that the frequency response is perfectly flat within the range of 20 ~ 20 kHz. Note that the frequency sweep range is a little wider than the frequency range of interest in order to minimize any potential boundary effects. Logarithmic frequency sweep mode spends more time on lower frequencies and thus requires less time to achieve the same resolution at the lower frequency end than linear frequency sweep mode.

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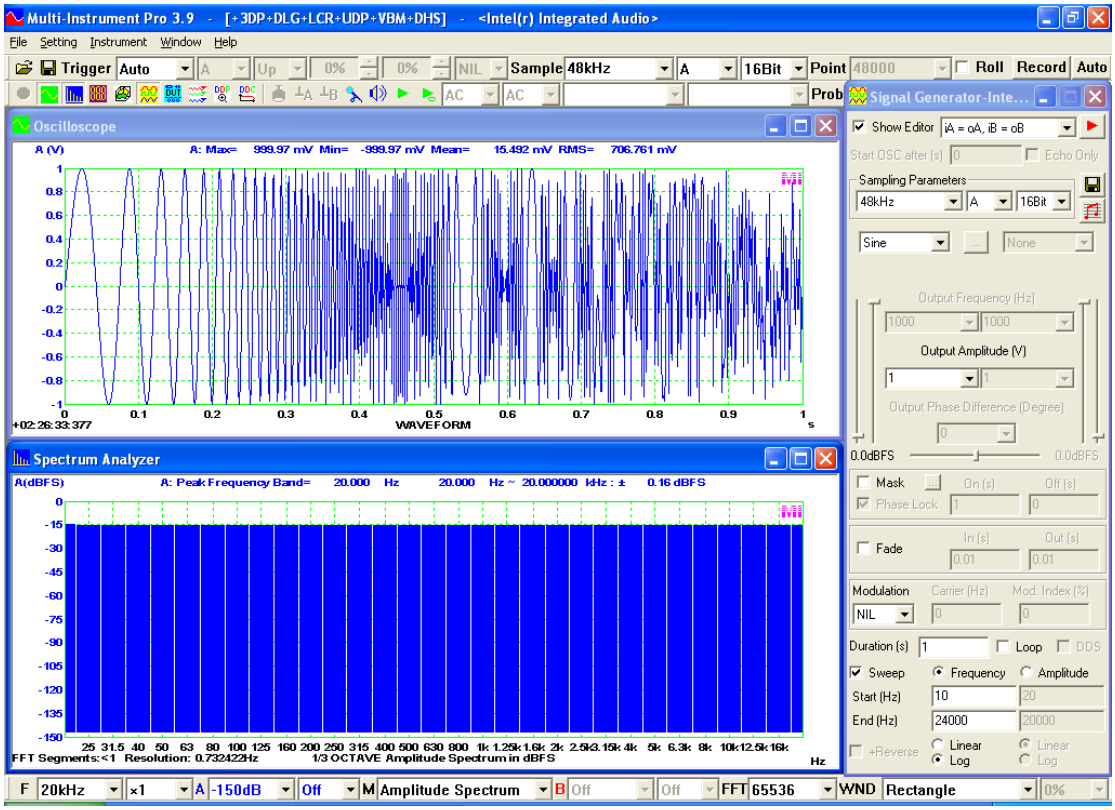

Magnitude frequency response tests can also be performed using the peak hold function of the Spectrum Analyzer. In this method, the oscilloscope frame width must be set to be much less than the frequency sweep duration such that the data sampled in an oscilloscope frame contains very little frequency variation. A smooth magnitude frequency response curve can be obtained by peak-holding a large number of consecutive frames of spectra through the inter-frame processing. The "Loop" checkbox in the Signal Generator panel must be ticked so that the sweep can be fired continuously. Moreover, to make multiple sweeps interconnected smoothly, the "Reverse" checkbox should be ticked. One advantage of this method is that it allows you to monitor the sweep process in real time. But its accuracy is not as good as that of the previous method.

4.5 Output Duration/Loop

Under non-sweep mode, the "Duration" specifies the duration of the output signal. If the "Loop" checkbox is ticked, the specified signal will be output continuously until the Signal Generator is stopped manually, and in this case, the "Duration" edit box will be disabled and become not applicable.

Under sweep mode, the "Duration" specifies the sweep duration instead. If the "Loop" checkbox is ticked, the specified signal will be output continuously until the Signal Generator is stopped manually.

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4.6 Output Signal Processing

The output signal will undergo the following processes sequentially before it is finally output. They are:

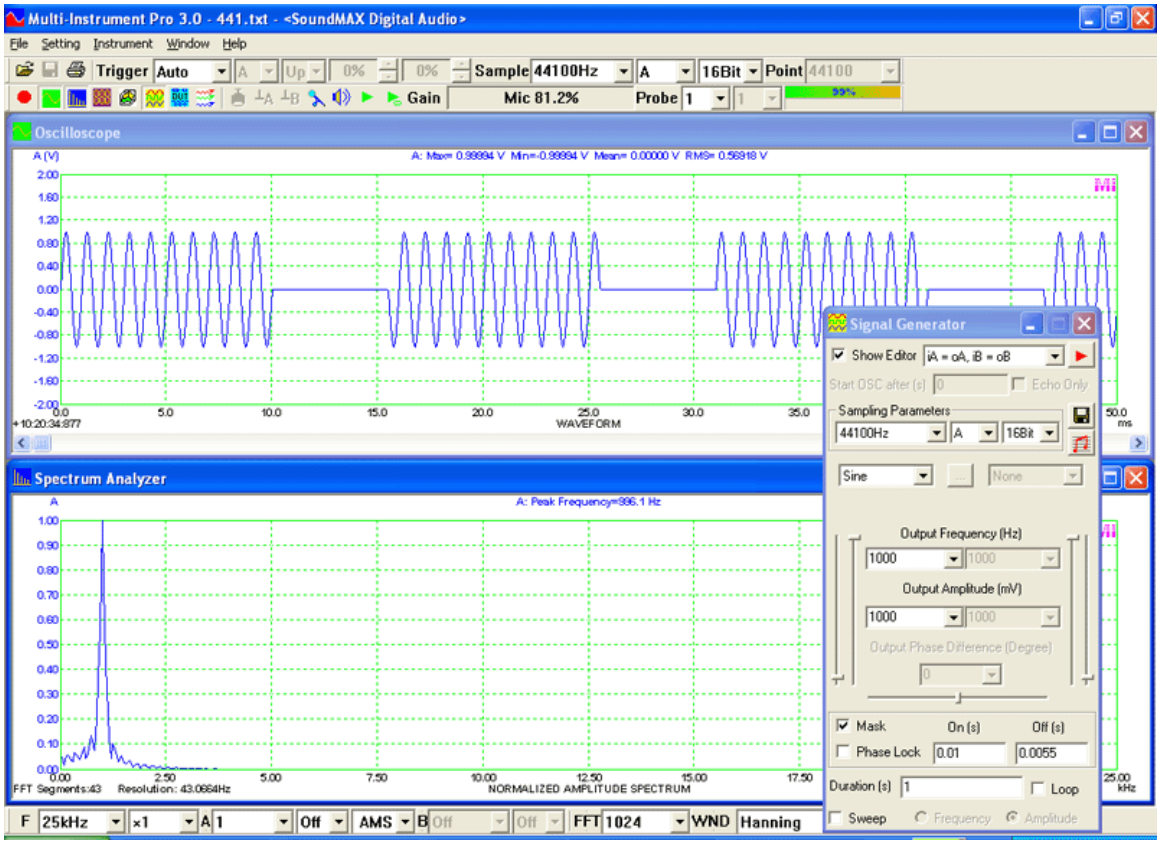
(1) Output Masking
(2) Output Fade In/ Fade Out
(3) Output Modulation

The above processing steps are usually performed by software and supported by streaming mode only.

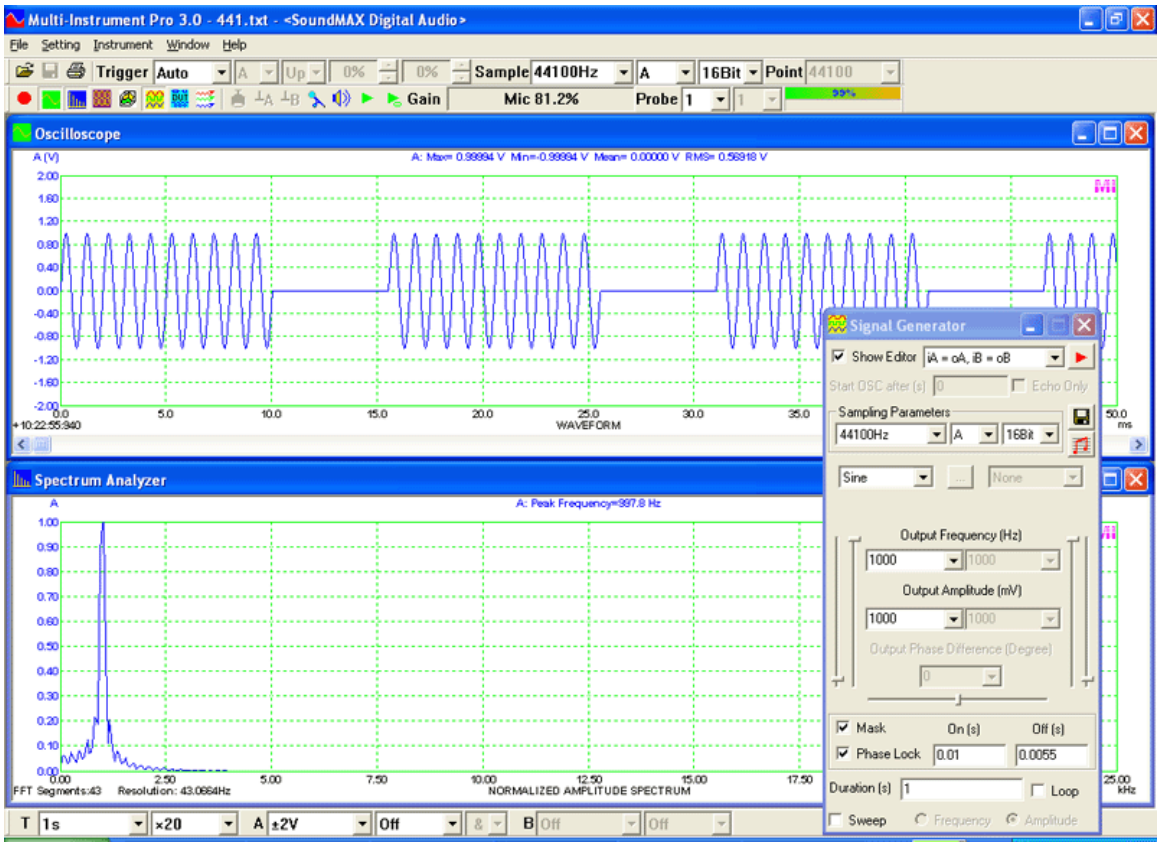
4.6.1 Output Masking (Window-Shaped Bursts)

The output mask is used to mask “on” or mask “off” the output signal periodically. If the “Mask” checkbox is ticked, you can specify the “on” time and “off” time in a masking cycle. “On” means to let the signal pass and "off" means to block the signal. You can use this feature to generate a burst-like signal. For signals with a “Sine”, “Rectangle”, “Triangle”, “Saw Tooth”, “MultiTones”, “WFLibrary”, or “DTMF” waveform, you can also force each burst to start at the same phase by ticking the “Phase Lock” checkbox.

The following figure shows a sine wave burst signal of 1000 Hz, with a mask-on time of 0.01 s, a mask-off time of 0.0055 s, and no phase lock. It can be seen that each burst is not started at the same phase angle.

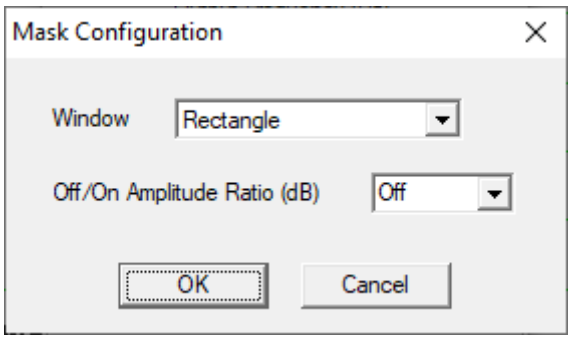


The following figure shows the same signal as above except that the phase lock is enforced here. It can be seen that each burst is started at exactly the same phase angle.



4.6.1.1 Window-Shaped Mask Configuration

In the above two examples, "On" means "100% pass" while "Off" means "0% pass". This is the default behavior of the mask function. More sophisticated behavior can be configured by clicking the button ... beside the "Mask" checkbox to open the following dialog box.



The above mask configuration dialog box allows you to specify a window function to be applied to the signal during the "on" period, and the off/on amplitude radio in dB. "Rectangle" and "Off" is selected by default.

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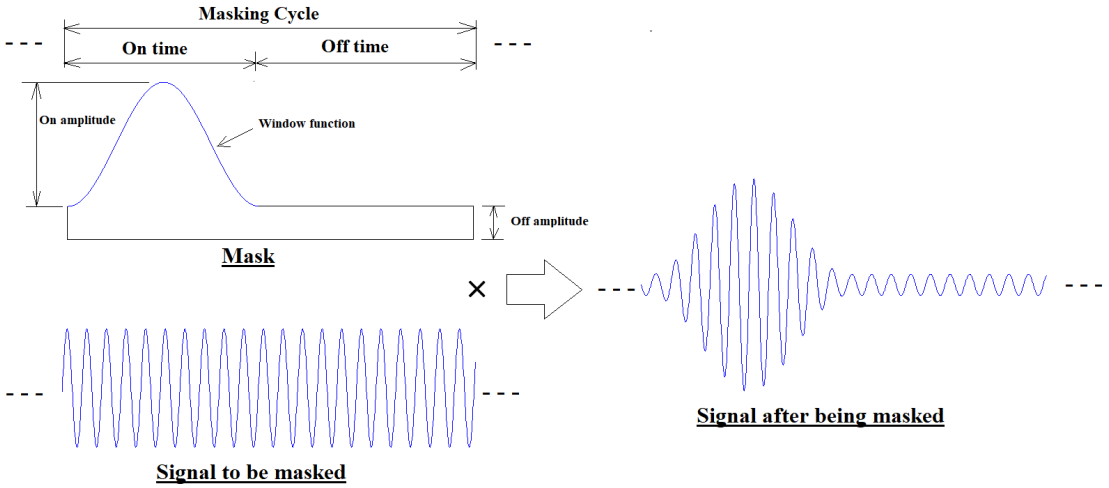
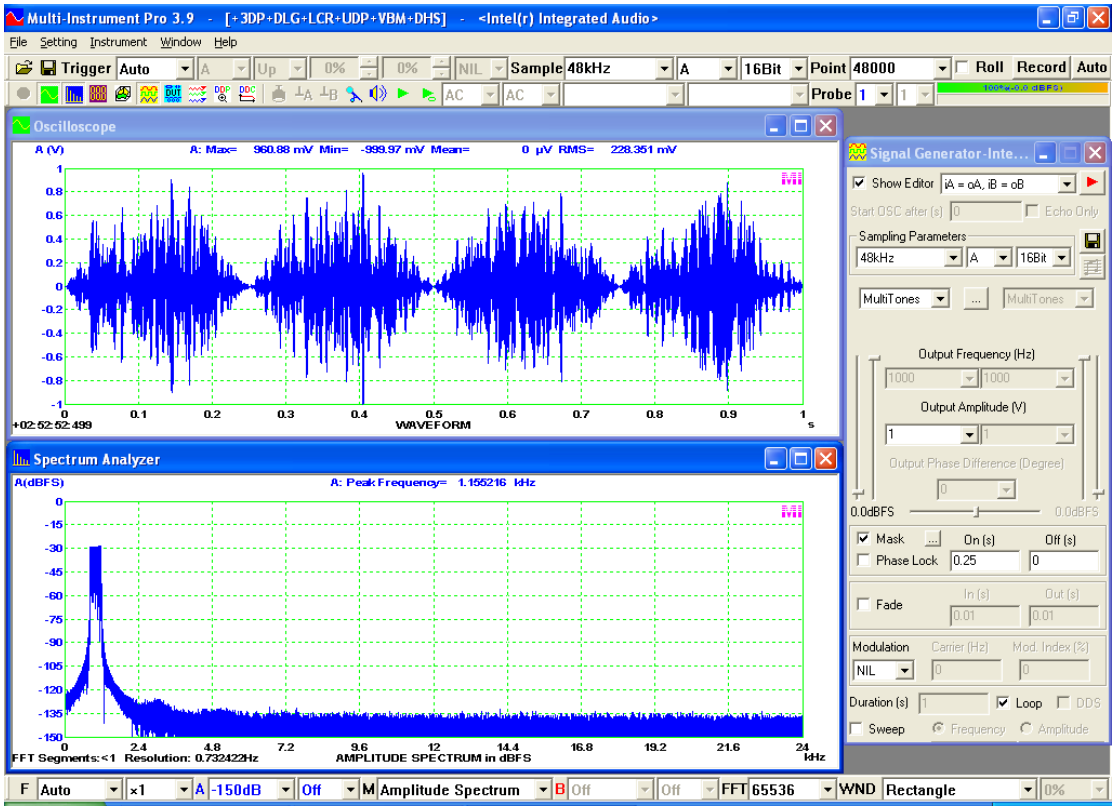
Generally speaking, the mask function works as an amplitude modulator. A masking cycle consists of an "On" time and an "Off" time. The masking cycle determines the modulating frequency. The window function and off/on amplitude ratio determine the modulating envelop. The signal to be masked is the carrier. It can be of any kind. This is in contrast to the output amplitude modulation function which will be introduced later. The following figure illustrates how the window-shaped mask function works.



4.6.1.2 Intensity-Modulated Band-Limited Pink Noise for Direct STI measurements

As described previously in the Oscilloscope chapter, in direct STI (Speech Transmission Index) measurements, a band-limited pink noise intensity-modulated by a low-frequency sine wave is used to measure the corresponding modulation transfer function (MTF). The Full STI can be derived after 98 such MTFs are measured. The following figure shows a 1 kHz pink noise with a half-octave bandwidth from 841 Hz to 1189 Hz, intensity-modulated by a 4 Hz sine wave. The band-limited pink noise is generated using automatic MultiTones configuration with all tones aligned to 1/96 octave bands. The 4 Hz intensity modulation is realized using Cosine 1.0 window-shaped mask with [On time] = 0.25 s, [Off time] = 0 s, [Off/On Amplitude Ratio] = Off. Note that if amplitude modulation is required instead, Hann window, a.k.a Cosine 2.0 window, should be used instead.



4.6.1.3 ANSI / CEA-2010 Subwoofer Peak SPL Test

ANSI / CEA-2010 standard defines a method to measure the peak SPL of a powered subwoofer within the specified harmonic distortion limits. The test signal is a 6.5-cycle Hann-windowed sinewave burst. The generated burst signal is captured and analyzed with narrow-band FFT. The resulting spectrum is checked against the harmonic distortion limits which are defined in the following table. The peak SPL can be obtained by stepping up the output power until the stipulated limits are about to be broken. The tests are usually performed individually at fundamental frequencies 20Hz, 25Hz, 31.5Hz, 40Hz, 50Hz, and 63Hz. The bandwidth of the test signal is exactly 1/3 octave.

Start Frequency (Hz)	End Frequency (Hz)	Limit (dB)	Notes
16	$1.59f_0$	0	Fundamental
$1.59f_0$	$2.52f_0$	-10	2^{nd} harmonic
$2.52f_0$	$3.78f_0$	-15	3^{rd} harmonic
$3.78f_0$	$5.61f_0$	-20	4^{th}, 5^{th} harmonic
$5.61f_0$	$8.50f_0$	-30	6^{th}, 7^{th} ,8^{th}
$8.50f_0$	10k	-40	9^{th} and above

Six panel setting files are provided in the PSF\Pro subdirectory of the software:

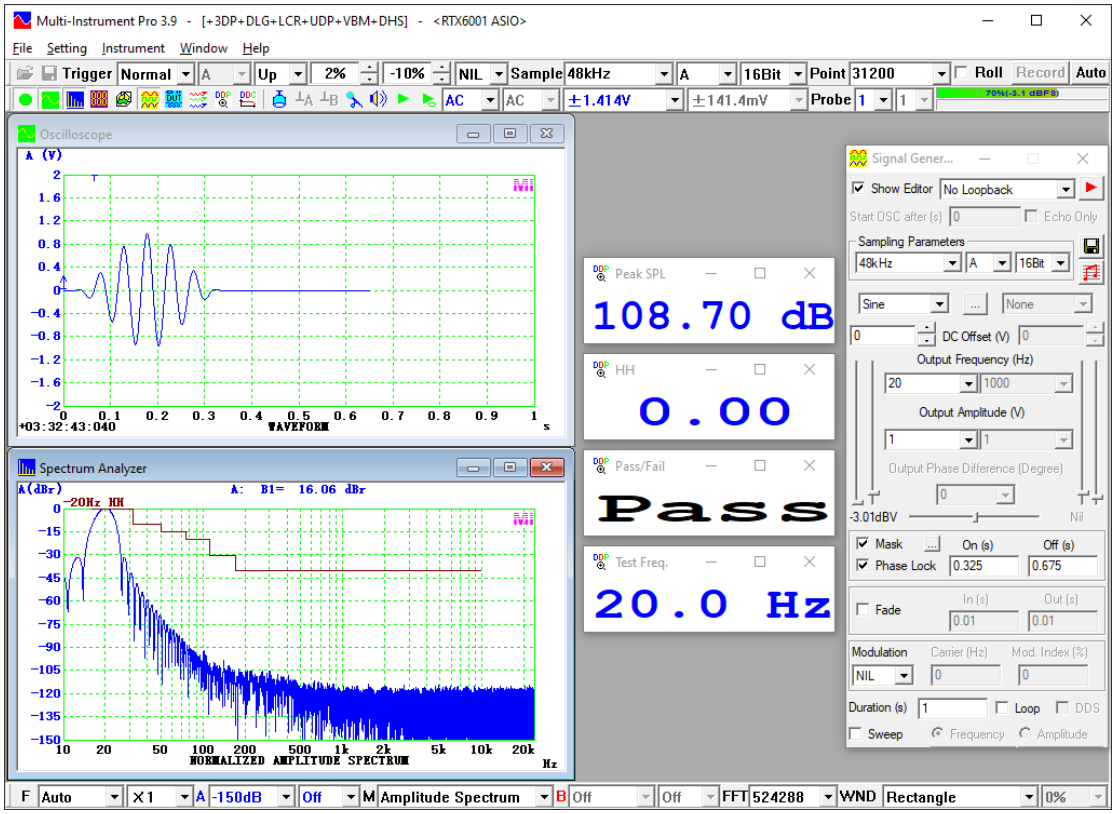
(1) CEA-2010-20Hz.psf
(2) CEA-2010-25Hz.psf
(3) CEA-2010-31.5Hz.psf
(4) CEA-2010-40Hz.psf
(5) CEA-2010-50Hz.psf

(6) CEA-2010-63Hz.psf

Correspondingly, six reference files are provided in the REF subdirectory of the software.

(1) CEA-2010-20Hz.ref
(2) CEA-2010-25Hz.ref
(3) CEA-2010-31.5Hz.ref
(4) CEA-2010-40Hz.ref
(5) CEA-2010-50Hz.ref
(6) CEA-2010-63Hz.ref

In the above settings, the Record Length of the Oscilloscope is twice the length of the sine burst. Noise and room reflections outside the Record Length are excluded. The trigger level and trigger delay are set such that the sine burst will be captured in the Oscilloscope and aligned left. 1/12 octave smoothing is recommended in the standard and can be added if necessary. The repetition rate of the burst must be greater than 1 second. The peak SPL is obtained through [Spectrum Analyzer Processing]>"Parameter Measurement">"Energy in User Defined Frequency Bands", in the 1/3 octave band around the fundamental, in accordance with CEA-2010B. The peak SPL values for Channels A&B are stored in DDPs: RMSDB_A(dB) and RMSDB_B(dB), when the Multimeter is not opened. The reference curve is configured as the High-High limit. Whether the limit is broken and how much it is exceeded can be accessed through DDPs: SpectrumComparisonH_A and SpectrumComparisonH_B for Channels A&B respectively. The following figure shows a sample test at 20Hz.

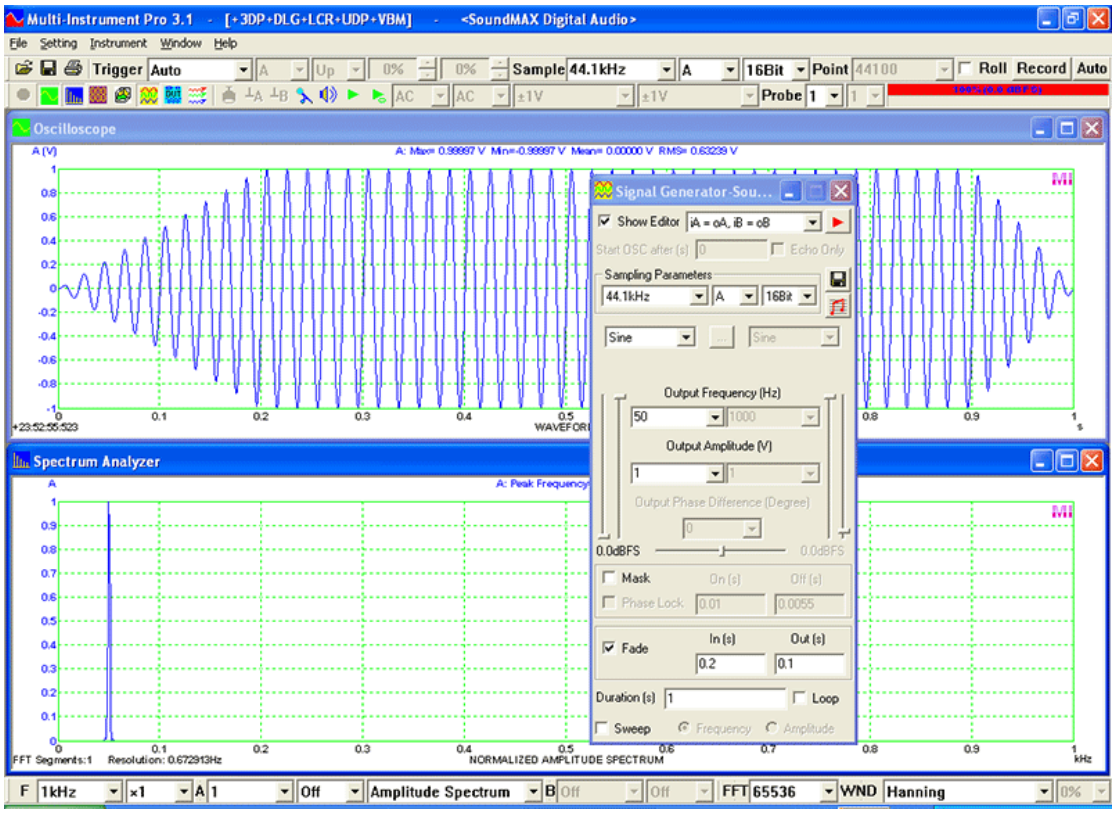


4.6.2 Output Fade In / Out

If "Fade" is ticked, you can specify the Fade In duration for the output signal. This duration starts as the signal output starts. During this duration, the signal amplitude increases linearly from zero to the specified Output Amplitude.

Furthermore, if the output signal is time limited (i.e. the "Loop" checkbox is not ticked), then you can also specify the Fade Out duration. This duration starts at the [End time of the output signal]-[Fade out duration]. During this duration, the signal amplitude decreases linearly from the specified Output Amplitude to zero.

The following figure illustrates a 1-second 50 Hz signal with a Fade In duration of 0.2 s and Fade Out duration of 0.1 s.



4.6.3 Output Modulation

Note that if an output modulation method is selected, all the signal parameters described previously refer to those of the modulating signal, except the output amplitude which refers to the modulated signal. The carrier signal here is a sine wave. For the cases of a non-sinewave carrier signal, consider using the window-shaped mask function described previously.

4.6.3.1 Amplitude Modulation (AM)

In Amplitude Modulation (AM), the amplitude of the carrier signal varies in proportion to the instantaneous amplitude of the modulating signal. The carrier frequency in the range from 0 to ½ of the sampling frequency and the modulation index in the range from 0% to 100% can be specified.

4.6.3.2 Frequency Modulation (FM)

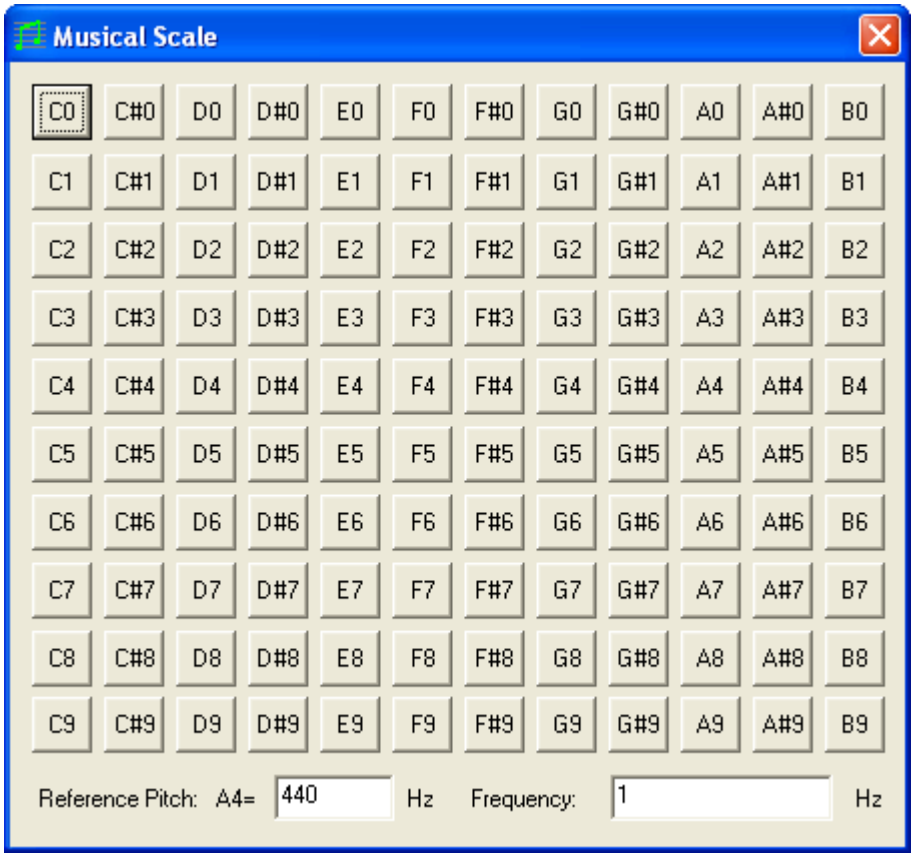
In Frequency Modulation (FM), the instantaneous frequency deviation from the carrier frequency of the modulated signal varies in proportion to the instantaneous amplitude of the modulating signal. The carrier frequency in the range from 0 to ½ of the sampling frequency and the maximum frequency deviation in the range from 0 to 1/2 of the carrier frequency can be specified.

4.6.3.3 Phase Modulation (PM)

In Phase Modulation (PM), the instantaneous phase deviation from the carrier phase (i.e. the phase if no modulation on the carrier signal is applied) of the modulated signal varies in proportion to the instantaneous amplitude of the modulating signal. The carrier frequency in the range from 0 to ½ of the sampling frequency and the maximum phase deviation in the range from 0 to 180° can be specified.

4.7 Musical Scale

The Musical Scale button is enabled when the signal is “Sine”, “Rectangle”, “Triangle”, “Saw Tooth”, or “WFLibrary”. If pressed, the following Musical Scale panel will pop up. The reference pitch can be adjusted if necessary.



4.8 Save Function

The specified output signal can be saved as a Wave File (*.wav) or Text File (*.txt) for a time duration specified by the “Duration” (The “Loop” checkbox will be ignored even if it is ticked). If the “Duration” is greater than 1000 seconds, it will be limited to 1000 seconds. If the Save button is pressed, a "Save as" window will be displayed. You can select a file type and enter a file name for the file to be saved.

The wave file format follows the standard Wave File PCM format and can be replayed using third party software such as Microsoft Windows Media Player or loaded back into the software itself. The Text File is a Comma Separated Value file and has the following format. It can be loaded back into the software using the [File]>[Import] command described previously in the Oscilloscope chapter.

- **One channel data format**

;Data Points
;Sampling Frequency (Hz) = 44100
;Sampling Bit Resolution (Bits) = 16
;Sampling Channels = 1
;A:Full-scale Voltage (V) = 1
;A:Sensor Sensitivity (V/V)=1
;Total Data Points = 44100
;No., Time (S), Channel A (V)

1, 0.000000, 0.000000
2, 0.000023, 0.141995
3, 0.000045, 0.281113
4, 0.000068, 0.414534
5, 0.000091, 0.539554
……

Apart from the 8 header lines, each row contains a sequential number, a time stamp (in second) and a value for Channel A (in Volt).

- **Two channel data format**

;Data Points
;Sampling Frequency (Hz) = 44100
;Sampling Bit Resolution (Bits) = 16
;Sampling Channels = 2
;A:Full-scale Voltage (V) = 1
;A:Sensor Sensitivity (V/V)=1
;B:Full-scale Voltage (V)=1
;B:Sensor Sensitivity (V/V) = 1
;Total Data Points = 44100
;No., Time (S), Channel A (V), Channel B (V)
1, 0.000000, 0.000000, 1.000000
2, 0.000023, 0.141732, 1.000000
3, 0.000045, 0.275591, 1.000000
4, 0.000068, 0.409449, 1.000000
5, 0.000091, 0.535433, 1.000000
……

Apart from the 10 header lines, each row contains a sequential number, a time stamp (in second), a value for Channel A (in Volt) and a value for Channel B (in Volt).

4.9 Run/Stop

Clicking this button will generate the signal. It toggles between Run and Stop if clicked. You can also use the Enter key to start/stop the signal generator when the signal generator panel is the current active window.

4.10 Loopback Mode

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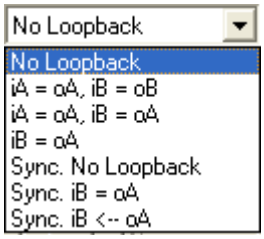


The available loopback modes depend on the DAC and ADC devices used. For a sound card based system, there are seven loopback modes available:

4.10.1 No Loopback

Under this mode, the Signal Generator and the DAQ-related instruments (e.g. Oscilloscope, Spectrum Analyzer, Multimeter, Spectrum 3D Plot, Data Logger…) work independent within the software. However, you can still establish a loopback via external means:

- Hardwired loopback via an external loopback cable

- Mixer-level loopback established via selecting "Wave Out Mix" , “Stereo Mix” or the like as the input source in the Recording Control under Windows Control Panel. From Windows Vista onwards, this can be selected via [Setting]>[ADC Device]>"Device No."

  The "Wave Out Mix" may not be listed as an input source in the Recording Control by default, but you should be able to find it in the Options of the Recording Control. This method feeds the sound card DAC output directly to its ADC input through the sound card mixer, so there is some hardware involved. Be sure to adjust the "Wave Out Mix" volume properly to avoid possible input saturation.

- External software loopback via virtual audio cable software, which is available as freeware or shareware such as the one from www.vb-audio.com. This method usually involves software resampling which introduces a small amount of distortion and noise.

4.10.2 iA = oA, iB = oB

Under this mode, a full loopback digital route is established at software level between the output channel and input channel, such that the signal from the Output Channel A is fed into the Input Channel A and the signal from the Output Channel B is fed into the Input Channel B.

Under this mode, the Trigger Parameters and Sampling Parameters will be automatically set by the software, and will be disabled and remain not adjustable until the end of the signal output. The frame width of the Oscilloscope has to be 1 second. It should be noted that the signal fed into the input channels is "ideal" in the sense that it does not pass through any hardware circuit, neither the sound card output channels,

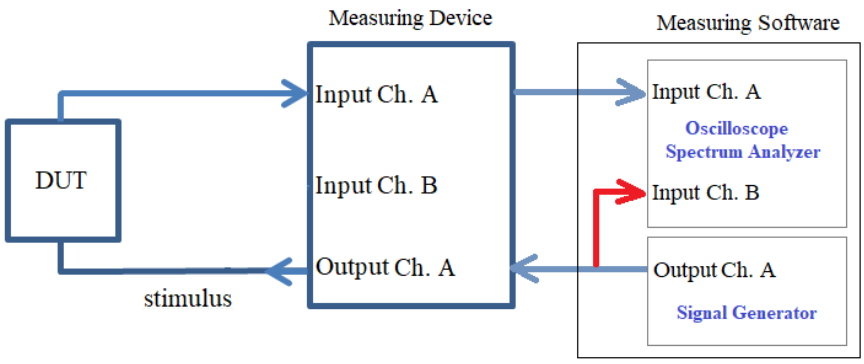
nor the sound card input channels. This is different from the hardwired loopback and mixer-level loopback modes described previously. This mode is especially useful for teaching.

4.10.3 iA = oA, iB = oA

This loopback mode is similar to the previous one except that both input channels are fed with the signal from the Output Channel A only. It is especially useful, for example, when you want to evaluate the characteristics of a digital filter in the Oscilloscope and Spectrum Analyzer. In such cases, you should generate a proper stimulus from the Channel A of the Signal Generator and apply the digital filter in the Oscilloscope to the Input Channel A only, leaving the same stimulus fed into the Input Channel B intact. This arrangement complies with the requirements of the cross correlation, transfer function, coherence function and impulse response modes of the Spectrum Analyzer.

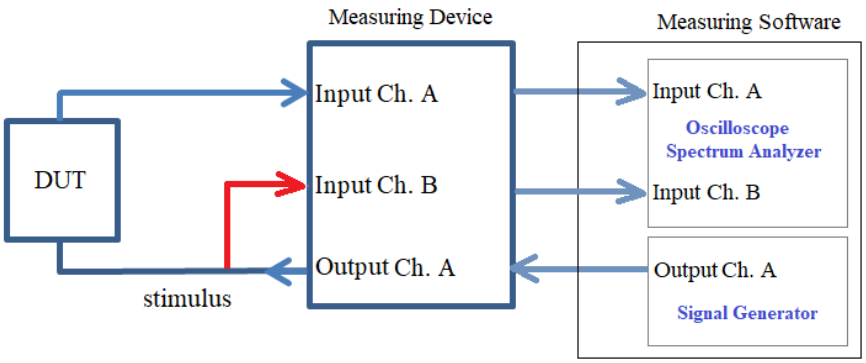
4.10.4 iB = oA

Under this mode, a partial loopback digital route is established at software level between the Output Channel A and the Input Channel B, such that the signal from the Output Channel A is fed into the Input Channel B and the Input Channel A is still available for field input. Only the very first part of data from the Output Channel A, with a length equal to the Record Length of the Oscilloscope, will be fed into the Input Channel B and remain unchanged there. In other words, if the oscilloscope continues to acquire the second frame of data, then the data in Channel A may not be in correspondence with the data in Channel B anymore. The Oscilloscope should be set in dual channel mode in order to see the signal from the Signal Generator. This mode is usually employed as a single-channel method for the measurements of cross correlation, transfer function, coherence function and impulse response, as illustrated as follows. The drawback of this method is that the measured transfer function / impulse response includes the contributions from the output channel A and the input channel A of the measuring device as well. Thus these two should be kept as "transparent" as possible.



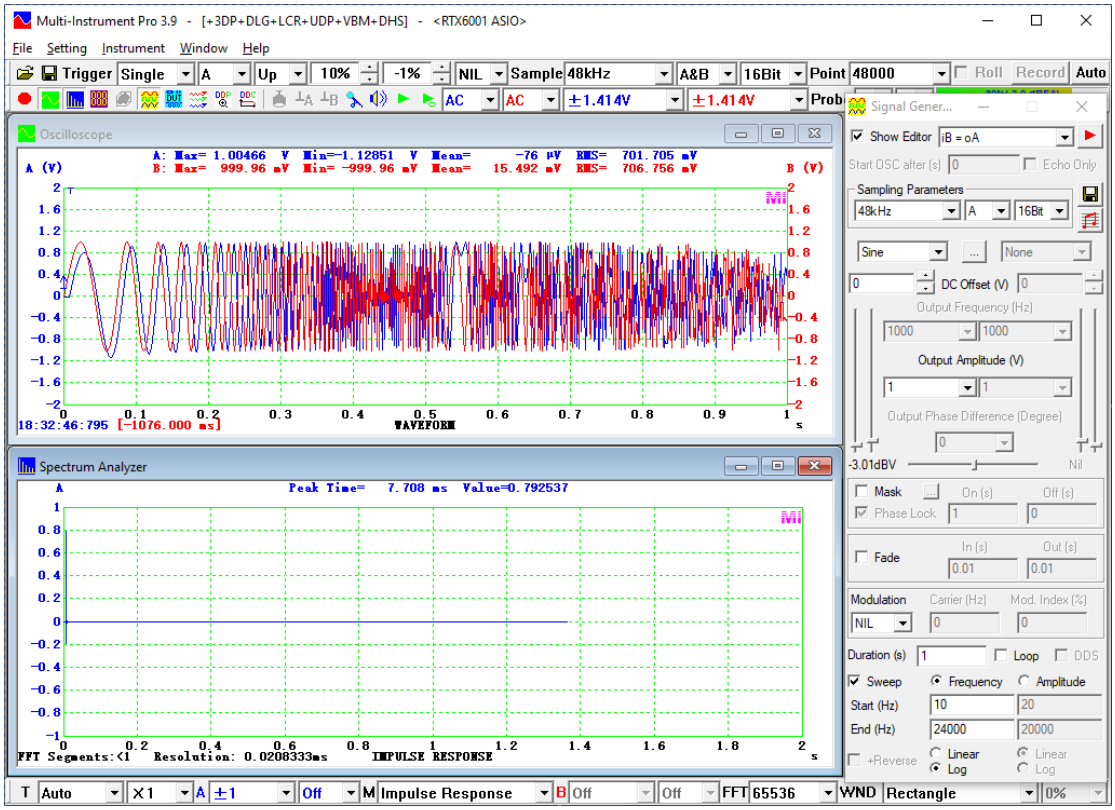
Single-Channel Method (iB=oA)
for Transfer Function / Impulse Response Measurement

In contrast to the above single-channel measurement method using the "iB=oA" mode, the following figure illustrates a dual-channel measurement method using the "No loopback" mode. The dual-channel method offers a higher accuracy as the measured transfer function / impulse response is only affected by the disparity between the input channels A&B of the measuring device, which is usually very small. Moreover, this disparity can be compensated in the software if necessary.



Dual-Channel Method (no software loopback)
for Transfer Function / Impulse Response Measurement

To make the single-channel measurement method work properly, "Loop" option in the Signal Generator panel should be unticked so that only a short-time signal will be generated. In the Oscilloscope, the Record Length should be long enough to hold the short-time test signal. Single trigger mode must be used and trigger parameters must be set properly such that the response from the DUT in the input channel A is more or less time aligned with the stimulus in the input channel B. "Time Delay Removal" function in the Oscilloscope can be used to remove the time alignment error if necessary. The time difference between the Signal generator is commanded to start and the arrival of the captured response in the Oscilloscope will be displayed at the bottom left of the Oscilloscope window. However, it should be considered as indicative only. The following figure shows an impulse response of a sound card loopback measured using frequency logarithmically swept sine wave under single-channel mode.



4.10.5 Sync. No Loopback

This mode is the same as the No Loopback mode, except that the start of the Signal Generator and the start of the Oscilloscope are synchronized. The synchronization works as follows:

(1) Stop the Oscilloscope manually if it is running
(2) Start the Signal Generator

The Oscilloscope will be started automatically after the Latency Time + the Delay Time specified by “Start OSC after (s)”. The latency time should be calibrated according to the procedure described previously in the section “Latency for Synchronized Output/Input”.

If you tick "Echo Only", the delay time will be automatically filled with the duration of the output signal and the “Loop” checkbox will be un-ticked once, so that the Oscilloscope will start sampling just after the direct output signal finishes, i.e. only the echo will be captured.

4.10.6 Sync. iB = oA

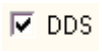
This mode is the same as the iB = oA mode, except the synchronization between the Signal Generator and the Oscilloscope as described previously.

4.10.7 Sync. iB<--oA

This loopback mode requires an external cable to feed the signal from the Output Channel A into the Input Channel B. Note that as far as the loopback modes are concern, "=" means a software-level loopback. And "←" means a hardwired loopback for synchronization and a software-level loopback for data.

You can specify when to start the Oscilloscope after the signal has been output from the output channel by the signal generator. The Latency Time has no effect on this mode, because the software uses the actual signal acquired from the Input Channel B to determine when the front of the output signal actually arrives. So this mode is more accurate than "Sync. iB = oA" mode, but the drawback is that it requires two physical input channels with two input jacks and a hardwired loopback. Note that the Trigger Edge, Trigger Level and Trigger Delay must be set properly to allow accurate detection of the front of the output signal. The Trigger Mode will be forced to "Single" and the Trigger Source will be forced to "B" automatically by the software.

4.11 DDS Mode vs Streaming Mode



The signal generator can work in two modes: streaming mode and DDS mode.

Under streaming mode, the output data are computed, continuously if necessary, by the software in the computer and then streamed continuously to the DAC device for output. One advantage of using streaming mode is that you can fully utilize the power of the computer software to generate very complicated signal at minimum hardware cost. However, limited by the software computing speed and the streaming speed, this mode is not able to provide high frequency output. Also, it may consume a lot of CPU time when the output sampling frequency is high or the output signal is complicated. Some DAC devices such as sound cards work only in streaming mode.

Under DDS mode, the computer only needs to send the output signal parameters (such as signal frequency, amplitude, waveform type or shape data, etc.) to the DAC device and the DAC device will take care of the rest. No actual output data will be sent from the computer to the DAC device. This greatly reduces the CPU work load and communication traffic between the computer and the DAC device. Very high signal frequencies can be generated if the DAC device is capable. One drawback is that not all functions supported by the streaming mode are supported by the DDS mode. Please refer to the respective hardware manual for details.

Some DAC devices such as VT DSO-2810 and VT DSO-2810E support both streaming mode and DDS mode. Tick/Untick the DDS checkbox in the above dialog to select the DDS mode/streaming mode. If the selection is disabled, try to select a lower sampling frequency first as the streaming mode may not support very high sampling frequency.

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5 Multimeter

5.1 Overview

This is a dual channel Multimeter, providing the following types of digital displays:

- RMS
- dBV
- dBu
- dB
- dB(A)
- dB(Z)
- dB(C)
- Frequency Counter
- RPM (Revolutions Per Minute)
- Counter
- Duty Cycle
- F/V (Frequency Voltage Conversion)
- Cycle RMS
- Cycle Mean
- Pulse Width
- Vibrometer

The above items from Frequency Counter to Pulse Width involves a pulse counting process, and the software allows you to configure the counter trigger level and trigger hysteresis in order to rectify the analog signal to rectangular pulses before counting. The frequency dividing ratio can also be configured. Modulation domain analysis is performed during the counting process to measure jitter statistics. Frequency / RPM deviation (or Jitter) Peak (%), Max. Negative (%), Max. Positive (%) and Normalized Standard Deviation (%) are stored in DDPs: WowAndFlutter_A(%), WowRMS_A(%), FlutterRMS_A(%), WowAndFlutterRMS_A (%) for Channel A and WowAndFlutter_B(%), WowRMS_B(%), FlutterRMS_B(%), WowAndFlutterRMS_B (%) for Channel B respectively, when Wow and Flutter option in [Spectrum Analyzer Processing] is not selected.

The toggle button in the Instrument Toolbar is used to open or close the Multimeter. You can also close it by clicking the "Close" button at the upper right corner of the Multimeter window.

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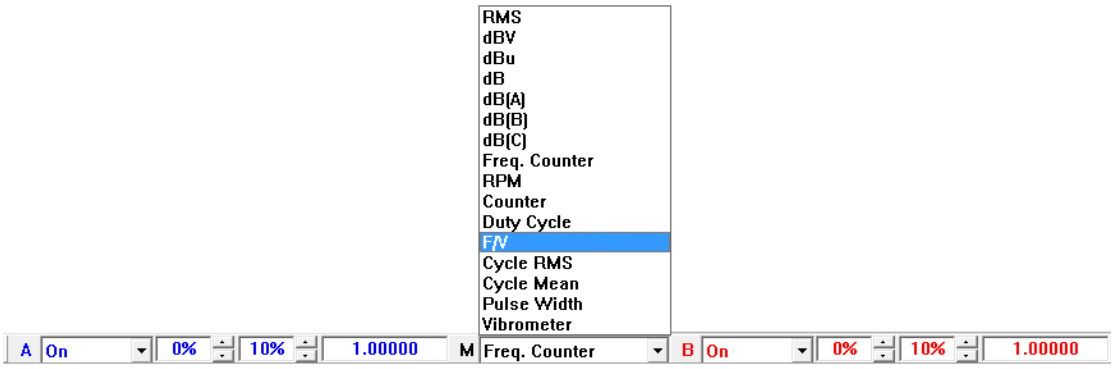
5.2 View Parameters



View parameters determine how the collected data are analyzed and displayed. There are 16 types of views, as shown in the above figure.

5.2.1 RMS

Under this mode, the Multimeter displays the RMS values of the current frame of data, which are the same as the ones displayed in the Oscilloscope but with bigger font. You can display or hide the value for any channel. The rest of the view parameters in the view parameter toolbar are not applicable.

The corresponding DDPs are: RMS_A(EU) for Channel A and RMS_B(EU) for Channel B.

5.2.2 dBV

Under this mode, the Multimeter displays the dBV values of the current frame of data. You can display or hide the value for any channel. The rest of the view parameters in the view parameter toolbar are not applicable. If the Sensor Engineering Unit is not voltage, then this option means dBEU.

The corresponding DDPs are: RMSDBV_A(dBEU) for Channel A and RMSDBV_B(dBEU) for Channel B.

5.2.3 dBu

Under this mode, the Multimeter displays the dBu values of the current frame of data. You can display or hide the value for any channel. The rest of the view parameters in the view parameter toolbar are not applicable.

The corresponding DDPs are: RMSDBu_A(dBu) for Channel A and RMSDBu_B(dBu) for Channel B.

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5.2.4 dB

Under this mode, the Multimeter displays the dB values of the current frame of data. You can display or hide the value for any channel. The rest of the view parameters in the view parameter toolbar are not applicable. The reference level for dB is user definable and can be set via [Setting]>[Calibration]>“0dB Reference Vr”.

The corresponding DDPs are: RMSDB_A(dB) for Channel A and RMSDB_B(dB) for Channel B.

5.2.5 dB(A)

Under this mode, the Multimeter displays the dB(A) values of the current frame of data. You can display or hide the value for any channel. The rest of the view parameters in the view parameter toolbar are not applicable. Note that the Spectrum Analyzer will be forced to display Amplitude Spectrum with A weighting.

The corresponding DDPs are: RMSDBA_A(dBA) for Channel A and RMSDBA_B(dBA) for Channel B. Note that this value is sharply bandlimited to 10Hz ~20kHz.

5.2.6 dB(Z)

Under this mode, the Multimeter displays the dB(Z) values of the current frame of data. You can display or hide the value for any channel. The rest of the view parameters in the view parameter toolbar are not applicable. Note that the Spectrum Analyzer will be forced to display Amplitude Spectrum with B weighting.

The corresponding DDPs are: RMSDBZ_A(dBZ) for Channel A and RMSDBZ_B(dBZ) for Channel B. Note that this value is sharply bandlimited to 10Hz ~20kHz.

5.2.7 dB(C)

Under this mode, the Multimeter displays the dB(C) values of the current frame of data. You can display or hide the value for any channel. The rest of the view parameters in the view parameter toolbar are not applicable. Note that the Spectrum Analyzer will be forced to display Amplitude Spectrum with C weighting.

The corresponding DDPs are: RMSDBC_A(dBC) for Channel A and RMSDBC_B(dBC) for Channel B. Note that this value is sharply bandlimited to 10Hz ~20kHz.

5.2.8 Frequency Counter

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Under this mode, the Multimeter displays the counted frequency values of the current frame of data. You can display or hide the value for any channel.

You can also set the Counter Trigger Level and Counter Trigger Hysteresis as well as the ratio of the Frequency Divider. The Counter Trigger Level and Counter Trigger Hysteresis are useful parameters under the Frequency Counter, RPM, Counter, Duty Cycle, F/V, Cycle RMS, Cycle Mean, Pulse Width modes. They are used as parameters to rectify the analog signal in the Oscilloscope to rectangular pulses before the counting process. They behave in a similar way as the Trigger Level and Trigger Hysteresis of an electronic Schmitt trigger, except that they are relative values which vary with the maximum and minimum values detected in the current frame of data. The Counter Trigger Level can be adjusted from –100% to 100%, with -100% corresponding to the minimum value and 100% corresponding to the maximum value. The Counter Trigger Hysteresis can be adjusted within 0 ~ 100% of the actual range of the frame of data, i.e. [the maximum value]-[the minimum value]. These two parameters are visualized as two markers along the vertical axes in the Oscilloscope, as shown in the following figure. The markers on A axis are for Channel A, and the markers on B axis are for Channel B.

In the figure, the signals in both channels are a 1 kHz sine wave mixed with a white noise at an amplitude ratio of 5:1. The Counter Trigger Level and Counter Trigger Hysteresis for Channel A are 50% and 0% respectively. In contrast, they are 50% and 20% respectively for Channel B. The Multimeter measures a frequency of 1457.355 Hz for Channel A which is incorrect due to the fluctuation caused by the white noise component. For Channel B, the measured frequency is 999.930 Hz which is very accurate. Thus, setting a proper hysteresis value is important when there are noises present in the signal.

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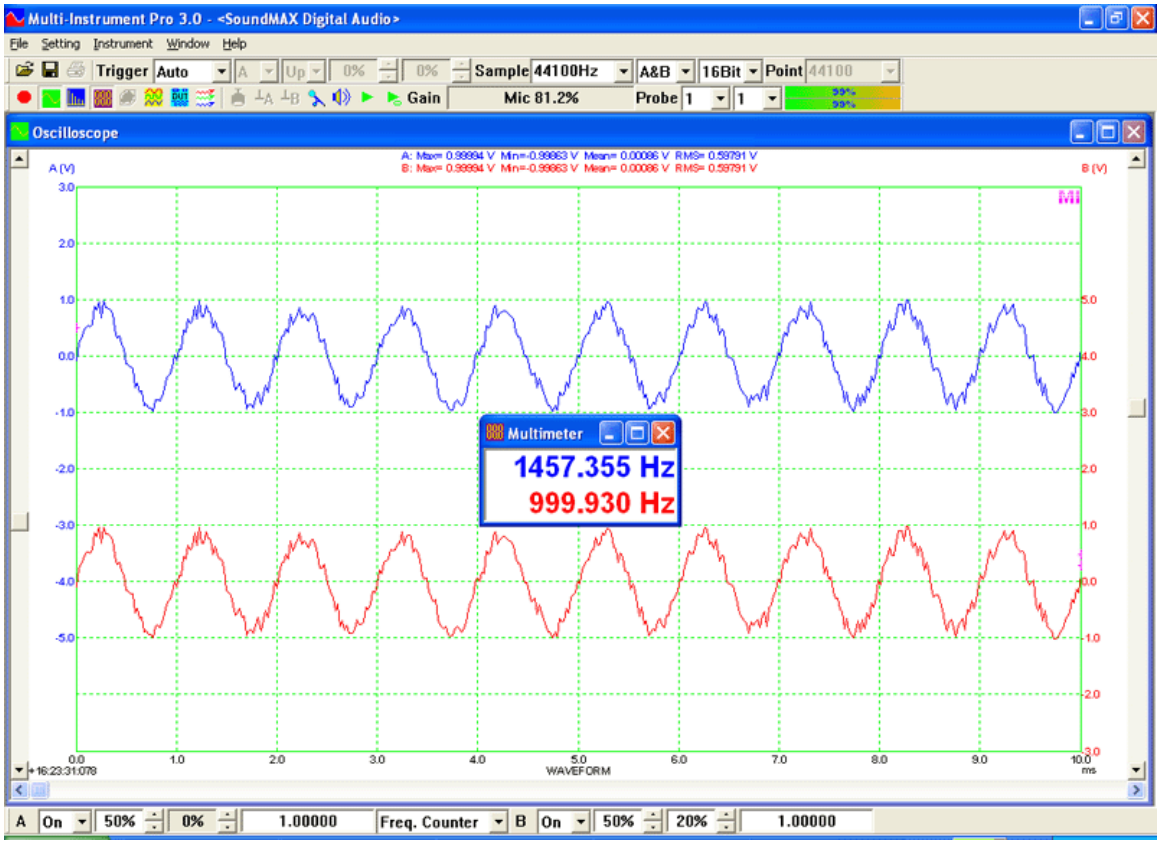

The corresponding DDPs are: Freq_A(Hz) for Channel A and Freq_B(Hz) for Channel B. These two are actually the counted mean frequencies in the current data frame. It is possible to display a Frequency Time Curve using DDC and jitter parameters: Peak Jitter (%), RMS Jitter (%), Max. Negative Jitter (%), Max. Positive Jitter (%) using DDP viewers. The following figure is such an example. A sample panel setting file DirectJitterMeasurement_10s.psf is provided in the PSF\Pro subdirectory of the software.

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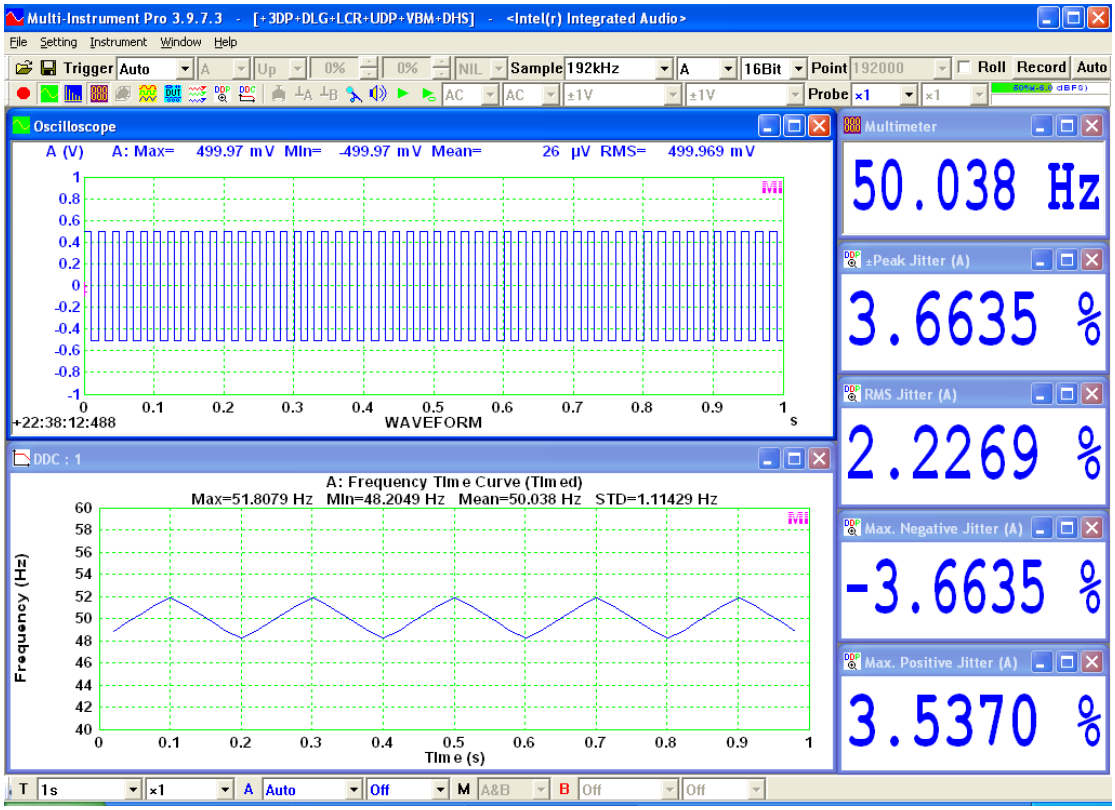


5.2.9 RPM

Under this mode, the Multimeter displays the RPM (Revolutions Per Minute) values of the current frame of data. You can display or hide the value for any channel.

The Counter Trigger Level, Counter Trigger Hysteresis and Frequency Divider are applicable to this mode.

The corresponding DDPs are: RPM_A(rpm) for Channel A and RPM_B(rpm) for Channel B.

5.2.10 Counter

Under this mode, the Multimeter displays the total counts of the current frame of data. You can display or hide the value for any channel.

The Counter Trigger Level, Counter Trigger Hysteresis and Frequency Divider are applicable to this mode.

The corresponding DDPs are: TotalCount_A for Channel A and TotalCount_B for Channel B.

5.2.11 Duty Cycle

Under this mode, the Multimeter displays the duty cycle values of the current frame of data. You can display or hide the value for any channel.

The Counter Trigger Level and Counter Trigger Hysteresis are applicable to this mode.

The corresponding DDPs are: DutyCycle_A(%) for Channel A and DutyCycle_B(%) for Channel B.

5.2.12 F/V

Under this mode, the Multimeter displays the voltage values after performing frequency counting and subsequent frequency/voltage conversion of the current frame of data. You can display or hide the value for any channel. The frequency/voltage conversion factors are configured as described previously in the section "Frequency Voltage Conversion Factor".

The Counter Trigger Level, Counter Trigger Hysteresis and Frequency Divider are applicable to this mode.

The corresponding DDPs are: FVC_A(V) for Channel A and FVC_B(V) for Channel B.

5.2.13 Cycle RMS

Under this mode, the Multimeter displays the RMS values calculated from an integer number of signal cycles within an Oscilloscope frame. The Cycle RMS is the "true" RMS value of a periodic signal. An Oscilloscope frame does not necessarily contain an integer number of signal cycles and thus the RMS value of the Oscilloscope frame may not be the same as the Cycle RMS value.

You can display or hide the value for any channel. The Counter Trigger Level and Counter Trigger Hysteresis are applicable to this mode.

The corresponding DDPs are: CycleRMS_A(EU) for Channel A and CycleRMS_B(EU) for Channel B. This display mode also generates the following DDPs: CyclePWR_A(W) for Channel A, which is the cycle power and equal to CycleRMS_A(EU)^2 / [Load Factor A], and CyclePWR_B(W) for Channel B, which is the cycle power and equal to CycleRMS_B(EU)^2 / [Load Factor B]. The load factors for Channels A and B can be specified via [Setting]>[Calibration].

5.2.14 Cycle Mean

Under this mode, the Multimeter displays the Mean values calculated from an integer number of signal cycles within an Oscilloscope frame. The Cycle Mean is the "true" mean value of a periodic signal. An Oscilloscope frame does not necessarily contain

an integer number of signal cycles and thus the Mean value of the Oscilloscope frame may not be the same as the Cycle Mean value.

You can display or hide the value for any channel. The Counter Trigger Level and Counter Trigger Hysteresis are applicable to this mode.

The corresponding DDPs are: CycleMean_A(EU) for Channel A and CycleMean_B(EU) for Channel B.

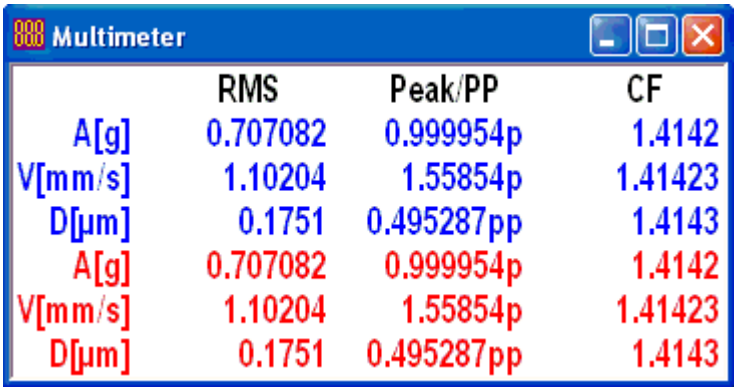
5.2.15 Pulse Width

Under this mode, the Multimeter displays the pulse width of the current frame of data. You can display or hide the value for any channel.

The Counter Trigger Level and Counter Trigger Hysteresis are applicable to this mode.

The corresponding DDPs are: PulseWidth_A(s) for Channel A and PulseWidth_B(s) for Channel B.

5.2.16 Vibrometer



If one of the sensors used is an/a acceleration, velocity or displacement sensor, then you can choose this mode to display the RMS, Peak or Peak-to-Peak (PP) and Crest Factor (CF) values of Acceleration, Velocity and Displacement.

You can display or hide the values for any channel. The rest of the view parameters in the view parameter toolbar are not applicable.

This display mode generates the following DDPs for Channel A:

(1) AccelerationRMS_A(g): Acceleration RMS value in g for Channel A
(2) AccelerationRMS_A(m/s^2): Acceleration RMS value in m/s^2 for Channel A
(3) AccelerationPeak_A(g): Acceleration Peak value in g for Channel A
(4) AccelerationPeak_A(m/s^2): Acceleration Peak value in m/s^2 for Channel A
(5) AccelerationCF_A: Acceleration Crest Factor for Channel A
(6) VelocityRMS_A(mm/s): Velocity RMS value for Channel A

(7) VelocityPeak_A(mm/s): Velocity Peak value for Channel A
(8) VelocityCF_A: Velocity Crest Factor for Channel A
(9) DisplacementRMS_A(μm): Displacement RMS value for Channel A
(10) DisplacementPP_A(μm): Displacement Peak-to-Peak value for Channel A
(11) DisplacementCF_A: Displacement Crest Factor for Channel A

It also generates similar DDPs for Channel B.

5.3 Menu

The menu for the Multimeter is only a subset of the menu for the Oscilloscope described previously.

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6 Data Logger

6.1 Overview

This is a dual channel parameter data logger. It provides long time data logging function for 254 derived data points and 16 user defined data points, including RMS value, Peak Frequency, Sound Pressure Level, RPM, THD, etc. Up to eight data logger windows can be opened and each window can trace up to 8 variables. The logged data can be reloaded for review.

The push button in the Instrument Toolbar is used to open a new data logger window. You can close it by clicking the "Close" button at the upper right corner of the window.

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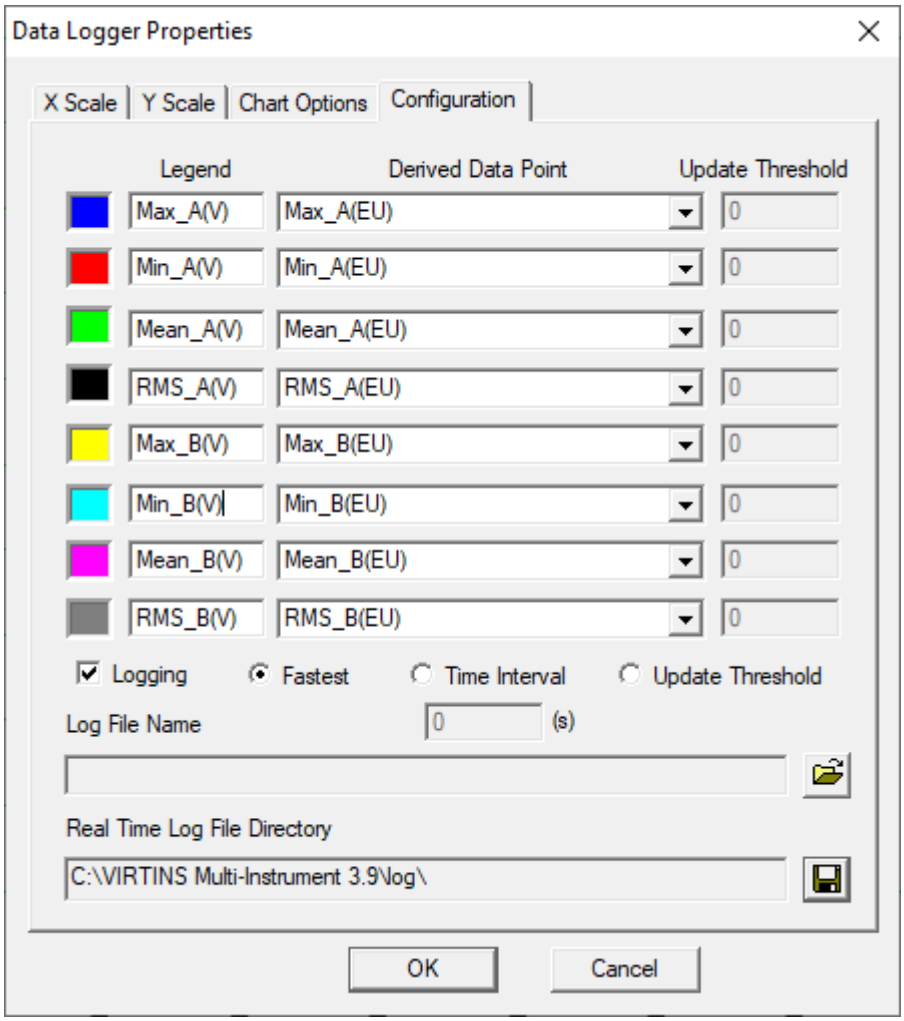
6.2 Configuration



Whenever the "Data Logger" button in the Instrument Toolbar is clicked, a new data logger window will be opened. A configuration dialog will pop up on the top of the data logger window to allow you to enter the configuration first. Parameters to be configured include the Derived Data Points (DDP) to be logged, Legend (color and label of each DDP), Update Threshold for each DDP (if "Update Threshold" is chosen for the Logging Method), Time Interval (if "Time Interval" is chosen for the Logging method), Logging Method, etc. By default, the log files will be stored in the LOG subdirectory of the software. This can be changed by re-selecting a "Real Time Log File Directory".

6.2.1 Derived Data Point and User Defined Data Point

In this software, a Derived Data Point (DDP) refers to a data point that is derived from a frame of raw DAQ data. The DDP values are updated whenever a new frame of data arrives. A User Defined Data Point (UDDP) is usually a function of certain DDPs. It is updated after all DDPs are updated except OverallPassFailCheck which is the last one to update. UDDPs can be configured by the user using DDP viewer. The following derived data points and user defined data points can be selected for logging.

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No.	DDP	Description	Source
1	Max_A(EU)	Maximum value for Channel A or combined	OSC
2	Min_A(EU)	Minimum value for Channel A or combined	OSC
3	PP_A(EU)	Peak-to-Peak value for Channel A or combined	OSC
4	Mean_A(EU)	Mean value for Channel A or combined	OSC
5	RMS_A(EU)	RMS value for Channel A or combined	OSC
6	PWR_A(W)	Power value (= RMS^2 / Load Factor) for Channel A or combined	OSC
7	Skewness_A	Skewness for Channel A	OSC
8	Kurtosis_A	Kurtosis for Channel A	OSC
9	PeakLevelPercent_A(%)	Peak Level in Percentage for Channel A	OSC
10	PeakLeveldBFS_A	Peak Level in dBFS for Channel A	OSC
11	f1Freq_A(Hz)	1. Peak / Fundamental Frequency for Channel A 2. Frequency at peak impedance for Channel A 3. 3.15 kHz for Channel A	1. SPEC(AMS) 2. SPEC (AMS+IM)/(AMS +OCT+IM) 3. SPEC (AMS+DIM)
12	f2Freq_A(Hz)	1. The 2nd Harmonic Frequency for Channel A 2. The 2nd Peak Frequency for Channel A 3. The 2nd Peak Frequency for Channel A 4. 15 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
13	f3Freq_A(Hz)	1. The 3rd Harmonic Frequency for Channel A 2. The 3rd Peak Frequency for Channel A 3. The 2nd order IMD lower frequency for Channel A 4. 0.75 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
14	f4Freq_A(Hz)	1. The 4th Harmonic Frequency for Channel A 2. The 4th Peak Frequency for Channel A 3. The 2nd order IMD higher frequency for Channel A 4. 2.40 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
15	f5Freq_A(Hz)	1. The 5th Harmonic Frequency for Channel A 2. The 5th Peak Frequency for Channel A 3. The 3rd order IMD lower frequency for Channel A 4. 3.90 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+IMD)
16	f6Freq_A(Hz)	1. The 6th Harmonic Frequency for Channel A 2. The 6th Peak Frequency for Channel A 3. The 3rd order IMD higher frequency for Channel A 4. 5.55 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)

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17	f7Freq_A(Hz)	1. The 7th Harmonic Frequency for Channel A 2. The 7th Peak Frequency for Channel A 3. 7.05 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+DIM)
18	fnFreq_A(Hz)_Array[?]	1. Harmonic Frequencies for Channel A 2. Peak Frequency for Channel A 3. 1st, 2nd, 3rd order IMD frequencies for Channel A 4. 3.15 kHz, 15 kHz, U1~U9 frequencies for Channel A "?" should be replaced by a number starting from 0.	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
19	f1RMS_A(EU)	1) RMS value at Peak / Fundamental Frequency for Channel A 2) Peak impedance for Channel A 3) RMS value at 3.15 kHz for Channel A	1. SPEC (AMS) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC (AMS+DIM)
20	f2RMS_A(EU)	1. RMS value of the 2nd harmonics for Channel A 2. RMS value of the 2nd peak for Channel A 3. RMS value at the 2nd peak frequency for Channel A 4. RMS value at 15 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
21	f3RMS_A(EU)	1. RMS value of the 3rd harmonics for Channel A 2. RMS value of the 3rd peak for Channel A 3. RMS value at the 2nd order IMD lower frequency for Channel A 4. RMS value at 0.75 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
22	f4RMS_A(EU)	1. RMS value of the 4th harmonics for Channel A 2. RMS value of the 4th peak for Channel A 3. RMS value at the 2nd order IMD higher frequency for Channel A 4. RMS value at 2.40 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
23	f5RMS_A(EU)	1. RMS value of the 5th harmonics for Channel A 2. RMS value of the 5th peak for Channel A 3. RMS value at the 3rd IMD lower frequency for Channel A 4. RMS value at 3.90 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
24	f6RMS_A(EU)	1. RMS value of the 6th harmonics for Channel A 2. RMS value of the 6th peak for Channel A 3. RMS value at the 3rd IMD higher frequency for Channel A 4. RMS value at 5.55 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC

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			(AMS+DIM)
25	f7RMS_A(EU)	1. RMS value of the 7th harmonics for Channel A 2. RMS value of the 7th peak for Channel A 3. RMS value at 7.05 kHz for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+DIM)
26	fnRMS_A(EU)_Array[?]	1. RMS values of harmonics for Channel A 2. RMS values of peaks for Channel A 3. RMS values of 1st, 2nd, 3rd order IMD frequencies for Channel A 4. RMS value in user defined frequency bands for Channel A 5. RMS value at 3.15kHz, 15kHz, or U1~U9 for Channel A 6. Sound intensity magnitude in user defined frequency bands "?" should be replaced by a number starting from 0.	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+EFB) 5. SPEC (AMS+DIM) 6. SPEC (AMS+SI+EFB)
27	f1Phase_A(D)	Phase of Peak / Fundamental frequency for Channel A	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak)
28	fnPhase_A(D)_Array[?]	1. Phases of harmonics for Channel A 2. Phases of peaks for Channel A 3. Direction of sound intensity "?" should be replaced by a number starting from 0.	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+SI+EFB)
29	THD_A(%)	THD in % for Channel A	SPEC(AMS)
30	THDN_A(%)	THD+N in % for Channel A	SPEC(AMS)
31	THDDB_A(dB)	THD in dB for Channel A	SPEC(AMS)
32	THDNDB_A(dB)	THD+N in dB for Channel A	SPEC(AMS)
33	SINAD_A(dB)	SINAD in dB for Channel A	SPEC(AMS)
34	SNR_A(dB)	SNR in dB for Channel A	SPEC(AMS)
35	NoiseLevel_A(EU)	Noise Level in EUrms for Channel A	SPEC(AMS)
36	ENOB_A(Bit)	Effective Number of Bits for Channel A	SPEC(AMS)
37	IMD_A(%)	IMD in % for Channel A	SPEC (AMS+IMD)
38	IMD_A(dB)	IMD in dB for Channel A	SPEC (AMS+IMD)
39	TDN_A(%)	TD+N in % for Channel A	SPEC (AMS+Peak+TDN)
40	TDN_A(dB)	TD+N in dB for Channel A	SPEC (AMS+Peak+TDN)
41	GedLeeMetric_A	GedLee Metric for Channel A	SPEC (AMS+AMS)
42	BandWidthLowLimit_A(Hz)	Band Width Low Limit for Channel A	SPEC (AMS+BW)/ (AMS+OCT)
43	BandWidthHighLimit_A(Hz)	Band Width High Limit for Channel A	SPEC (AMS+BW)/ (AMS+OCT)
44	fBand1RMS_A(EU)	1. RMS value of frequency band 1 for Channel A 2. Sound intensity magnitude of frequency band 1	1.SPEC (AMS+EFB) 2.SPEC (AMS+SI+EFB)

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45	fBand2RMS_A(EU)	1. RMS value of frequency band 2 for Channel A 2. Sound intensity magnitude of frequency band 2	1.SPEC (AMS+EFB) 2.SPEC (AMS+SI+EFB)
46	fBand3RMS_A(EU)	1.RMS value of frequency band 3 for Channel A 2. Sound intensity magnitude of frequency band 3	1.SPEC (AMS+EFB) 2.SPEC (AMS+SI+EFB)
47	fBand4RMS_A(EU)	1.RMS value of frequency band 4 for Channel A 2. Sound intensity magnitude of frequency band 4	1.SPEC (AMS+EFB) 2.SPEC (AMS+SI+EFB)
48	fBand5RMS_A(EU)	1.RMS value of frequency band 5 for Channel A 2. Sound intensity magnitude of frequency band 5	1.SPEC (AMS+EFB) 2.SPEC (AMS+SI+EFB)
49	fBand6RMS_A(EU)	1.RMS value of frequency band 6 for Channel A 2. Sound intensity magnitude of frequency band 6	1.SPEC (AMS+EFB) 2.SPEC (AMS+SI+EFB)
50	fBand7RMS_A(EU)	1.RMS value of frequency band 7 for Channel A 2. Sound intensity magnitude of frequency band 7	1.SPEC (AMS+EFB) 2.SPEC (AMS+SI+EFB)
51	fBand8RMS_A(EU)	1.RMS value of frequency band 8 for Channel A 2. Sound intensity magnitude of frequency band 8	1.SPEC (AMS+EFB) 2.SPEC (AMS+SI+EFB)
52	SFDR_A(dB)	Spurious Free Dynamic Range for Channel A	SPEC (AMS+Peak)
53	RMSDBV_A(dBEU)	RMS value in dBEU for Channel A or combined	MM(dBV)
54	RMSDBu_A(dBu)	RMS value in dBu for Channel A or combined	MM(dBu)
55	RMSDB_A(dB)	RMS value in dB for Channel A or combined	MM(dB)
56	RMSDBA_A(dBA)	RMS value in dB(A) for Channel A	MM(dBA) SPEC(AMS+A)
57	RMSDBZ_A(dBZ)	RMS value in dB(Z) for Channel A	MM(dBZ) SPEC(AMS)
58	RMSDBC_A(dBC)	RMS value in dB(C) for Channel A	MM(dBC) SPEC(AMS+C)
59	Freq_A(Hz)	Counted Frequency for Channel A	MM (Freq. Counter)
60	RPM_A(rpm)	Counted RPM for Channel A	MM(RPM)
61	TotalCount_A	Total Count for Channel A	MM(Counter)
62	DutyCycle_A(%)	Duty Cycle for Channel A	MM(Duty Cycle)
63	FVC_A(V)	Voltage after F/V conversion for Channel A	MM(F/V)
64	CycleRMS_A(EU)	Cycle RMS value for Channel A	MM(CycleRMS)
65	CycleMean_A(EU)	Cycle Mean value for Channel A	MM(CycleMean)
66	CyclePWR_A(W)	Cycle Power value (= $CycleRMS^2$ / Load Factor) for Channel A	MM(CycleRMS)
67	PulseWidth_A(s)	Pulse Width for Channel A	MM(Pulse Width)
68	AccelerationRMS_A(g)	Acceleration RMS value in g for Channel A	MM(Vibrometer)
69	AccelerationRMS_A(m/s^2)	Acceleration RMS value in m/s^2 for Channel A	MM(Vibrometer)
70	AccelerationPeak_A(g)	Acceleration Peak value in g for Channel A	MM(Vibrometer)
71	AccelerationPeak_A(m/s^2)	Acceleration Peak value in m/s^2 for Channel A	MM(Vibrometer)

72	AccelerationCF_A	Acceleration Crest Factor for Channel A	MM(Vibrometer)
73	VelocityRMS_A(mm/s)	Velocity RMS value for Channel A	MM(Vibrometer)
74	VelocityPeak_A(mm/s)	Velocity Peak value for Channel A	MM(Vibrometer)
75	VelocityCF_A	Velocity Crest Factor for Channel A	MM(Vibrometer)
76	DisplacementRMS_A(μm)	Displacement RMS value for Channel A	MM(Vibrometer)
77	DisplacementPP_A(μm)	Displacement Peak-to-Peak value for Channel A	MM(Vibrometer)
78	DisplacementCF_A	Displacement Crest Factor for Channel A	MM(Vibrometer)
79	WaveformComparisonH_A	Waveform High or High-High Limit Alarm for Channel A	OSC
80	WaveformComparisonL_A	Waveform Low or Low-Low Limit Alarm for Channel A	OSC
81	WaveformComparisonSimilarity_A	Waveform Similarity Coefficient for Channel A (Reserved)	OSC
82	SpectrumComparisonH_A	Spectrum High or High-High Limit Alarm for Channel A	SPEC
83	SpectrumComparisonL_A	Spectrum Low or Low-Low Limit Alarm for Channel A	SPEC
84	SpectrumComparisonSimilarity_A	Spectrum Similarity Coefficient for Channel A (Reserved)	SPEC
85	InBandMinF_A(Hz)	1. Frequency at which the minimum RMS value is located in the specified frequency band in Channel A 2. Frequency at which the minimum impedance value is located in the specified frequency band in Channel A 3. Frequency at which the minimum gain value is located in the specified frequency band 4. Frequency at which the minimum impedance value is located in the specified frequency band	1. SPEC (AMS)/(AMS+OCT) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC(TF) 4. SPEC(TF+IM)
86	InBandMinRMS_A(EU)	1. The minimum RMS value in the specified frequency band in Channel A 2. The minimum impedance value in the specified frequency band in Channel A 3. The minimum gain value in the specified frequency band 4. The minimum impedance value in the specified frequency band	1. SPEC(AMS) /(AMS+OCT) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC(TF) 4. SPEC(TF+IM)
87	InBandMaxF_A(Hz)	1. Frequency at which the maximum RMS value is located in the specified frequency band in Channel A 2. Frequency at which the maximum impedance value is located in the specified frequency band in Channel A 3. Frequency at which the maximum gain value is located in the specified frequency band 4. Frequency at which the maximum impedance value is located in the specified frequency band	1. SPEC(AMS) /(AMS+OCT) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC(TF) 4. SPEC(TF+IM)
88	InBandMaxRMS_A(EU)	1.The maximum RMS value in the specified frequency band in Channel A 2.The maximum impedance value in the specified frequency band in Channel A 3.The maximum gain value in the specified frequency band 4.The maximum impedance value in the	1. SPEC(AMS) /(AMS+OCT) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC(TF) 4. SPEC(TF+IM)

		specified frequency band	
89	oBandRMS_A(EU)_Array[?]	1. RMS value in a particular octave band in Channel A. 2. Impedance value in a particular octave band in Channel A. 3. Sound intensity magnitude in a particular octave band "?" should be replaced by the actual band number starting from 0.	1.SPEC (AMS+OCT) 2.SPEC (AMS+OCT+IM) 3.SPEC (AMS+OCT+SI)
90	WowAndFlutter_A(%)	Peak Wow & Flutter for Channel A Peak Jitter for Channel A	SPEC(AMS) MM(FC~PW)
91	WowAndFlutterRMS_A(%)	RMS Wow & Flutter for Channel A RMS Jitter for Channel A	SPEC(AMS) MM(FC~PW)
92	Drift_A(%)	RMS Drift for Channel A	SPEC(AMS)
93	WowRMS_A(%)	RMS Wow for Channel A Max. Negative Jitter for Channel A	SPEC(AMS) MM(FC~PW)
94	FlutterRMS_A(%)	RMS Flutter for Channel A Max. Positive Jitter for Channel A	SPEC(AMS) MM(FC~PW)
95	SecondPeakACFTimeDelay_A(s)	Time Delay at the second peak of the Auto-Correlation Function for Channel A	SPEC(ACF)
96	SecondPeakACFCoef_A	The second peak Auto-Correlation Function Coefficient for Channel A	SPEC(ACF)
97	Loudness_A(SONE)	Sound Loudness for Channel A	SPEC(AMS)
98	LoudnessLevel_A(PHON)	Sound Loudness Level for Channel A	SPEC(AMS)
99	Sharpness_A(ACUM)	Sound Sharpness for Channel A	SPEC(AMS)
100	OutputAmplitude_A(dBFS)	Output Amplitude in dBFS for Channel A	GEN
101	AcousticParameter_A_Array[?]	Acoustic Parameters for Channel A. "?" should be replaced by the actual array index starting from 0.	OSC(RT60)
102	MaxDiscontinuityValue_A(%)	Maximum Discontinuity Value for Channel A	OSC(DIS)
103	MaxDiscontinuityTime_A(s)	Time at Maximum Discontinuity for Channel A	OSC(DIS)
104	StepResponseLowLevel_A(EU)	Low Level in Step Response for Channel A	OSC(STR)
105	StepResponseHighLevel_A(EU)	High Level in Step Response for Channel A	OSC(STR)
106	StepResponseOvershoot_A(%)	Overshoot in Step Response for Channel A	OSC(STR)
107	StepResponseUndershoot_A(%)	Undershoot in Step Response for Channel A	OSC(STR)
108	StepResponseRiseTime_A(s)	Rise Time in Step Response for Channel A	OSC(STR)
109	StepResponseFallTime_A(s)	Fall Time in Step Response for Channel A	OSC(STR)
110	StepResponsePeakTime_A(s)	Peak Time in Step Response for Channel A	OSC(STR)
111	StepResponsePeakSettle_A(s)	Peak Settling Time in Step Response for Channel A	OSC(STR)
112	StepResponseTroughTime_A(s)	Trough Time in Step Response for Channel A	OSC(STR)
113	StepResponseTroughSettle_A(s)	Trough Settling Time in Step Response for Channel A	OSC(STR)
114	EchoTime_A(s)	Echo Time for Channel A	OSC(ECH)
115	DampingRatio_A(s)	Damping Ratio for Channel A	OSC(DPR)
116	ArticulationIndex_A(%)	Articulation Index for Channel A	SPEC(AMS)
117	OpenArticulationIndex_A(%)	Open Articulation Index for Channel A	SPEC(AMS)
118	NoiseRating_A	Noise Rating Value for Channel A	SPEC(AMS)
119	NoiseCriterion_A	Noise Criterion Value for Channel A	SPEC(AMS)
120	Max_B(EU)	Maximum value for Channel B	OSC(&, |)

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121	Min_B(EU)	Minimum value for Channel B	OSC(&, |)
122	PP_B(EU)	Peak-to-Peak value for Channel B	OSC(&, |)
123	Mean_B(EU)	Mean value for Channel B	OSC(&, |)
124	RMS_B(EU)	RMS value for Channel B	OSC(&, |)
125	PWR_B(W)	Power value (= RMS^2 / Load Factor) for Channel B	OSC(&, |)
126	Sknewness_B(%)	Sknewness for Channel B	OSC
127	Kurtosis_B(%)	Kurtosis for Channel B	OSC
128	PeakLevelPercent_B(%)	Peak Level in Percentage for Channel B	OSC
129	PeakLeveldBFS_B	Peak Level in dBFS for Channel B	OSC
130	f1Freq_B(Hz)	1. Peak / Fundamental Frequency for Channel B 2. Frequency at peak impedance for Channel B 3. 3.15 kHz for Channel B	1. SPEC(AMS) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC (AMS+DIM)
131	f2Freq_B(Hz)	1. The 2nd Harmonic Frequency for Channel B 2. The 2nd Peak Frequency for Channel B 3. The 2nd Peak Frequency for Channel B 4. 15 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
132	f3Freq_B(Hz)	1. The 3rd Harmonic Frequency for Channel B 2. The 3rd Peak Frequency for Channel B 3. The 2nd order IMD lower frequency for Channel B 4. 0.75 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
133	f4Freq_B(Hz)	1. The 4th Harmonic Frequency for Channel B 2. The 4th Peak Frequency for Channel B 3. The 2nd order IMD higher frequency for Channel B 4. 2.40 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
134	f5Freq_B(Hz)	1. The 5th Harmonic Frequency for Channel B 2. The 5th Peak Frequency for Channel B 3. The 3nd order IMD lower frequency for Channel B 4. 3.90 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
135	f6Freq_B(Hz)	1. The 6th Harmonic Frequency for Channel B 2. The 6th Peak Frequency for Channel B 3. The 3nd order IMD higher frequency for Channel B 4. 5.55 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
136	f7Freq_B(Hz)	1. The 7th Harmonic Frequency for Channel B 2. The 7th Peak Frequency for Channel B	1. SPEC (AMS+HARM) 2. SPEC

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		3. 7.05 kHz for Channel B	(AMS+Peak) 3. SPEC (AMS+DIM)
137	fnFreq_B(Hz)_Array[?]	1. Harmonic frequencies for Channel B 2. Peak frequency for Channel B 3. 1st, 2nd, 3rd order IMD frequencies for Channel B 4. 3.15 kHz, 15 kHz, U1~U5 frequencies for Channel B "?" should be replaced by a number starting from 0.	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
138	f1RMS_B(EU)	1. RMS value at Peak / Fundamental Frequency for Channel B 2. Peak impedance for Channel B 3. RMS value at 3.15 kHz for Channel B	1. SPEC(AMS) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC (AMS+DIM)
139	f2RMS_B(EU)	1. RMS value of the 2nd harmonics for Channel B 2. RMS value of the 2nd peak for Channel B 3. RMS value at the 2nd peak frequency for Channel B 4. RMS value at 15 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
140	f3RMS_B(EU)	1. RMS value of the 2nd harmonics for Channel B 2. RMS value of the 2nd peak for Channel B 3. RMS value at the 2nd order IMD lower frequency for Channel B 4. RMS value at 0.75 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
141	f4RMS_B(EU)	1. RMS value of the 2nd harmonics for Channel B 2. RMS value of the 2nd peak for Channel B 3. RMS value at the 2nd order IMD higher frequency for Channel B 4. RMS value at 2.40 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
142	f5RMS_B(EU)	1. RMS value of the 2nd harmonics for Channel B 2. RMS value of the 2nd peak for Channel B 3. RMS value at the 3nd order IMD lower frequency for Channel B 4. RMS value at 3.90 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
143	f6RMS_B(EU)	1. RMS value of the 2nd harmonics for Channel B 2. RMS value of the 2nd peak for Channel B 3. RMS value at the 3nd order IMD higher frequency for Channel B 4. RMS value at 5.55 kHz for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+DIM)
144	f7RMS_B(EU)	1. RMS value of the 2nd harmonics for Channel B	1. SPEC (AMS+HARM)

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		2. RMS value of the 2nd peak for Channel B 3. RMS value at 7.05 kHz for Channel B	2. SPEC (AMS+Peak) 3. SPEC (AMS+DIM)
145	fnRMS_B(EU)_Array[?]	1. RMS value of harmonics for Channel B 2. RMS value of peaks for Channel B 3. RMS values of 1st, 2nd, 3rd order IMD frequencies for Channel B 4. RMS value in user defined frequency bands for Channel B 5. RMS value at 3.15 kHz, 15kHz, or U1~U9 for Channel B "?" should be replaced by a number starting from 0.	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak) 3. SPEC (AMS+IMD) 4. SPEC (AMS+EFB) 5. SPEC (AMS+DIM)
146	f1Phase_B(D)	Phase of Peak / Fundamental Frequency for Channel B	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak)
147	fnPhase_B(D)_Array[?]	1. Phases of harmonic frequencies for Channel B 2. Phases of peak frequencies for Channel B "?" should be replaced by a number starting from 0.	1. SPEC (AMS+HARM) 2. SPEC (AMS+Peak)
148	THD_B(%)	THD in % for Channel B	SPEC(AMS)
149	THDN_B(%)	THD+N in % for Channel B	SPEC(AMS)
150	THDDB_B(dB)	THD in dB for Channel B	SPEC(AMS)
151	THDNDB_B(dB)	THD+N in dB for Channel B	SPEC(AMS)
152	SINAD_B(dB)	SINAD in dB for Channel B	SPEC(AMS)
153	SNR_B(dB)	SNR in dB for Channel B	SPEC(AMS)
154	NoiseLevel_B(EU)	Noise Level in EUrms for Channel B	SPEC(AMS)
155	ENOB_B(Bit)	Effective Number of Bits for Channel B	SPEC(AMS)
156	IMD_B(%)	IMD in % for Channel B	SPEC (AMS+IMD)
157	IMD_B(dB)	IMD in dB for Channel B	SPEC (AMS+IMD)
158	TDN_B(%)	TD+N in % for Channel B	SPEC (AMS+Peak+TDN)
159	TDN_B(dB)	TD+N in dB for Channel B	SPEC (AMS+Peak+TDN)
160	GedLeeMetric_B	GedLee Metric for Channel B	SPEC (AMS+AMS)
161	BandWidthLowLimit_B(Hz)	Band Width Low Limit for Channel B	SPEC (AMS+BW)/ (AMS+OCT)
162	BandWidthHighLimit_B(Hz)	Band Width High Limit for Channel B	SPEC (AMS+BW)/ (AMS+OCT)
163	fBand1RMS_B(EU)	RMS value of frequency band 1 for Channel B	SPEC (AMS+EFB)
164	fBand2RMS_B(EU)	RMS value of frequency band 2 for Channel B	SPEC (AMS+EFB)
165	fband3RMS_B(EU)	RMS value of frequency band 3 for Channel B	SPEC (AMS+EFB)
166	fband4RMS_B(EU)	RMS value of frequency band 4 for Channel B	SPEC (AMS+EFB)

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167	fBand5RMS_B(EU)	RMS value of frequency band 5 for Channel B	SPEC (AMS+EFB)
168	fBand6RMS_B(EU)	RMS value of frequency band 6 for Channel B	SPEC (AMS+EFB)
169	fBand7RMS_B(EU)	RMS value of frequency band 7 for Channel B	SPEC (AMS+EFB)
170	fBand8RMS_B(EU)	RMS value of frequency band 8 for Channel B	SPEC (AMS+EFB)
171	SFDR_B(dB)	Spurious Free Dynamic Range for Channel B	SPEC (AMS+Peak)
172	RMSDBV_B(dBEU)	RMS value in dBEU for Channel B	MM(dBV)
173	RMSDBu_B(dBu)	RMS value in dBu for Channel B	MM(dBu)
174	RMSDB_B(dB)	RMS value in dB for Channel B	MM(dB)
175	RMSDBA_B(dBA)	RMS value in dB(A) for Channel B	MM(dBA) SPEC (AMS+A)
176	RMSDBZ_B(dBZ)	RMS value in dB(Z) for Channel B	MM(dBZ) SPEC(AMS)
177	RMSDBC_B(dBC)	RMS value in dB(C) for Channel B	MM(dBC) SPEC(AMS+C)
178	Freq_B(Hz)	Counted Frequency for Channel B	MM (Freq. Counter)
179	RPM_B(rpm)	Counted RPM for Channel B	MM(RPM)
180	TotalCount_B	Total Count for Channel B	MM(Counter)
181	DutyCycle_B(%)	Duty Cycle for Channel B	MM(Duty Cycle)
182	FVC_B(V)	Voltage after F/V conversion for Channel B	MM(F/V)
183	CycleRMS_B(EU)	Cycle RMS for Channel B	MM(CycleRMS)
184	CycleMean_B(EU)	Cycle Mean for Channel B	MM(CycleMean)
185	CyclePWR_B(W)	Cycle Power (= CycleRMS^2 / Load Factor) for Channel B	MM(CycleRMS)
186	PulseWidth_B(s)	Pulse Width for Channel B	MM(Pulse Width)
187	AccelerationRMS_B(g)	Acceleration RMS value in g for Channel B	MM(Vibrometer)
188	AccelerationRMS_B(m/s^2)	Acceleration RMS value in m/s^2 for Channel B	MM(Vibrometer)
189	AccelerationPeak_B(g)	Acceleration Peak value in g for Channel B	MM(Vibrometer)
190	AccelerationPeak_B(m/s^2)	Acceleration Peak value in m/s^2 for Channel B	MM(Vibrometer)
191	AccelerationCF_B	Acceleration Crest Factor for Channel B	MM(Vibrometer)
192	VelocityRMS_B(mm/s)	Velocity RMS value for Channel B	MM(Vibrometer)
193	VelocityPeak_B(mm/s)	Velocity Peak value for Channel B	MM(Vibrometer)
194	VelocityCF_B	Velocity Crest Factor for Channel B	MM(Vibrometer)
195	DisplacementRMS_B(μm)	Displacement RMS value for Channel B	MM(Vibrometer)
196	DisplacementPP_B(μm)	Displacement Peak-to-Peak value for Channel B	MM(Vibrometer)
197	DisplacementCF_B	Displacement Crest Factor for Channel B	MM(Vibrometer)
198	WaveformComparisonH_B	Waveform High or High-High Limit Alarm for Channel B	OSC
199	WaveformComparisonL_B	Waveform Low or Low-Low Limit Alarm for Channel B	OSC
200	WaveformComparisonSimilarity_B	Waveform Similarity Coefficient for Channel B (Reserved)	OSC
201	SpectrumComparisonH_B	Spectrum High or High-High Limit Alarm for Channel B	SPEC
202	SpectrumComparisonL_B	Spectrum Low or Low-Low Limit Alarm for Channel B	SPEC
203	SpectrumComparisonSimilarity_B	Spectrum Similarity Coefficient for Channel B (Reserved)	SPEC
204	InBandMinF_B(Hz)	1. Frequency at which the minimum RMS value is located in the specified frequency band in Channel B	1. SPEC(AMS) /(AMS+OCT) 2. SPEC

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		2. Frequency at which the minimum impedance value is located in the specified frequency band in Channel B 3. Frequency at which the minimum gain value is located in the specified frequency band 4. Frequency at which the minimum impedance value is located in the specified frequency band	(AMS+IM)/ /(AMS+OCT+IM) 3. SPEC(TF) 4. SPEC(TF+IM)
205	InBandMinRMS_B(EU)	1. The minimum RMS value in the specified frequency band in Channel B 2. The minimum impedance value in the specified frequency band in Channel B 3. The phase at the minimum gain value in the specified frequency band 4. The phase at the minimum impedance value in the specified frequency band	1. SPEC(AMS) /(AMS+OCT) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC(TF) 4. SPEC(TF+IM)
206	InBandMaxF_B(Hz)	1. Frequency at which the maximum RMS value is located in the specified frequency band in Channel B 2. Frequency at which the maximum impedance value is located in the specified frequency band in Channel B 3. Frequency at which the maximum gain value is located in the specified frequency band 4. Frequency at which the maximum impedance value is located in the specified frequency band	1. SPEC(AMS) /(AMS+OCT) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC(TF) 4. SPEC(TF+IM)
207	InBandMaxRMS_B(EU)	1.The maximum RMS value in the specified frequency band in Channel B 2.The maximum impedance value in the specified frequency band in Channel B 3.The phase at the maximum gain value in the specified frequency band 4.The phase at the maximum impedance value in the specified frequency band	1. SPEC(AMS) /(AMS+OCT) 2. SPEC (AMS+IM) /(AMS+OCT+IM) 3. SPEC(TF) 4. SPEC(TF+IM)
208	oBandRMS_B(EU)_Array[?]	1. RMS value in a particular octave band in Channel B. 2. Impedance value in a particular octave band in Channel B. "?" should be replaced by the actual band number starting from 0.	1.SPEC (AMS+OCT) 2.SPEC (AMS+OCT+IM)
209	WowAndFlutter_B(%)	Peak Wow & Flutter for Channel B Peak Jitter for Channel B	SPEC(AMS) MM(FC~PW)
210	WowAndFlutterRMS_B(%)	RMS Wow & Flutter for Channel B RMS Jitter for Channel B	SPEC(AMS) MM(FC~PW)
211	Drift_B(%)	RMS Drift for Channel B	SPEC(AMS)
212	WowRMS_B(%)	RMS Wow for Channel B Max. Negative Jitter for Channel B	SPEC(AMS) MM(FC~PW)
213	FlutterRMS_B(%)	RMS Flutter for Channel B Max. Positive Jitter for Channel B	SPEC(AMS) MM(FC~PW)
214	SecondPeakACFTimeDelay_B(s)	Time Delay at the second peak of the Auto-Correlation Function for ChannelB	SPEC(ACF)
215	SecondPeakACFCoef_B	The second peak Auto-Correlation Function Coefficient for Channel B	SPEC(ACF)
216	Loudness_B(SONE)	Sound Loudness for Channel B	SPEC(AMS)
217	LoudnessLevel_B(PHON)	Sound Loudness Level for Channel B	SPEC(AMS)
218	Sharpness_B(ACUM)	Sound Sharpness for Channel B	SPEC(AMS)

219	OutputAmplitude_B(dBFS)	Output Amplitude in dBFS for Channel B	GEN
220	AcousticParameter_B_Array[?]	Acoustic Parameters for Channel B. "?" should be replaced by the actual array index starting from 0.	OSC(RT60)
221	MaxDiscontinuityValue_B(%)	Maximum Discontinuity Value for Channel B	OSC(DIS)
222	MaxDiscontinuityTime_B(s)	Time at Maximum Discontinuity for Channel B	OSC(DIS)
223	StepResponseLowLevel_B(EU)	Low Level in Step Response for Channel B	OSC(STR)
224	StepResponseHighLevel_B(EU)	High Level in Step Response for Channel B	OSC(STR)
225	StepResponseOvershoot_B(%)	Overshoot in Step Response for Channel B	OSC(STR)
226	StepResponseUndershoot_B(%)	Undershoot in Step Response for Channel B	OSC(STR)
227	StepResponseRiseTime_B(s)	Rise Time in Step Response for Channel B	OSC(STR)
228	StepResponseFallTime_B(s)	Fall Time in Step Response for Channel B	OSC(STR)
229	StepResponsePeakTime_B(s)	Peak Time in Step Response for Channel B	OSC(STR)
230	StepResponsePeakSettle_B(s)	Peak Settling Time in Step Response for Channel B	OSC(STR)
231	StepResponseTroughTime_B(s)	Trough Time in Step Response for Channel B	OSC(STR)
232	StepResponseTroughSettle_B(s)	Trough Settling Time in Step Response for Channel B	OSC(STR)
233	EchoTime_B(s)	Echo Time for Channel B	OSC(ECH)
234	DampingRatio_B	Damping Ratio for Channel B	OSC(DPR)
235	ArticulationIndex_B(%)	Articulation Index for Channel B	SPEC(AMS)
236	OpenArticulationIndex_B(%)	Open Articulation Index for Channel B	SPEC(AMS)
237	NoiseRating_B	Noise Rating Value for Channel B	SPEC(AMS)
238	NoiseCriterion_B	Noise Criterion Value for Channel B	SPEC(AMS)
239	PeakCCFTimeDelay_AB(s)	Time Delay at the peak of the Cross-Correlation Function	SPEC(CCF)
240	PeakCCFCoef_AB	Peak Cross-Correlation Function Coefficient	SPEC(CCF)
241	PeakCHFFreq_AB(Hz)	Frequency at the peak of the Coherence Function	SPEC(CHF)
242	PeakCHFCoef_AB	Peak Coherence Function Coefficient	SPEC(CHF)
243	TNCD_AB(%)	Total Non-Coherent Distortion and Noise	SPEC(CHF)
244	PeakGainFreq_AB(Hz)	1. Frequency at the peak gain 2. Frequency at the peak impedance	1. SPEC(TF) 2. SPEC(TF+IM)
245	PeakGainValue_AB(dB)	1. Peak gain Value 2. Peak impedance Value	1. SPEC(TF) 2. SPEC(TF+IM)
246	PeakGainPhase_AB(D)	1. Phase at the peak gain 2. Phase at the peak impedance	1. SPEC(TF) 2. SPEC(TF+IM)
247	PeakIPRTime_AB(s)	Time at the peak of the Impulse Response	SPEC(IPR)
248	PeakIPRValue_AB	Peak Impulse Response Value	SPEC(IPR)
249	GainAtGeneratedFreq(dB)	1. Gain at the frequency generated by Channel A of the Signal Generator 2. Impedance at the frequency generated by Channel A of the Signal Generator	1. SPEC (TF,GEN) 2. SPEC (TF+IM,GEN)
250	PhaseAtGeneratedFreq(D)	1. Phase at the frequency generated by Channel A of the Signal Generator 2. Phase at the frequency generated by Channel A of the Signal Generator	1.SPEC (TF,GEN) 2. SPEC (TF+IM,GEN)
251	CrossTalkAB(dB)	Crosstalk from Channel A to Channel B	SPEC(AMS,CT)
252	CrossTalkBA(dB)	Crosstalk from Channel B to Channel A	SPEC(AMS,CT)
253	NumberOfFrames	Number of data frames sampled	Always
254	OverallPassFailCheck	Overall pass/fail check for all opened DDP viewers	DDP Viewer

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255 ~ 270	UDDP1(UU) ~ UDDP16(UU)	User Defined Data Points	DDP Viewer

Legend: OSC---Oscilloscope, SPEC---Spectrum Analyzer, GEN---Signal Generator, MM---Multimeter AMS---Amplitude Spectrum, CCF---Cross Correlation Function, CHF---Coherence Function, TF---Transfer Function, IPR---Impulse Response, BW---Bandwidth, EFB---Energy in User Defined Frequency Bands, HARM---Harmonics, CT---Crosstalk, A---A Weighting, Z---Z Weighting, C---C Weighting, OCT---Octave Analysis, IM---Impedance Mode, IMD---Intermodulation, DIM---Dynamic Intermodulation, TDN---Total Distortion+Noise, RT60---Reverberation and Speech Intelligibility, DIS---Discontinuity, STR---Step Response, ECH---Echo, DPR---Damping Ratio, FC-Frequency Counter, PW-Pulse Width, SI---Sound Intensity.

6.2.2 Legend

Each DDP can be assigned a unique color and label.

6.2.3 Logging Method

Three logging methods are supported:

- Fastest
  The data logger will take one record whenever a new data frame is available

- Time Interval
  The data logger will take one record whenever a new data frame is available and the specified time duration has elapsed since the last update.

- Update Threshold
  The data logger will take one record whenever a new data frame is available and the change of any DDP compared with its last update exceeds the specified update threshold for that DDP.

6.2.4 Log File Name

For real time logging, there is no need to specify the log file name as it will be assigned automatically by the software, according to the following naming convention:

Example: 2007-10-31-09-55-45-180-1.log

where:
2007 -----Year: 2007
10---------Month: October
31---------Day: 31
09---------Hour: 9
55---------Minute: 55
45---------Second: 45
180--------Millisecond: 180
1-----------Data Logger Window No.: 1
.log--------File Extension

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The above time stamp corresponds to the first data point captured in the data logger window. The logged data will be automatically written to the hard disk once the number of data points per trace exceeds 32767 (configurable via [Setting]>[Display]>"Data Logger">"Number of records per log file"). A new log file will be created automatically afterwards to continue the logging process. Note that the logged data will be lost if the DDP configuration is changed before the data in memory are saved to the hard disk. If you do not want to lose the data, you should perform "Data Logger Export" or "Data Logger Copy As Text" command first. When you close the data logger window, the logged data points that are still in the computer memory will be automatically written to the hard disk.

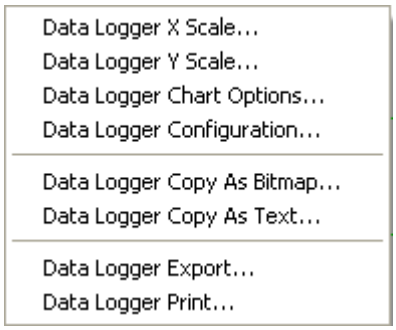
The log file name is always displayed in the title bar of the data logger window. A "*" sign will be displayed on the right of the log file name if the log file has not been persisted in the hard disk yet. This is to indicate that it is a real-time log file (e.g. 2020-07-22-23-29-51-878-2.log*). Otherwise, it is a historical log file (e.g. 2020-07-22-23-29-51-878-2.log).

Pressing the "File Open" button in the Configuration dialog will allow you to select and load a historical log file for review. Clicking the "Logging" checkbox will switch it back to the real-time logging mode.

The log file is a Comma Separated Value (CSV) TXT file. It has the following format which is self-explanatory. All data are logged with an accuracy of 1 millisecond.

No., Time(s), Max_A(V), Min_A(V), Mean_A(V)
1, 2007-10-31 12:04:52:021, -0.0132446, -0.0142517, -0.0137236
2, 2007-10-31 12:04:53:022, -0.0132751, -0.0144348, -0.0139354
……

6.3 Context Menu



Data Logger does not have its own menu and view parameter toolbar. Its functions are accessed via its context menu. The context menu pops up when you right click anywhere within the data logger window.

6.3.1 X Scale

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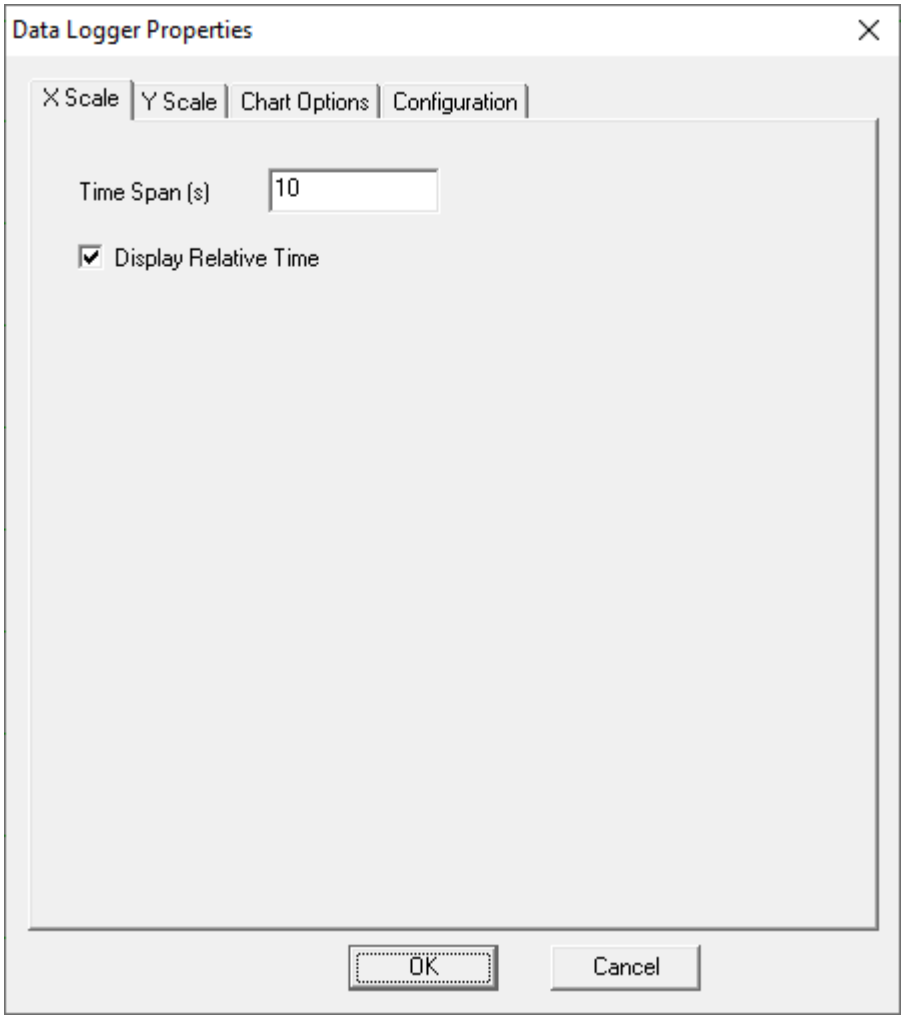

The X Scale dialog can be accessed via the context menu or by double clicking on X axis of the window. You can specify the span of X axis, the default value is 10 s. If the data spans a larger time range, then a horizontal scrollbar will appear at the bottom of the window to allow you to scroll over the full time range.

During real time logging, the screen will automatically scroll to show the latest data as new data are continuously fed into the right of the window.

"Display Relative Time" option is used to display time difference relative to the starting time of current log file rather than the absolute time. It is ticked by default.

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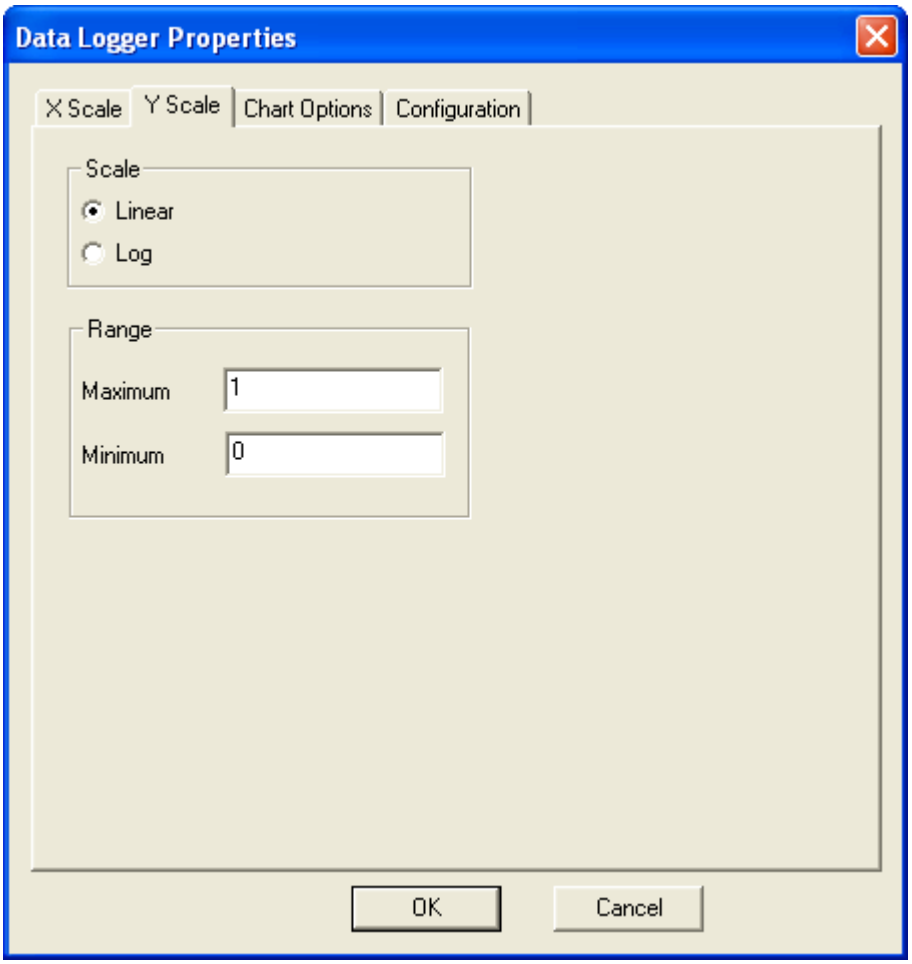
6.3.2 Y Scale



The Y Scale dialog can be accessed via the context menu or by double clicking on Y axis. You can specify the range of Y axis and choose between linear or logarithmic scales. Note that the software will disallow logarithmic scale if the range covers negative or zero values.

6.3.3 Chart Options

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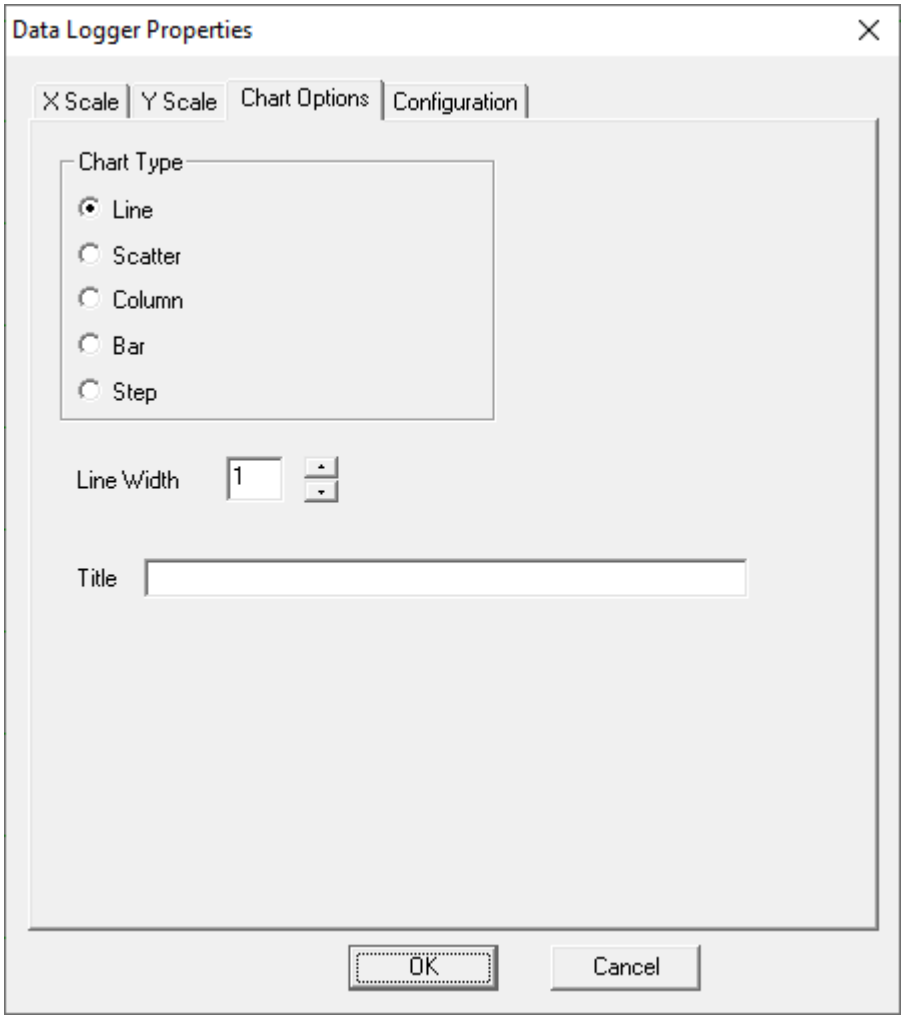

Same as Oscilloscope, except that you can also specify a title here.

6.3.4 Copy As Bitmap

Same as Oscilloscope

6.3.5 Copy As Text

Same as Oscilloscope

6.3.6 Export

Same as Oscilloscope

6.3.7 Print

Same as Oscilloscope

6.4 Cursor reader and Markers

Similar to those in the Oscilloscope, except that the cursor reader can read out the measurement points for all traces (up to 8) and the markers mark the point exactly at the point of mouse click rather than sticking to the nearest measurement point.

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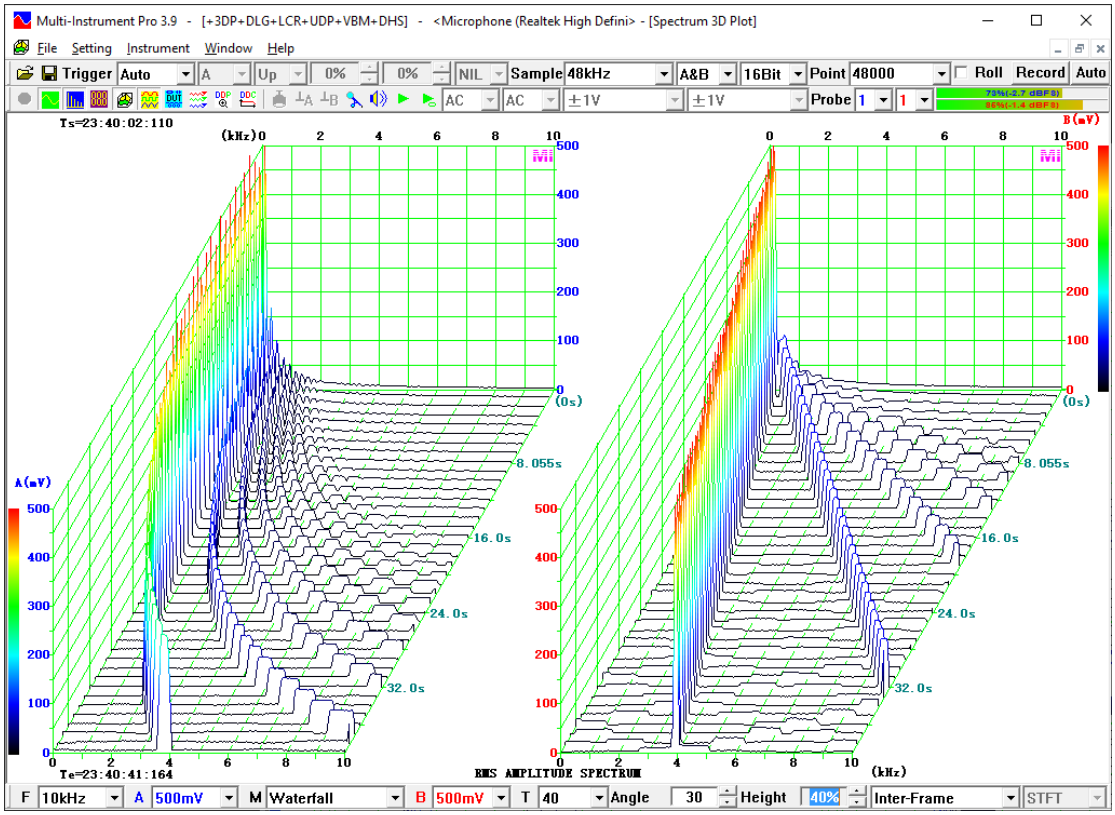
7 Spectrum 3D Plot

7.1 Overview

Spectrum 3D Plot is used to trace the spectrum variation with time. It is a very useful tool for time-frequency analysis. Two types of plots are provided: waterfall plot and spectrogram. Two processing modes are supported: Inter-frame and Intra-frame. Two analyzing modes are available: STFT (Short-Time Fourier Transform) and CSD (Cumulative Spectral Decay).

The Spectrum 3D Plot can be generated by multiple frames of data (Inter-frame Mode) or one single frame of data (Intra-frame Mode). The former supports STFT only while the latter supports both STFT and CSD.

The following figure shows a waterfall plot, with an adjustable tilt angle of T axis, an adjustable height of Y axis, and selectable color palettes.



The following figure shows a spectrogram, with selectable color palettes and a smoothing option.

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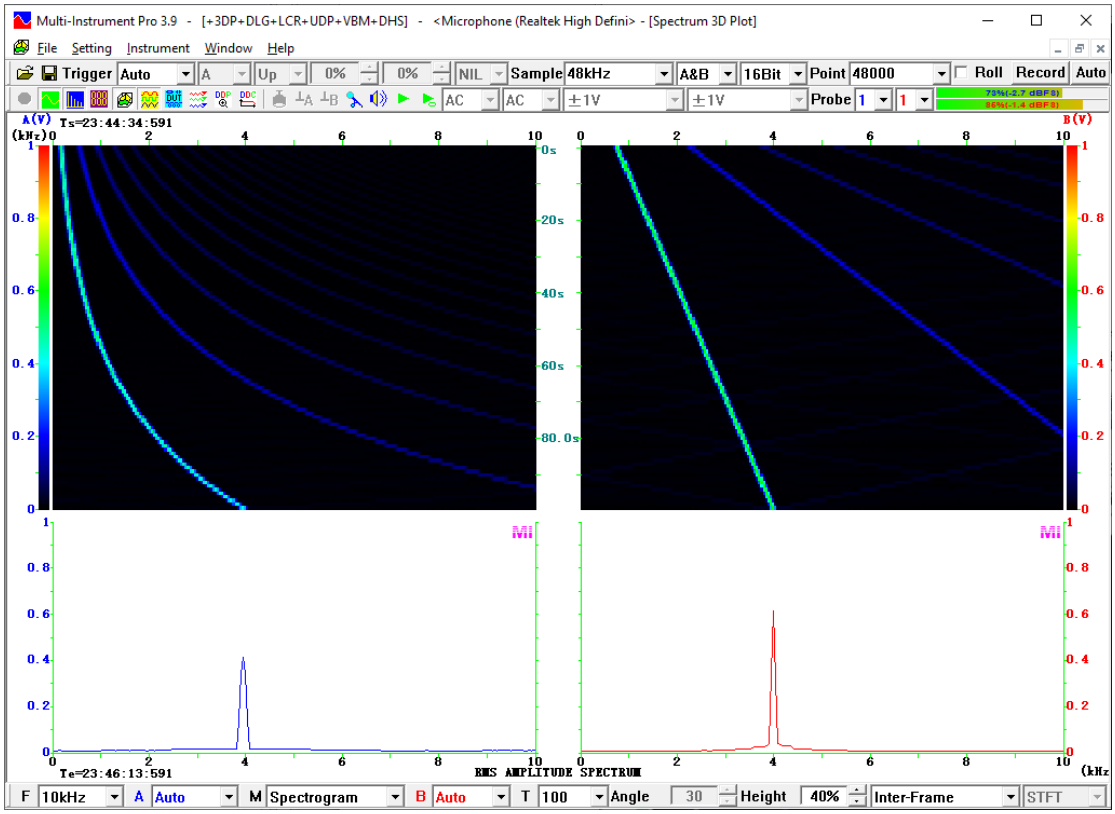


The toggle button in the Instrument Toolbar is used to open or close the Spectrum 3D Plot. The button is enabled only when the Spectrum Analyzer is under Amplitude Spectrum mode without “Cross Correlation” inter-frame processing, because the spectral profiles are fed from the Spectrum Analyzer under that mode. All intra-frame processing options set in the Spectrum Analyzer are applicable to the Spectrum 3D Plot too.

You can also close the Spectrum 3D Plot window by clicking the "Close" button at its upper right corner.

7.2 View Parameters

View parameters determine how the collected data are analyzed and displayed.

7.2.1 Frequency Range (F)

Same as Amplitude Spectrum display in the Spectrum Analyzer.

7.2.2 Channel A Display Range

Same as Amplitude Spectrum display in the Spectrum Analyzer.

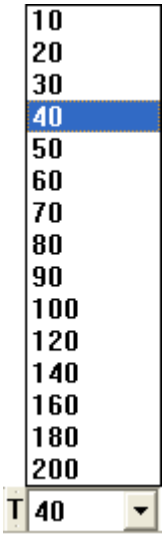
7.2.3 View Type

Two types of views are supported: Waterfall Plot and Spectrogram.

7.2.4 Channel B Display Range

Same as Amplitude Spectrum display in the Spectrum Analyzer.

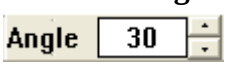
7.2.5 Number of Spectral Profiles (T axis)



This parameter determines the maximum number of spectral profiles to be displayed in one graph. All spectral profiles are computed and fed by the Spectrum Analyzer through either inter-frame processing or intra-frame processing. The oldest spectral profile will be dropped when a new spectral profile comes in.

This parameter is available for selection from the above options under inter-frame mode, while under intra-frame mode, its value will be automatically determined by the software and thus prohibited from user selection.

7.2.6 Tilt Angle of T Axis



The Tilt Angle of T axis is the angle between T axis and the vertical axis. It can be adjusted from 0 degree to a maximum value (less than 90 degrees) which is determined by the Height Percentage of Y axis.

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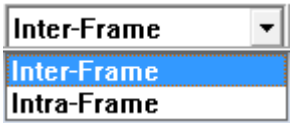
This parameter is only applicable to the Waterfall Plot and is not applicable to the Spectrogram.

7.2.7 Height Percentage of Y Axis

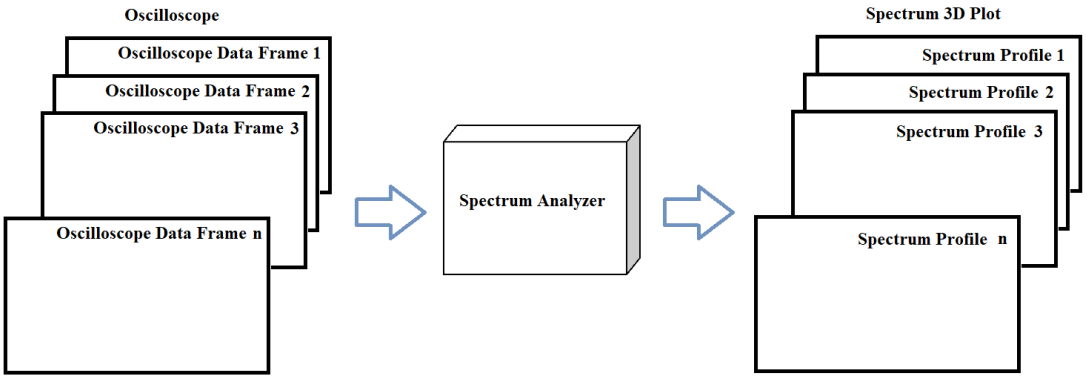
The Height Percentage is the ratio of the height of Y axis to the height of the plot area. It can be adjusted from 5% to 90%. Lowering the Height Percentage will cause the tilt angle of T axis go down accordingly if the tilt angle exceeds its upper limit determined by the Height Percentage.

7.2.8 Processing Mode



Two processing modes are supported: inter-frame mode and intra-frame mode.

7.2.8.1 Inter-frame Mode



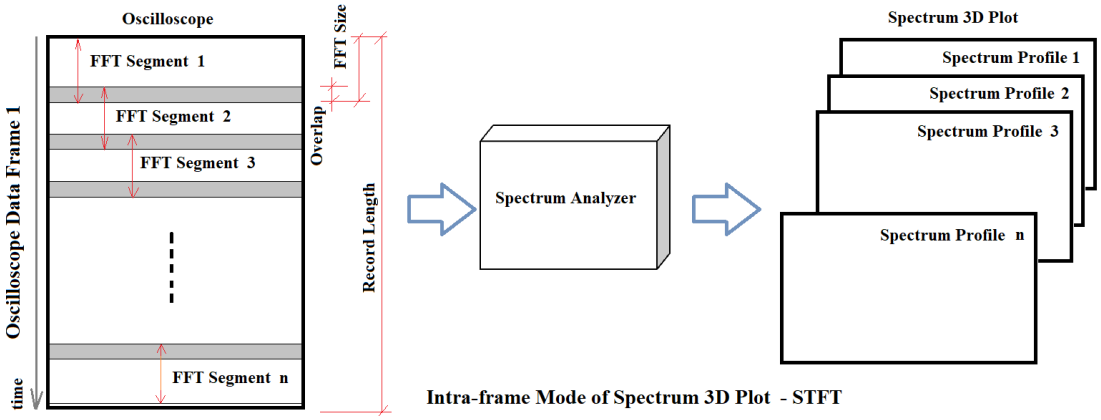
Inter-frame Mode of Spectrum 3D Plot - STFT

Under the inter-frame mode, the Spectrum Analyzer derives one spectrum profile per oscilloscope data frame and feed it into the Spectrum 3D Plot. Multiple consecutive oscilloscope data frames are required in order to fill up the entire Spectrum 3D Plot. The inter-frame mode supports only STFT.

7.2.8.2 Intra-frame Mode

Under the intra-frame mode, the Spectrum Analyzer splits an oscilloscope data frame into multiple FFT segments, derives one spectrum profile per FFT segment, and fill up the entire Spectrum 3D Plot in one shot. Two analyzing modes are supported: STFT and CSD.

7.2.8.2.1 Short-Time Fourier Transform (STFT)



Short-Time Fourier Transform (STFT) is a time-frequency analysis technique suited for non-stationary signals. It uses a short-time data window to perform Fourier transform in order to get the spectrum profile during that period. The spectral contents are considered quasi-stationary inside the data window. Spectral variation with time can be obtained by sliding or hopping the window along the time axis. This can be realized through either the intra-frame processing or the inter-frame processing described previously.

When conceiving your own time-frequency analysis strategy, the uncertainty principle should be noted. There exists a trade-off between time and frequency resolution in STFT. Although a narrow-width window results in a better resolution in time domain, it generates a poor resolution in frequency domain, and vice versa.

The number of FFT segments is automatically determined by the software as follows:

[Number of FFT segments] =
([Record Length]-[Overlap Size]) / ([FFT Size]-[Overlap Size])

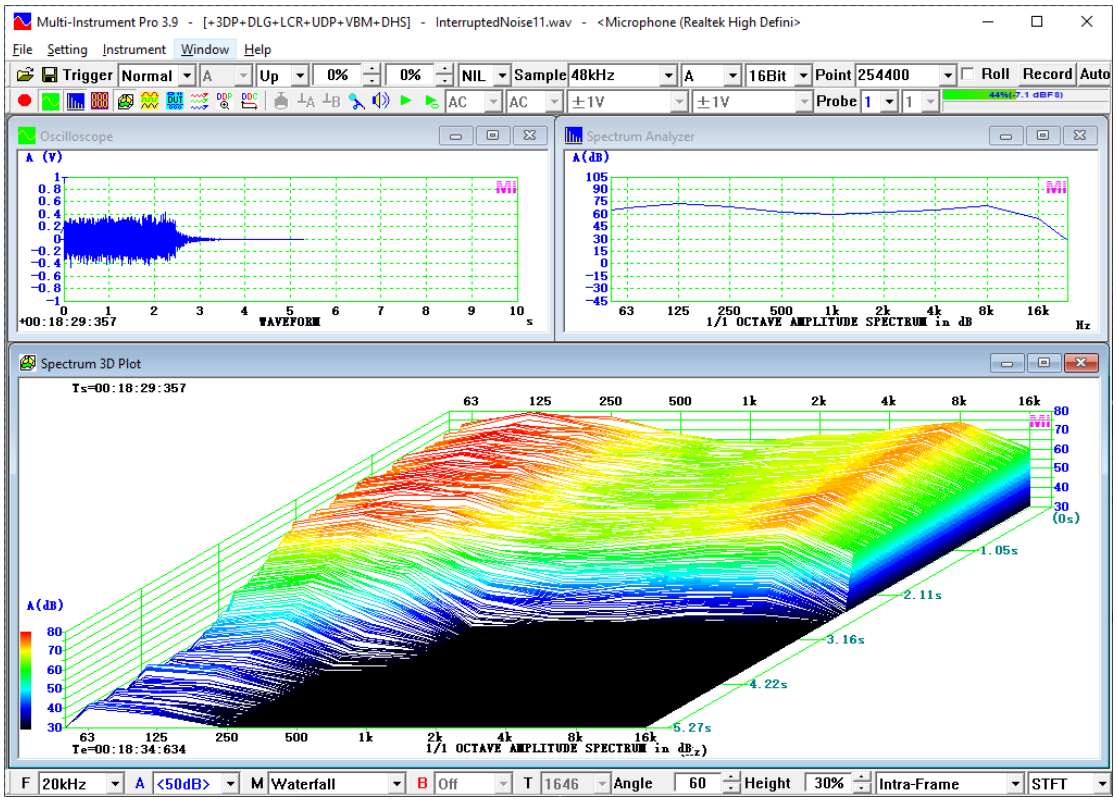
where [Overlap Size]= [FFT Size]×[Window Overlap Percentage]).

Note that:

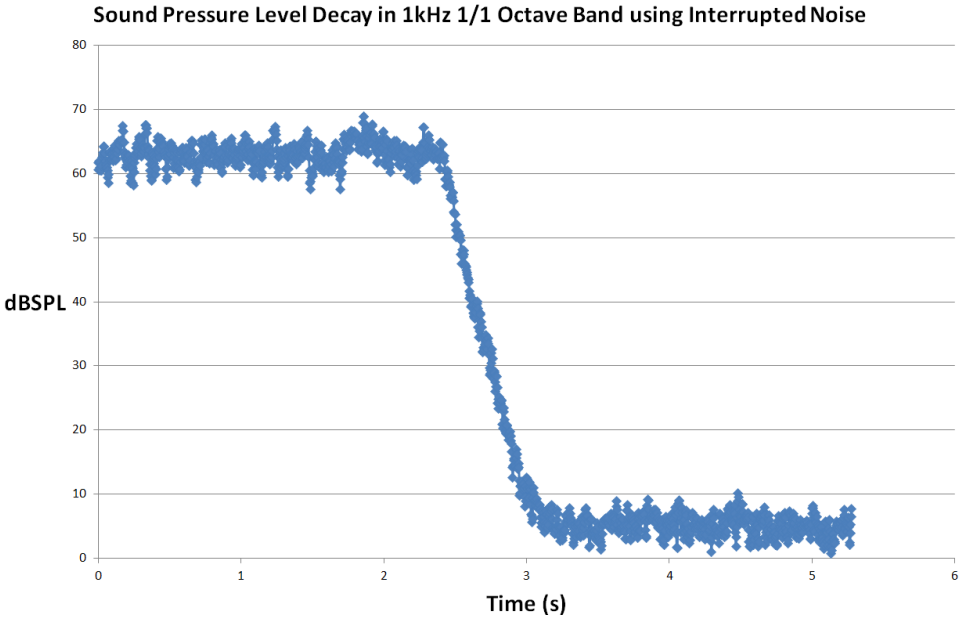
(1) If [Record Length]<[FFT Size], then the number of spectrum profiles will be forced to 1.
(2) The maximum number of spectrum profiles allowed is 2000. Data used for the Spectrum 3D Plot will be truncated if this maximum number is exceeded.
(3) No inter-frame processing is allowed in the Spectrum Analyzer.
(4) Window Overlap Percentage should generally be set to a value less than 99%.

The following figure illustrates how to use Spectrum 3D Plot to perform reverberation analysis. The sound pressure decay data were recorded using the interrupted noise method introduced previously. The parameters are as follows: [Sampling Frequency]=48kHz, [Record length]=254400, [FFT Size]=1024, [Window Overlap Percentage]=85%, [Overlap Size]=1024×85%=870. The number of spectrum profiles is then: (254400-870)/[1024-870]=1646. The time length of a FFT segment is

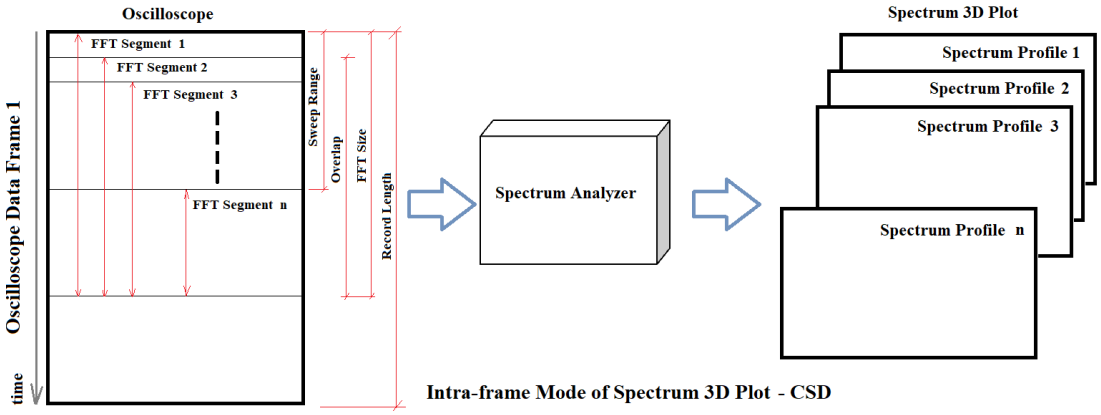
1024/48000=0.021333s. The time interval between two consecutive spectrum profiles is (1024-870)/48000=0.003208s. Hann window is used in the Spectrum Analyzer and the X scale of the Spectrum 3D Plot is set to 1/1 octave.



The data in the above figure can be exported to EXCEL for further analysis. The exported data can be either time-sliced or frequency-sliced. The following figure shows the sound pressure level decay in the 1kHz 1/1 octave band using the exported frequency-sliced data.



7.2.8.2.2 Cumulative Spectral Decay (CSD)



Cumulative Spectral Decay (CSD) is similar to STFT. However, unlike STFT, CSD does not use a fixed-size window and does not move the entire window. Instead, it anchors the end side of the data window at the end of the useful part of the signal, and only moves the start side of the data window along the time axis. As the window width decreases, the frequency resolution, which is equal to 1/[window width], becomes coarser, the unresolvable low frequency region of the spectrum becomes wider. This is often reflected by the boundary line along the low frequency end in the CSD graph. In contrast to STFT where the actual number of samples used in a FFT segment must be a power of 2 and no zero-padding is applied, CSD does not impose such a restriction and zero-padding is used when necessary.

CSD is often used when the signal duration is too short to have space for STFT window sliding. It is true that the signal can be recorded for a longer duration to solve this problem. However, this usually means more noises will be incorporated into the analysis as the signal may have already decayed significantly. Using CSD prevents the incorporation of the noise where the signal-to-noise ratio becomes bad. Due to this reason, CSD is mainly used to analyze loudspeaker impulse response which spans a very short duration. It is a useful tool for loudspeaker resonance detection.

The number of FFT segments in CSD is automatically determined by the software as follows:

[Number of FFT segments] = 1+[Sweep Range] / ([FFT Size]-[Overlap Size])

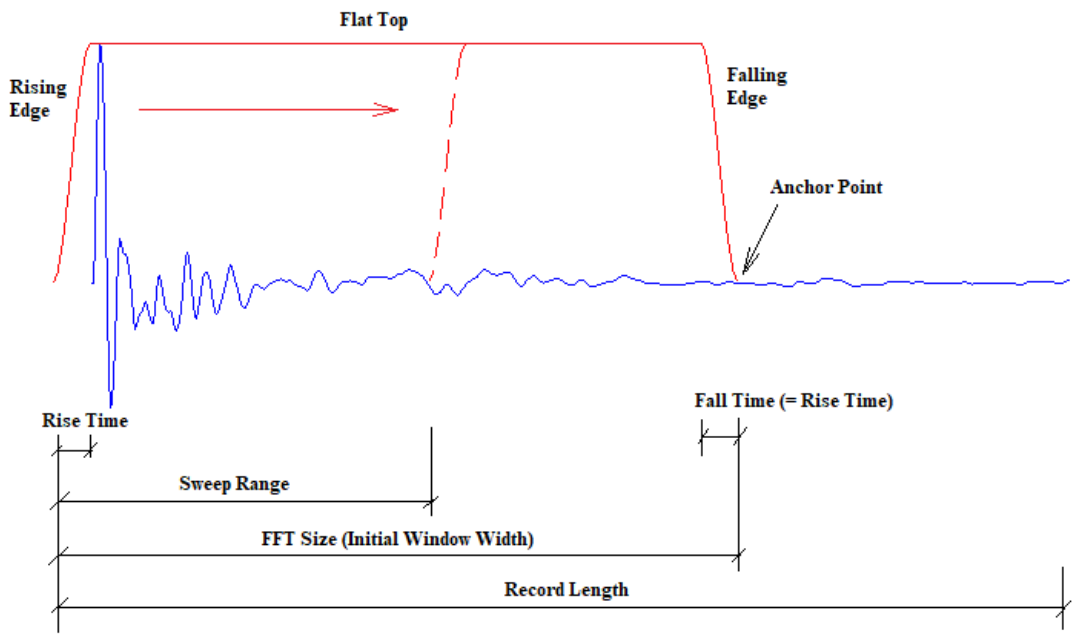
where:
[Sweep Range] = min([FFT Size],[Record Length]) × [Sweep Percentage]
[Overlap Size] = [FFT Size] × [Window Overlap Percentage]

In the CSD of this software, the FFT size determines the initial data window width. The moving range of the start side of the data window is called Sweep Range, which is equal to the product of the FFT size or Record Length, whichever is smaller, and the Sweep Percentage. The Sweep Percentage can be specified by the users and is 90% by default.

Note that:

(1) The number of spectrum profiles will be forced to be at least 1.
(2) The maximum number of spectrum profiles allowed is 2000. Data used for the Spectrum 3D Plot will be truncated if this maximum number is exceeded.
(3) No inter-frame processing is allowed in the Spectrum Analyzer.
(4) Window Overlap Percentage should generally be set to a value greater than 95%.

In Multi-Instrument, the window function is applied to the real samples in a FFT segment. If the signal is very much "left aligned", which is usually the case for an impulse response, then the useful information will be significantly tapered by a non-rectangle window function. To carter for the special need of CSD, a composite window function is used (see figure below). The window function consists of three parts: rising edge, flat top, and falling edge. The rising edge and falling edge are formed respectively by the left and right parts of the window function selected in the Spectrum Analyzer. The widths of the rising and falling edges are equal, fixed and configurable. The width of the flat top portion of the composite window decreases as the start side of the window moves along the time axis.



Composite Window Function used in CSD

The following figure shows a CSD waterfall plot of an impulse response of a loudspeaker box, with the following parameters: [Record Length]=6314, [Sampling Frequency]=44100Hz, [FFT Size]=1024, [Window Function]=Hann, [Window Overlap Percentage]=99%, [Rise Time]=0.35ms, [Sweep Percentage]=90%. Then we have: [Sweep Range] = 1024 × 90% =921, [Overlap Size] = 1024 × 99% =1013. The number of spectrum profiles is then: 1 + 921 / (1024-1013) = 84.

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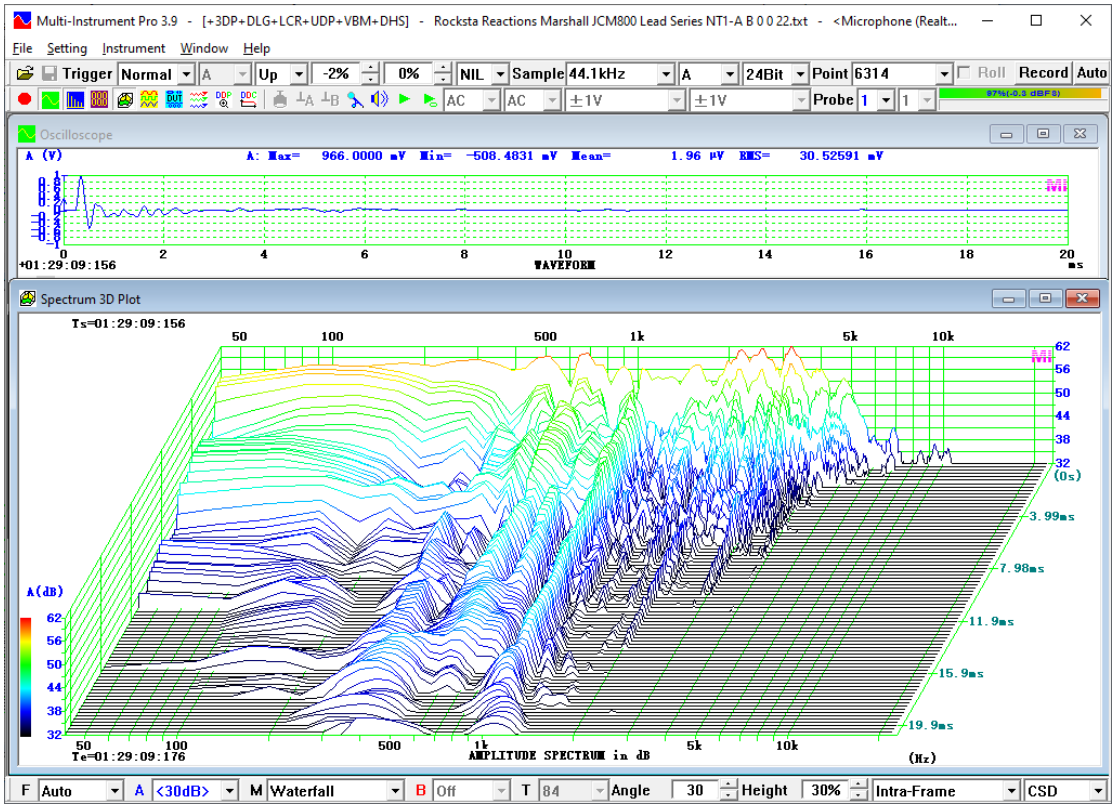
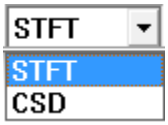


7.2.9 Analyzing Mode

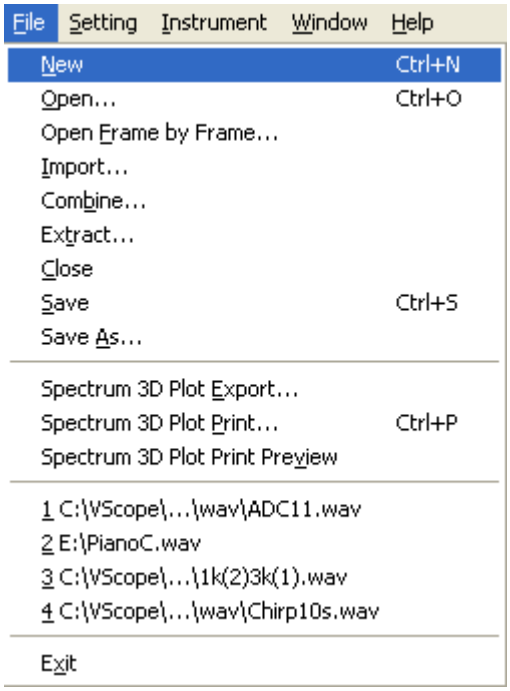


Two analyzing modes are available: STFT and CSD, as described previously.

7.3 Menu

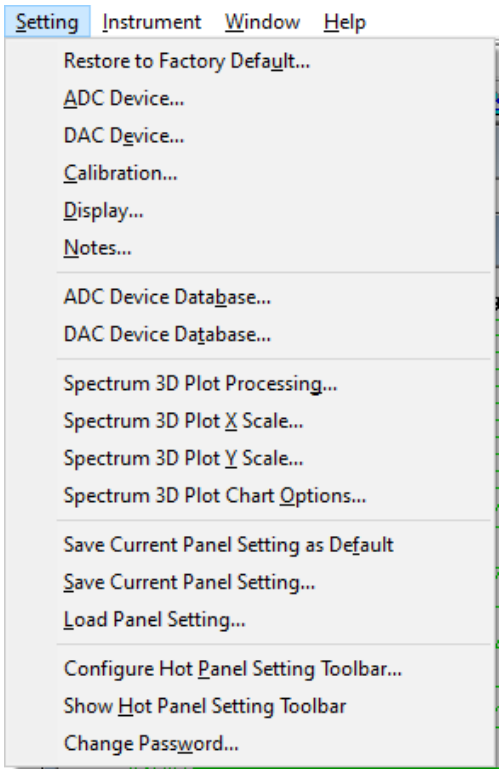
Spectrum 3D Plot has its own menu and additional functions can be accessed through the menu items in each submenu.

7.3.1 File SubMenu



This submenu provides access to the file operation and printing functions, and is similar to that of the Oscilloscope.

7.3.2 Setting SubMenu



This submenu provides access to various setting functions, and is similar to that of the Oscilloscope, except the following items.

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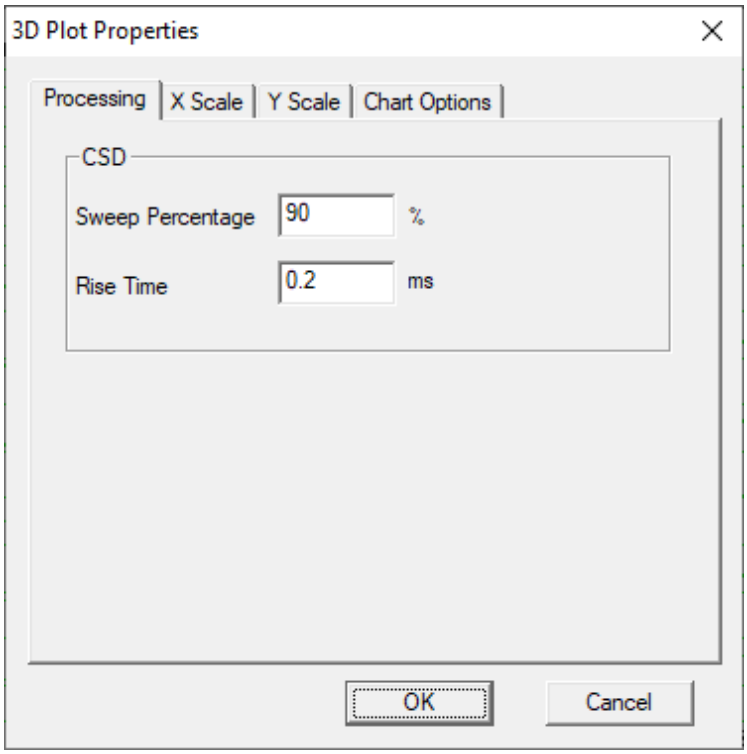
7.3.2.1 Spectrum 3D Plot Processing (Setting SubMenu) (ALT-S-G)



Under the Processing Tab, the Sweep Percentage and Rise Time of CSD can be configured. These two parameters have been introduced previously.

7.3.2.2 Spectrum 3D Plot X Scale (Setting SubMenu) (ALT-S-X)

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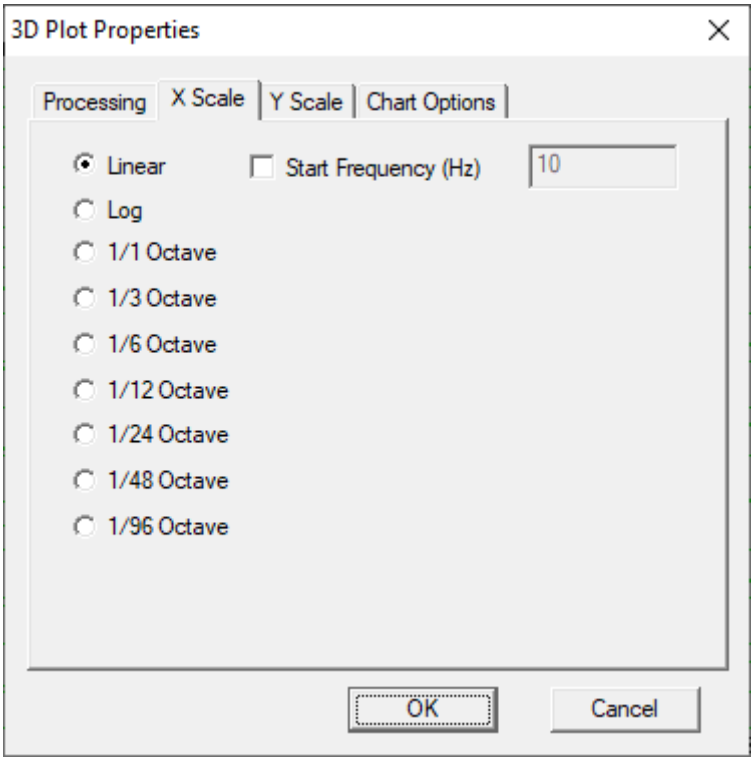

X axis (or F axis) can be set to linear, log, or 1/1~1/96 octave scale. The "Start Frequency" option allows you to specify the start frequency of the X axis. If it is not used, the start frequency will be determined by the software automatically.

This function can also be accessed by double clicking on X axis.

7.3.2.3 Spectrum 3D Plot Y Scale (Setting SubMenu) (ALT-S-Y)

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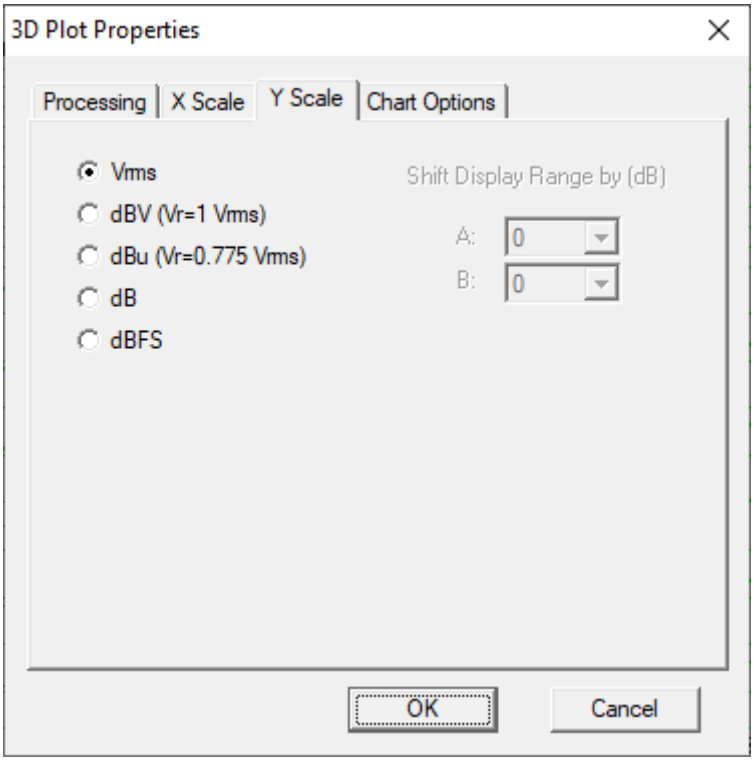
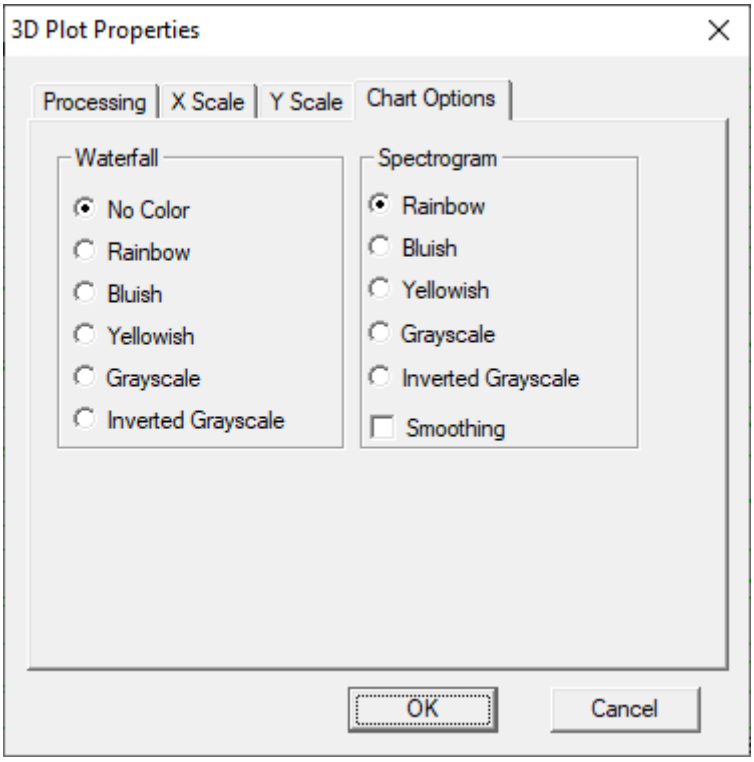


There is only one display mode for the vertical axis (Y): Absolute Mode. All spectral data points are plotted based on their absolute values, in V(rms), dBV, dBu, dB, or dBFS. "Shift Display Range by (dB)" option is enabled when the unit of Y axis is dBV, dBu, dB or dBFS. It is used to modify the display range of Y axis. For example, if the original display range is 120dB~20dB and "-10dB" is selected in this option, then the display range will be changed to 110dB~10dB. In effect, the display is "amplified" by 10dB. This function can be more conveniently manipulated by the axis magnifying glass introduced in the Oscilloscope chapter.

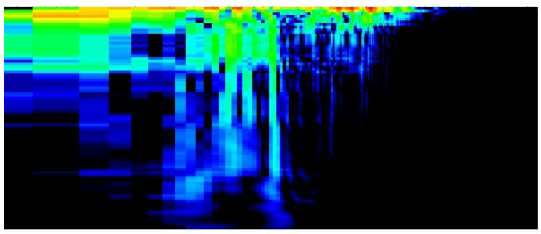
This function can also be accessed by double clicking on Y axis.

7.3.2.4 Spectrum 3D Plot Chart Options (Setting SubMenu) (ALT-S-O)

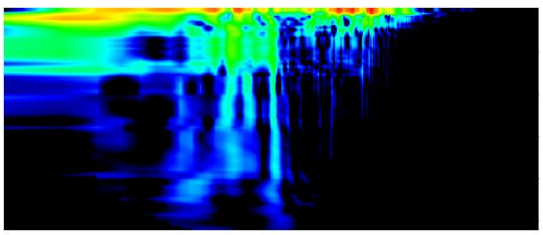


Six color palettes are available for the Waterfall Plot: No Color, Rainbow, Bluish, Yellowish, Grayscale, Inverted Grayscale. Note that except the first option, the rest options may slow down the process a bit, but you can choose to dye the waterfall plot after you stop the data acquisition.

Five color palettes are available for the Spectrogram: Rainbow, Bluish, Yellowish, Grayscale, Inverted Grayscale. The "Smoothing" option can make the spectrogram look more "continuous" in space, as illustrated as follows.



Not Smoothed



Smoothed

7.3.3 Instrument SubMenu

Same as Oscilloscope.

7.3.4 Window SubMenu

Same as Oscilloscope.

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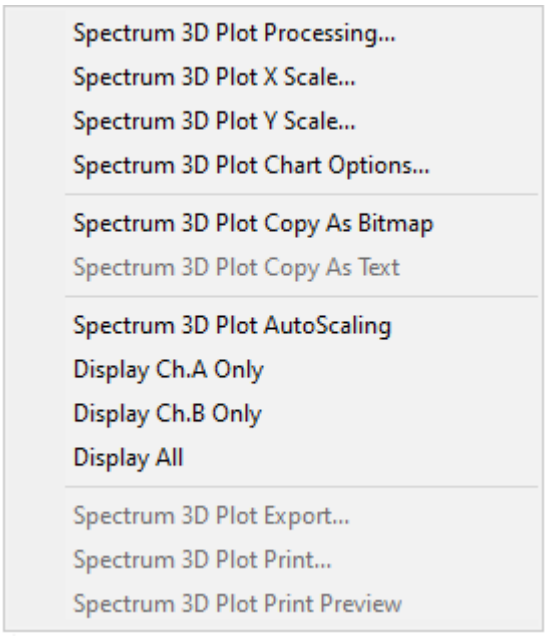
7.3.5 Help SubMenu

Same as Oscilloscope.

7.4 Context Menu

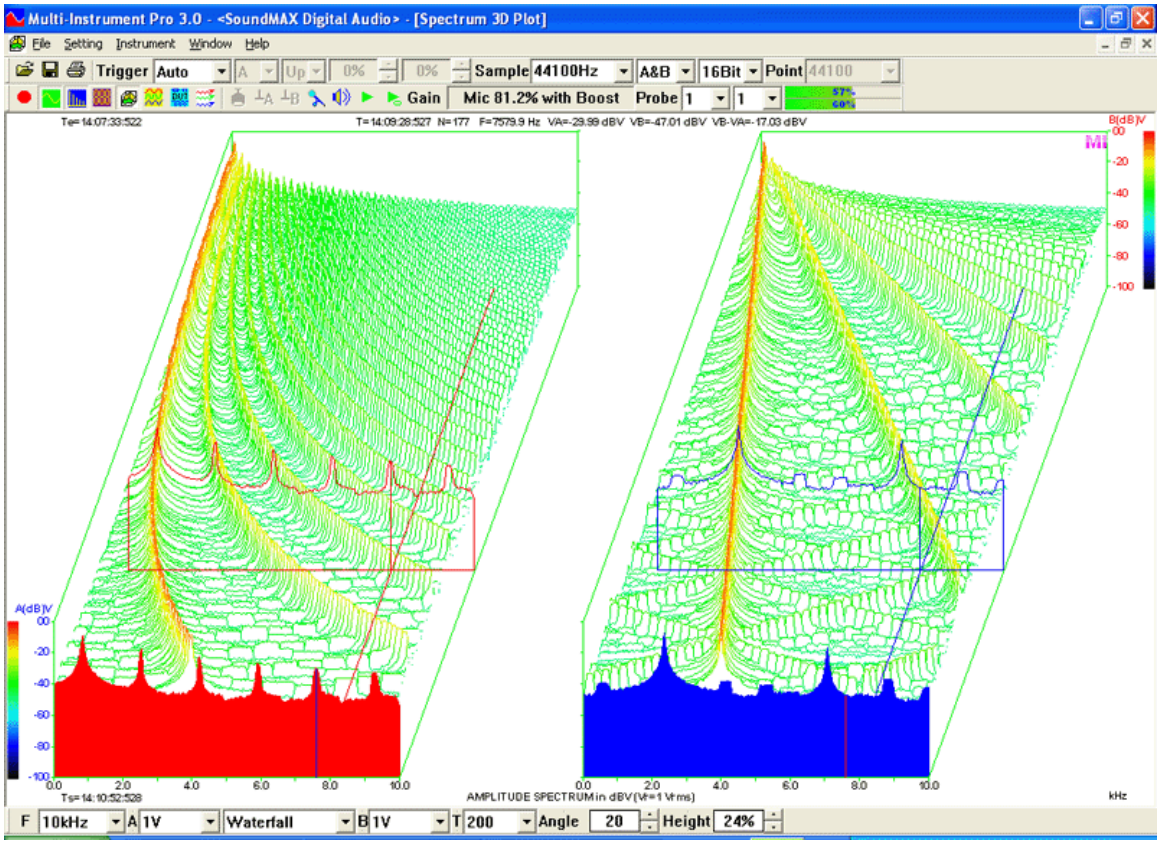


The above context menu will be shown when right clicking anywhere within the Spectrum 3D Plot. It provides additional convenience to you. All menu items in the context menu can also be found in the Spectrum 3D Plot main menu except the following items: Copy As Bitmap, Copy As Text, AutoScaling, Display Ch.A Only, Display Ch.B Only, Display All.

7.5 3D Cursor Reader

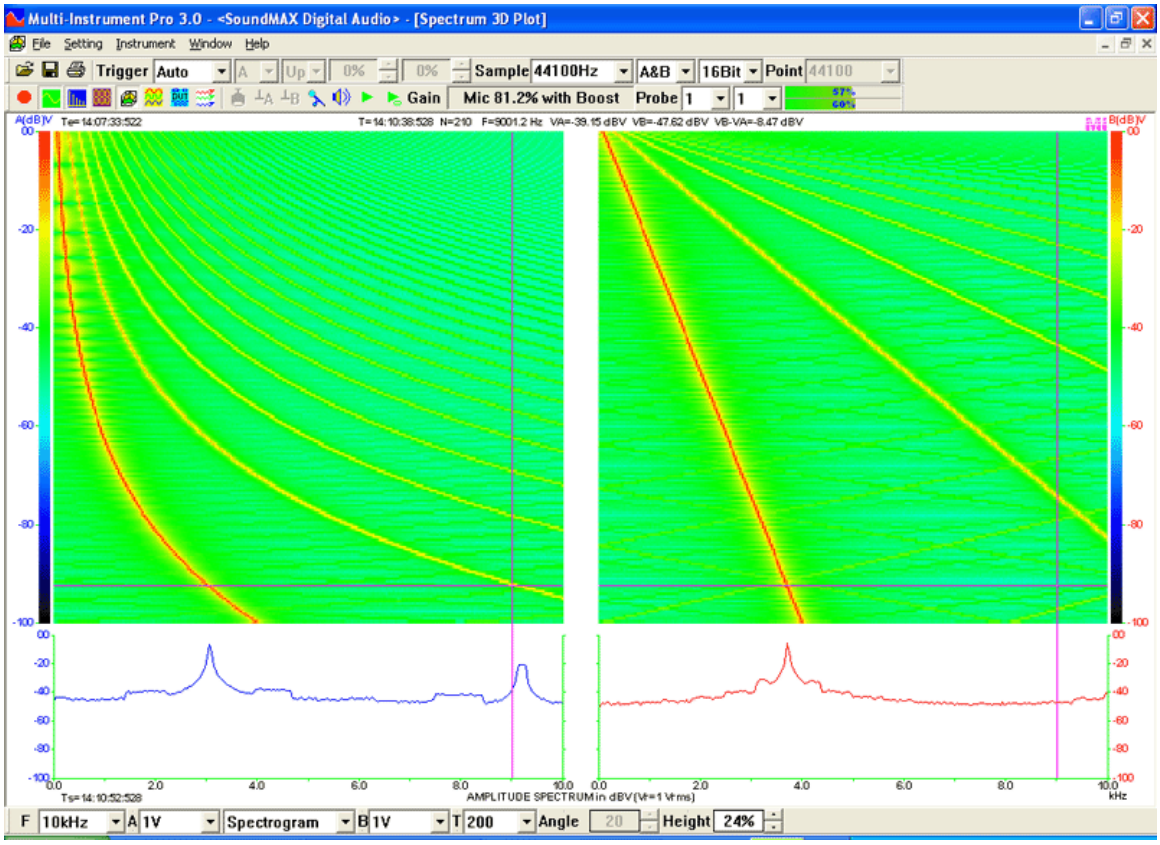
The Waterfall Plot and the Spectrogram have different 3D cursor readers.

7.5.1 3D Cursor Reader for Waterfall Plot



The cursor reader will be shown when you left click anywhere within the plot area of the bottom plane (X-T). The contour of the spectrum profile with a time stamp nearest to the point of mouse click will be highlighted (thus T is determined). This profile will also be projected onto the front plane (X-Y) so that it can be easily viewed against the X and Y scales. Within this profile, the cursor reader will point to the measurement point with a frequency nearest to the point of mouse click (thus X is determined). With T and X determined, Y is thus determined from the measurement results. The cursor reader always appears as a pair, one for each channel. The cursor's (X, Y, T) readings for both channels are displayed on the top the graph. Their difference in Y is also shown.

7.5.2 3D Cursor Reader for Spectrogram



The cursor reader will be shown when you left click anywhere within the Spectrogram (X-T). The spectrum profile with a time stamp nearest to the point of mouse click will be extracted and displayed at the bottom of the Spectrogram (thus T is determined). Within this profile, the cursor reader will point to the measurement point with a frequency nearest to the point of mouse click (thus X is determined). With T and X determined, Y is thus determined from the measurement results. The cursor reader always appears as a pair, one for each channel. The cursor's (X, Y, T) readings for both channels are displayed on the top the graph. Their difference in Y is also shown.

7.6 Time Stamp

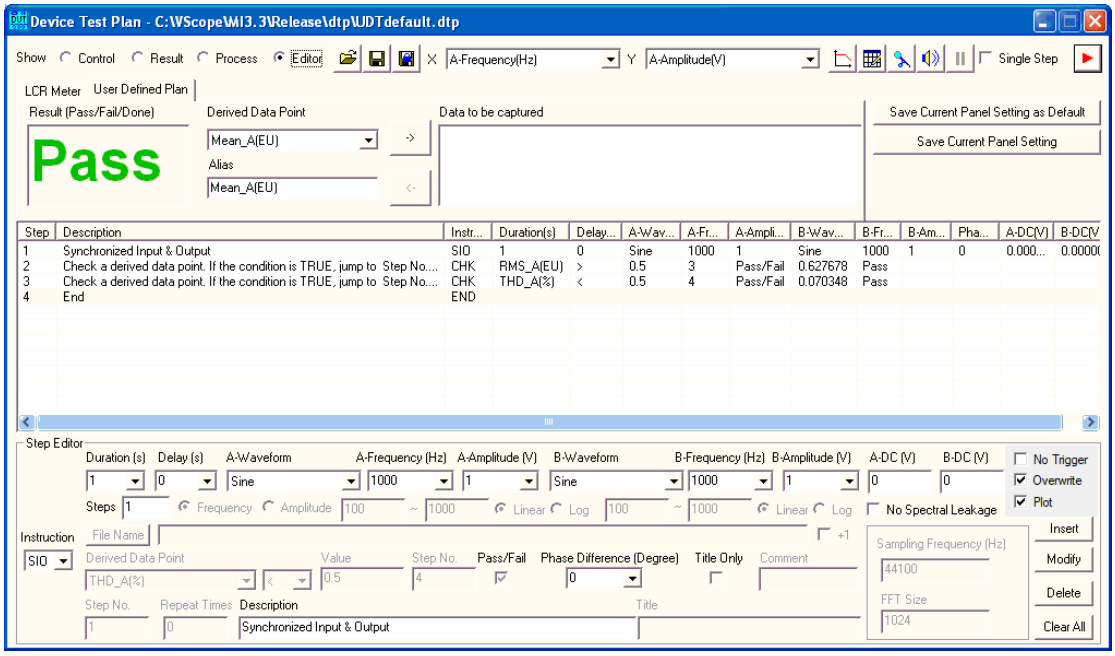
The time stamp of the latest spectral profile is displayed at the bottom of the 3D graph, and the time stamp of the oldest spectral profile is displayed on the top the 3D graph. The time axis labels are marked with a time offset relative to the time of the oldest spectrum profile.

7.7 Ordering of Output Data

When outputting data using either "Export" or "Copy As Text" command, the message "Press YES to slice data by Time, or press NO to slice data by Frequency" will pop up. The former outputs spectrum profiles at particular times one by one, while the latter output time variation profiles at particular frequencies one by one.

8 Device Test Plan

8.1 Overview



Device Test Plan provides a mechanism for you to configure and conduct your own device test steps. It takes the advantage of the sound card's (or other ADC/DAC hardware's) capability of simultaneous input and output, to generate a stimulus to the Device Under Test (DUT) and acquire the response from that device at the same time. Different stimuli can be generated and the response can be analyzed in different ways. The DUT can be marked as Pass or Fail after a sequence of test steps and a test report can be generated. Device Test Plan supports 30 instructions with corresponding parameters. Test results (e.g. Gain vs Frequency, Phase vs Frequency, etc.) can be plotted in up to 8 X-Y plots and reported in one textual log window in real time. Device Test Plan supports connection with external systems through serial communication.

Device Test Plan can also be used to perform other functions such as data file batch processing, batch signal event capturing and storing, etc.

The toggle button in the Instrument Toolbar is used to open or close the Device Test Plan panel. You can also close it by clicking the "Close" button at its upper right corner.

Device Test Plan includes User Defined Plan and Dedicated Test Plan (e.g. LCR Meter). Unlike the User Defined Plan which is fully configurable by the user, the Dedicated Test Plan has some built-in non-configurable algorithm dedicated for testing certain types of devices. Only the User Defined Plan will be described in this chapter.

The screen layout of the Device Test Plan panel is divided into four parts: Control Bar, Result/Option Area, Process Viewer, and Step Editor.

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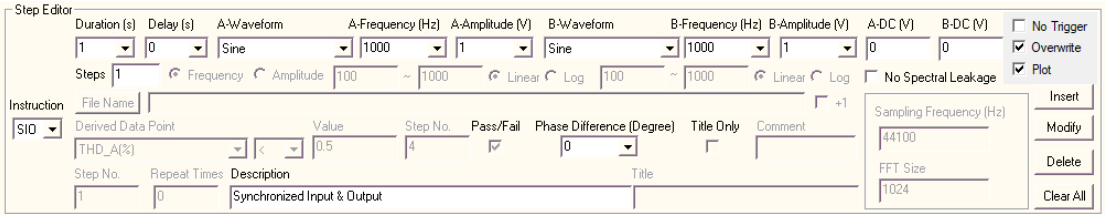
8.2 Step Editor



A device test plan consists of a number of test steps. Step Editor is used to configure each single step. A test step consists of a number of parameters. Instruction is the key parameter in any test step, because it determines what other parameters are required for that step.

8.2.1 Insert a Step

Pressing the "Insert" button will insert a step in the Process Viewer (described later). If no step is selected (highlighted) in the Process Viewer, the new step will be added at the end of the plan, otherwise it will be added just before the selected (highlighted) step.

8.2.2 Modify a Step

Pressing the "Modify" button will overwrite the entire content of the selected (highlighted) step in the Process Viewer (described later). Nothing will happen if no step is selected (highlighted) in the Process Viewer.

8.2.3 Delete a Step

Pressing the "Delete" button will delete the entire content of the selected (highlighted) step in the Process Viewer (described later). Nothing will happen if no step is selected (highlighted) in the Process Viewer.

8.2.4 Clear All

Pressing the "Clear All" button will delete all steps in the current plan.

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8.2.5 Instruction

30 instructions are supported. They are described as follows.

8.2.5.1 SIO

SIO (Synchronize Input with Output) is used to command the Signal Generator to generate a specified stimulus to the DUT and use the DAQ-related instruments (e.g. Oscilloscope, Spectrum Analyzer, Multimeter ……) to acquire and analyze the response from the DUT. Unlike the SOI instruction, which will be described later, the SIO instruction commands the Signal Generator to start in "Loop" output mode first. The Oscilloscope is then commanded to start, and is automatically stopped after data acquisition. Finally, the Signal is commanded to stop.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Synchronize Input with Output". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: [TimeoutStepNo]: specifies Step No. to go to when timeout error occurs. TimeoutStepNo must be surrounded by a pair of "[" and "]". A number without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards

		"(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at. By default, the plan will pop up a message and stop upon timeout error unless this parameter is specified. <TimeoutSeconds>: specifies the timeout seconds. TimeoutSeconds must be surrounded by a pair of "<" and ">". It is in additional to the instruction's duration. By default, 10 seconds will be used, unless this parameter is specified. Example: Synchronized Input with Output {[5]<20>}
Instruction	Instruction	SIO
Duration(s)	Duration(s)	It defines how long the specified signal will last.
Delay(s)	Delay(s)	It defines the idle time between the end of the signal output and the start of the next step.
A-Waveform	A-Waveform	Waveform to be generated in Channel A, available options are: None, Sine, Rectangle, Triangle, Saw Tooth, White Noise, Pink Noise, MultiTones, WFLibrary, MLS. For those parameters that cannot be set on the Step Editor, e.g. the duty cycle for Rectangle waveform and length for MLS, the values set before the opening of the Device Test Plan (if any) or otherwise the default values will be used. Loading a Panel Setting File in the test plan will also update these values. For MultiTones and WFLibrary, you will be prompted to select a file name, and you can also check/uncheck the flag "+1" on the right of the File Name edit box. This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc. This parameter is ignored when the Overwrite checkbox is not selected.
A-Frequency(Hz)	A-Frequency(Hz)	Frequency to be generated in Channel A. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
A-Amplitude(V)	A-Amplitude(V)	Amplitude to be generated in Channel A. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
B-Waveform	B-Waveform	Same as that of Channel A. This parameter is ignored when the Overwrite checkbox is not selected.
B-Frequency(Hz)	B-Frequency(Hz)	Frequency to be generated in Channel B. It can be a real number or variable (x1~x5).

		This parameter is ignored when the Overwrite checkbox is not selected.
B-Amplitude(V)	B-Amplitude(V)	Amplitude to be generated in Channel B. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
Phase Difference(D)	Phase Difference (Degree)	Phase difference between the signals to be generated in Channel A and Channel B. Applicable only if none of the waveforms in Channel A and Channel B is None, White Noise, Pink Noise, MultiTones, WFLibrary, or MLS, and the frequencies in both Channel are equal. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
A-DC(V)	A-DC(V)	DC offset of Channel A. Applicable only to those DAC devices that support DC offset adjustment. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
B-DC(V)	B-DC(V)	DC offset of Channel B. Applicable only to those DAC devices that support DC offset adjustment. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
DDPs to be captured	N.A. They are configured from the Result/Options Area instead.	They define the DDPs to be obtained from the DAQ-related instruments after the response from the DUT is captured. The values of the DDPs will be displayed in these fields at runtime. In case the value is not available due to any reason (e.g. incorrect settings), then "NA" will be filled instead.

If the "No Trigger" checkbox is not selected, the Oscilloscope's Trigger Mode will be forced to Single mode, and its Record Length will be forced to be half of the stimulus signal duration if it is greater than [Duration-0.05] seconds. In other words, the stimulus signal must last at least 50ms longer than the Record Length. The Oscilloscope will be commanded to start after 4/5 of [Duration - Record Length] elapses. This is to compensate the latency between the time when the Signal Generator is commanded to start and the time when the generated signal is received at the input, and also to ensure that the DUT reaches its stable state after being excited by the stimulus. The timing diagram is shown as follows.

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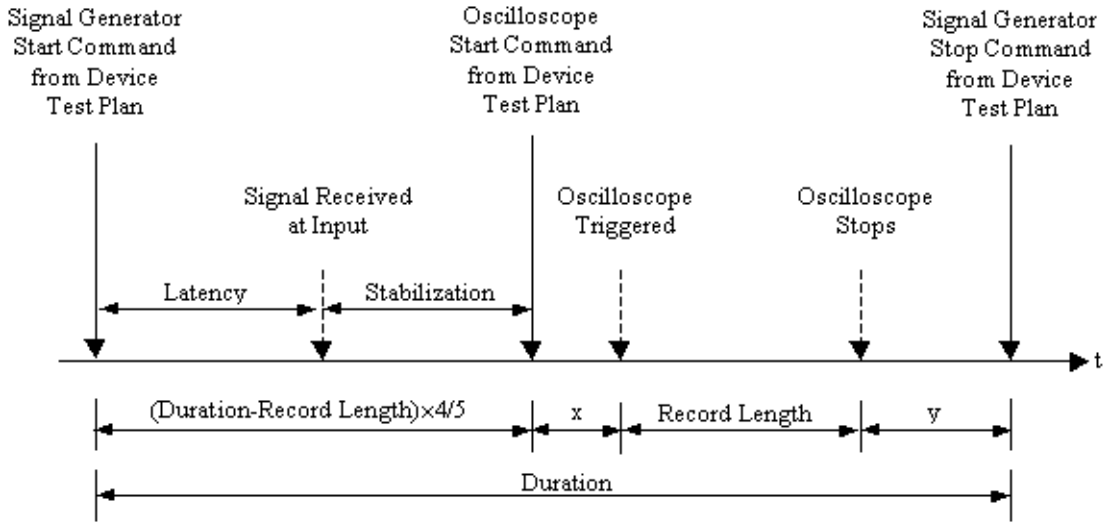

y in the above figure accounts for 1/5 of [Duration – Record Length] while the minimum value of x is zero. If the Oscilloscope does not trigger, x and Duration will be automatically extended. Different DUTs and ADC/DAC devices have varying latencies and stabilization times, so designing an appropriate timing scheme is critical for the SIO instruction.

If the Oscilloscope remains un-triggered for [TimeoutSeconds+2 × RecordLength] after the signal duration elapses due to any reasons, a time out error will be generated and the test plan will be stopped. By default, TimeoutSeconds is set to 10s.

The rest parameters for the DAQ-related instruments will be kept as they were before the opening of the Device Test Plan (if any) or otherwise the system default values will be used. Loading a Panel Setting File in the test plan will also update these values. It is recommended to load a properly configured Panel Setting File in the beginning of each test plan to ensure all settings are in good shape.

When the "Overwrite" checkbox is not selected, the signal generator parameters (waveform, frequency, amplitude, phase difference between the two channels, DC offset) on the Signal Generator panel will be kept as they are and will not be overwritten by the parameters here. This feature is useful if you want to load a Panel Setting File and then execute the SIO command without changing the above parameters.

If the "No Trigger" checkbox is selected, the Oscilloscope's Trigger Mode will be forced to Auto mode. The "No Trigger" option allows the SIO command to measure a DC voltage, which does not contain an edge triggering point. Moreover, it supports inter-frame peak hold or linear average for a specified number of frames in the Spectrum Analyzer. The Oscilloscope will stop automatically after the specified inter-frame processing is done. In these cases, the duration parameter in the SIO command is for one frame only. Under any other situations, only one frame of data will be acquired before the Oscilloscope stops automatically.

If the "Plot" checkbox is unticked, then no "data to be captured" will be logged and displayed in the X-Y Plots. This option helps to differentiate between trial and actual measurements.

8.2.5.2 OUT

OUT (Signal Output) is used to command the Signal Generator to generate a specified signal. The Signal Generator will be set to non-Loop output mode. Unlike SIO, the DAQ will not be started under this instruction.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Output".
Instruction	Instruction	OUT
Duration(s)	Duration(s)	It defines how long the specified signal will last.
Delay(s)	Delay(s)	It defines the idle time between the end of the signal output and the start of the next step.
A-Waveform	A-Waveform	Waveform to be generated in Channel A, available options are: None, Sine, Rectangle, Triangle, Saw Tooth, White Noise, Pink Noise, MultiTones, WFLibrary, MLS. For those parameters that cannot be set on the Step Editor, e.g. the duty cycle for Rectangle waveform and length for MLS, the values set before the opening of the Device Test Plan (if any) or otherwise the default values will be used. Loading a Panel Setting File in the test plan will also update these values. For MultiTones and WFLibrary, you will be prompted to select a file name, and you can also check/uncheck the flag "+1" on the right of the File Name edit box. This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc. This parameter is ignored when the Overwrite checkbox is not selected.
A-Frequency(Hz)	A-Frequency(Hz)	Frequency to be generated in Channel A. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
A-Amplitude(V)	A-Amplitude(V)	Amplitude to be generated in Channel A. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
B-Waveform	B-Waveform	Same as that of Channel A. This parameter is ignored when the Overwrite checkbox is not selected.

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B-Frequency(Hz)	B-Frequency(Hz)	Frequency to be generated in Channel B. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
B-Amplitude(V)	B-Amplitude(V)	Amplitude to be generated in Channel B. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
Phase Difference(D)	Phase Difference (Degree)	Phase difference between the signals to be generated in Channel A and Channel B. Applicable only if none of the waveforms in Channel A and Channel B is None, White Noise, Pink Noise, MultiTones, WFLibrary, or MLS, and the frequencies in both channels are equal. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
A-DC(V)	A-DC(V)	DC offset of Channel A. Applicable only to those DAC devices that support DC offset adjustment. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
B-DC(V)	B-DC(V)	DC offset of Channel B. Applicable only to those DAC devices that support DC offset adjustment. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
DDPs to be captured	N.A.	N.A.

8.2.5.3 STI

STI (Single Triggered Input) is used to command the DAQ to start in Single trigger mode if the "No Trigger" checkbox is not selected, or in Auto trigger mode if that it is selected. The rest parameters for the DAQ-related instruments will be kept as they were before the opening of the Device Test Plan (if any) or otherwise the system default values will be used. Loading a Panel Setting File in the test plan will also update these values.

The step finishes after a frame of data is captured.

This instruction can be used together with other instructions to capture and store a sequence of events automatically.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Single Triggered Input".

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		In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: [TimeoutStepNo]: specifies Step No. to go to when timeout error occurs. TimeoutStepNo must be surrounded by a pair of "[" and "]". A number without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at. By default, this instruction will wait until success, unless this parameter is specified. <TimeoutSeconds>: specifies the timeout seconds. TimeoutSeconds must be surrounded by a pair of "<" and ">". It is in additional to the sampling duration. By default, 10 seconds will be used unless this parameter is specified. Example: Single Triggered Input {[5]<20>}
Instruction	Instruction	STI
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A. They are configured from the Result/Options Area instead.	They define the DDPs to be obtained from the DAQ-related instruments after a frame of data is captured. The values of the DDPs will be displayed in these fields at runtime. In case the value is not available due to any reason (e.g. incorrect settings), then "NA" will be filled instead.

If the "Plot" checkbox is unticked, then no "data to be captured" will be logged and displayed in the X-Y Plots.

8.2.5.4 RPT

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RPT (Repeat) is used to control the execution of steps in a test plan. It can be used to repeat a range of steps for a specified number of times. Note that a repetition loop should not contain or overlap with another repetition loop.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Repeat Step No. x for y times".
Instruction	Instruction	RPT
Duration(s)	Step No.	It specifies the Step No. to jump to. Only backward jump is allowed. A number without a sign indicates an absolute jump while a number with a preceding negative sign indicates a backward relative jump. Step Label can also be used. Examples: "3": jump to Step 3 "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at.
Delay(s)	Repeat Times	It specifies the number of times to repeat (exclusive of the very first time). Note that a zero value means "repeat forever".
A-Waveform	N.A.	This field will be used to indicate the number of times that has been repeated during runtime. It will be cleared after the repetition process finished.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.5 LDP

LDP (Load Panel Setting File) is used to load a pre-stored Panel Setting File, in order to preset the parameters for DAQ and data analysis. When it is selected in the Instruction combo box, a File Open window will pop up requesting for the path and name of the Panel Setting File. It can also be set using the File Name button.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Load Panel Setting".
Instruction	Instruction	LDP
Duration(s)	File Name	It specifies the path and name of the Panel Setting File to be loaded.
Delay(s)	+1	This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly,

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		The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.6 LDF

LDF (Load File) is used to load a time-domain data file, either in WAV format or in TXT format (must has a file extension of txt), for analysis. It is usually used together with the exporting instructions for bath file processing. When it is selected in the Instruction combo box, a File Open window will pop up requesting for the path and name of the file. It can also be set using the File Name button. By default, the current panel setting will be reset after the file is loaded, unless "{[]}" is placed in the Description field.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Load WAV File". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: []: There is no parameter within the pair of "[" and "]". This is to indicate that the file should be loaded with the current panel setting.
Instruction	Instruction	LDF
Duration(s)	File Name	It specifies the path and name of the time-domain data file to be loaded.
Delay(s)	+1	This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

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8.2.5.7 SVF

SVF (Save File) is used to save a WAV file. The file path and name can be specified by using the File Name button and/or the File Name edit box. If the specified file already exists, it will be overwritten without any warning message at runtime.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Save WAV File".
Instruction	Instruction	SVF
Duration(s)	File Name	It specifies the path and name of the WAV file to be saved.
Delay(s)	+1	This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.8 OET

OET (Oscilloscope Export TXT file) is used to export the data in the Oscilloscope to a TXT file. It is the same as the Export command in the Oscilloscope. The file path and name can be specified by using the File Name button and/or the File Name edit box. If the specified file already exists, it will be overwritten without any warning message at runtime.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Oscilloscope Export TXT File".
Instruction	Instruction	OET
Duration(s)	File Name	It specifies the path and name of the TXT file to be exported.
Delay(s)	+1	This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc.

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A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.9 SET

SET (Spectrum Analyzer Export TXT file) is used to export the analysis result in the Spectrum Analyzer to a TXT file. It is the same as the Export command in the Spectrum Analyzer. The file path and name can be specified by using the File Name button and/or the File Name edit box. If the specified file already exists, it will be overwritten without any warning message at runtime.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Spectrum Analyzer Export TXT File".
Instruction	Instruction	SET
Duration(s)	File Name	It specifies the path and name of the TXT file to be exported.
Delay(s)	+1	This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.10 DLY

DLY (Delay) is used to pause the execution of the plan for a specified number of seconds.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Delay x Second (s)".
Instruction	Instruction	DLY
Duration(s)	Duration(s)	It specifies the number of seconds to wait before executing the next step. The value must be a

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		multiple of 0.1 second.
Delay(s)	N.A.	This field will be used to indicate the number of seconds that has been delayed during runtime. It will be cleared after the step is finished.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.11 JMP

JMP (Jump) is used to redirect the execution of the plan from the current step to a specified step.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Jump to Step No. x".
Instruction	Instruction	JMP
Duration(s)	Step No.	It specifies the Step No. to jump to. A number without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.12 CHK

CHK (Check) is used to evaluate the value of a specified Derived Data Point (refer to the relevant section in the chapter for Data Logger) against a preset value. If the specified condition is met, then jump to the specified Step, otherwise continue to execute the next step.

Title in Process Viewer	Title in Step Editor	Description

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Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Check a derived data point. If the condition is TRUE, jump to Step No. x".
Instruction	Instruction	CHK
Duration(s)	Derived Data Point	It specifies the Derived Data Point to be evaluated with the following exception: 1. TimeStamp, current date and time (s). 2. UserInput, latest text input by the user using IPT instruction. 3. UserName, current user priority 4. TestResult, current test result: DONE=0, PASS=1, FAIL=-1 5. VolatileInt1~VolatileInt5 and NonVolatileInt1~NonVolatileInt5 6. ReceivedBytes_Array[?], an indexed byte in the received bytes by RCM instruction. "?" should be replaced by the actual array index starting from 0. 7. x1~x5
Delay(s)	Comparison Operator: >, >=, <, <=, ==, !=	>: greater than >=: greater than or equal to <: less than <=: less than or equal to ==: equal to !=: not equal to
A-Waveform	Value	It specifies the value to be compared with
A-Frequency(Hz)	Step No.	It specifies the Step No. to jump to if the condition is met. A number without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at.
A-Amplitude(V)	Pass/Fail	If ticked, a Pass/Fail check will be performed.
B-Waveform	N.A.	The actual value of the specified Derived Data Point will be shown in this field at runtime.
B-Frequency(Hz)	N.A.	"Pass" will be indicated in this field if the specified condition is true, Otherwise "Fail" will be indicated.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

If the Derived Data Point is not available due to any reason (e.g. the settings are incorrect, etc.), the step will fail.

8.2.5.13 LOG

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LOG is used to generate one textual record in the Device Test Plan Log window or a specified text file.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Log the Title, Derived Data Point Value, Comment". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: [FileName]: specifies the path and name of the log file. Both an absolute path, e.g. C:\log\1.log, and a path relative to the software's root directory, e.g. log\1.log, are supported. File Name must be surrounded by a pair of "[" and "]". By default, the Device Test Plan Log window will be used for output, unless this parameter is specified. The following formats are supported: (1)[FileName]: a fixed file name. FileName can also be "Panel1", "Panel2", or "Panel3" which represents a textual display area in the Result window on the panel. (2)[FileName, Date and Time Format, xxx] or [FileName, Date and Time Format]: a file name that changes automatically with date and time. Date and Time Format supported are: "YYYY-MM-DD-HH", "YYYY-MM-DD", "YYYY-MM" and "YYYY". The date and time will be automatically added at the end of the FileName but before "." if any. New files will be automatically created as time progresses. If the new file does not contain any header lines, then xxx should be "0" which means that the new file can start from the current step. If the new file contains 1 header line, then only for the header line, xxx should be "1"; for non-header lines, xxx should not be specified. If the new file contains N header lines, for header line 1, xxx should be "1-N"; for header line 2, xxx should be "2-N"; for header line N, xxx should be "N-N"; for non-header lines, xxx should not be specified. If the new file contains 1 header line and it also shows up every N lines, then xxx should be "1/N". If a non-header line needs to show up every N lines, then xxx should be "/N". The above mechanism ensures that every new file automatically created according to date and time starts from a correct position automatically in a test plan.

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		(3)[FileName, UserInput] or [FileName, UserInput, xxx]: a file name that changes with UserInput. UserInput must be "UserInput". xxx can be used if it shows up every N lines. In this case, xxx should be "/N". Header lines should not be defined through xxx as the test plan should know which step to create a new file. (4)[FileName, UserName] or [FileName, UserName, xxx]: a file name that changes with UserName. UserName must be "UserName". xxx can be used if it shows up every N lines. In this case, xxx should be "/N". Header lines should not be defined through xxx as the test plan should know which step to create a new file. (5) [FileName, (any combination of Date and Time Format, UserInput and UserName)] or [FileName, (any combination of Date and Time Format, UserInput and UserName), xxx], e.g. [C:\log\1.log, UserName-UserInput-YYYY-MM,1-3] (Date and Time Format): Year:YYYY or YY; Month: MM; Day: DD; Hour:HH; Minute: mm; Second: SS. Date and Time format must be surrounded by a pair of "(" and ")". By default, the Date and Time Format is "YYYY-MM-DD HH:mm:SS", unless this parameter is specified. <Record Separator>: specifies the separator between two consecutive records. Record Separator is surrounded by a pair of "<" and ">". By default, "Carriage Return" is used, unless this parameter is specified. 'RedValue, GreenValue, BlueValue': specifies the RGB values for the color of the texts when the target of this command (i.e. FileName) is "Panel1", "Panel2", or "Panel3". RedValue, GreenValue, BlueValue must be surrounded by a pair of "" and "" Each value ranges from 0 to 255. For example, '0,0,0' represents black. '255,0,0' represents red. |CodePage|: specifies the code page of the texts. Its value ranges from 1~7. 1-ANSI, 2-Macintosh, 3-OEM, 4-Symbol, 5-Current Thread's ANSI, 6-UTF-7, 7-Unicode. CodePage must be surrounded by a pair of "|" and "|". By default, UTF-8 is used, , unless this parameter is specified. /EngineeringUnit/: specifies the engineering unit of the DDP to be logged. EngineeringUnit must be surrounded by a pair of "/" and "/". By default, it is assigned by the software automatically, unless this parameter is specified. Use "//" if you do not want to log the unit.

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		\DDP Separator\: indicates that all the specified DDPs under "Data to be captured" will be logged. It also specifies the separator between two consecutive DDPs. DDP Separator is surrounded by a pair of "\" and "\". If DDP Separator is NULL (i.e. "\\"), "Carriage Return" will be used. The single DDP specified in the Step editor will be ignored. Example: Log the Title, Derived Data Point Value, Comment {[C:\log\1.log](YYYYMMDDHHmmSS)< >} If the file does not exist, a new file will be created with new content. If the file exists, new content will be appended at the end of the existing file.
Instruction	Instruction	LOG
Duration(s)	Title	It is just a textual description, usually the title of the DDP.
Delay(s)	Derived Data Point	It specifies the Derived Data Point whose value to be logged with the following exception: 1. TimeStamp, current date and time. 2. UserInput, latest text input by the user using IPT instruction. 3. UserName, current user name 4. TestResult, current test result: DONE, PASS, FAIL 5. VolatileInt1~VolatileInt5 and NonVolatileInt1~NonVolatileInt5 6. x1~x5
A-Waveform	Comment	Any comment can be put here.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.14 END

END (End) is used to mark the end of a plan. It is especially useful when you want to conditionally end the plan in the middle of steps.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "End".
Instruction	Instruction	END
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.

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B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.15 IPT

IPT (Input) is used to get the user input through a pop-up dialog box containing only one text input field. The dialog box will be closed automatically, if the number of characters entered is within the range specified, upon receiving a Carriage Return. The input text can be accessed through the extended DDP point "UserInput". It is especially useful when you want to read a DUT's barcode with a barcode reader.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Get user input".
Instruction	Instruction	IPT
Duration(s)	From (A)	Minimum number of characters
Delay(s)	To (A)	Maximum number of characters
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.16 SEL

SEL (Selection) is used to get the user selection, Yes or No, through a pop-up dialog box. If Yes is selected, then jump to the specified Step, otherwise continue to execute the next step.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Popup for user selection, Y or N. If Y, jump to Step No. x". The description will be shown in the popup dialog.
Instruction	Instruction	SEL
Duration(s)	Step No.	It specifies the Step No. to jump to if Yes is selected. A number without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at

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Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.17 OPT

OPT (Operation) is used to perform mathematic operation on 5 volatile integer variables (VolatileInt1~VolatileInt5), 5 non-volatile integer variables (NonVolatileInt1~NonVolatileInt5) and 5 volatile floating-point variables (x1~x5). These variables can be used for various purposes such as controlling a loop, counting PASS and FAIL, varying frequencies, amplitudes and phases, etc. VolatileInt1~VolatileInt5 are volatile integers. They are initialized to zero at the start of a test plan. NonVolatileInt1~NonVolatileInt5 are non-volatile integers. They are initialized through a CSV text file named "NonVolatile.txt" under the "\dtp" directory. If the file does not exist, then they will be initialized to zero in the beginning. After that, any change of these non-volatile integers will be persisted in the file "NonValatile.txt" and only the Reset button can be used to manually reset them to zeros.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Volatile/Non-Volatile variable +, -, *, /, or = x".
Instruction	Instruction	OPT
Duration(s)	Derived Data Point	Only volatile integers: VolatileInt1~ VolatileInt5, non-volatile integers: NonVolatileInt1~ NonVolatileInt5, and volatile floating-point variables: x1~x5 are allowed
Delay(s)	Value	Five operations are allowed: (1) +x (2) –x (7) *x (8) /x (9) =x where x is an integer or floating-point number. x can also be a DDP or extended DDP point, e.g [RMS_A(EU)]
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.18 CLR

CLR (Clear) is used to clear the intermediate test results such as PASS and FAIL, captured DDP values, etc., in the Process Viewer or Device Test Plan Log window. After this command, the Process Viewer will resume to the status when a plan is just loaded, and the Device Test Plan Log window will be cleared. The content in the window can be stored in a file on the hard disk before it is cleared.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Clear test results". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: [Target]: specifies the target on which the clear command is performed. If Target ="Log", then the texts in the Device Test Plan Log window will be cleared. By default, the process viewer is the target and the test results in it will be cleared, unless this parameter is specified. (FileName): specifies the path and name of the file in which the texts in the Device Test Plan Log window will be appended. Both an absoute path, e.g. C:\log\1.log, and a path relative to the software's root directory, e.g. log\1.log, are supported. File Name must be surrounded by a pair of "(" and ")". The above two options can be used together to store the texts in the Device Test Plan Log window into a file on the hard disk and then clear the window.
Instruction	Instruction	CLR
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.19 OEB

OEB (Oscilloscope Export BMP file) is used to export the graph in the Oscilloscope to a BMP file. It is the same as the Export command in the Oscilloscope. The file path and name can be specified by using the File Name button and/or the File Name edit

box. If the specified file already exists, it will be overwritten without any warning message at runtime.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Oscilloscope Export BMP File". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: (GraphToExport): specifies the graph to export. GraphToExport must be surrounded by a pair of "(" and ")". GraphToExport: 0: Screen occupied by Multi-Instrument Example: {(0)}
Instruction	Instruction	OEB
Duration(s)	File Name	It specifies the path and name of the BMP file to be exported.
Delay(s)	+1	This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.20 SEB

SEB (Spectrum Analyzer Export BMP file) is used to export the graph in the Spectrum Analyzer to a BMP file. It is the same as the Export command in the Spectrum Analyzer. The file path and name can be specified by using the File Name button and/or the File Name edit box. If the specified file already exists, it will be overwritten without any warning message at runtime.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Spectrum Analyzer Export BMP File".

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		In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: (GraphToExport): specifies the graph to export. GraphToExport must be surrounded by a pair of "(" and ")". GraphToExport: 0: Screen occupied by Multi-Instrument Example: {(0)}
Instruction	Instruction	SEB
Duration(s)	File Name	It specifies the path and name of the BMP file to be exported.
Delay(s)	+1	This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.21 RCM

RCM (Read Communication port) is used to open, configure and read data from a communication port. If the command contains configuration parameters and the communication port has not been opened yet, then the port will be opened and configured first, otherwise the command will read data from the port only. Once the port is opened, it will be closed only when the test plan is stopped or errors occur. The receiving buffer of the communication port will be purged first each time the command is executed. If the command is successful, the received data can be accessed through the extended DDP point "ReceivedBytes_Array[?]".

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Read Communication Port". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character

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		pair. The supported optional parameters are: (Port No., Baud Rate, Stop Bits, Parity, Byte Size): contains the communication port configuration parameters. They must be surrounded by a pair of "(" and ")" and the parameters must be separated by ",". Stop Bits: 1,5,2 represent 1, 1.5 and 2 stop bits respectively. Parity: 0~4 represent no, odd, even, mark, space parity respectively. <BytesToReceive>: specifies the number of bytes to receive. BytesToReceive must be surrounded by a pair of "<" and ">". This is a compulsory parameter. |TimeoutMiniSeconds, TimeoutStepNo|: specifies Timeout in ms and Step No. to go to when timeout error occurs. These two parameters must be surrounded by a pair of "|"and "|" and separated by ",". TimeoutStepNo: A number without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at. 'CheckSum': specify the CheckSum type. "CRC" represents CRC-16. If the checksum of the received message is incorrect, the test plan will jump to TimeoutStepNo specified in the Timeout parameter session. CheckSum must be surrounded by a pair of "" and "". Example: Read Communication Port {(3,19200,1,0,8)<1>}
Instruction	Instruction	RCM
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

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8.2.5.22 WCM

WCM (Write Communication port) is used to open, configure and write data to a communication port. If the command contains configuration parameters and the communication port has not been opened yet, then the port will be opened and configured first, otherwise the command will write data to the port only. Once the port is opened, it will be closed only when the test plan is stopped or errors occur. The transmitting buffer of the communication port will be purged first each time the command is executed.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Write Communication Port". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: (Port No., Baud Rate, Stop Bits, Parity, Byte Size): contains the communication port configuration parameters. They must be surrounded by a pair of "(" and ")" and the parameters must be separated by ",". Stop Bits: 1,5,2 represent 1, 1.5 and 2 stop bits respectively. Parity: 0~4 represent no, odd, even, mark, space parity respectively. [BytesToSend]: specifies the bytes to send. Each byte is represented by two HEX characters. Multiple bytes are separated by empty spaces. BytesToSend must be surrounded by a pair of "[" and "]". This is a compulsory parameter. 'CheckSum': specify the CheckSum type. "CRC" represents CRC-16. If this parameter is specified, the checksum byte(s) will be calculated and automatically appended at the end of the BytesToSend. CheckSum must be surrounded by a pair of "" and "". Example: Write Communication Port {(3,19200,1,0,8)[01 F1 A0]}
Instruction	Instruction	WCM
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.

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B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.23 PRT

PRT (Print) is used to print screen, or the graph in the Oscilloscope or Spectrum Analyzer.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Print". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: (GraphToPrint): specifies the graph to print. GraphToPrint must be surrounded by a pair of "(" and ")". GraphToPrint: 0: (Default) Screen occupied by Multi-Instrument 1: Oscilloscope 2: Spectrum analyzer Example: {(1)}
Instruction	Instruction	PRT
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.24 SIR

SIR (Set Input Range) is used to set the input range of the ADC device.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Set Input Range".
Instruction	Instruction	SIR
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.

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A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	A-Amplitude(V)	Input Range to be set in Channel A. It can be a real number or variable (x1~x5).
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	B-Amplitude(V)	Input Range to be set in Channel B. It can be a real number or variable (x1~x5).
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.25 SOR

SOR (Set Output Range) is used to set the output range of the DAC device.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Set Output Range".
Instruction	Instruction	SOR
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	A-Amplitude(V)	Output Range to be set in Channel A. It can be a real number or variable (x1~x5).
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	B-Amplitude(V)	Output Range to be set in Channel B. It can be a real number or variable (x1~x5).
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.26 RMD

RMD (Read MIDI device) is used to open and read data from a MIDI input device. If the MIDI input device has not been opened yet, then it will be opened first, otherwise the command will read data from the MIDI device only. Once the MIDI device is opened, it will be closed only when the test plan is stopped or errors occur. Only the latest MIDI message will be kept in the receiving buffer. If the command is successful, the received data can be accessed through the extended DDP point "ReceivedBytes_Array[?]". All types of MIDI messages, including System Exclusive (SysEx) messages, are supported.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Read MIDI Port". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also

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		be surrounded by its own identification character pair. The supported optional parameters are: (Device No.): contains the MIDI input device number. It must be surrounded by a pair of "(" and ")". |TimeoutMiniSeconds, TimeoutStepNo|: specifies Timeout in ms and Step No. to go to when timeout error occurs. These two parameters must be surrounded by a pair of "|"and "|" and separated by ",". TimeoutStepNo: A number without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at. Example: Read MIDI Port {(0)|300,-1|}
Instruction	Instruction	RMD
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.27 WMD

WMD (Write MIDI device) is used to open a MIDI output device, write data to it, and then close it. All types of MIDI messages, including System Exclusive (SysEx) messages, are supported.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Write MIDI Port". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are:

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		(Device No.): contains the MIDI output device number. It must be surrounded by a pair of "(" and ")". [BytesToSend]: specifies the bytes to send. Each byte is represented by two HEX characters. Multiple bytes are separated by empty spaces. BytesToSend must be surrounded by a pair of "[" and "]". This is a compulsory parameter. Example: Write MIDI Port {(0)[90 3C 40]}
Instruction	Instruction	WMD
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.28 SUB

SUB (Subroutine) is used to redirect the execution of the plan from the current step to the start step of a subroutine. In this regard, it functions similarly to the JMP instruction. A subroutine consists of a block of steps, which must be placed after the main routine and terminated with a RTN (Return) instruction, as described in the next section. It is possible to nest subroutines up to 100 levels deep.

Step	Instruction / Description
1	Start of Main Routine
…	…
…	SUB(L+1)
…	…
L	END
L+1	Start of Subroutine
…	…
…	SUB(M+1)
…	…
M	RTN
M+1	Start of Sub Subroutine
…	…
N	RTN

Nesting of Subroutines

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Call a Subroutine at Step No. x".
Instruction	Instruction	SUB
Duration(s)	Step No.	It specifies the Step No. to jump to. A number

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		without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.29 RTN

RTN (Return) is used to mark the end of a subroutine. It must be paired with a corresponding SUB instruction so that the software knows where to return after the subroutine completes.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Return from a Subroutine".
Instruction	Instruction	RTN
Duration(s)	N.A.	N.A.
Delay(s)	N.A.	N.A.
A-Waveform	N.A.	N.A.
A-Frequency(Hz)	N.A.	N.A.
A-Amplitude(V)	N.A.	N.A.
B-Waveform	N.A.	N.A.
B-Frequency(Hz)	N.A.	N.A.
B-Amplitude(V)	N.A.	N.A.
Phase Difference(D)	N.A.	N.A.
DDPs to be captured	N.A.	N.A.

8.2.5.30 SOI

SOI (Synchronize Output with Input) is used to command the Signal Generator to generate a specified stimulus to the DUT and use the DAQ-related instruments (e.g. Oscilloscope, Spectrum Analyzer, Multimeter ……) to acquire and analyze the response from the DUT. Unlike the SIO instruction described previously, the SOI instruction commands the Oscilloscope to start in Single Trigger Mode first, after about 30ms's delay, the Signal Generator is commanded to start in non-loop output mode (i.e. single-shot mode). The Signal Generator stops automatically after signal

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generation. The Oscilloscope stops automatically after a frame of data is captured. Unlike the SIO instruction, the generated signal length can be precisely controlled and it can be very short (e.g. less than 50ms). The trigger parameters of the Oscilloscope MUST be configured so that it is triggered only by the generated signal, not by background noise. The trigger delay and oscilloscope frame width can be adjusted to select the portion of the generated signal to capture.

Title in Process Viewer	Title in Step Editor	Description
Step	N.A.	Step number is assigned automatically.
Description	Description	You can enter a description that best describes this step. The default description is "Synchronize Output with Input". In addition to the description, it is possible to add optional parameters for this instruction. All these parameters should be put within a single pair of "{" and "}". Each parameter should also be surrounded by its own identification character pair. The supported optional parameters are: [TimeoutStepNo]: specifies Step No. to go to when timeout error occurs. TimeoutStepNo must be surrounded by a pair of "[" and "]". A number without a sign indicates an absolute jump, while a number with a preceding negative or positive sign indicates a backward or forward relative jump respectively. Step Label can also be used. Examples: "3": jump to Step 3 "+3": jump 3 steps forwards "-3": jump 3 steps backwards "(abc)": jump to Step Label "abc" regardless of which Step No. "abc" is at. By default, the plan will pop up a message and stop upon timeout error unless this parameter is specified. <TimeoutSeconds>: specifies the timeout seconds. TimeoutSeconds must be surrounded by a pair of "<" and ">". It is in additional to the instruction's duration. By default, 10 seconds will be used, unless this parameter is specified. Example: Synchronized Input with Output {[5]<20>}
Instruction	Instruction	SOI
Duration(s)	Duration(s)	It defines how long the specified signal will last.
Delay(s)	Delay(s)	It defines the idle time between the end of the operation and the start of the next step.
A-Waveform	A-Waveform	Waveform to be generated in Channel A, available options are: None, Sine, Rectangle, Triangle, Saw Tooth, White Noise, Pink Noise, MultiTones, WFLibrary, MLS. For those parameters that cannot be set on the Step Editor,

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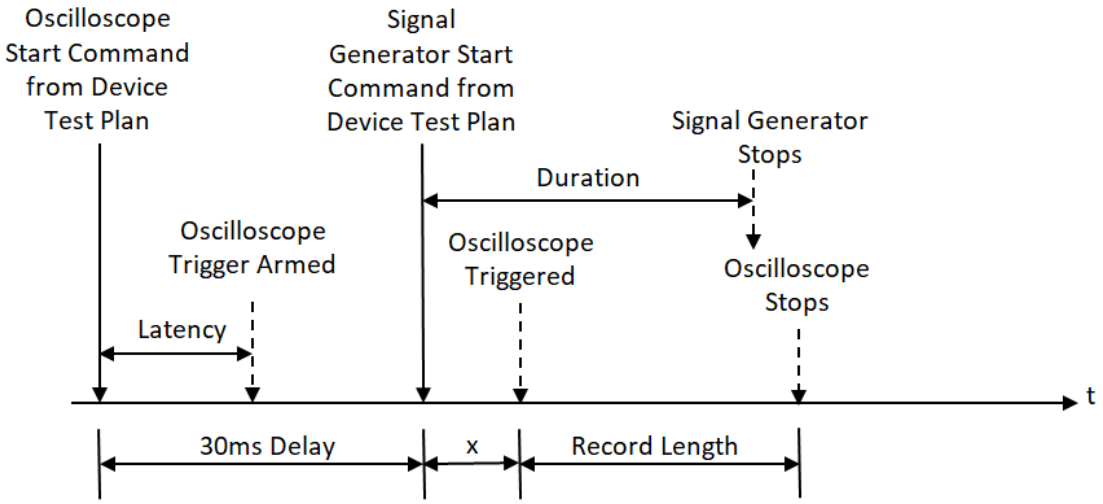
		e.g. the duty cycle for Rectangle waveform and length for MLS, the values set before the opening of the Device Test Plan (if any) or otherwise the default values will be used. Loading a Panel Setting File in the test plan will also update these values. For MultiTones and WFLibrary, you will be prompted to select a file name, and you can also check/uncheck the flag "+1" on the right of the File Name edit box. This flag takes effect only when this step is within a repetition loop. The file name will "plus" one automatically in every repetition during runtime. To make it function correctly, The file name should have at least one numerical digit at the end before the file extension, e.g. xxx0.xxx, xxx888.xxx, etc. This parameter is ignored when the Overwrite checkbox is not selected.
A-Frequency(Hz)	A-Frequency(Hz)	Frequency to be generated in Channel A. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
A-Amplitude(V)	A-Amplitude(V)	Amplitude to be generated in Channel A. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
B-Waveform	B-Waveform	Same as that of Channel A. This parameter is ignored when the Overwrite checkbox is not selected.
B-Frequency(Hz)	B-Frequency(Hz)	Frequency to be generated in Channel B. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
B-Amplitude(V)	B-Amplitude(V)	Amplitude to be generated in Channel B. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
Phase Difference(D)	Phase Difference (Degree)	Phase difference between the signals to be generated in Channel A and Channel B. Applicable only if none of the waveforms in Channel A and Channel B is None, White Noise, Pink Noise, MultiTones, WFLibrary, or MLS, and the frequencies in both Channel are equal. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
A-DC(V)	A-DC(V)	DC offset of Channel A. Applicable only to those DAC devices that support DC offset adjustment. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite

		checkbox is not selected.
B-DC(V)	B-DC(V)	DC offset of Channel B. Applicable only to those DAC devices that support DC offset adjustment. It can be a real number or variable (x1~x5). This parameter is ignored when the Overwrite checkbox is not selected.
DDPs to be captured	N.A. They are configured from the Result/Options Area instead.	They define the DDPs to be obtained from the DAQ-related instruments after the response from the DUT is captured. The values of the DDPs will be displayed in these fields at runtime. In case the value is not available due to any reason (e.g. incorrect settings), then "NA" will be filled instead.

The timing diagram is shown as follows.



The minimum value of x is about zero. If the Oscilloscope does not trigger, x will be automatically extended, while the duration remains unchanged.

If the Oscilloscope remains un-triggered for [TimeoutSeconds+2 × RecordLength] after the signal duration elapses due to any reasons, a time out error will be generated and the test plan will be stopped. By default, TimeoutSeconds is set to 10s.

The rest parameters for the DAQ-related instruments will be kept as they were before the opening of the Device Test Plan (if any) or otherwise the system default values will be used. Loading a Panel Setting File in the test plan will also update these values. It is recommended to load a properly configured Panel Setting File in the beginning of each test plan to ensure all settings are in good shape.

When the "Overwrite" checkbox is not selected, the signal generator parameters (waveform, frequency, amplitude, phase difference between the two channels, DC offset) on the Signal Generator panel will be kept as they are and will not be overwritten by the parameters here. This feature is useful if you want to load a Panel Setting File and then execute the SOI command without changing the above parameters.

If the "Plot" checkbox is unticked, then no "data to be captured" will be logged and displayed in the X-Y Plots. This option helps to differentiate between trial and actual measurements.

8.2.6 Multi-Step Generation

For the SIO, SOI and OUT instructions, the Step Editor provides a way to generate multiple steps in one shot if frequency or amplitude is the only parameter that varies among them. You need to enter the number of steps to be generated first in the "Steps" field, and then choose either frequency or amplitude as the variable, specify its varying range and incrementing method (linear or logarithmic) for each channel, press the "Insert" button in the end.

8.2.7 No Spectral Leakage

If this option is selected, you can further specify the Sampling Frequency of the Oscilloscope and the FFT size of the Spectrum Analyzer. The only use of these two parameters are to fine tune the output signal frequency to the nearest integer multiple of [Sampling Frequency of the Oscilloscope] / [FFT Size of the Spectrum Analyzer], in order to avoid the spectrum leakage issue. For example, if the signal frequency you specify is 1 kHz, the Sampling Frequency of the Oscilloscope is 48 kHz, and the FFT Size of the Spectrum Analyzer is 16384, then the signal frequency will be changed to 999.0234375 Hz. Combing this option with the Multi-Step Generation function would be very useful if you want to design the test frequencies for THD vs Frequency plot.

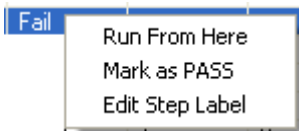
8.3 Process Viewer

Step	Descript...	Instructi...	Duratio...	Delay(s)	A-Wave...	A-Frequ...	A-Amplit...	B-Wave...	B-Frequ...	B-Amplit...	Pha...	A-RMS	A-
1	Load P...	LDP	psf\TH...										
2	Synchro...	SIO	1	0	Sine	50	1000	Sine	50	1000	0		
3	Synchro...	SIO	1	0	Sine	100	1000	Sine	100	1000	0		
4	Synchro...	SIO	1	0	Sine	200	1000	Sine	200	1000	0		
5	Synchro...	SIO	1	0	Sine	500	1000	Sine	500	1000	0		
6	Synchro...	SIO	1	0	Sine	1000	1000	Sine	1000	1000	0		
7	Synchro...	SIO	1	0	Sine	2000	1000	Sine	2000	1000	0		
8	Synchro...	SIO	1	0	Sine	5000	1000	Sine	5000	1000	0		
9	Synchro...	SIO	1	0	Sine	10000	1000	Sine	10000	1000	0		
10	Synchro...	SIO	1	0	Sine	20000	1000	Sine	20000	1000	0		
11	Repeat ...	RPT	2	0									

Process Viewer is used to view a test plan during its editing and running process. The first 12 columns are fixed and the rest of columns are used for the Derived Data Points to be captured, which are configurable and only applicable to the SIO, SOI and STI instructions. From the 4th column to the 12th column, each column may have different meanings for different instructions, and the displayed titles of these columns are only applicable to the SIO and SOI instructions. Please refer to the instructions described previously for the actual meanings.

During the execution of a test plan, the current running step will be highlighted.

8.3.1 Context Menu



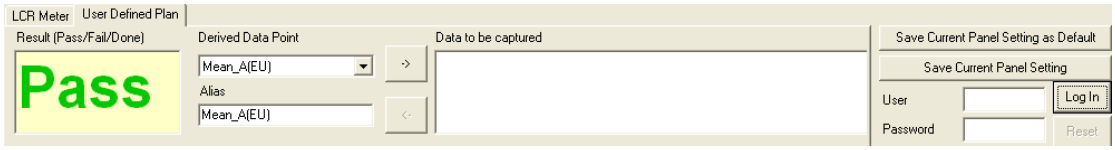
Process Viewer has its own context menu where additional functions can be accessed. The context menu pops up when you right click a step in the Process Viewer if the Device Test Plan is not running. There are three menu items in the context menu.

- Run From Here
  Unlike the Run/Stop button for plan execution which always starts from Step 1, this option allows you to start from any step chosen.

- Mark as PASS
  This option is available only if the step has already been marked as FAIL, e.g. as a result of a CHK instruction. You can overwrite the automated test result and mark it as PASS according to your own judgement using this command.

- Edit Step Label
  This option allows you to add, edit or remove Step Label. Step Label is used to label a particular step. Unlike Step No., it does not change when you add or delete a step in the test plan. It can be used in those instructions that have a Step No. as its parameter, such as CHK, JMP, and RPT. A dialog window with an edit box for input will pop up once this option is selected. You can then enter a label consisting of any characters. Each Step Label must be unique in a test plan. The Step Label will be automatically bracketed and put right behind the Step No. You do not need to enter the pair of brackets "(" and ")" manually. The Step Label will be removed if nothing is entered.

8.4 Result/Options Area



The left most read-only edit box (Result window) is used to display the test result, such as: "Pass", "Fail", "Done", "…" (in progress). During a test plan execution, it will be marked as "…". If the plan contains one or more Pass/Fail checks, the final result will be marked as "Pass" only if all the Pass/Fail checks succeed, otherwise it will be marked as "Fail". If the plan contains no Pass/Fail check, it will be marked as "Done" after the plan is finished.

You can specify the Derived Data Points to be captured in a test plan and they are only applicable to those DAQ-related steps (i.e. SIO, SOI and STI). The values of these DDPs will be displayed in the Process Viewer from the 15th column onwards.

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You can also assign an alias for the DDP to make it more understandable. The alias will appear in the column title in the Process Viewer as well as the axis title in the X-Y Plot. Pressing "->" button will add the selected DDP into the list of data to be captured. Pressing "<-" button will remove the selected DDP from that list.

On the right hand side, there are two buttons: "Save Current Panel Setting as Default" and "Save Current Panel Setting". The functionality of these two buttons is the same as that of the "Save Current Panel Setting as Default" and "Save Current Panel Setting" menu item described previously, except that the Device Test Plan panel will be automatically launched with the current device test plan loaded when the saved Panel Setting File is loaded.

8.4.1 User Log In / Off

It is possible to configure a user database and then allow user log in and log off. The user database is just a text file containing three comma separate variables for each user entry. The three variables are: user name, password, priority. The first two variables can consist of any characters, but the latest one can only be a character from "0" to "9", with "0" being the highest priority. Each line in the file contains one user entry only. The file should be put in the software's root directory and named as "Users.udb". It can be edited by any text editor such as Windows Notepad and should be managed by the administrator. The following illustrates the format of the file.

NS3390,1234,0
NS3391,5678,0
NS3392,90ab,0
…

If the user database file (User.udb) exists and it contains at least one user entry, then the edit boxes for "User" and "Password" and the button for "Log In / Off" will become visible at the lower right corner of the Result/Options area.

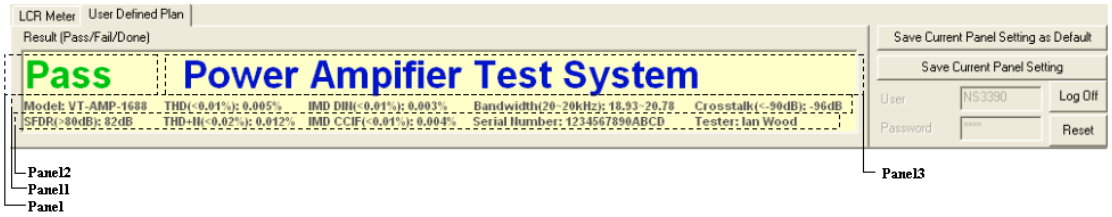
If the user database file (User.udb) exists, then only a user with a priority equal to "0" is allowed to edit a plan and reset those non-volatile integers.

8.4.2 Non Volatile Variable Reset

The Reset button is used to reset those non-volatile integers (NonVolatile1~ NonVolatile5) persisted in the CSV text file "NonVolatile.txt" under "\dtp" diretory.

8.4.3 Expanded Result Window at Runtime

The Result window will expand to the right at runtime in order to allow more space for programmed display. It is split into four areas as shown below: Panel, Panel1, Panel2 and Panel3. The texts and their colors in Panel1, Panel2 and Panel3 are controllable by the instruction LOG. Each Panel can accommodate one line only. Concatenation of texts can be achieved using a <Record Separator> rather than the default Line Feed + Carriage Return.



8.5 Control Bar

The above control bar contains (from left to right):

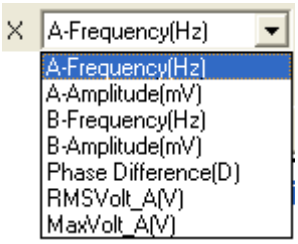
- Four radio buttons allow you to choose to show Control, show Control+Result, show Control+Result+Process, or show Control+Result+Process+Editor.

- File Open button is used to load a device test plan. The path and file name of the plan will be displayed in the title bar of the Device Test Plan panel.

- File Save button is used to save the current device test plan. Not only the content of the device test plan but also the parameters of the X-Y plots and Device Test Plan Log will be saved.

- File Save & Lock button is used to save the current device test plan as a locked plan. A locked plan is not editable when reloaded and the "Editor" radio button will be disabled. This feature is useful when the test plan is configured by an engineer and operated by an operator.

- X combo box for selecting the variable for X axis.



  The first five variables (i.e. Frequency of Channel A, Amplitude of Channel A, Frequency of Channel B, Amplitude of Channel B, Phase difference between Channel A and Channel B) are pre-fixed and are always available for selection. The rest variables will be shown in the list only if they are configured in the "data to be captured" list.

- Y combo box for selecting the variable for Y axis.

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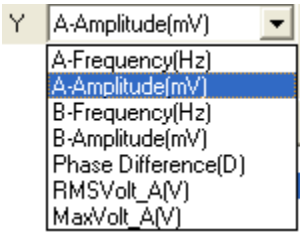


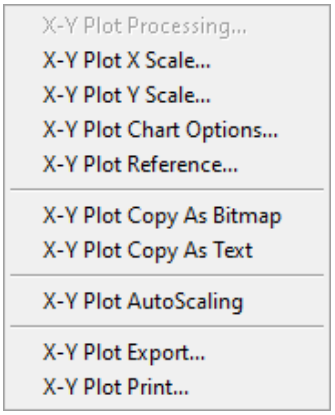
The available variables in the list are the same as those in the X combo box.

- X-Y Plot button is used to open a new X-Y plot window. Maximum eight X-Y Plot windows can be opened. And these plots will be updated in real time.
- Device Test Plan Log button is used to open the Device Test Plan Log window. One line of texts will be logged into this window each time when the instruction LOG is executed.
- Windows Recording Control button
- Windows Volume Control button
- Pause button for plan execution
- Single Step button Single Step for plan execution
- Run/Stop button for plan execution

8.6 X-Y Plot

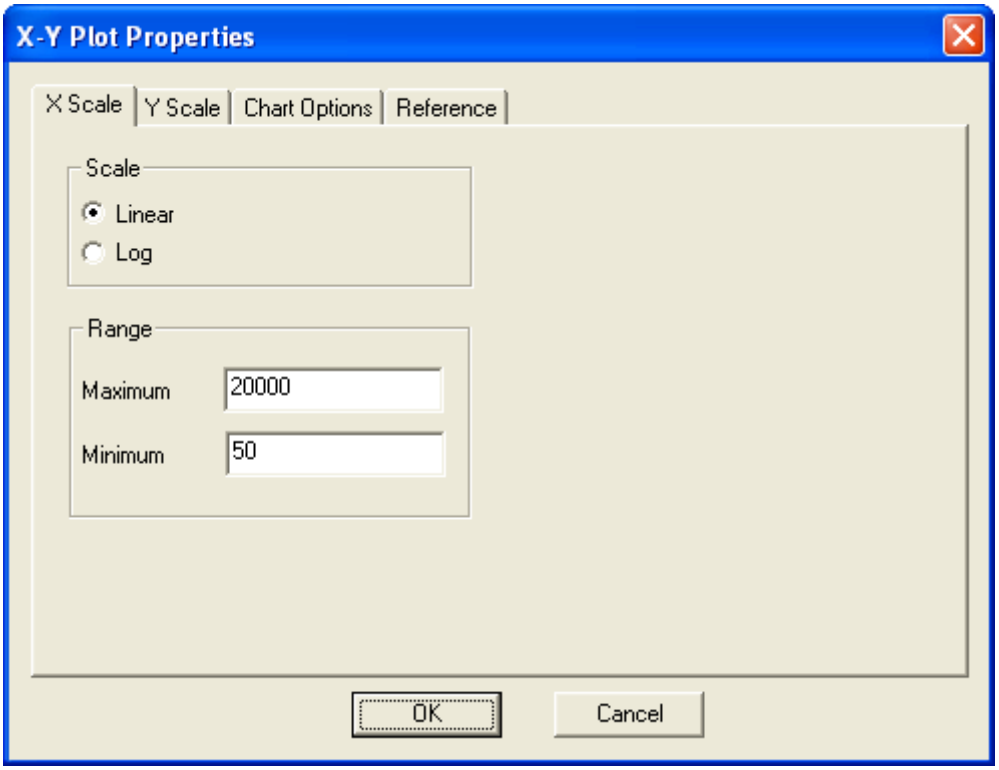
X-Y Plot is used to display the result of a test plan. It can be opened before, in the middle, or after the execution of the test plan. An X-Y Plot can be generated even when variables (x1~x5) instead of actual numbers are used in the output frequencies, amplitudes or phases.

8.6.1 Context Menu



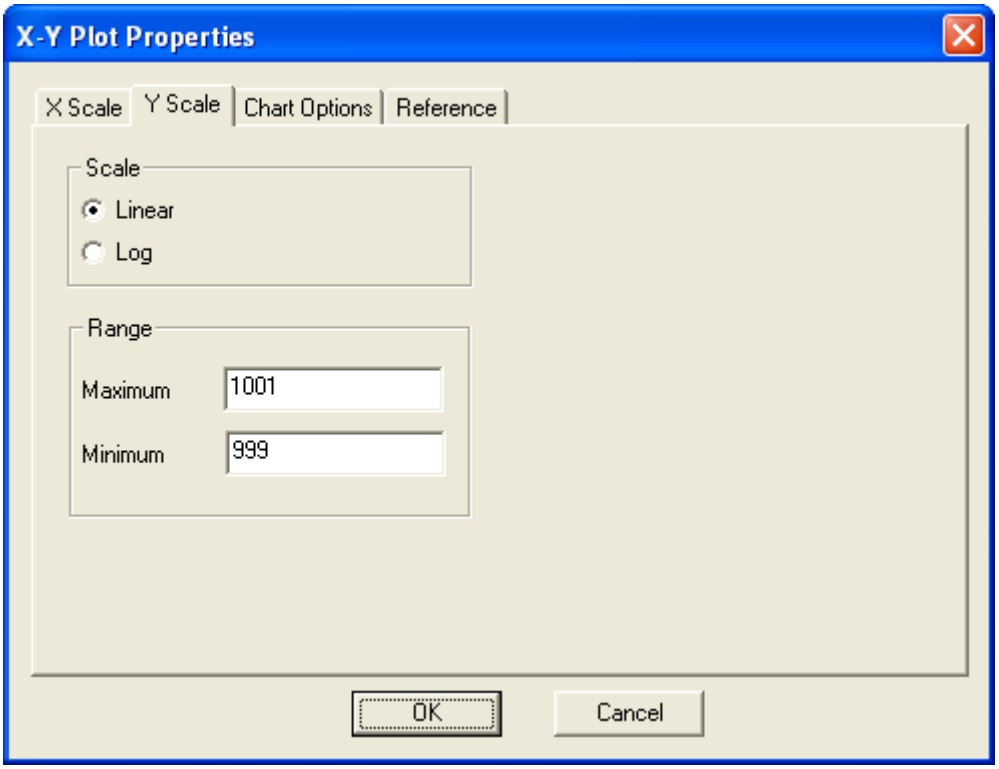
X-Y Plot does not have its own menu and view parameter toolbar. Its functions are accessed via its context menu. The context menu pops up when you right click anywhere within the X-Y Plot window. The X-Y Plot Processing option is only available in Derived Data Curve (DDC) and thus will be described there later.

8.6.1.1 X Scale



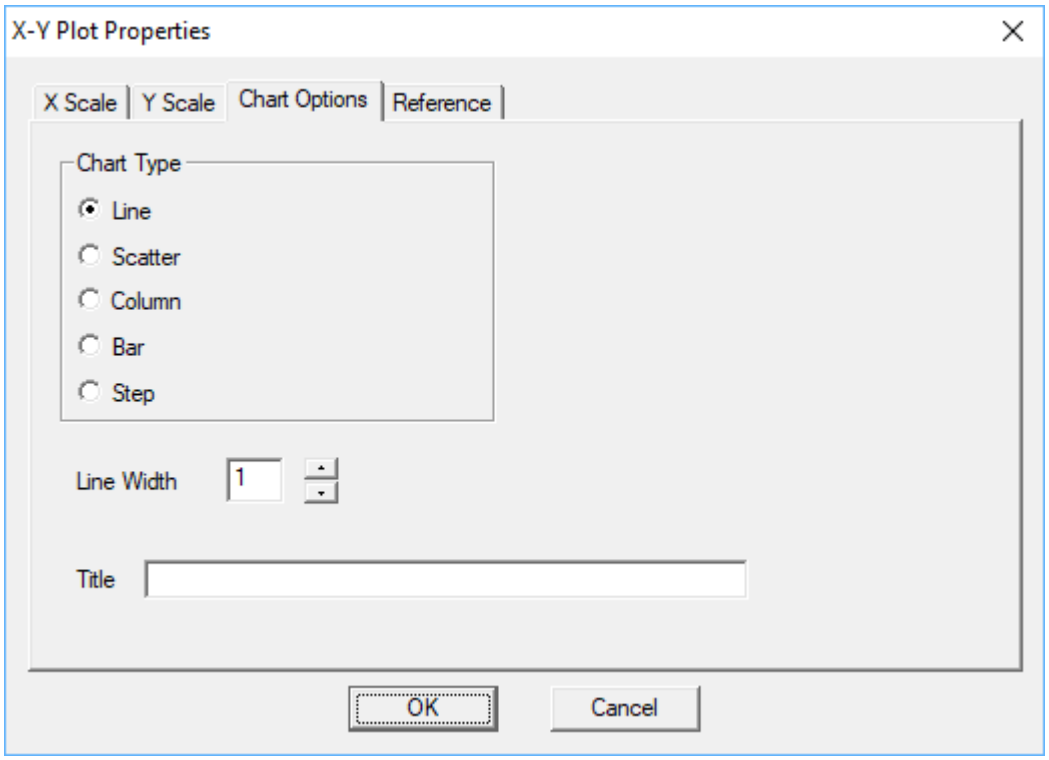
The X Scale dialog can be accessed via the context menu or by double clicking on X axis of the plot. You can specify the range of X axis and choose between linear or logarithmic scales. Note that the software will disallow logarithmic scale if the range covers negative or zero values.

8.6.1.2 Y Scale



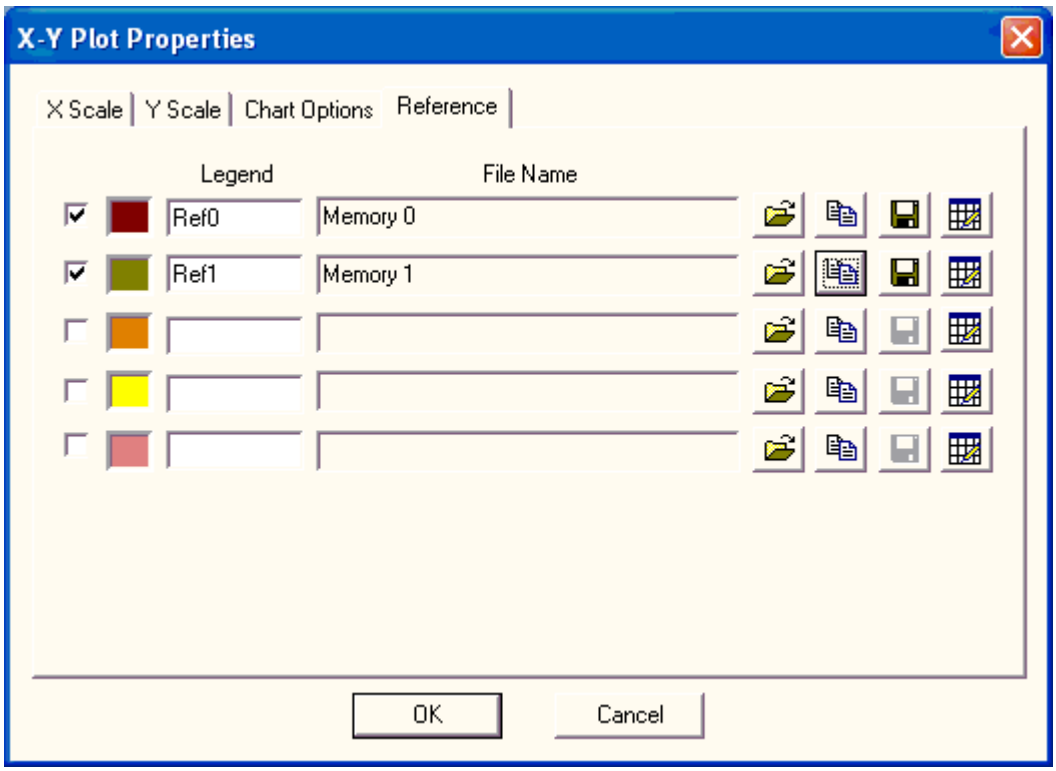
The Y Scale dialog can be accessed via the context menu or by double clicking on Y axis. You can specify the range of Y axis and choose between linear or logarithmic scales. Note that the software will disallow logarithmic scale if the range covers negative or zero values.

8.6.1.3 Chart Options



Same as Oscilloscope, except that you can also specify a title for the X-Y Plot here.

8.6.1.4 Reference



Same as Oscilloscope except that only one channel of data is available.

8.6.1.5 Copy As Bitmap

Same as Oscilloscope

8.6.1.6 Copy As Text

Same as Oscilloscope

8.6.1.7 AutoScaling

Same as Oscilloscope

8.6.1.8 Export

Same as Oscilloscope

8.6.1.9 Print

Same as Oscilloscope

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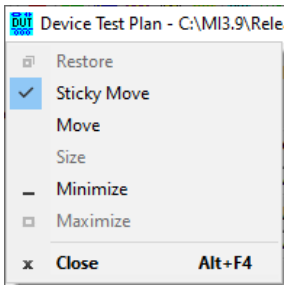
8.6.2 Cursor Reader and Marker

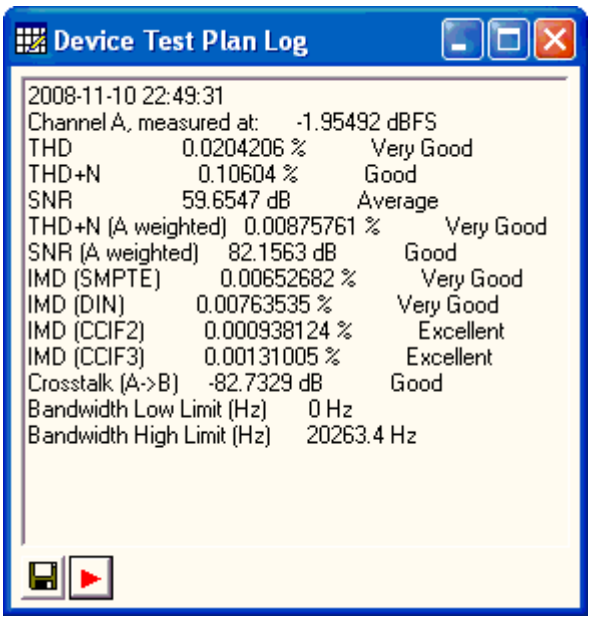
Similar to those in the Oscilloscope, except that the cursor reader and marker mark the point exactly at the point of mouse click rather than sticking to the nearest measurement point.

8.6.3 Sticky Move



"Sticky Move" option can be accessed by clicking the icon at the upper left corner of the Device Test Plan panel. It is ticked by default. With this option selected, all X-Y Plots will move together if you drag the title bar of the Device Test Plan panel.

8.7 Device Test Plan Log



Device Test Plan Log can be used to log the test results or generate a test report. It should be used with the instruction LOG.

Two buttons are available at the bottom of the log:

- File Save
  To save the logged data.

- Run/Stop button for plan execution
  It has the same functionality as the one in the Device Test Plan main window.

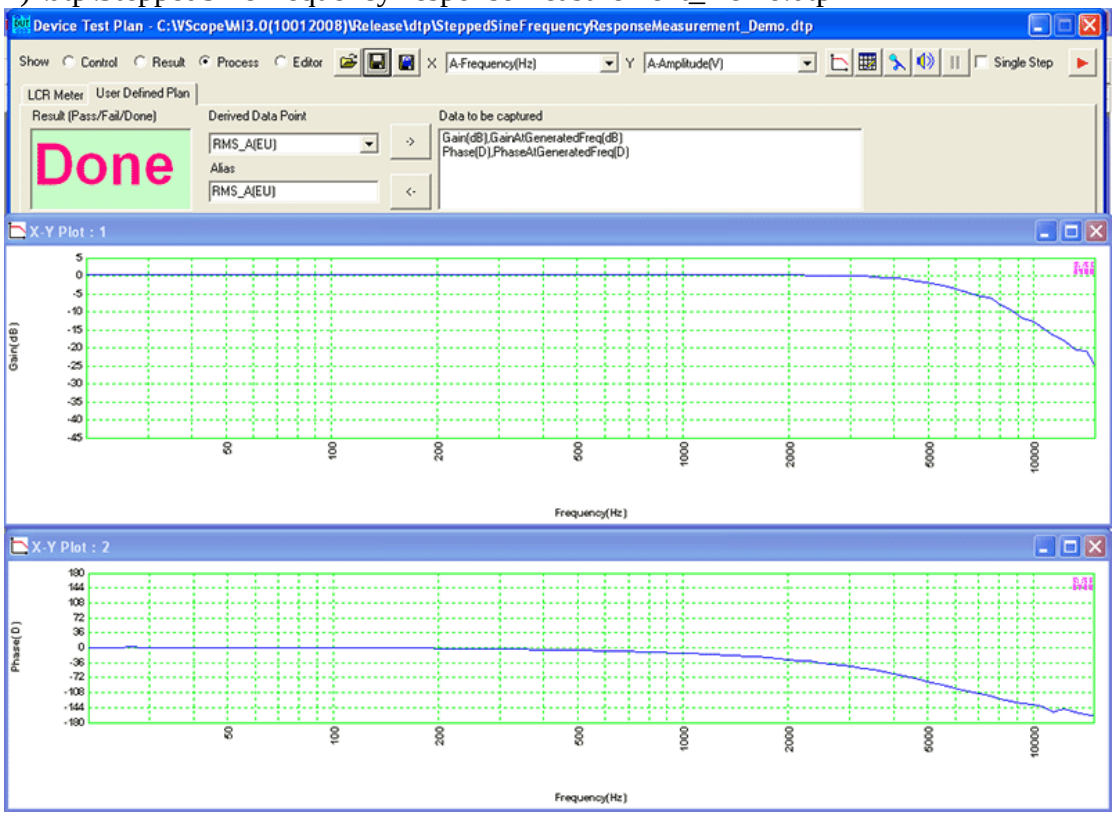
8.8 Device Test Plan Examples

Some sample test plans are provided in the DTP directory of the software and can be used as templates. The following are a few examples.

8.8.1 Transfer Function Measurement using Frequency Stepped Sine Signal

The following figure illustrates the transfer function (i.e. Gain and Phase Plot, or Bode Plot) of a 5513 Hz second order Butterworth low pass filter, measured using a frequency stepped sine stimulus, with the stimulus data stored in Channel B and the response data stored in Channel A. It shows that the gain maintains at nearly 0 dB from 0 Hz to about 5513 Hz, and then start to drop down very quickly, meanwhile the phase changes gradually from 0 degree at 0 Hz towards –180 degree as the frequency goes to infinity. At the cutoff frequency 5513 Hz, the gain is about –3dB and the phase is about –90 degree.

The sample DTP files are at:
1) \dtp\SteppedSineFrequencyResponseMeasurement.dtp
2) \dtp\SteppedSineFrequencyResponseMeasurement_Demo.dtp



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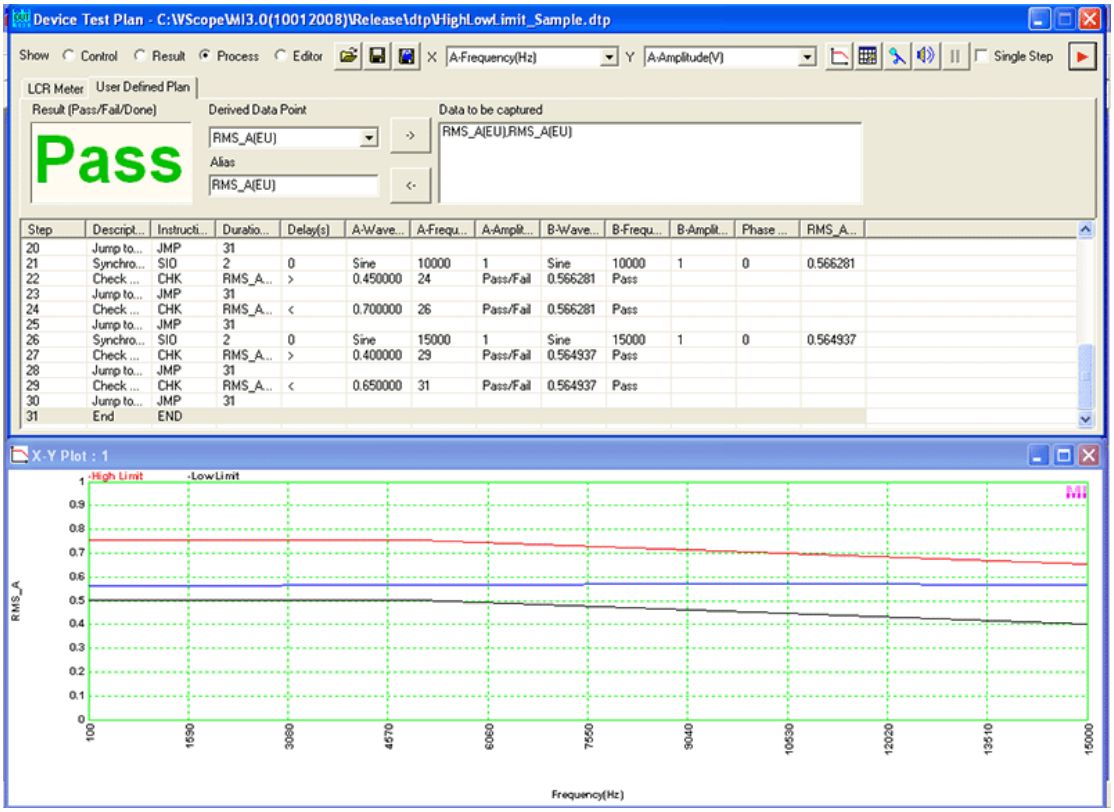
8.8.2 Pass/Fail Test

The following figure illustrates a Pass/Fail test on magnitude frequency response using a stepped sine signal. The measured RMS voltage values (blue) at the specified frequencies are checked against the preset high and low limits at those frequencies in order to determine "Pass" or "Fail". The high (red) and low (black) limits are also loaded as reference in the X-Y Plot. The final result is displayed as "Pass" only if all limit checks have been passed.

The sample DTP file is at: \dtp\HighLowLimit.dtp.

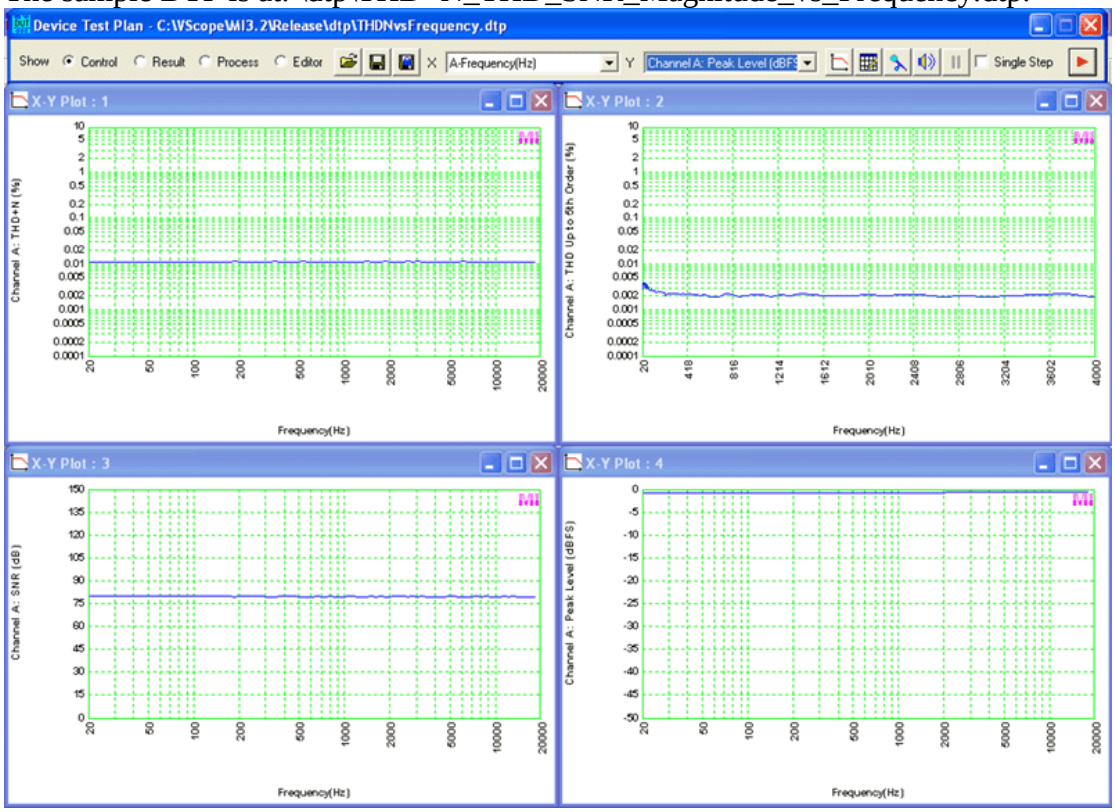


8.8.3 THD+N, THD, SNR, Magnitude vs Frequency Plots

The following figure shows the THD+N, THD, SNR, Magnitude vs Frequency Plots measured using frequency stepped sine signals. The frequency of the test signal increases step by step logarithmically from 20 Hz to 20 kHz while its amplitude is kept constant such that the input peak level is at around –1 dBFS. The test frequencies are fine tuned using the aforementioned "No Spectral Leakage" option to avoid the spectral leakage issue. The THD calculation range is set to 10 Hz ~ 20 kHz and up to the 5th harmonic frequencies is taken into account in THD. This means that, for the fundamental frequency 20 Hz, only its harmonic frequencies 40 Hz, 60 Hz, 80Hz and 100 Hz are used in the THD calculation; for the fundamental frequency 4 kHz, only its harmonics frequencies 8 kHz, 12 kHz, 16 kHz and 20 kHz are used; but for a fundamental frequency higher than 4 kHz, only those harmonic frequencies less than 20 kHz, i.e. less than the 5th order, are used. Therefore only the THDs in the frequency range from 20 Hz to 4 kHz are comparable because the same order of

harmonic frequencies is used in their THD calculation. On the other hand, THD+N values are comparable in the full frequency range from 20 Hz to 20 kHz. The plot at the upper left corner is the THD+N vs Frequency plot; the plot at the upper right corner is the THD vs Frequency plot; the plot at the bottom left corner is the SNR vs Frequency plot; the plot at the bottom right corner is the Magnitude vs Frequency plot. Note that, although the magnitude of the signal output towards the DUT is kept constant, the magnitude of the response signals output by the DUT may vary with frequency. Therefore the Magnitude vs Frequency plot actually reflects the magnitude frequency response of the DUT.

The sample DTP is at: \dtp\THD+N_THD_SNR_Magnitude_vs_Frequency.dtp.

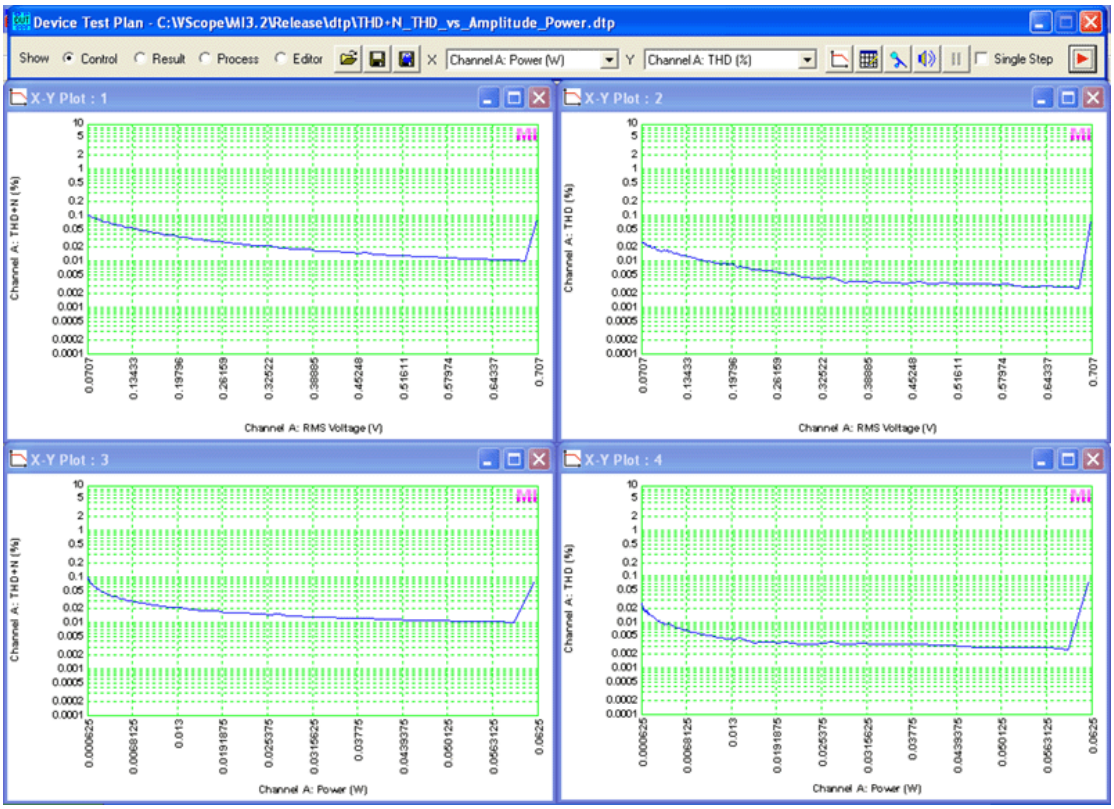


8.8.4 THD+N, THD vs Magnitude, Power Plots

The following figure shows the THD+N, THD, vs Magnitude, Power Plots measured using amplitude stepped sine signals. The amplitude of the test signal increases step by step linearly from 0.1 V to 1 V while its frequency is kept constant at 999.0234375 Hz. The frequency is fine tuned using the aforementioned "No Spectral Leakage" option to avoid the spectral leakage issue. The THD calculation range is set to 10 Hz ~ 20 kHz. The plot at the upper left corner is the THD+N vs RMS voltage plot; the plot at the upper right corner is the THD vs RMS voltage plot; the plot at the bottom left corner is the THD+N vs Power plot; the plot at the bottom right corner is the THD vs Power plot. Note that, the power value is obtained from the Derived Data Point PWR_A(W), which is equal to [RMS Voltage]^2 / [Load Factor], where the load factor is set via [Setting]>[Calibration]> "Load Factor for Power Calculation". For better

measurement accuracy, it is recommended the input peak level should not be lower than –20 dBFS during the amplitude sweep.

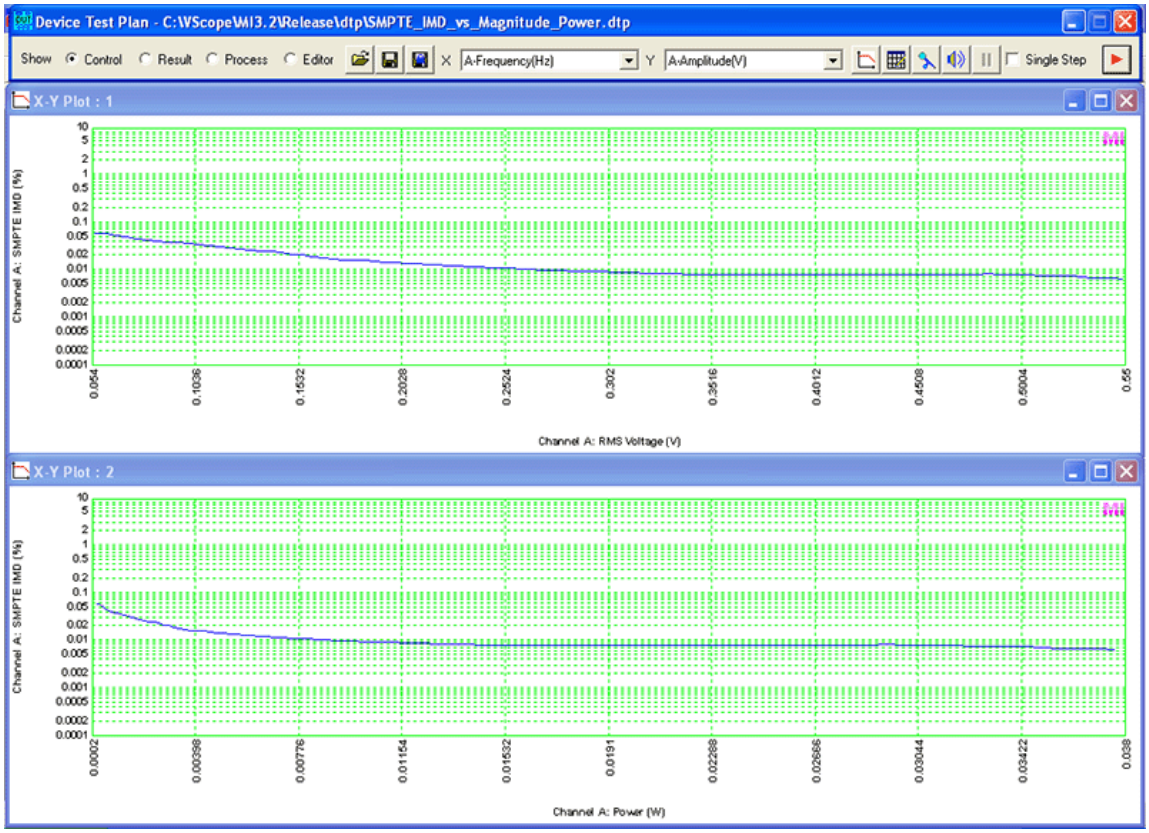
The sample DTP is at: \dtp\THD+N_THD_vs_Magnitude_Power.dtp.



8.8.5 SMPTE IMD vs Magnitude, Power Plots

The following figure shows the SMPTE IMD vs Magnitude, Power Plots measured using amplitude stepped SMPTE sine signals (60 Hz + 7kHz, Amplitude Ratio 4: 1). The amplitude of the test signal increases step by step linearly from 0.1 V to 1 V. The upper plot is the SMPTE IMD vs RMS voltage plot; the lower plot is the SMPTE IMD vs Power plot. Note that, the power value is obtained from the Derived Data Point PWR_A(W), which is equal to [RMS Voltage]^2 / [Load Factor], where the load factor is set via [Setting]>[Calibration]> "Load Factor for Power Calculation". For better measurement accuracy, it is recommended the input peak level should not be lower than –20 dBFS during the amplitude sweep.

The sample DTP is at: \dtp\SMPTE_IMD_vs_Magnitude_Power.dtp.



8.8.6 Crosstalk vs Frequency Plots

The following figure shows the Crosstalk vs Frequency Plot measured using frequency stepped sine signals. The frequency of the test signal increases step by step logarithmically from 20 Hz to 20 kHz while its amplitude is kept constant such that the input peak level is at around –1 dBFS. The test frequencies are fine tuned using the aforementioned “No Spectral Leakage” option to avoid the spectral leakage issue. .

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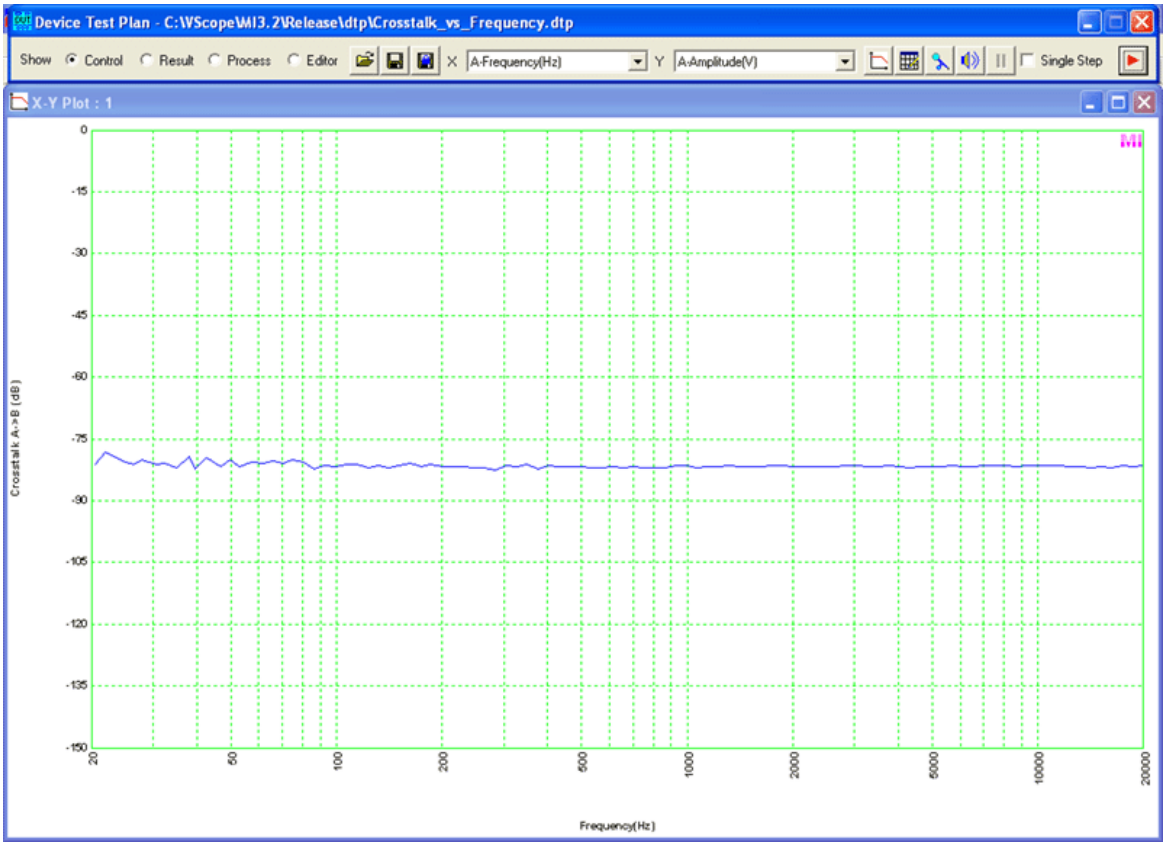


The sample DTP is at: \dtp\Crosstalk_vs_Frequency.dtp.

8.8.7 Automated Audio Parameter Test

The test is to measure THD, THD+N, THD+N (A-Weighted), SNR, SNR (A-Weighted), SMPTE IMD, DIN IMD, CCIF2 IMD, CCIF3 IMD, Crosstalk, and -3dB Bandwidth automatically through a sequence of automated steps. The test results will be summarized in the log window. The following criteria are used to evaluate the results. The test can be used to evaluate a sound card's own audio performance through the loopback test. It can also be used to measure a device's audio parameters automatically when the test setup's own audio performance is one order better than the device under test.

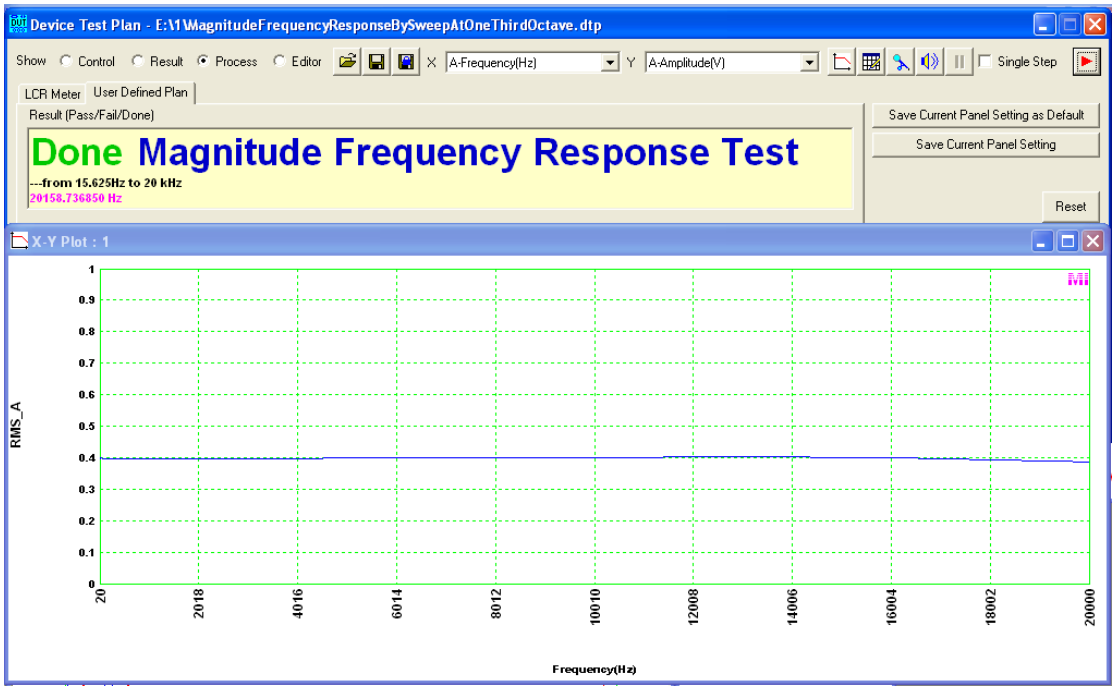
	Excellent	Very Good	Good	Average	Poor	Very Poor
THD	<=0.003%	<=0.03%	<=0.3%	<=3%	<=10%	>10%
THD+N	<=0.005%	<=0.05%	<=0.5%	<=5%	<=20%	>20%
THD+N (A-Weighted)	<=0.004%	<=0.04%	<=0.4%	<=4%	<=15%	>15%
SNR	>=110 dB	>=90 dB	<=70 dB	>=50 dB	>=30dB	<30dB
SNR (A-Weighted)	>=115 dB	>=95 dB	<=75 dB	>=55 dB	>=35dB	<35dB
SMPTE IMD	<=0.003%	<=0.03%	<=0.3%	<=3%	<=10%	>10%
DIN IMD	<=0.003%	<=0.03%	<=0.3%	<=3%	<=10%	>10%
CCIF2 IMD	<=0.003%	<=0.03%	<=0.3%	<=3%	<=10%	>10%
CCIF3 IMD	<=0.003%	<=0.03%	<=0.3%	<=3%	<=10%	>10%
Crosstalk	<=-110 dB	<=-90 dB	<=-70 dB	<= -50 dB	<=-30 dB	>-30 dB

The sample DTPs are at: \dtp\AudioParameter_SR44100_A.dtp
\dtp\AudioParameter_SR44100_AB.dtp
\dtp\AudioParameter_SR48000_A.dtp
\dtp\AudioParameter_SR48000_AB.dtp

8.8.8 Magnitude Frequency Response Measurement using Variable Based Frequency Stepped Sine Signal

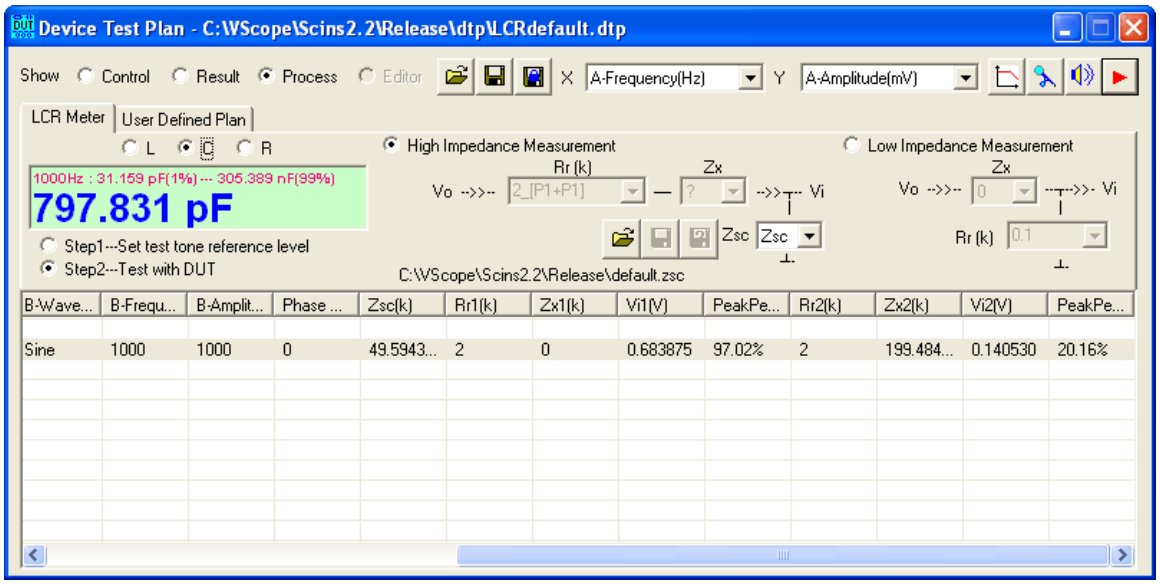
The following figure shows the magnitude frequency response measured using variable based frequency stepped sine signal. The test frequencies in this example are not specified explicitly in the device test plan. Instead, they are generated dynamically by manipulating the variable x1 during the execution of the plan. The following shows the related pseudo codes. It can be seen that using variables can greatly reduce the number of steps and increase the flexibility.

Step 1: OPT x1 = 15.625; //Initialize x1
Step 2: SIO (x1); //Output frequency x1
Step 3: OPT x1 = x1*1.25992105; //Increase by 1/3 octave, 2^(1/3)=1.25992105
Step 4: CHK x1 > 20159, goto Step 6 //Exit if x1 is greater than 20159
Step 5: JMP to Step 2; //Repeat
Step 6: END



9 LCR Meter

9.1 Overview



LCR Meter is one of the dedicated test plans. It is used to measure the value of an inductor, a capacitor or a resistor, or the impedance of a network of them. Two types of external connections are supported:

- Serial connection for high impedance measurement
- Parallel connection for low impedance measurement

An external reference resistor Rr is required in both cases. In the former one, the reference resistor Rr is in series to the sound card input impedance Zsc, and in the latter one, Rr is parallel to Zsc.

The screen layout of the LCR Meter is similar to that of the Device Test Plan and the difference will be described in the following sections.

9.2 Step Editor

By default, the LCR Meter will load predefined locked dedicated test plans, therefore the Step Editor will always be disabled. However, you are allowed to save a locked test plan as an unlocked test plan and subsequently load it for editing. In such case, the Step Editor will be shown.

9.3 Progress Viewer

As a dedicated test plan, the LCR Meter has unique content in the Progress Viewer from the 13th column onwards. They are:

Column Title	Description
Zsc (k)	Sound Card Input Impedance (kΩ)
Rr1 (k)	Reference Resistor Value (kΩ) used in Step 1
Zx1 (k)	Always be zero (kΩ) because Zx should not be connected in this step
Vi1 (V)	RMS voltage (V) measured in Channel A in Step 1
PeakPercent1	Peak level percentage measured in Channel A in Step 1
Rr2 (k)	Reference Resistor Value (kΩ) used in Step 2
Zx2 (k)	Measured impedance value (kΩ) in Step 2
Vi2 (V)	RMS voltage (V) measured in Channel A in Step 2
PeakPercent2	Peak level percentage measured in Channel A in Step 2

The above columns will be filled with the actual values by the software automatically as the process goes. However, if there are exceptions encountered, they will be filled with one of the following texts instead:

"NA"-----Not Available/Not Applicable
"BAD"---The quality of the test tone is considered as bad if the peak level measured in Channel A exceeds 99.9% or the THD value measured in Channel A exceeds 3%, in which case, you should adjust the output level, the reference resistor value, or the input gain.

If the value is currently unknown and is to be measured, then the field will be filled with "?".

The meaning of each of the above columns will be further explained in the following sections.

9.4 Result/Options Area

The left most read-only edit box (Result window) is used to display the test result. In Step 1 (Set test tone reference level), it displays the peak level percentage at each test frequency. In Step 2 (Test with DUT), it displays the measured impedance value of the DUT. You can choose to display the impedance as inductance, capacitance, or resistance, by selecting the corresponding radio buttons "L", "C", "R". The measurement range is also indicated. Adjusting the reference resistor value will change the measurement range.

In the right part of this area, you can choose between High Impedance Measurement method or Low Impedance Measurement method. The corresponding mimic connection diagram will then be enabled. Details of these two methods will be described later.

9.5 Control Bar

Same as that in the Device Test Plan. The File Open, File Save, File Save & Lock buttons are still enabled to allow advanced users to modify the default LCR test plans.

However, it should be noted that the LCR meter may function incorrectly if a non-LCR test plan is used.

9.6 High Impedance Measurement

High Impedance Measurement uses the sound card input impedance as one of the reference resistor. Thus, to enable this type of measurement, the sound card input impedance, which is sound card dependent, must be calibrated first. After calibration, the LCR Meter will be able to display the measurement range based on the reference resistor value and the test frequency used. The software uses 1% and 99% variation from the test tone reference level to suggest the measurement range, within which good measurement accuracy can be achieved. This is similar to the case of resistor measurement using an analog multimeter whereby the middle region of the swing of the needle has good measurement accuracy.

The procedure to make a LCR measurement is also similar to the procedure to measure a resistor using an analog multimeter, as shown as follows.

Step	Description	LCR Meter Implementation	Analog Multimeter Implementation
0	Choose a proper measurement range	By connecting a proper reference resistor Rr and enter the corresponding value in the Rr combo box on the screen. Then start the plan.	Via the switch which switches to a corresponding reference resistor inside the multimeter
1	Set the test tone reference level such that the maximum allowable level is reached to ensure sufficient measurement accuracy	By shorting the two test leads and then adjusting the output level via Windows Volume Control, and/or the input gain via Windows Recording Control. A peak level above 85% is recommended.	By shorting the two test leads and then adjusting the zeroing potentiometer such that the needle points to zero ohms.
2	Connect the DUT and make the measurement. You can repeat this step to test other DUTs as long as there is no need to change the measurement range.	By connecting the DUT between the two test leads and choose Step 2 radio button on the screen	By connecting the DUT between the two test leads

9.6.1 Connection for High Impedance Measurement

The connection diagram for high impedance measurement is shown as follows.

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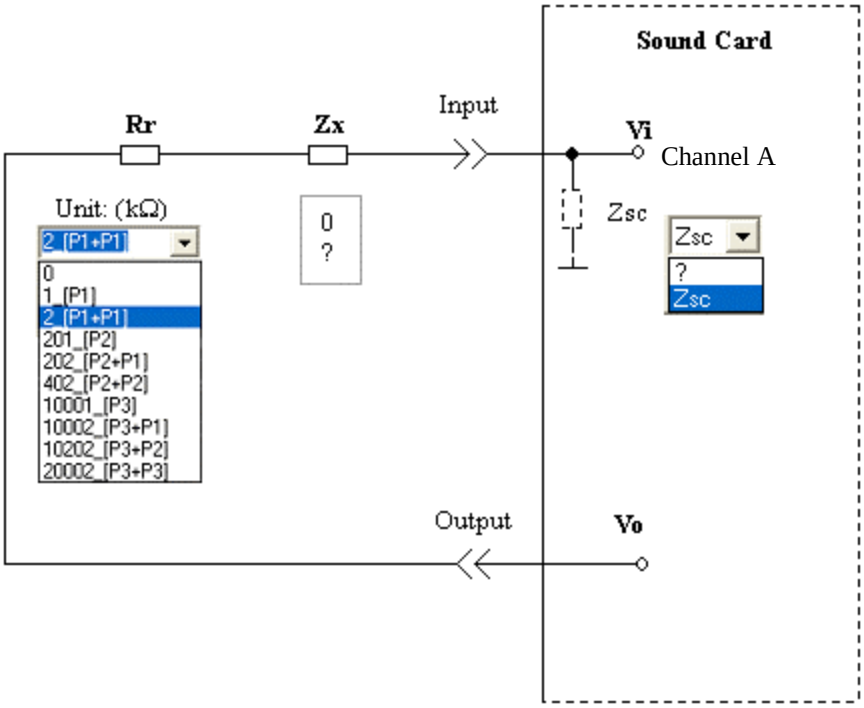

where:
Rr is the reference resistor.
Zx is the impedance to be measured.
Zsc is the sound card input impedance
Vo is the output RMS voltage.
Vi is the input RMS voltage.

Rr, Zx and Zsc form a voltage divider and thus we have Vi/Vo=Zsc/(Rr+Zx+Zsc). This connection should be used for high impedance measurement where Zx is comparable to or higher than Zsc. **It should be noted that the sound card output impedance, which typically ranges from nearly zero ohm to a few ohms for Speaker/Headphone Out, is ignored in the above formula. This simplification has negligible effect on the measurement accuracy as long as the output impedance is negligibly small compared with the value of Rr+Zx+Zsc, which holds true in almost all cases. It can be taken into account by simply adding its value (if it is known) to the value of Rr.**

Zsc is typically in the range of 600 Ω ~ 50 kΩ, depending on the sound card and channels (MIC In or Line In) used. It must be calibrated before you can start the real LCR measurements. The calibration data can be saved so that you do not have to calibrate it again as long as the same sound card is used. In the Zsc combo box, two options are available: "?" and "Zsc". Choose "?" if you want to conduct Zsc calibration, and choose "Zsc" if you want to conduct a normal LCR measurement. Only "?" will be available if the software cannot find any calibration data in the current software directory. This is to force you to conduct Zsc calibration first.

Rr is an external reference resistor. You need to find the resistor and make the connection by yourself. An easier way is to use two VIRTINS Sound Card Oscilloscope Probes (P601), one connected to the input jack of the sound card and the other connected to the output jack of the sound card. Then you can connect the

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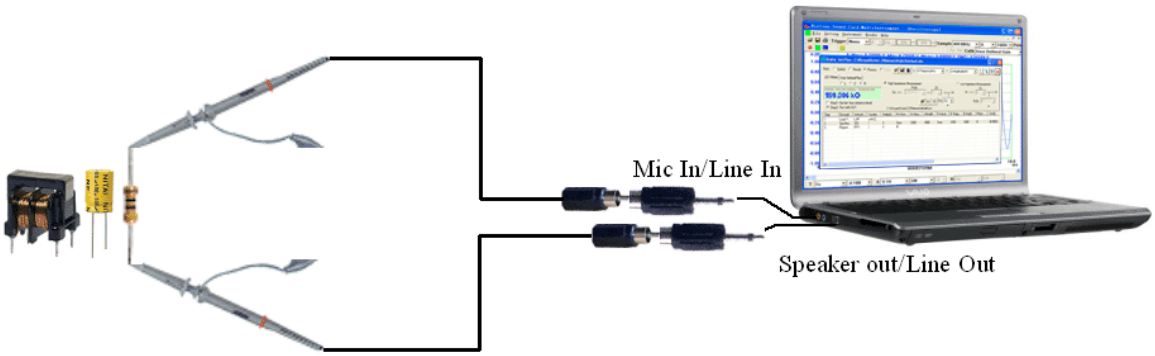
inductor/capacitor/resistor to be measured between them easily (see figure below, you do not have to connect the ground lead if they are connected within the sound card itself.). The impedance of the probes will act as the reference resistor. Note that for VIRTINS Sound Card Oscilloscope Probe (P601), the input impedance is 1 k, 201k and 10.001M for switch position 1, 2 and 3 respectively. In the Rr combo box, you should enter the actual value (in kΩ) you used. Different test steps or different measurement ranges desired may require different resistor values.



Some pre-configured resistance values are available for selection, most of them are relevant to the case where two VIRTINS Sound Card Oscilloscope Probes (P601) are used as both the test leads and reference resistor.

Options	Resistance (kΩ)	Description
0	0	Two zero-resistance test leads are used.
1_[P1]	1	One VIRTINS Sound Card Oscilloscope Probe and one zero-resistance test lead are used. The probe is in Position 1.
2_[P1+P1]	2	Two VIRTINS Sound Card Oscilloscope Probes are used. Both of them are in Position 1.
201_[P2]	201	One VIRTINS Sound Card Oscilloscope Probe and one zero-resistance test lead are used. The probe is in Position 2.
202_[P2+P1]	202	Two VIRTINS Sound Card Oscilloscope Probes are used, one in Position 2 and the other in Position 1.
402_[P2+P2]	402	Two VIRTINS Sound Card Oscilloscope Probes are used. Both of them are in Position 2.
10001_[P3]	10001	One VIRTINS Sound Card Oscilloscope Probe and one zero-resistance test lead are used. The probe is in Position 3.
10002_[P3+P1]	10002	Two VIRTINS Sound Card Oscilloscope Probes are used, one in Position 3 and the other in Position 1.
10202_[P3+P2]	10202	Two VIRTINS Sound Card Oscilloscope Probes are used, one in Position 3 and the other in Position 2.
20002_[P3+P3]	20002	Two VIRTINS Sound Card Oscilloscope Probes are used. Both of them are in Position 3.

If you are not using the above values, then you have to enter the resistance value manually. For advanced users, if you have your own set of reference resistors, you can modify the TXT file named “ResistorRef1.txt” under the software root directory so that your can have your own set of reference resistor values available in the Rr combo box. In the following sections, to simplify the explanation of the measurement procedure, we will assume that two VIRTINS Sound Card Oscilloscope Probes are used. But bear in mind that you can always use your own test leads and reference resistors.

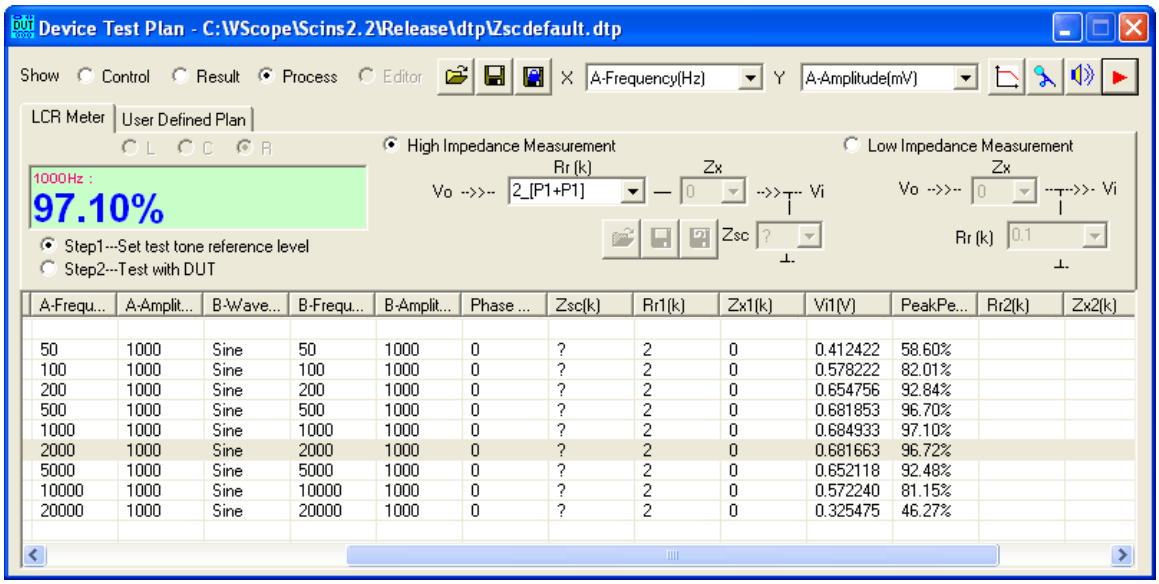
Zx is the impedance to be measured. When "?" is chosen in the Zsc combo box, the Zx combo box will always display "0", which means during Zsc calibration, Zx should not be connected. When "Zsc" is chosen in the Zsc combo box, the Zx combo box will display "0" for Step 1 and "?" for Step 2. In other word, only the last step of the actual LCR measurement requires Zx to be connected.

9.6.2 Calibration of Sound Card Input Impedance

The software does not assume Zsc is a constant with regard to frequency. Thus the value of Zsc should be calibrated against a number of frequencies covering the entire range within which the test frequencies will be chosen. During the actual LCR measurement, the actual Zsc values at the actual test frequencies will be interpolated from the calibration result. To avoid the interpolation error, use the same frequencies during calibration and actual measurement.

The procedure to calibrate Zsc is similar to the procedure to make a LCR measurement. You need to take two steps to complete the calibration.

9.6.2.1 Step1---Set Test Tone Reference Level



A-Frequ...	A-Amplit...	B-Wave...	B-Frequ...	B-Amplit...	Phase ...	Zsc(k)	Rr1(k)	Zx1(k)	Vi1(V)	PeakPe...	Rr2(k)	Zx2(k)
50	1000	Sine	50	1000	0	?	2	0	0.412422	58.60%		
100	1000	Sine	100	1000	0	?	2	0	0.578222	82.01%		
200	1000	Sine	200	1000	0	?	2	0	0.654756	92.84%		
500	1000	Sine	500	1000	0	?	2	0	0.681853	96.70%		
1000	1000	Sine	1000	1000	0	?	2	0	0.684933	97.10%		
2000	1000	Sine	2000	1000	0	?	2	0	0.681663	96.72%		
5000	1000	Sine	5000	1000	0	?	2	0	0.652118	92.48%		
10000	1000	Sine	10000	1000	0	?	2	0	0.572240	81.15%		
20000	1000	Sine	20000	1000	0	?	2	0	0.325475	46.27%		

To start the calibration, you need to choose "?" in the Zsc combo box. It is automatically chosen if this is the first time you conduct the calibration. Then, a locked dedicated test plan named Zscdefault.dtp will be automatically loaded (see figure above). By factory default, this plan will measure the Zsc values at 50 Hz, 100 Hz, 200 Hz, 500 Hz, 1000 Hz, 2000 Hz, 5000 Hz, 10000 Hz, 20000Hz.

"Step 1-Set test tone reference level" should be selected. The purpose of this step is to set the test tone reference level which is similar to the zeroing step when you want to measure a resistor using an analog multimeter. The reference level should be set as high as possible in order to get sufficient measurement accuracy provided no signal clipping occurs.

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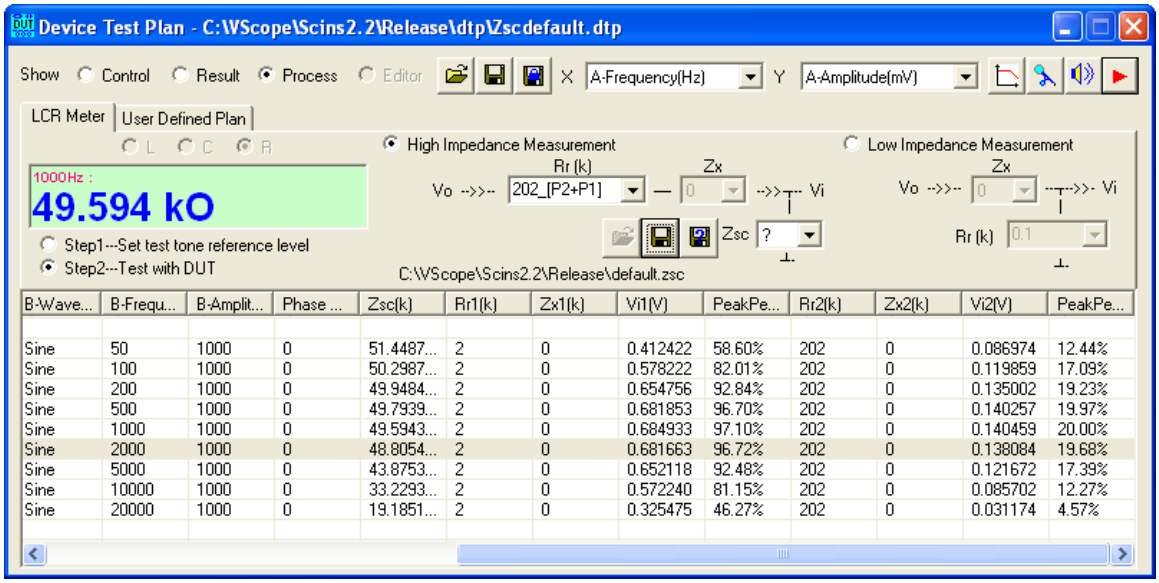
Before you start the test plan, make sure the test loop is connected correctly and the Rr value in the Rr combo box corresponds to the actual value used. For this step, a relatively low (compared with the one in the next step) Rr value should be used. The value can be zero. In the example, we used 2_[P1+P1] (=2 kΩ) for convenience purpose.

Zx is bypassed and its value is zero.

Once the test plan is started, it will generate each test frequency one by one and display the corresponding peak level in the PeakPercent1 column as well as the Result window. After finishing the last test frequency, it will go back to the first test frequency again and repeat this process until the plan is stopped manually. During this process, you should adjust the output level and input gain using the Windows Volume Control and Recording Control such that the maximum peak level among all test frequencies is close to 100% without any clipping. You may have to change the Rr value if the peak levels are too low.

Having set the desired test tone level, press the Start/Stop button to stop the plan.

9.6.2.2 Step2---Test with DUT (Zsc)



B-Wave...	B-Frequ...	B-Amplit...	Phase ...	Zsc(k)	Rr1(k)	Zx1(k)	Vi1(V)	PeakPe...	Rr2(k)	Zx2(k)	Vi2(V)	PeakPe...
Sine	50	1000	0	51.4487...	2	0	0.412422	58.60%	202	0	0.086974	12.44%
Sine	100	1000	0	50.2987...	2	0	0.578222	82.01%	202	0	0.119859	17.09%
Sine	200	1000	0	49.9484...	2	0	0.654756	92.84%	202	0	0.135002	19.23%
Sine	500	1000	0	49.7939...	2	0	0.681853	96.70%	202	0	0.140257	19.97%
Sine	1000	1000	0	49.5943...	2	0	0.684933	97.10%	202	0	0.140459	20.00%
Sine	2000	1000	0	48.8054...	2	0	0.681663	96.72%	202	0	0.138084	19.68%
Sine	5000	1000	0	43.8753...	2	0	0.652118	92.48%	202	0	0.121672	17.39%
Sine	10000	1000	0	33.2293...	2	0	0.572240	81.15%	202	0	0.085702	12.27%
Sine	20000	1000	0	19.1851...	2	0	0.325475	46.27%	202	0	0.031174	4.57%

Keep the test tone reference level intact, then choose “Step 2-Test with DUT”. The DUT in this step is in fact Zsc. The value of Rr must be changed in this step. In the example, we changed it to 202_[P2+P1] (=202 kΩ).

Once the test plan is started, it will generate each test frequency one by one and display the corresponding Zsc value measured in the Zsc column as well as the Result window. After finishing the last test frequency, it will go back to the first test frequency again and repeat this process until the plan is stopped manually. The output level and input gain set in the previous step must be maintained in this step.

You should press the File Save button beside "Zsc" to save the calibration result as default.

9.6.2.3 Save, Save As, Open a Zsc calibration file

The File Open button beside "Zsc" can be used to load a sound card impedance file (*.zsc).

The File Save button beside "Zsc" is used to save the current calibration data to the default sound card impedance file named default.zsc. The LCR Meter refers to this file for the value of Zsc by default. You can use the File Open button to load a non-default Zsc calibration file if necessary.

The File Save As button beside "Zsc" is used to save the current calibration data to a specified sound card impedance file.

The path and name of the current sound card impedance file in use, if any, is displayed below the above three buttons.

9.6.3 Make a LCR Measurement

As described previously, the procedure to make a LCR measurement is similar to the procedure to measure a resistor using an analog multimeter. You need to take two steps to complete a measurement at first, and then the first step can be skipped if the measurement range has not been changed.

9.6.3.1 Step1--- Set Test Tone Reference Level

To start the actual measurement, you need to choose "Zsc" in the Zsc combo box. Then, a locked dedicated test plan named LCRdefault.dtp will be automatically loaded (see figure below). By factory default, this plan will measure the Zx values at 1000 Hz only.

Then choose a proper measurement range by connecting a proper reference resistor Rr and enter the corresponding value in the Rr combo box on the screen. Then short the two test leads to bypass Zx, select Step 1, start the plan and adjust the output level via Windows Volume Control, and/or the input gain via Windows Recording Control, in order to set the test tone reference level such that the maximum allowable level is reached to ensure sufficient measurement accuracy. A peak level above 85% is recommended. Stop the plan after the test tone reference level is set.

In the following example, we used 2_[P1+P1] (=2 kΩ) for Rr and the peak level at the test frequency (1000 Hz) was set to 97.02%.

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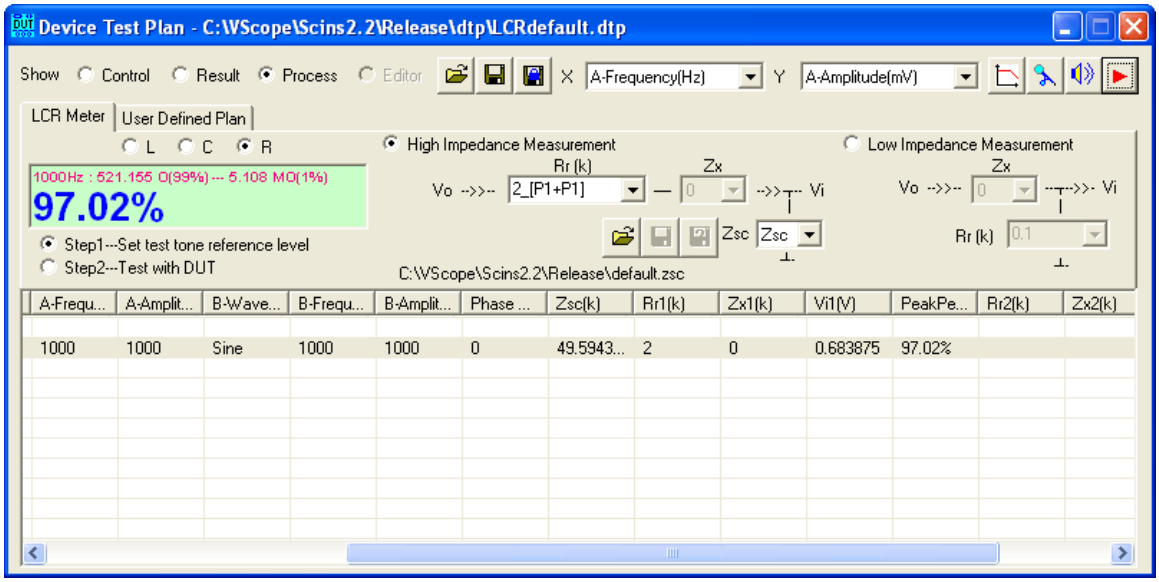
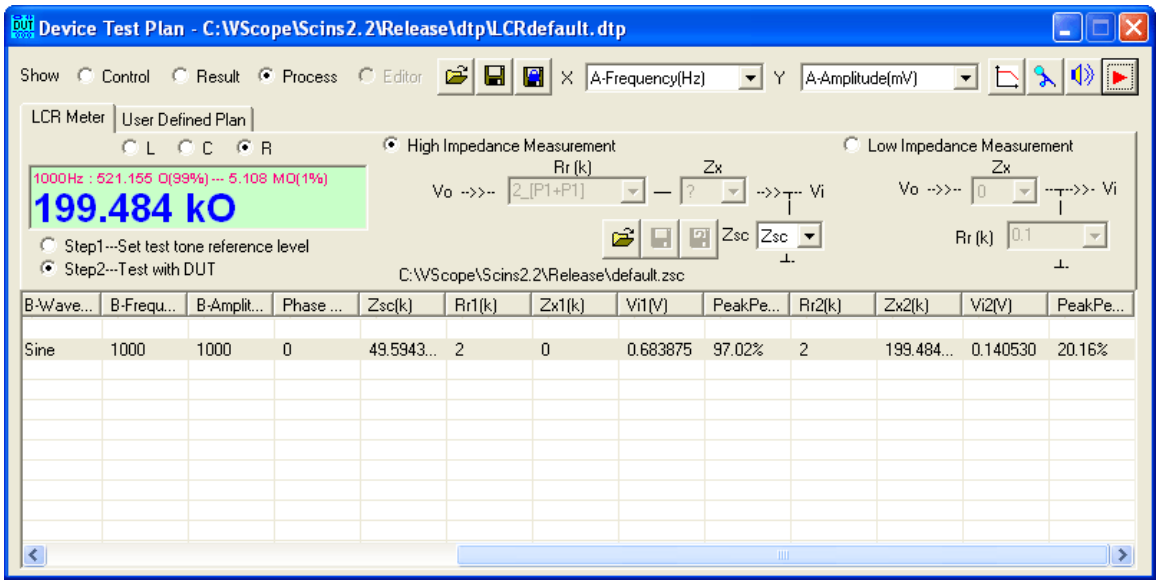


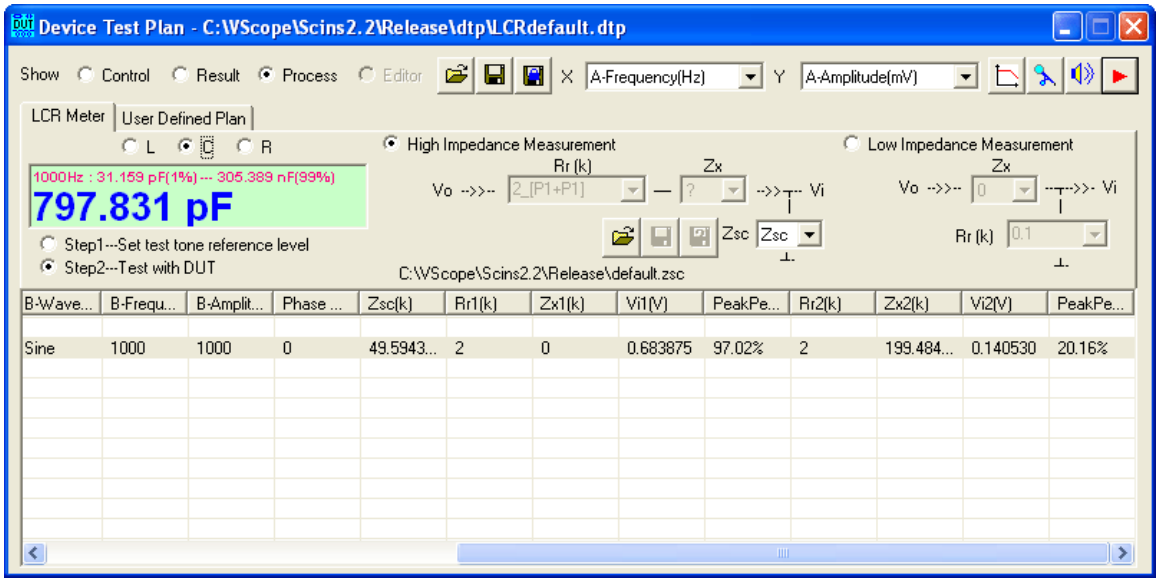
9.6.3.2 Step2---Test with DUT

Keep the test tone reference level intact, connect the DUT (Zx) between the two test leads, select Step 2 and start the plan. The impedance value measured will then be displayed in the Result window and Zx2 column.

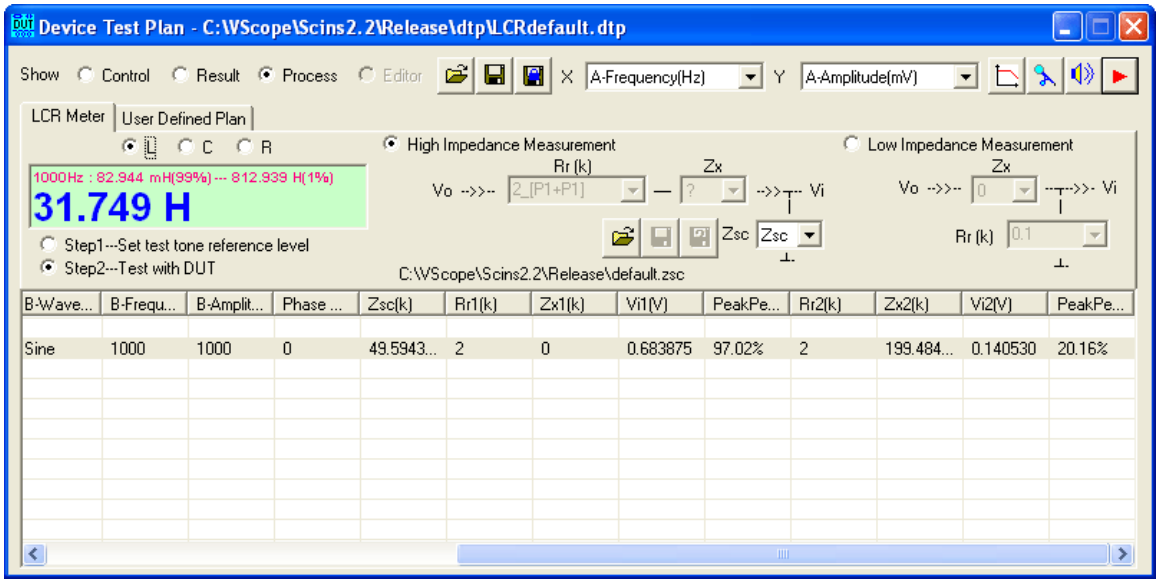
In the following example, we used 200 kΩ resistor with a tolerance value of 1% for Zx, and the measured value was 199.484 kΩ, which is very accurate. Note that the sound card input and output impedance here were about 50 kΩ and 100 Ω respectively and the output impedance was ignored without any compensation.



If Zx is a capacitor with the same impedance value at 1000 Hz, then the capacitance would be 797.831 pF.



If Zx is an inductor with the same impedance value at 1000 Hz, then the inductance would be 31.749 H.



9.7 Low Impedance Measurement

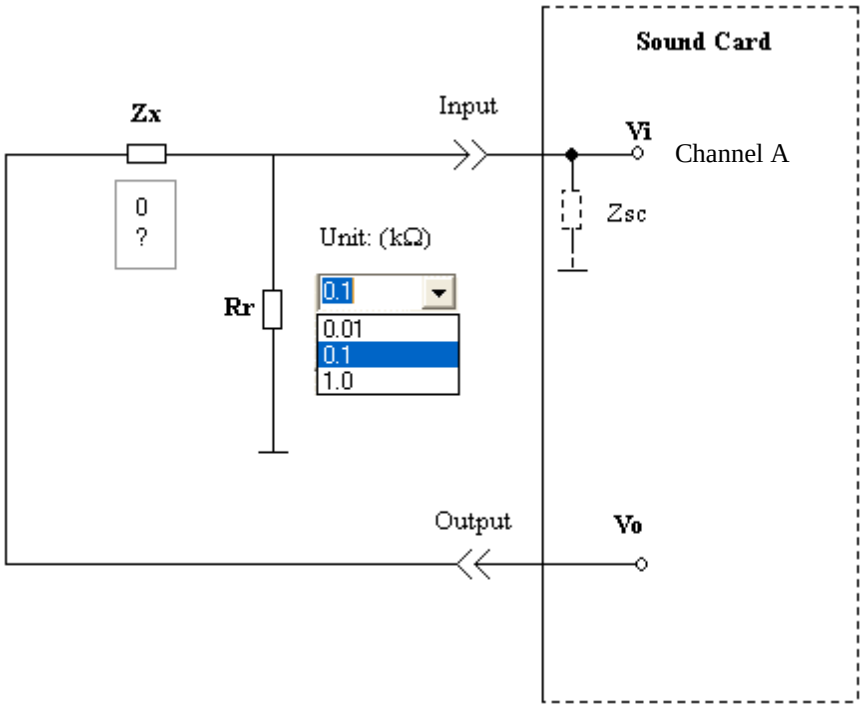
Low Impedance Measurement is used to measure impedance in a lower range. The reference resistor is placed in parallel to the sound card input impedance, and if its value is much lower than the sound card input impedance, which holds true in most of cases, the sound card input impedance can be ignored. Thus, it is not necessary to calibrate the sound card input impedance before measurement. However, if the default sound card input impedance file exists or a sound card input impedance file has been loaded, the sound card input impedance will be automatically taken into account for better accuracy.

The LCR Meter will be able to display the measurement range based on the reference resistor value and the test frequency used. The software uses 1% and 99% variation from the test tone reference level to suggest the measurement range, within which good measurement accuracy can be achieved. This is similar to the case of resistor measurement using an analog multimeter whereby the middle region of the swing of the needle has good measurement accuracy.

The procedure to make a LCR measurement is also similar to the procedure to measure a resistor using an analog multimeter, as described previously.

9.7.1 Connection for Low Impedance Measurement

The connection diagram for low impedance measurement is shown as follows.



where:
Rr is the reference resistor.
Zx is the impedance to be measured.
Zsc is the sound card input impedance.
Vo is the output RMS voltage.
Vi is the input RMS voltage.

(Rr|Zsc) and Zx form a voltage divider and thus we have Vi/Vo=(Rr|Zsc)/((Rr|Zsc)+Zx), where (Rr|Zsc) is the resultant resistance when Rr and Zsc are connected in parallel. **It should be noted that the sound card output impedance, which typically ranges from nearly zero ohm to a few ohms for Speaker/Headphone Out, is ignored in the above formula. This simplification has negligible effect on the measurement accuracy as long as the output impedance is negligibly small compared with the value of Zx+(Rr|Zsc), which holds true in most of the cases.**

Rr is an external reference resistor. You need to find the resistor and make the connection by yourself. In the Rr combo box, you should enter the actual value (in kΩ) you used. Different measurement ranges requires different reference resistor values. Some pre-configured resistance values are available for selection: 0.01 k, 0.1 k, 1.0 k.

If you are not using the above values, then you have to enter the resistance value manually. For advanced users, if you have your own set of reference resistors, you can modify the TXT file named "ResistorRef2.txt" under the software root directory so that your can have your own set of reference resistor values available in the Rr combo box.

Zx is the impedance to be measured. The Zx combo box will display "0" for Step 1 and "?" for Step 2. In other word, in Step 1, Zx should be bypassed and in Step 2, Zx should be connected.

9.7.2 Make a LCR Measurement

As described previously, the procedure to make a LCR measurement is similar to the procedure to measure a resistor using an analog multimeter. You need to take two steps to complete a measurement at first, and then the first step can be skipped if the measurement range has not been changed.

9.7.2.1 Step1--- Set Test Tone Reference Level

Choose a proper measurement range by connecting a proper reference resistor Rr and enter the corresponding value in the Rr combo box on the screen. Then, short the two test leads to bypass Zx, select Step 1, start the plan and adjust the output level via Windows Volume Control, and/or the input gain via Windows Recording Control, in order to set the test tone reference level such that the maximum allowable level is reached to ensure sufficient measurement accuracy. A peak level above 85% is recommended. Stop the plan after the test tone reference level is set.

In the following example, we used a 1kΩ resistor for Rr and the peak level at the test frequency (1000 Hz) was set to 98.33%.

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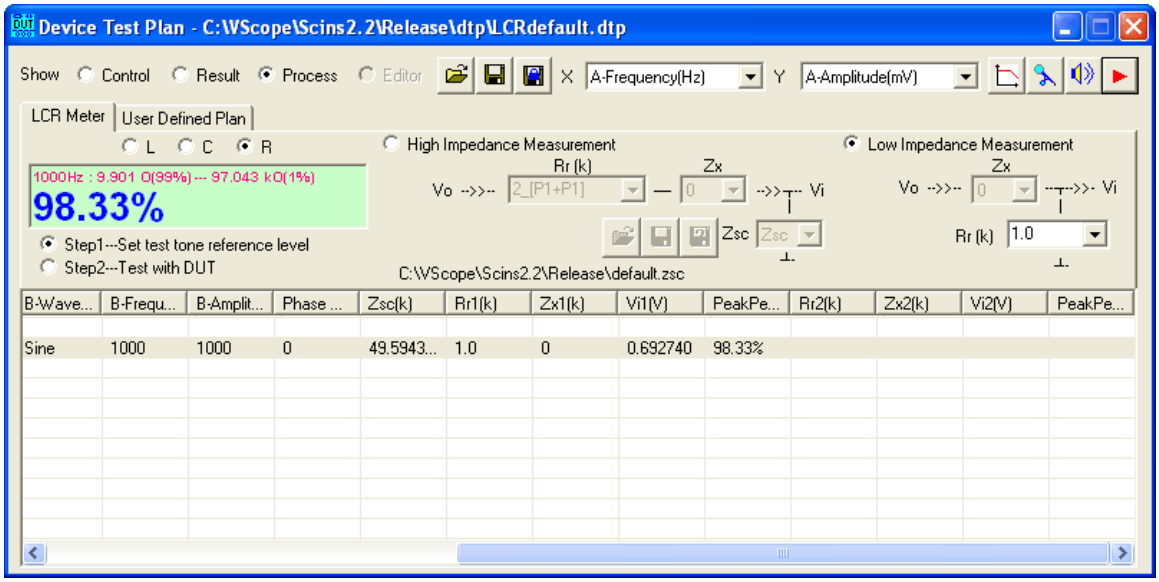
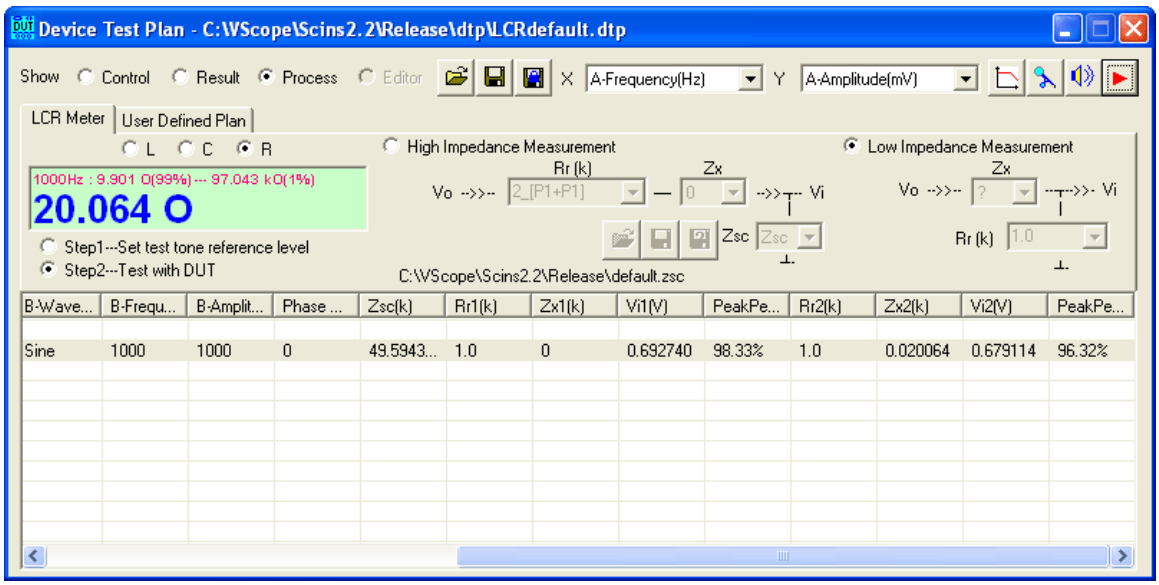


9.7.2.2 Step2---Test with DUT

Keep the test tone reference level intact, connect the DUT (Zx) between the two test leads, select Step 2 and start the plan. The impedance value measured will then be displayed in the Result window and Zx2 column.

In the following example, we used 20 Ω resistor with a tolerance value of 1% for Zx, and the measured value was 20.064 kΩ, which is very accurate. Note that the sound card input and output impedance here were about 50 k Ω and 100 Ω respectively and the output impedance was ignored without any compensation.



9.8 Measurement Accuracy

The following points should be noted in order to achieve high accuracy in LCR measurements:

- High-precision reference resistor should be used.
- Use a sound card with low output impedance.
- Set measurement range correctly. Change the reference resistor value or the test frequency will change the range of measurement. The highest measurement accuracy is expected if the peak level in Step 2 is half of that in Step 1 and the peak level in Step 1 is close to 100%. This is similar to the case of measuring a resistor using an analog multimeter whereby the middle of the needle swing range has the highest measurement accuracy.

On the other hand, the frequency response of the sound card does not affect the accuracy in LCR measurements. Also, calibration of the sound card input and output channels is not required here.

9.9 Measurement with Multiple Test Frequencies

If you configure a LCR test plan with multiple test frequencies in multiple test steps, the averaged resistance, capacitance or inductance value will be displayed in the Result window. Some sample LCR test plans beside the default one for LCR measurements are provided under the DTP (Device Test Plan) directory of the software. These test plans (with prefix "LCR" in their file names) are configured with different test frequencies which allow you the set different measurement ranges without changing the reference resistor.

The following figure shows a measurement of a capacitor using multiple test frequencies. The X-Y Plot illustrates its impedance variation with regard to frequency.

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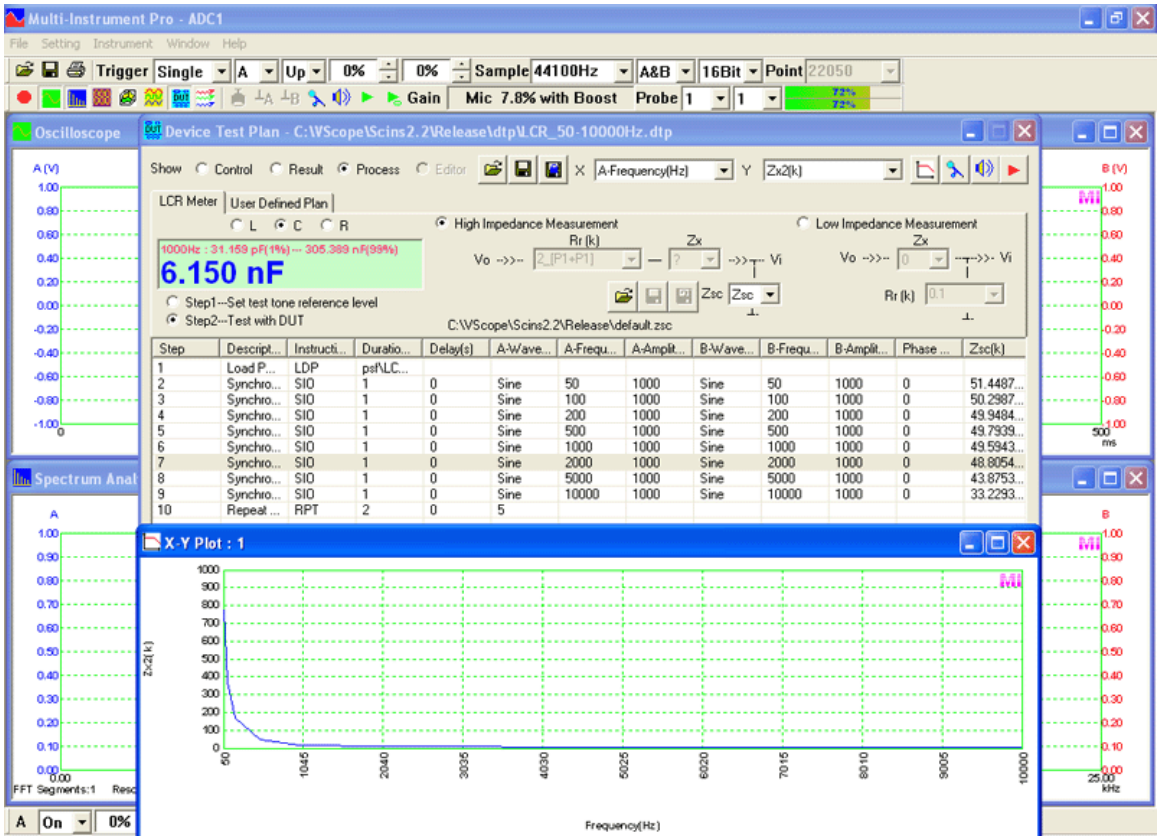

Multi-Instrument Pro - ADC1
Oscilloscope
Device Test Plan - C:\VScope\Scins2.2\Release\dtp\LCR_50-10000Hz.dtp
LCR Meter
User Defined Plan
6.150 nF
Spectrum Anal
X-Y Plot : 1
Frequency(Hz)

10 Derived Data Point (DDP) Viewer

10.1 Overview

DDP viewer is used to display the value of a DDP (Derived Data Point) in a dedicated window with a bigger font size. It is also possible to specify the DDP's high-high, high, low, low-low limits for alarming and the number of decimal places for display. Different alarm sounds can be configured for different types of alarms. Alarm acknowledgement is supported. Up to 32 DDP viewers can be opened. These DDP viewers can also be used to define and display UDDPs (User Defined Data Points)

The push button in the Instrument Toolbar is used to open a new DDP viewer window. You can close it by clicking the "Close" button at the upper right corner of the window.

DDP array viewer is used to display arrays of DDPs. It can be launched from a DDP viewer's configuration dialog box. Only one DDP array viewer is supported in the software.

10.2 Configuration

10.2.1 Settings for both DDP and UDDP

Whenever the "DDP Viewer" button in the Instrument Toolbar is clicked, a new DDP Viewer window will be opened. A configuration dialog box (see figure below) will pop up on the top of the DDP Viewer window to allow you to enter the configuration first. Parameters to be configured include the DDP to be viewed, the alias of the DDP which will be displayed in the title bar of the DDP Viewer, the high-high, high, low, low-low limits for alarming and their respective background colors. The alarms will be enabled only if the respective checkboxes are ticked. The number of decimal places will be set according to the DDP's source precision by default. You can change it thereafter.

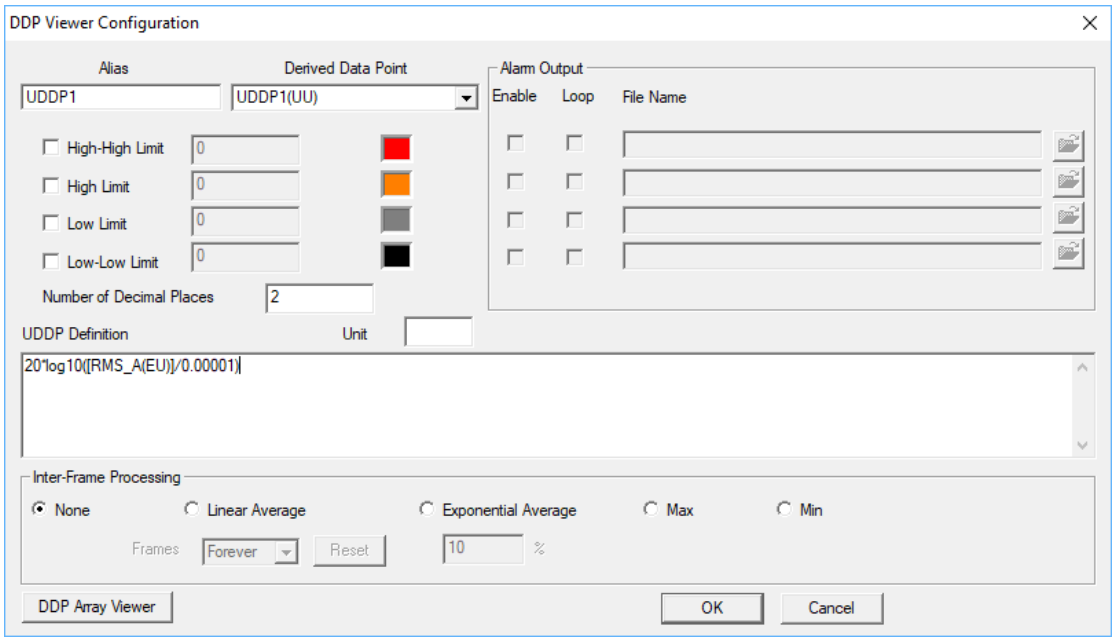
Every alarm limit can be assigned with a particular alarm sound. The alarm sound can be defined using a WAV file. If no WAV file is configured, a default alarm sound will be used. The alarm sound can be configured to play once or cyclically when alarm occurs. If the alarm output is enabled, it will be activated when the alarm limit is broken. High-High Limit alarm has a higher priority than High Limit alarm. Low-Low Limit alarm has a higher priority than Low Limit alarm. To prevent different

types of alarm sounds from toggling too frequently, a 3 seconds' hysteresis is used internally. If the alarm sound is configured to play cyclically, a speaker icon will appear at the lower right corner of the DDP viewer when the alarm occurs, clicking that icon will acknowledge the alarm and stop the alarm sound. The alarm sound will be played cyclically until it is acknowledged even though the actual DDP value has returned to normal.

The alarm statuses of all opened DDP viewers are summarized into one DDP: OverallPassFailCheck. It is evaluated by the software after all DDPs and UDDPs have already been updated. It has three values: "Nil", "Pass", "Fail". If none of the opened DDP viewer is configured for alarm checking, "Nil" will be displayed (=0). If any opened DDP viewer has an alarm, "Fail" will be displayed (<0, the absolute value represents the number of fails). If no alarm in all the opened DDP viewers, then "Pass" will be displayed (>0, the absolute value represents the number of passes).

10.2.2 Define a UDDP

A UDDP is usually a function of certain DDPs. It gets updated after all DDPs are updated. Each DDP Viewer can be used to define and display one UDDP. For example, DDP Viewers 1 and 16 can be used to define and display UDDP1 and UDDP16 respectively. UDDPs are updated in a predefined order with UDDP1 being the first and UDDP16 being the last. The earlier-updated UDDPs can be used as variables in the functions that define the later-updated UDDPs.



A UDDP can be expressed as a function of DDPs, Panel Setting Parameters and earlier-updated UDDPs. The mathematical expression may consist of constants, DDPs, Panel Setting Parameters, UDDPs, mathematical operators, and mathematical functions. Spaces and Carriage Returns in the expression will be ignored.

Constants must be written in the forms of -x, -x.x, x, or x.x, e.g. –25, -49.55, 38, 71.45. Constants expressed in exponential forms such as 2E10 are not accepted.

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DDPs, Panel Setting Parameters and UDDPs must be referred to using their names bracketed by "[]", e.g. [Max_A(EU)], [RMS_B(EU)], [SamplingFrequency], [UDDP1_(UU)]. The following Panel Setting Parameters are supported:

Name	Note
SamplingFrequency	Unit: Hz
SamplingChannels	Number of Sampling Channels
SamplingResolution	Unit: Bit
RecordLength	Unit: Sample
SpectrumAnalyzerType	Index starting from zero
FFTSize	Unit: Point
WindowFunction	Index starting from zero
WindowOverlapPercentage	Unit: %, Range: 0~100
SharedMemorySize	Unit: Byte

Mathematical operators supported are: +, -, *, /, ^, where * represents × and ^ is used to raise a number to the power of an exponent (number ^ exponent). Parentheses "()" can be used to specify the order of operation. Arbitrary nesting of parentheses is allowed.

Mathematical functions supported are listed in the following table. The variables of a function can be constants, DDPs, UDDPs, or functions. A function can be called recursively.

No.	Function	Description
1	SIN(x)	Sine of x, where x is in radians
2	COS(x)	Cosine of x, where x is in radians
3	TAN(x)	Tangent of x, where x is in radians
4	SUM(x_1,x_2,…x_n)	Sum of x_1, x_2, ...x_n
5	LOG(x)	Base-e logarithm of x
6	LOG10(x)	Base-10 logarithm of x
7	ABS(x)	Absolute value of x
8	MAX(x_1,x_2,…x_n)	Maximum value of x_1, x_2, ...x_n
9	MIN(x_1,x_2,…x_n)	Minimum value of x_1, x_2, ...x_n
10	POW(x,y)	X raised to a power of y
11	SQRT(x)	Square root of x
12	IFGT(x,y,z,w)	If x>y, return z, otherwise return w. z and w can be either a number or a ASCII character string enclosed in "".
13	IFGE(x,y,z,w)	If x≥y, return z, otherwise return w. z and w can be either a number or a ASCII character string enclosed in "".
14	IFLT(x,y,z,w)	If x<y, return z, otherwise return w. z and w can be either a number or a ASCII character string enclosed in "".
15	IFLE(x,y,z,w)	If x≤y, return z, otherwise return w. z and w can be either a number or a ASCII character string enclosed in "".

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16	IFEQ(x,y,z,w)	If x=y, return z, otherwise return w. z and w can be either a number or a ASCII character string enclosed in “”.
17	IFNE(x,y,z,w)	If x≠y, return z, otherwise return w. z and w can be either a number or a ASCII character string enclosed in “”.
18	IFIN(x,l,h,z,w)	If l≤x≤h, return z, otherwise return w. z and w can be either a number or a ASCII character string enclosed in “”.
19	ASIN(x)	Arc sine of x, where -1≤x≤1
20	ACOS(x)	Arc cosine of x, where -1≤x≤1
21	ATAN(x)	Arc tangent of x

A UDDP can also be set by an external ActiveX automation client program via SetUDDP(). No definition is required for the UDDP in this case. Please refer to Multi-Instrument_Automation_Server_Interfaces.pdf in the subdirectory \AutomationAPIs for details.

The unit of a UDDP can be specified by the user. The UDDP function greatly expands the software’s capability in direct measurements of various parameters. Some UDDP definition examples are shown in the following sections.

10.2.2.1 Calculate Total dBV from 8 User Defined Frequency Bands

Settings:
- [Spectrum Analyzer Y Scale]: Vrms
- [Spectrum Analyzer Processing]: Energy in User Defined Frequency Bands

UDDP1:
20*LOG10(SQRT(SUM([fBand1RMS_A(EU)]^2,[fBand2RMS_A(EU)]^2,
[fBand3RMS_A(EU)]^2,[fBand4RMS_A(EU)]^2,[fBand5RMS_A(EU)]^2,
[fBand6RMS_A(EU)]^2,[fBand7RMS_A(EU)]^2,[fBand8RMS_A(EU)]^2)))

10.2.2.2 DTMF Decoder in a Noisy Background

Settings:
[Spectrum Analyzer Processing]: Peaks with [DeadBand]=50, [Number of Peaks]=6
It is assumed that the dual tone peaks are among the detected 6 frequency peaks.

UDDP1:
IFIN([f1Freq_A(Hz)],692,702,1,0) +
IFIN([f1Freq_A(Hz)],765,775,2,0) +
IFIN([f1Freq_A(Hz)],847,857,3,0) +
IFIN([f1Freq_A(Hz)],936,946,4,0) +
IFIN([f1Freq_A(Hz)],1204,1214,10,0) +
IFIN([f1Freq_A(Hz)],1331,1341,20,0) +
IFIN([f1Freq_A(Hz)],1472,1482,30,0) +
IFIN([f1Freq_A(Hz)],1628,1638,40,0) +

IFIN([f2Freq_A(Hz)],692,702,1,0) +
IFIN([f2Freq_A(Hz)],765,775,2,0) +
IFIN([f2Freq_A(Hz)],847,857,3,0) +
IFIN([f2Freq_A(Hz)],936,946,4,0) +
IFIN([f2Freq_A(Hz)],1204,1214,10,0) +
IFIN([f2Freq_A(Hz)],1331,1341,20,0) +
IFIN([f2Freq_A(Hz)],1472,1482,30,0) +
IFIN([f2Freq_A(Hz)],1628,1638,40,0) +

IFIN([f3Freq_A(Hz)],692,702,1,0) +
IFIN([f3Freq_A(Hz)],765,775,2,0) +
IFIN([f3Freq_A(Hz)],847,857,3,0) +
IFIN([f3Freq_A(Hz)],936,946,4,0) +
IFIN([f3Freq_A(Hz)],1204,1214,10,0) +
IFIN([f3Freq_A(Hz)],1331,1341,20,0) +
IFIN([f3Freq_A(Hz)],1472,1482,30,0) +
IFIN([f3Freq_A(Hz)],1628,1638,40,0) +

IFIN([f4Freq_A(Hz)],692,702,1,0) +
IFIN([f4Freq_A(Hz)],765,775,2,0) +
IFIN([f4Freq_A(Hz)],847,857,3,0) +
IFIN([f4Freq_A(Hz)],936,946,4,0) +
IFIN([f4Freq_A(Hz)],1204,1214,10,0) +
IFIN([f4Freq_A(Hz)],1331,1341,20,0) +
IFIN([f4Freq_A(Hz)],1472,1482,30,0) +
IFIN([f4Freq_A(Hz)],1628,1638,40,0) +

IFIN([f5Freq_A(Hz)],692,702,1,0) +
IFIN([f5Freq_A(Hz)],765,775,2,0) +
IFIN([f5Freq_A(Hz)],847,857,3,0) +
IFIN([f5Freq_A(Hz)],936,946,4,0) +
IFIN([f5Freq_A(Hz)],1204,1214,10,0) +
IFIN([f5Freq_A(Hz)],1331,1341,20,0) +
IFIN([f5Freq_A(Hz)],1472,1482,30,0) +
IFIN([f5Freq_A(Hz)],1628,1638,40,0) +

IFIN([f6Freq_A(Hz)],692,702,1,0) +
IFIN([f6Freq_A(Hz)],765,775,2,0) +
IFIN([f6Freq_A(Hz)],847,857,3,0) +
IFIN([f6Freq_A(Hz)],936,946,4,0) +
IFIN([f6Freq_A(Hz)],1204,1214,10,0) +
IFIN([f6Freq_A(Hz)],1331,1341,20,0) +
IFIN([f6Freq_A(Hz)],1472,1482,30,0) +
IFIN([f6Freq_A(Hz)],1628,1638,40,0)

UDDP2:
IFEQ([UDDP1(UU)],24,1,0) +
IFEQ([UDDP1(UU)],11,2,0) +
IFEQ([UDDP1(UU)],21,3,0) +

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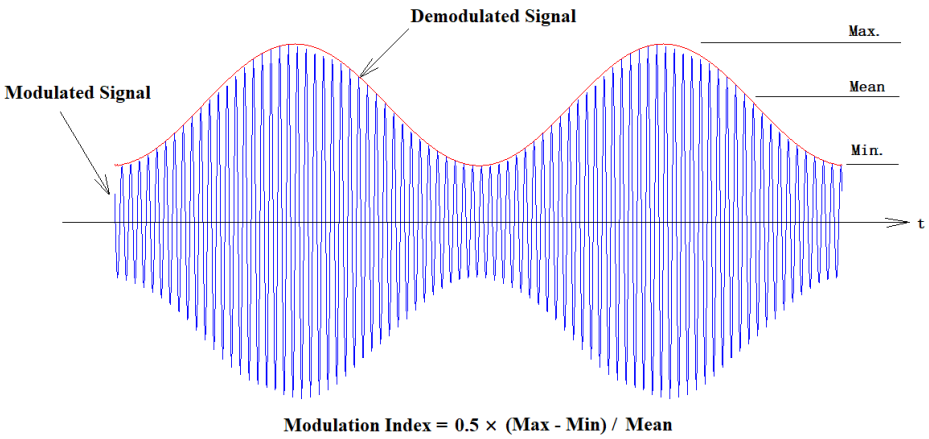
```
IFEQ([UDDP1(UU)],31,4,0) +
IFEQ([UDDP1(UU)],12,5,0) +
IFEQ([UDDP1(UU)],22,6,0) +
IFEQ([UDDP1(UU)],32,7,0) +
IFEQ([UDDP1(UU)],13,8,0) +
IFEQ([UDDP1(UU)],23,9,0) +
IFEQ([UDDP1(UU)],33,10,0) - 1
```

UDDP1 is an intermediate variable. UDDP2 will display the decoded DTMF character "0" ~ "9". "-1" will be displayed otherwise.

10.2.2.3 Amplitude Modulation Index Measurement



To measure the amplitude modulation index (or amplitude modulation depth), the modulated signal needs to be demodulated first. Then the modulation index can be obtained as follows:

[Modulation Index] = 0.5×(Max-Min)/Mean

where Max, Min and Mean are the maximum, minimum and mean values of the demodulated signal. One issue with the above formula is that the Max and Min values are sensitive to noise and the boundary effect of the demodulation process. A low pass filter after demodulation can be used to alleviate this issue. However, here we use the following formula instead. It can be derived from the above formula.

$$[Modulation\ Index] = [Crest\ Factor] \times \sqrt{\left(\frac{RMS}{Mean}\right)^2 - 1}$$

The above formula is more robust as it uses RMS value instead of Max and Min. For amplitude modulation using a sine wave, the crest factor is 1.414.

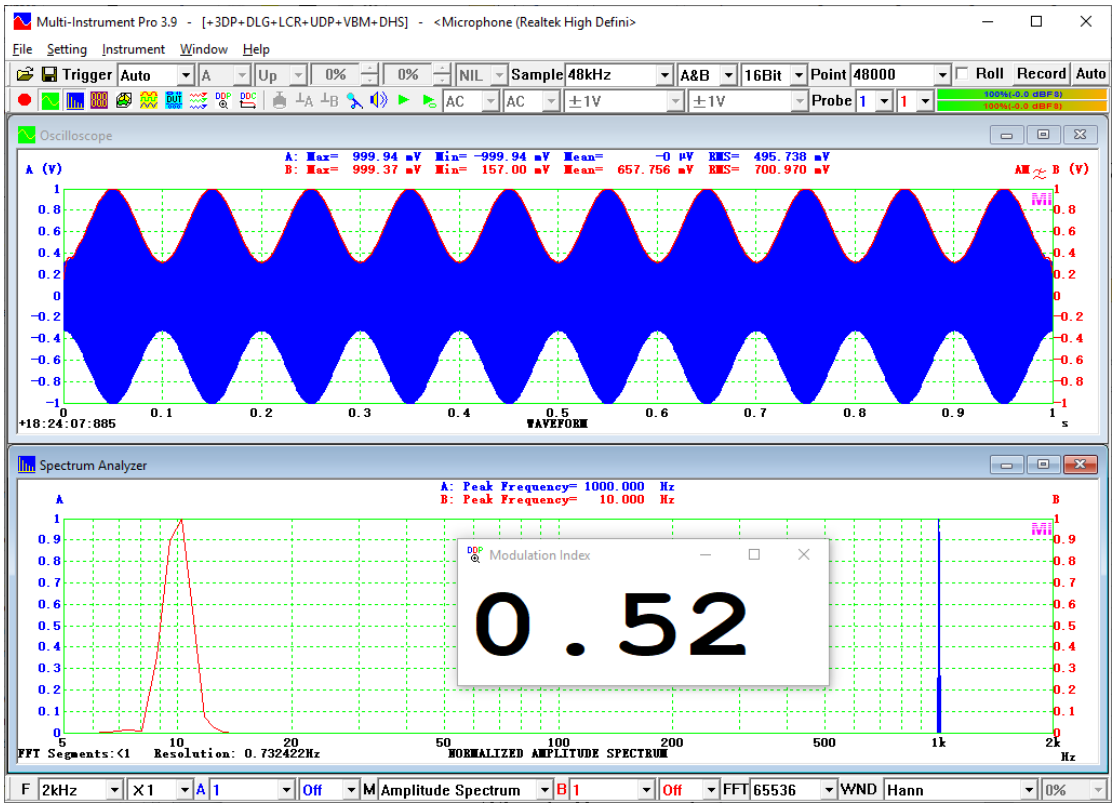
Prerequisite: For comparison purpose, the input channels A and B are fed with the same amplitude-modulated signal. Modulating frequency is 10Hz.

Settings:

[Oscilloscope Processing]: AM Demodulation and applied to Ch. B only.
[Oscilloscope Processing]: 100Hz low pass filter and applied to Ch. B only.

UDDP1:
1.414*SQRT(([RMS_B(EU)]/[Mean_B(EU)])^2-1)

The following figure shows a measurement example. The carrier used is 1kHz sine wave. Other carriers such as white noise and pink noise can also be used.



10.2.3 Inter-frame Processing

The DDP value will be refreshed accordingly when a new frame of data is sampled. Three options are available for the inter-frame processing of the DDP: None, Linear Average and Exponential Average. It should be noted that the inter-frame processed DDP value is only local to the specific DDP viewer and is not accessible by any other instrument such as Data Logger and Device Test Plan. The number of frame averaged can be accessed through DDP: NumberOfFrames.

10.2.3.1 Linear Average

If Linear Average is selected, the DDP viewer will keep track of the DDP values and only display the averaged value of the specified number of data frames acquired most recently. The number of frames averaged will be displayed in the title bar of the DDP viewer. The average is simply an arithmetic mean.

You can specify the number of contiguous frames to be processed. The available options are: 2, 3, 4, 5, 6, 7, 8, 9, 10, 20, 30, 40, 50, 60, 70, 80, 90, 100, 120, 140, 160, 180, 200 and forever. You can also enter any number between 1 and 200 manually. When "Forever" is selected, you can reset the process using the Reset button when necessary.

10.2.3.2 Exponential Average

Unlike Linear Average where all data frames used for average are given equal weights, in Exponential Average, the weighting factor for each data frame decreases with time exponentially, giving much more importance to recent observations while still not discarding older observations entirely. The degree of weighting decrease is expressed as a constant α in percentage. The greater the α, the faster the decrease. Alternatively, α may be expressed in terms of N, where $\alpha = 2/(N+1)$, and $N \times$ [time interval between the start times of two adjacent data frames] is called the time constant.

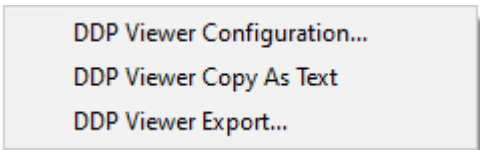
10.2.3.3 Max

The Max option is similar to the Linear Average option. However, instead of displaying the linearly averaged value, it displays the maximum value during the specified period.

10.2.3.4 Min

The Min option is similar to the Linear Average option. However, instead of displaying the linearly averaged value, it displays the minimum value during the specified period.

10.3 Context Menu



DDP Viewer does not have its own menu and view parameter toolbar. Its functions are accessed via its context menu. The context menu pops up when you right click anywhere within the DDP Viewer window. There are three menu items in the context menu:

- DDP Viewer Configuration
  It has been described in the previous section.

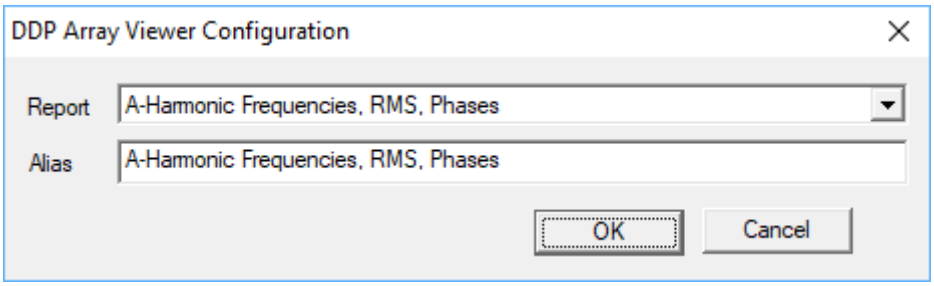
- DDP Viewer Copy As Text
  This command copies the data in all opened DDP Viewers to the clipboard. They can be subsequently pasted out into other programs such as Microsoft Excel.

- DDP Viewer Export
  This command exports the data in all opened DDP Viewers to a text file.

10.4 DDP Array Viewer

There are a few DDP arrays defined in Multi-Instrument, such as fnFreq_A(Hz)_Array[?], fnRMS_A(EU)_Array[?], fnPhase_A(D)_Array[?], oBandRMS_A(EU)_Array[?] for Channel A. It is possible to use the DDP viewer to view an element of an array. However, if you want to view the whole array, or a combination of a few arrays, then DDP array viewer should be used instead. To open the DDP array viewer, click the "DDP Viewer" button in the Instrument Toolbar, and then in the pop-up DDP viewer configuration dialog box, click "DDP Array Viewer". The following configuration dialog box will pop up.

10.4.1 Configuration



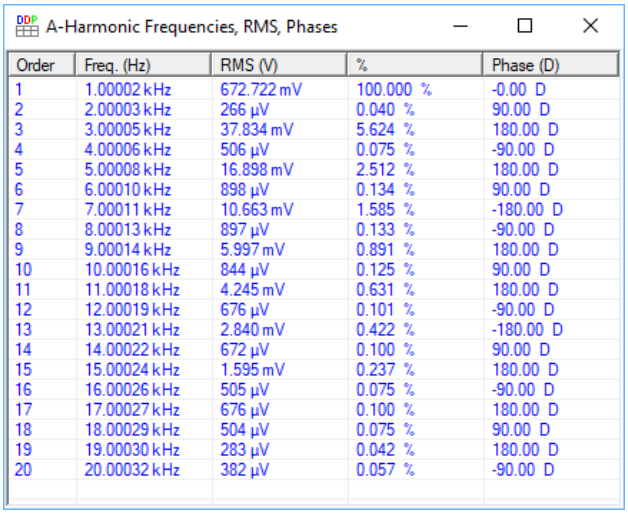
A report consists of one or more DDP arrays and/or derived arrays from them. There are 12 preconfigured reports available for selection. It is possible to assign an alias to the report title.

10.4.2 Reports

10.4.2.1 Reports for Harmonic Frequencies, RMS, Phases

Three reports are available for Channel A, Channel B and Channels A&B respectively:

(1) A-Harmonic Frequencies, RMS, Phases
(2) B-Harmonic Frequencies, RMS, Phases
(3) A&B-Harmonic Frequencies, RMS, Phases



A-Harmonic Frequencies, RMS, Phases

Order	Freq. (Hz)	RMS (V)	%	Phase (D)
1	1.00002 kHz	672.722 mV	100.000 %	-0.00 D
2	2.00003 kHz	266 µV	0.040 %	90.00 D
3	3.00005 kHz	37.834 mV	5.624 %	180.00 D
4	4.00006 kHz	506 µV	0.075 %	-90.00 D
5	5.00008 kHz	16.898 mV	2.512 %	180.00 D
6	6.00010 kHz	898 µV	0.134 %	90.00 D
7	7.00011 kHz	10.663 mV	1.585 %	-180.00 D
8	8.00013 kHz	897 µV	0.133 %	-90.00 D
9	9.00014 kHz	5.997 mV	0.891 %	180.00 D
10	10.00016 kHz	844 µV	0.125 %	90.00 D
11	11.00018 kHz	4.245 mV	0.631 %	180.00 D
12	12.00019 kHz	676 µV	0.101 %	-90.00 D
13	13.00021 kHz	2.840 mV	0.422 %	-180.00 D
14	14.00022 kHz	672 µV	0.100 %	90.00 D
15	15.00024 kHz	1.595 mV	0.237 %	180.00 D
16	16.00026 kHz	505 µV	0.075 %	-90.00 D
17	17.00027 kHz	676 µV	0.100 %	180.00 D
18	18.00029 kHz	504 µV	0.075 %	90.00 D
19	19.00030 kHz	283 µV	0.042 %	180.00 D
20	20.00032 kHz	382 µV	0.057 %	-90.00 D

Using the above report for Channel A as an example, the five columns are: Order of harmonics, Frequency of harmonics, RMS values of harmonics, Normalized harmonic amplitudes with respect to that of the fundamental (in percentage), Phases of harmonics (of a sine function).

10.4.2.2 Report for RMS values in Octave Bands

Three reports are available for Channel A, Channel B and Channels A&B respectively:

(1) A-Octave Bands, RMS
(2) B-Octave Bands, RMS
(3) A&B-Octave Bands, RMS

Using the following report for Channel A as an example, the two columns are: Octave band center frequencies and RMS values of the octave bands.

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A-Octave Bands, RMS

Center Freq. (Hz)	RMS (dBV)
1.00 Hz	-64.73 dBV
1.25 Hz	-63.73 dBV
1.60 Hz	-62.72 dBV
2.00 Hz	-61.72 dBV
2.50 Hz	-46.06 dBV
3.15 Hz	-45.02 dBV
4.00 Hz	-45.12 dBV
5.00 Hz	-46.18 dBV
6.30 Hz	-51.18 dBV
8.00 Hz	-45.26 dBV
10.00 Hz	-47.62 dBV
12.50 Hz	-44.11 dBV
16.00 Hz	-44.57 dBV
20.00 Hz	-40.57 dBV
25.00 Hz	-46.39 dBV
31.50 Hz	-39.38 dBV
40.00 Hz	-39.84 dBV
50.00 Hz	-39.04 dBV
63.00 Hz	-38.50 dBV
80.00 Hz	-35.96 dBV
100.00 Hz	-34.86 dBV
125.00 Hz	-33.91 dBV
160.00 Hz	-32.32 dBV
200.00 Hz	-33.06 dBV
250.00 Hz	-31.19 dBV
315.00 Hz	-29.71 dBV
400.00 Hz	-29.09 dBV
500.00 Hz	-27.93 dBV
630.00 Hz	-26.61 dBV
800.00 Hz	-26.19 dBV
1.00000 kHz	-25.05 dBV
1.25000 kHz	-23.86 dBV
1.60000 kHz	-23.10 dBV
2.00000 kHz	-22.12 dBV
2.50000 kHz	-20.61 dBV
3.15000 kHz	-20.10 dBV
4.00000 kHz	-18.68 dBV
5.00000 kHz	-18.07 dBV
6.30000 kHz	-17.00 dBV
8.00000 kHz	-15.87 dBV
10.00000 kHz	-14.81 dBV
12.50000 kHz	-14.06 dBV
16.00000 kHz	-12.85 dBV
20.00000 kHz	-11.72 dBV

10.4.2.3 Report for Peak Frequencies, RMS, Phases

Three reports are available for Channel A, Channel B and Channels A&B respectively:

(1) A-Peak Frequencies, RMS, Phases
(2) B-Peak Frequencies, RMS, Phases
(3) A&B-Peak Frequencies, RMS, Phases

These reports are similar to those harmonic reports described previously except that the peak frequencies here are not necessary to be harmonically related.

10.4.2.4 Report for RMS values in User Defined Frequency Bands

Three reports are available for Channel A, Channel B and Channels A&B respectively:

(1) A-Frequency Bands, RMS
(2) B-Frequency Bands, RMS
(3) A&B-Frequency Bands, RMS

These reports are similar to those reports for RMS values in octave bands described previously except that the frequency bands here are defined freely by the users.

10.4.2.5 Report for Reverberation / Speech Intelligibility

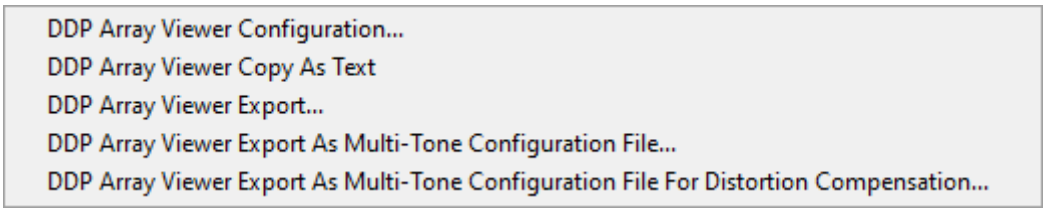
Four reports are available: Two for Channel A and two for Channel B.

(1) A-Reverberation / Speech Intelligibility (1/1 Octave)
(2) A-Reverberation / Speech Intelligibility (1/3 Octave)
(3) B-Reverberation / Speech Intelligibility (1/1 Octave)
(4) B-Reverberation / Speech Intelligibility (1/3 Octave)

These reports have been introduced in the Oscilloscope chapter.

10.4.3 Context Menu

DDP Array Viewer does not have its own menu and view parameter toolbar. Its functions are accessed via its context menu. The context menu pops up when you right click anywhere within the DDP Array Viewer window.



- DDP Array Viewer Configuration
  This has been described in the previous section.

- DDP Array Viewer Copy As Text
  This command copies the data in the report to the clipboard. They can be subsequently pasted out into other programs such as Microsoft Excel. Note the data in the clipboard is Tab separated.

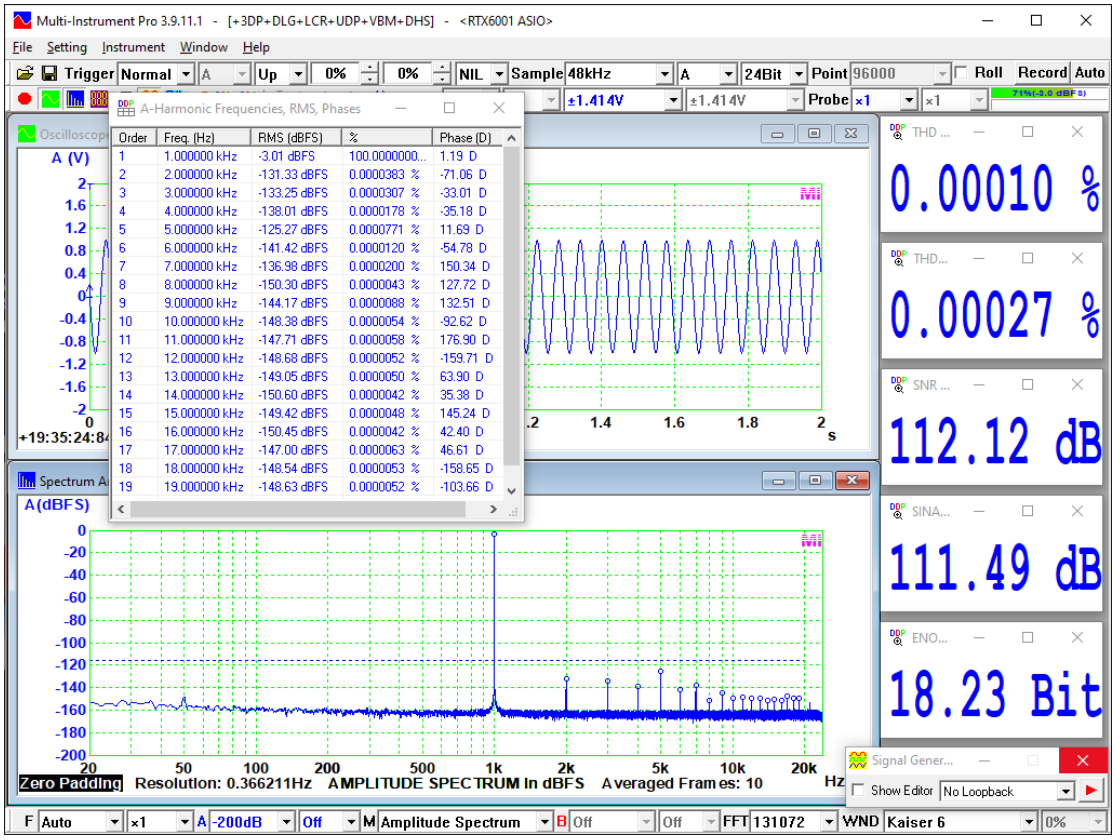
- DDP Array Viewer Export
  This command exports the data in the report to a text file which can be imported into third party software such as Microsoft Excel for further processing and analysis.

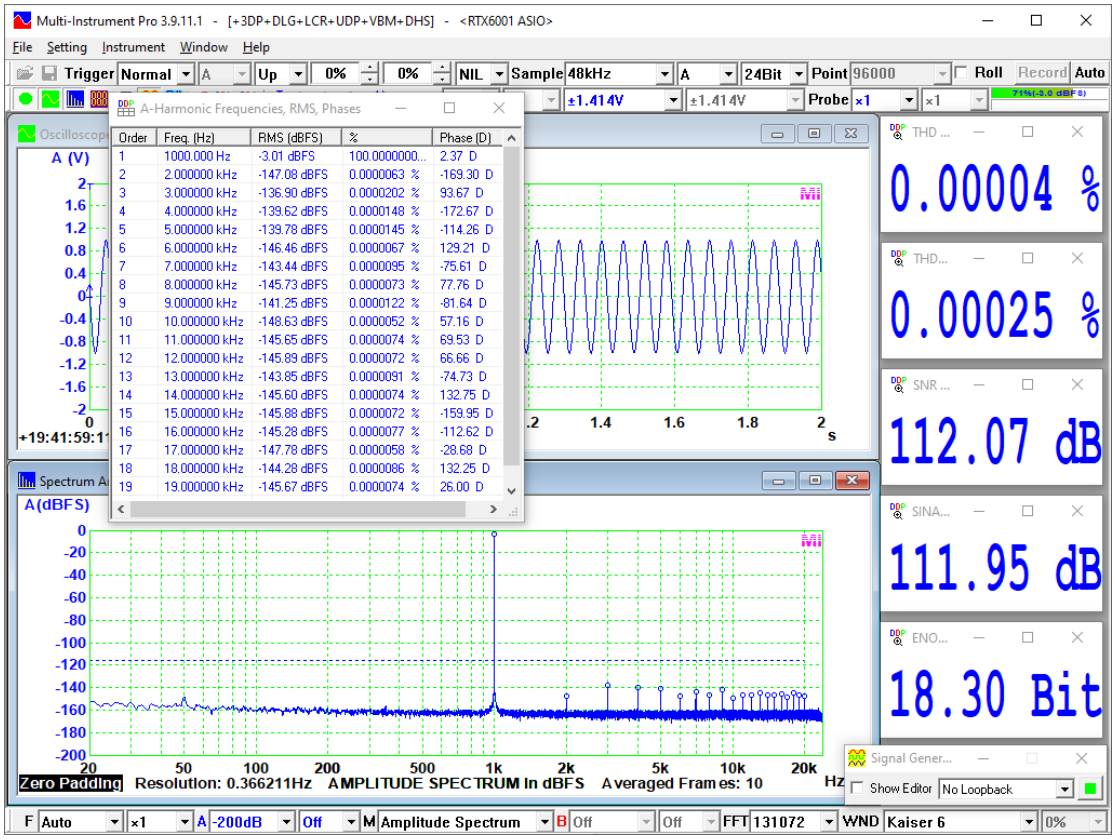
- DDP Array Viewer Export As Multi-Tone Configuration File
  This option will be enabled in the harmonic / peak analysis reports. It exports the data in the report to a Tone Configuration File (*.tcf) which can be imported into the Multitone Configuration dialog box in order to generate a multitone from the Signal Generator. In other words, a signal can be decomposed into a series of single frequency components, with only their amplitudes and phases recorded in the Tone Configuration File (much smaller

in size compared with a WAV file). It can be then reconstructed from that file and regenerated from the Signal Generator.

- DDP Array Viewer Export As Multi-Tone Configuration File for Distortion Compensation
  The residual non-linear distortion of a measurement setup can be measured and calibrated. The distortion components, if they are static, can be counteracted by pre-adding the same components with inverted phases. This can be done manually in the Tone Configuration File, but the option here can facilitate this process.

  The following is an example of harmonic distortion compensation. The first screenshot shows a THD loopback test of an RTX6001 audio analyzer with a 1 kHz test tone, where the measured THD is 0.00010% (-120dB). A Tone Configuration File was then exported using this option. The second screenshot displays the results of the same loopback test with a distortion-compensated multitone, generated using the Tone Configuration File, where the measured THD is reduced to 0.000040% (-128dB).




Multi-Instrument Pro 3.9.11.1 - [+3DP+DLG+LCR+UDP+VBM+DHS] - <RTX6001 ASIO>
File Setting Instrument Window Help
A-Harmonic Frequencies, RMS, Phases
Order Freq. (Hz) RMS (dBFS) % Phase (D)
1 1000.000 Hz -3.01 dBFS 100.0000000... 2.37 D
2 2.000000 kHz -147.08 dBFS 0.0000063 % -169.30 D
3 3.000000 kHz -136.90 dBFS 0.0000202 % 93.67 D
4 4.000000 kHz -139.62 dBFS 0.0000148 % -172.67 D
5 5.000000 kHz -139.78 dBFS 0.0000145 % -114.26 D
6 6.000000 kHz -146.46 dBFS 0.0000067 % 129.21 D
7 7.000000 kHz -143.44 dBFS 0.0000095 % -75.61 D
8 8.000000 kHz -145.73 dBFS 0.0000073 % 77.76 D
9 9.000000 kHz -141.25 dBFS 0.0000122 % -81.64 D
10 10.000000 kHz -148.63 dBFS 0.0000052 % 57.16 D
11 11.000000 kHz -145.65 dBFS 0.0000074 % 69.53 D
12 12.000000 kHz -145.89 dBFS 0.0000072 % 66.66 D
13 13.000000 kHz -143.85 dBFS 0.0000091 % -74.73 D
14 14.000000 kHz -145.60 dBFS 0.0000074 % 132.75 D
15 15.000000 kHz -145.88 dBFS 0.0000072 % -159.95 D
16 16.000000 kHz -145.28 dBFS 0.0000077 % -112.62 D
17 17.000000 kHz -147.78 dBFS 0.0000058 % -28.68 D
18 18.000000 kHz -144.28 dBFS 0.0000086 % 132.25 D
19 19.000000 kHz -145.67 dBFS 0.0000074 % 26.00 D
A (V)
A(dBFS)
Zero Padding
Resolution: 0.366211Hz
AMPLITUDE SPECTRUM in dBFS
Averaged Frames: 10
THD ...
0.00004 %
THD...
0.00025 %
SNR ...
112.07 dB
SINA...
111.95 dB
ENO...
18.30 Bit
Signal Gener...
Show Editor
No Loopback
Amplitude Spectrum
Kaiser 6

11 Derived Data Curve (DDC)

11.1 Overview

Derived Data Curve (DDC) is used to derive a dedicated data curve from one or multiple frames of sampled data and display it in a dedicated graph window. The push button

in the Instrument Toolbar is used to open a new DDC window. You can close it by clicking the "Close" button at the upper right corner of the window. Up to 8 DDC windows can be opened. The supported DDCs include:

(1) Energy Time Curve (Log-Squared)
(2) Energy Time Curve (Envelop)
(3) Energy Time Curve (dBSPL)
(4) Impulse Response Schroeder Integration Curve
(5) Step Response Curve (via Impulse Response Integration)
(6) Frequency Time Curve (Demodulated)
(7) X-Y Plot
(8) Shock Response Spectrum
(9) Frequency Time Curve (Timed)
(10) RPM Time Curve
(11) Spectrum Analysis on Selection

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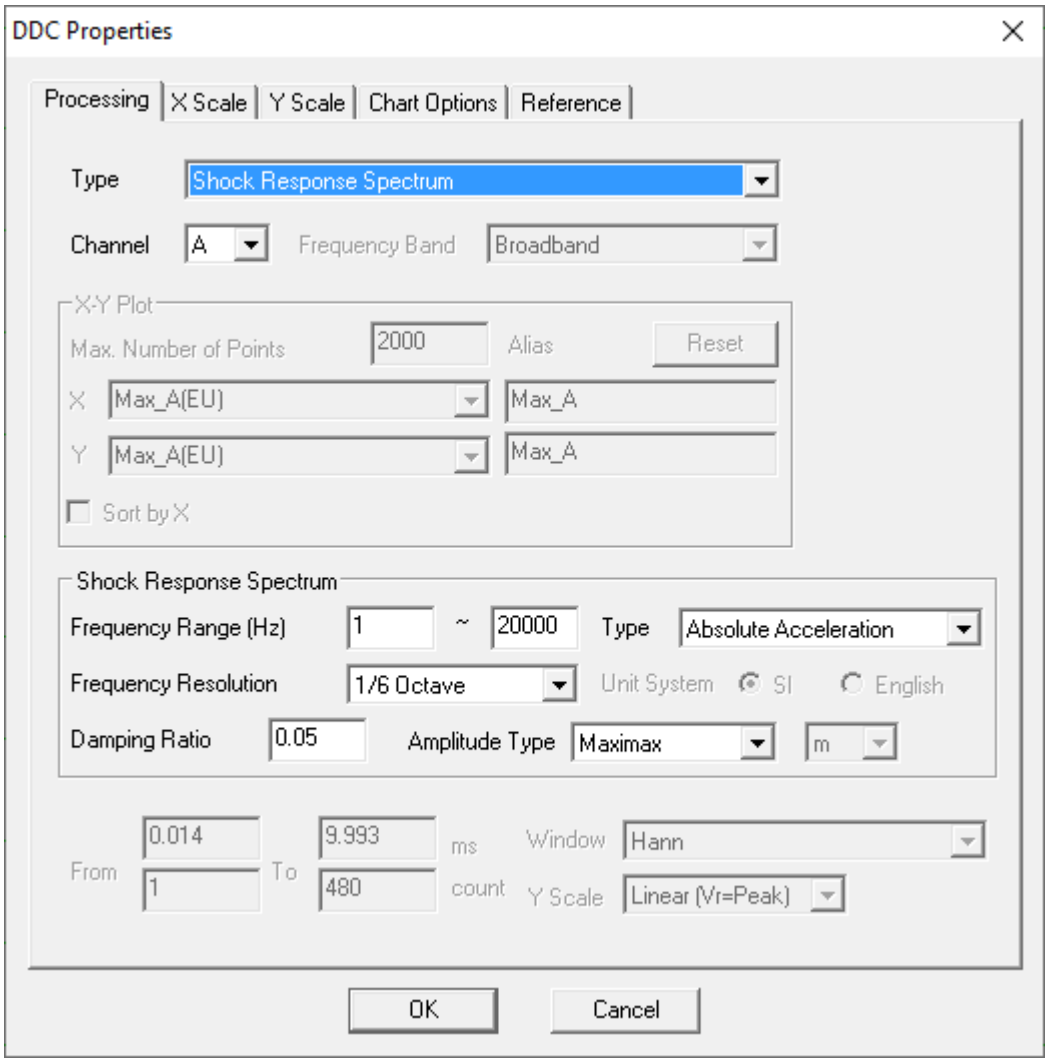
11.2 DDC Processing



Whenever the "DDC" button in the Instrument Toolbar is clicked, a new DDC window will be opened. The above Processing dialog box will pop up on the top of the DDC window to prompt you to configure the processing parameters first, which includes:

- Type
  It specifies the DDC to be generated.

- Channel
  It specifies the input channel from which the DDC is generated.

- Frequency Band (applicable to DDC Type 1~4 only)
  It specifies the band pass filter to be applied to the data before the DDC is generated. The supported band pass filter includes: broadband (i.e. not filtered), 1/1 octave band filters (63Hz, 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz, 8kHz), 1/3 octave band filters (50Hz, 63Hz, 80Hz, 100Hz, 125Hz,

160Hz, 200Hz, 250Hz, 315Hz, 400Hz, 500Hz, 630Hz, 800Hz, 1kHz, 1.25Hz, 1.6kHz, 2kHz, 2.5kHz, 3.15kHz, 4kH, 5kHz, 6.3kHz, 8kHz, 10kHz).

- Max. Number of Points (applicable to X-Y Plot only)
- DDP for X axis (applicable to X-Y Plot only)
- DDP alias for X axis (applicable to X-Y Plot only)
- DDP for Y axis (applicable to X-Y Plot only)
- DDP alias for Y axis (applicable to X-Y Plot only)
- Reset Button (applicable to X-Y Plot only)
- Sort by X (applicable to X-Y Plot only)

- Frequency Range (Hz) (applicable to Shock Response Spectrum)
- Frequency Resolution (applicable to Shock Response Spectrum)
- Damping Ratio (applicable to Shock Response Spectrum)
- Amplitude Type (applicable to Shock Response Spectrum)
- Type (applicable to Shock Response Spectrum)
- Unit System (applicable to Shock Response Spectrum)

- Start Point/Time of Selected Waveform (applicable to Spectrum Analysis on Selection)
- End Point/Time of Selected Waveform (applicable to Spectrum Analysis on Selection)
- Window Function (applicable to Spectrum Analysis on Selection)
- Y Scale (applicable to Spectrum Analysis on Selection)

11.3 DDC Types

11.3.1 Energy Time Curve (Log-Squared)

Energy Time Curve (ETC) (Log-Squared) shows the energy variation with time, where the individual samples in the Oscilloscope are squared and then plotted on a logarithmic scale in dB. The following figure shows such an example.

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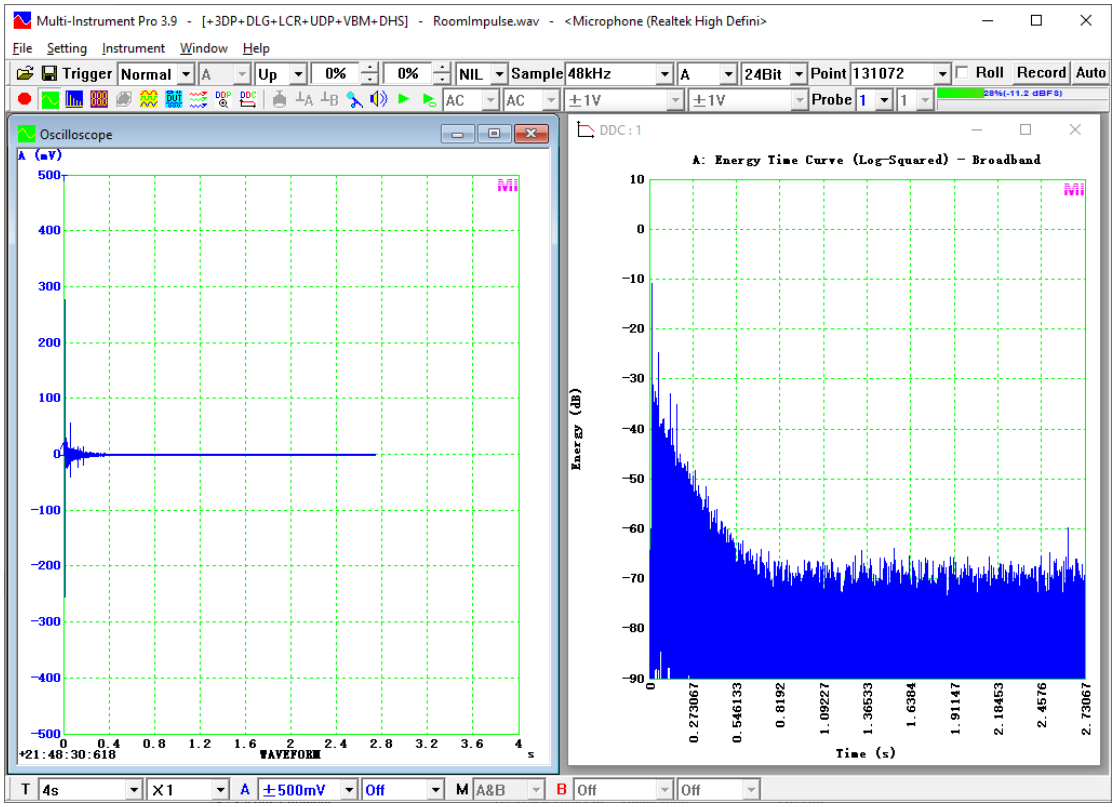
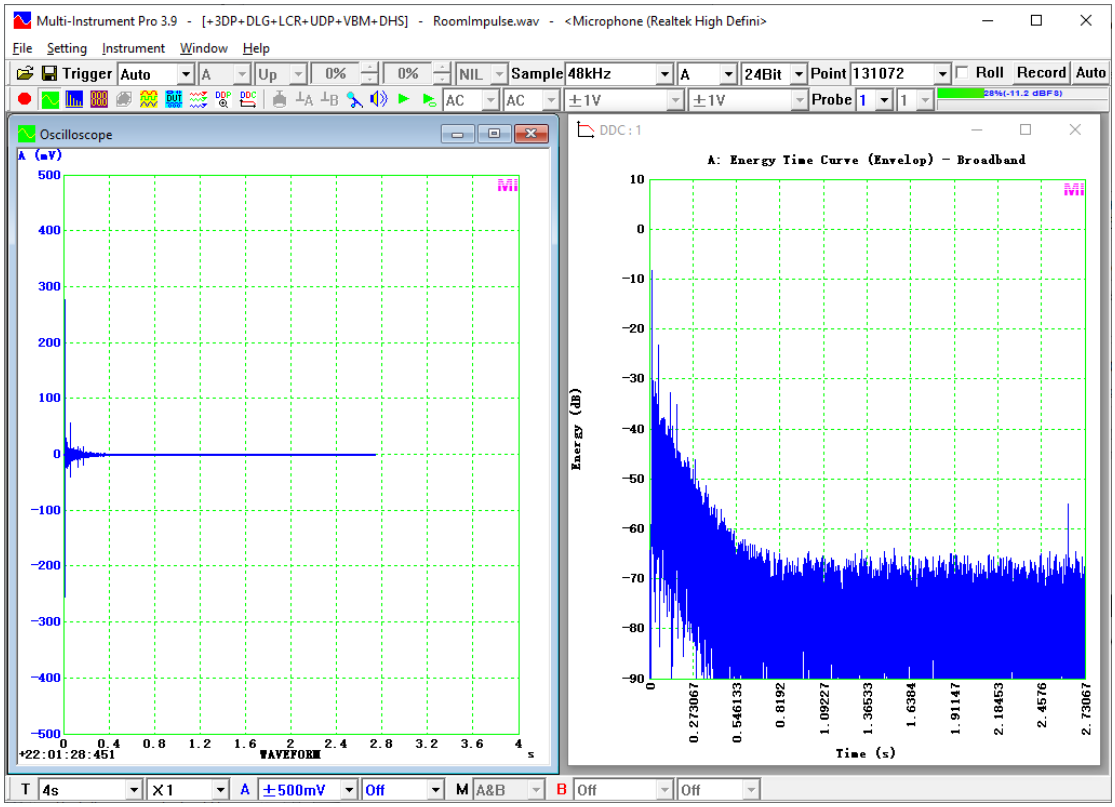


This DDC is derived from one frame of Oscilloscope data.

11.3.2 Energy Time Curve (Envelop)

Energy Time Curve (ETC) (Envelop) shows the energy variation with time, where an envelop of the data in the Oscilloscope is constructed by the analytic function through Hilbert Transform and then plotted on a logarithmic scale in dB. The method is similar to the amplitude demodulation method introduced previously in the Oscilloscope chapter. This representation of ETC is often referred to as Heyser ETC. The following figure shows such an example.



This DDC is derived from one frame of Oscilloscope data.

11.3.3 Energy Time Curve (dBSPL)

Energy Time Curve (ETC) (dBSPL) shows the sound energy variation with time, where the sound energy values in dBSPL are derived from the RMS values over a local integration time. The software determines the integration time automatically based on the band pass filter selected. The integration process serves as a low pass filter in effect and makes the curve smooth. This DDC forms the basis for deriving the reverberation parameters using the interrupted noise method, as introduced previously in the Oscilloscope chapter. The following figure shows such an example.

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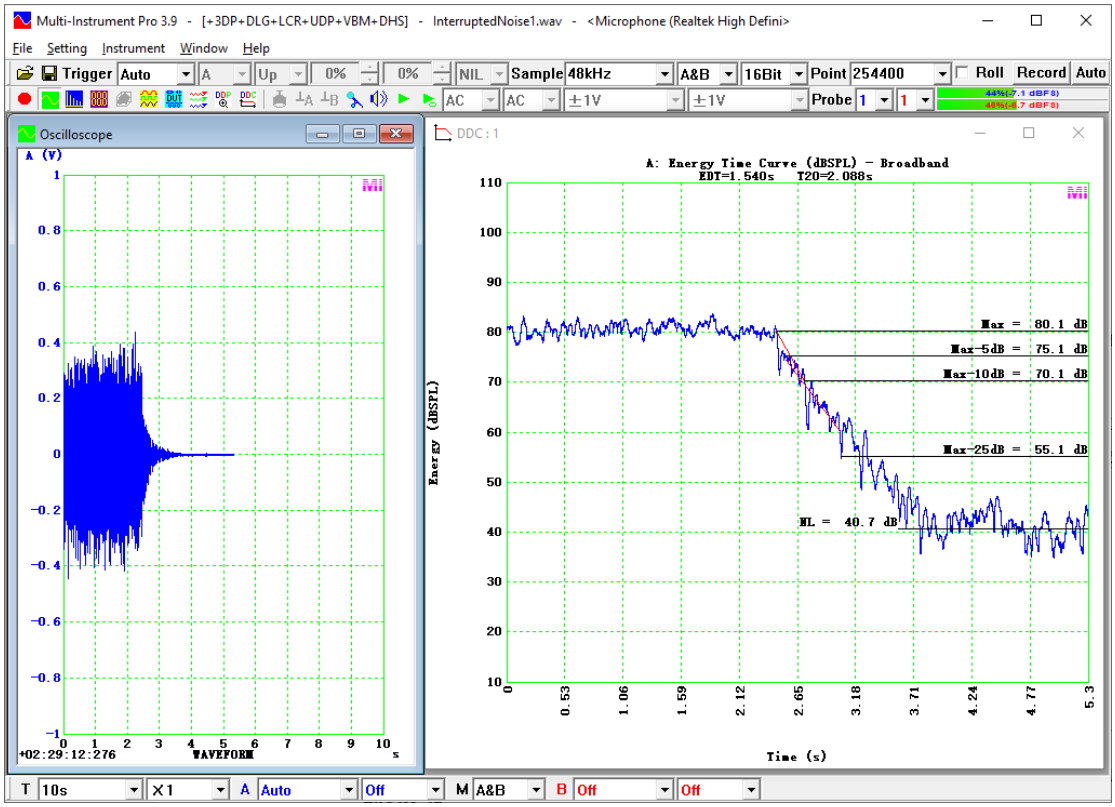
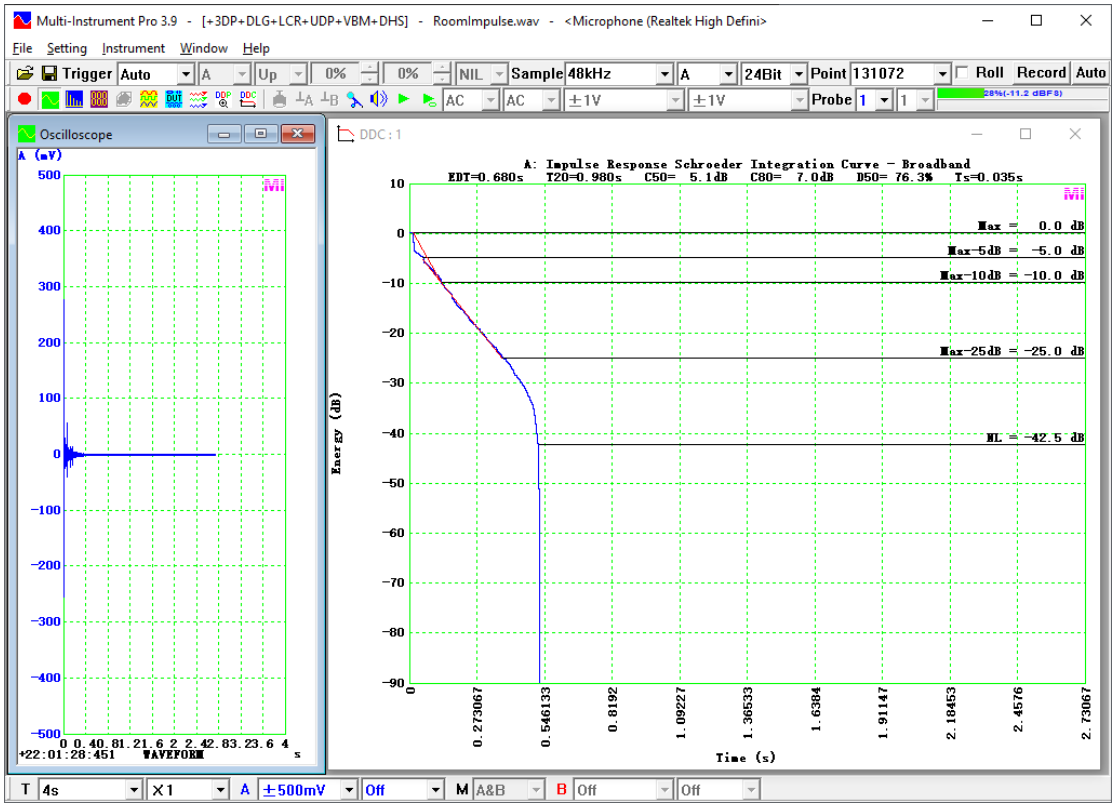


This DDC is derived from one frame of Oscilloscope data.

11.3.4 Impulse Response Schroeder Integration Curve

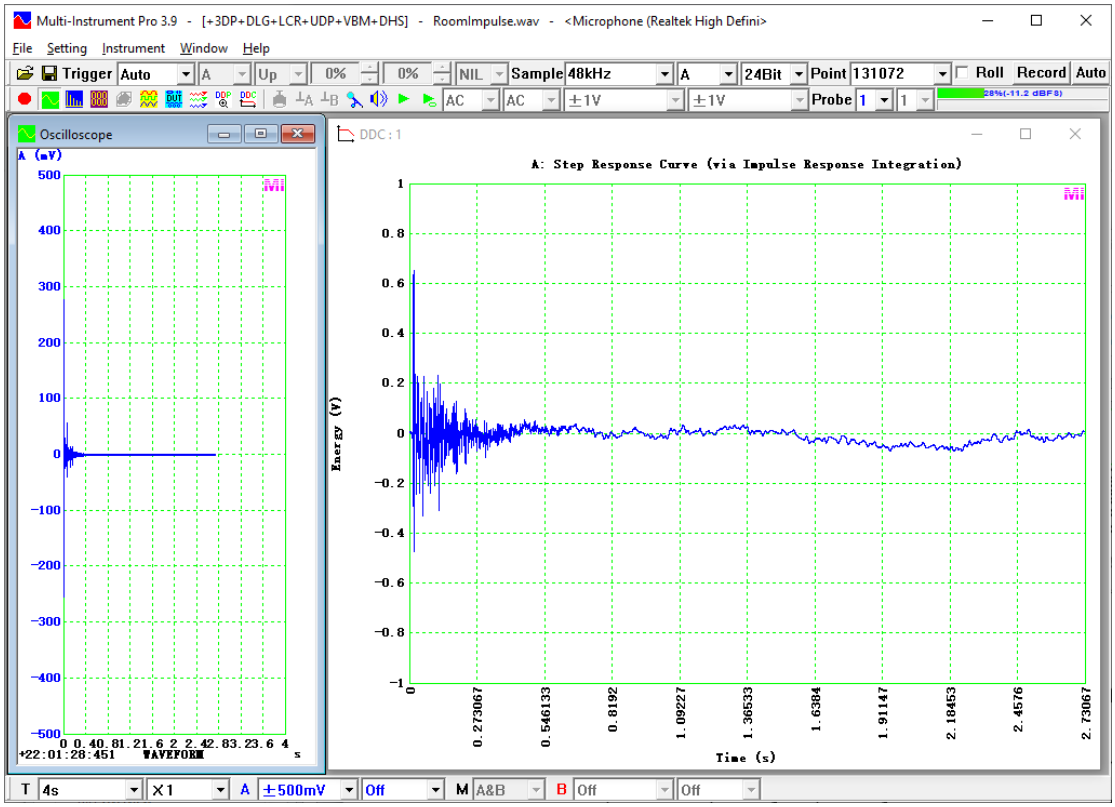
Impulse Response Schroeder Integration Curve serves as the basis for deriving the reverberation parameters using the integrated impulse response method, as introduced previously in the Oscilloscope chapter. The following figure shows such an example.



This DDC is derived from one frame of Oscilloscope data.

11.3.5 Step Response Curve (via Impulse Response Integration)

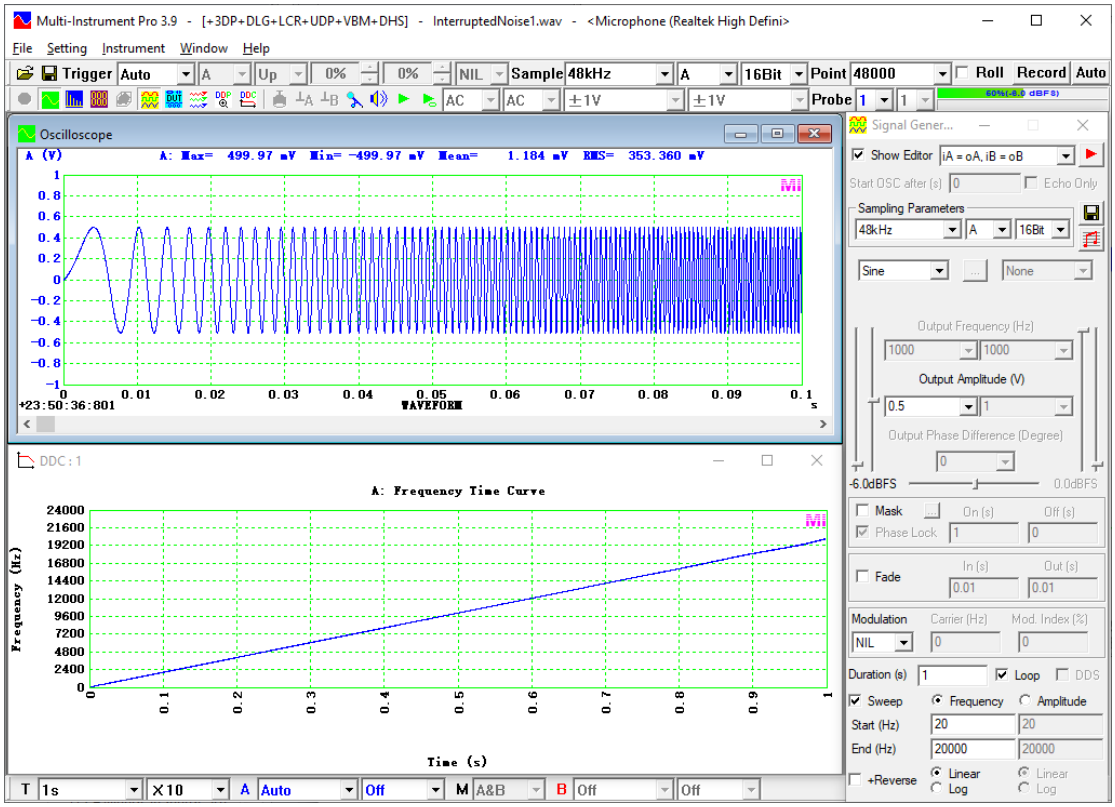
Step Response shows the reaction of a system to a step stimulus. It is obtained here by integrating the impulse response against time. The following figure shows such an example.



This DDC is derived from one frame of Oscilloscope data.

11.3.6 Frequency Time Curve (Demodulated)

Frequency Time Curve here shows the instantaneous frequency variation with time. It is obtained using the frequency demodulation method introduced previously in the Oscilloscope chapter. The following figure shows the frequency time curve of a frequency linearly swept sine signal.



This DDC is derived from one frame of Oscilloscope data.

11.3.7 X-Y Plot

X-Y Plot shows the variation of one DDP with another DDP. Both DDPs can be selected in the DDC Processing page described previously. Each DDP can be assigned with an alias which is used as the axis label. Each oscilloscope data frame generates only one point (a pair of X and Y values) on the X-Y plot. Multiple consecutive oscilloscope data frames are required in order to plot a curve. The Max. Number of Points is set to 2000 by default and can be changed in the DDC Processing page. Once the max. number is reached, the oldest point will be dropped whenever a new point comes. The plotting process can be reset by clicking the Reset button in the DDC Processing page or restarting the oscilloscope. The number of collected points is displayed at the bottom right corner of the plot. No point will be added if any one of the two DDPs is invalid. It is possible to sort the collected points according to their X values by ticking the "Sort by X" checkbox. The following figure shows a sound level vs rotational speed (in RPM) plot.

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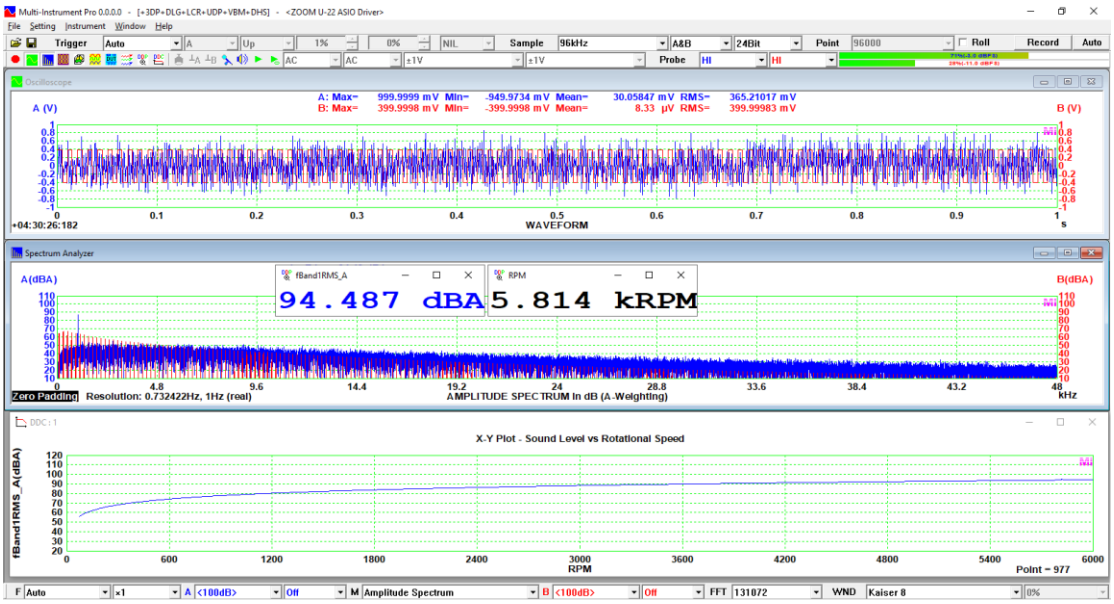


This DDC is derived from multiple frames of Oscilloscope data.

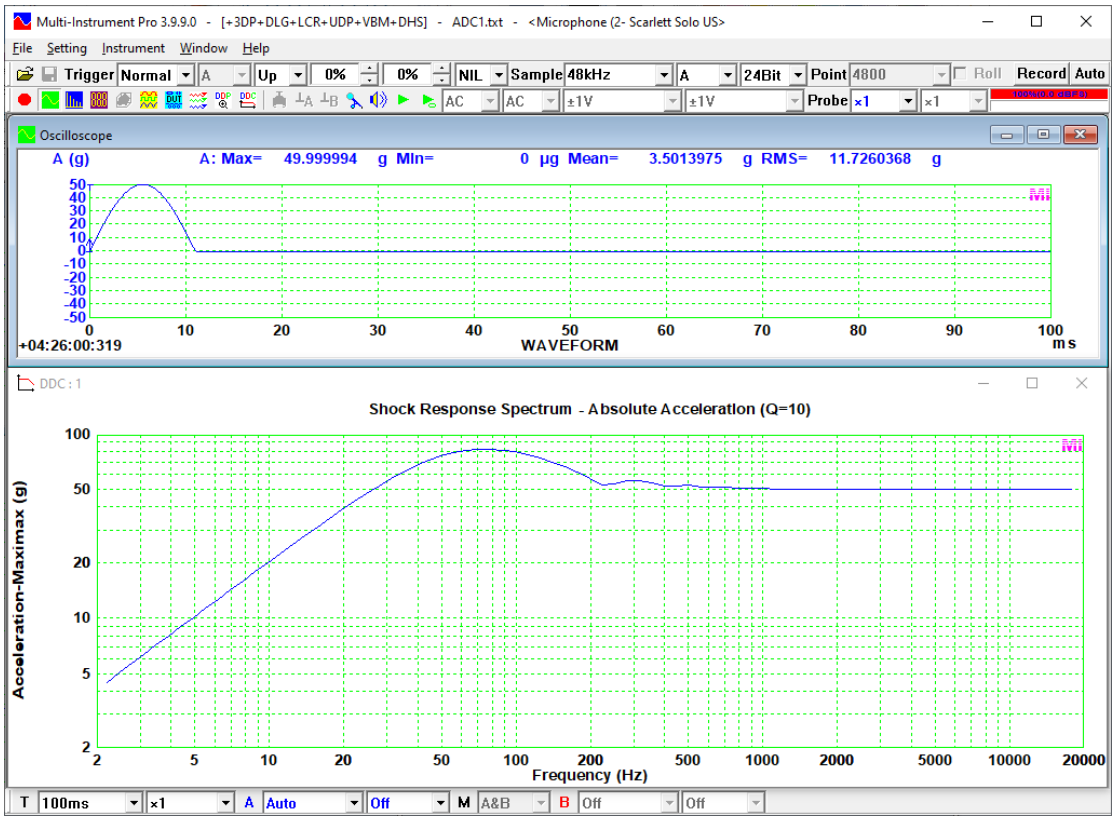
11.3.8 Shock Response Spectrum

Shock Response Spectrum (SRS) is a graphical representation of an arbitrary transient acceleration input, such as a shock, in terms of how a Single Degree Of Freedom (SDOF) system (like a mass on a spring) would respond to that input. The horizontal axis shows the natural frequency of a hypothetical SDOF system, and the vertical axis shows the peak acceleration, velocity or displacement, which this SDOF system would undergo as a consequence of the shock input. Given the time history of a transient acceleration input, a series of SDOF systems with different natural frequencies and the same damping ratio can be used to calculate its SRS. Five types of SRS can be calculated: Absolute Acceleration, Relative Velocity, Relative Displacement, Pseudo Velocity, and Equivalent Static Acceleration. They can be represented in either SI or English units. For SI units, options for velocity and displacement scaling include 'm' (meters), 'dm' (decimeters), 'cm' (centimeters), and 'mm' (millimeters).

In Multi-Instrument, the Ramp Invariant Method is used for SRS calculation, adhering to ISO 18431-4. The natural frequencies used for SRS calculation are chosen with the specified octave or fractional octave spacing within the specified frequency range. The ramp invariant algorithm contains a bias error that increases as the ratio of the signal frequency to the sampling frequency grows. Another source of error arises from the fact that the peak in the sampled response signal is only an estimation of the “true” peak, which can fall between samples. The combination of these two errors results in an overall error of about 2% at 5% of the sampling frequency, and 8% at 10% of the sampling frequency. Therefore, the sampling frequency should be at least 10 times the highest significant frequency content of the input signal. Multi-Instrument features a special algorithm that can up-sample the input signal and then calculate the SRS at a ×32 sampling frequency internally. Thus, the calculated SRS will maintain its high accuracy even at the Nyquist frequency, which is half of the sampling rate.

The damping ratio is configurable, typically 5% should be used. Three amplitude types are supported: Maximax (Maximum Absolute), Max+ (Maximum Positive) and Max- (Maximum Negative). The time record of the transient acceleration input should be long enough to be able to accommodate the important part (which contains the Maximum Positive and Maximum Negative) of the response of any of the SDOF systems used in the calculation. Usually, 1/6 octave spacing is used for the natural frequencies when the damping ratio is in the range of 0.02~0.1, while 1/12 and 1/3 octave spacings are used when the damping ratio is below and beyond that range respectively.

The following figures show the five types of Shock Response Spectrum of a half-sine pulse with a duration of 11 ms and a peak acceleration of 50g. The damping ratio used is 5% (i.e. Q=10).



Absolute Acceleration Shock Response Spectrum

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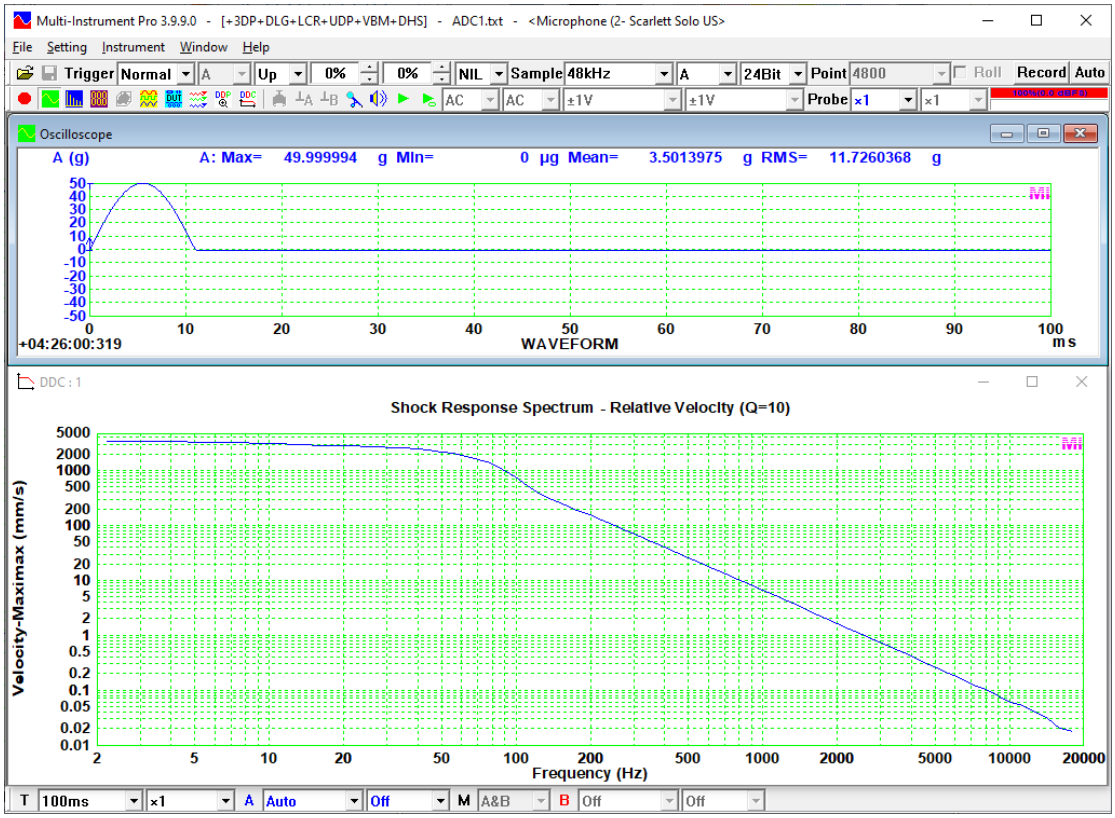
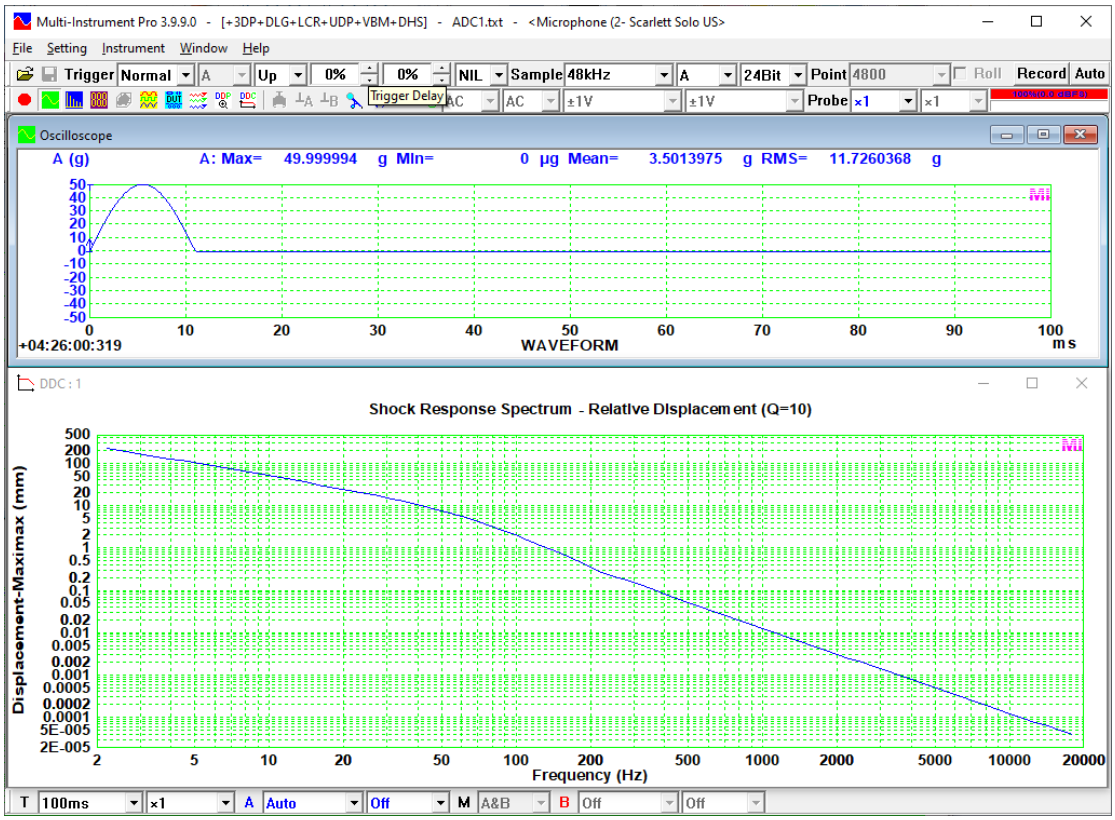
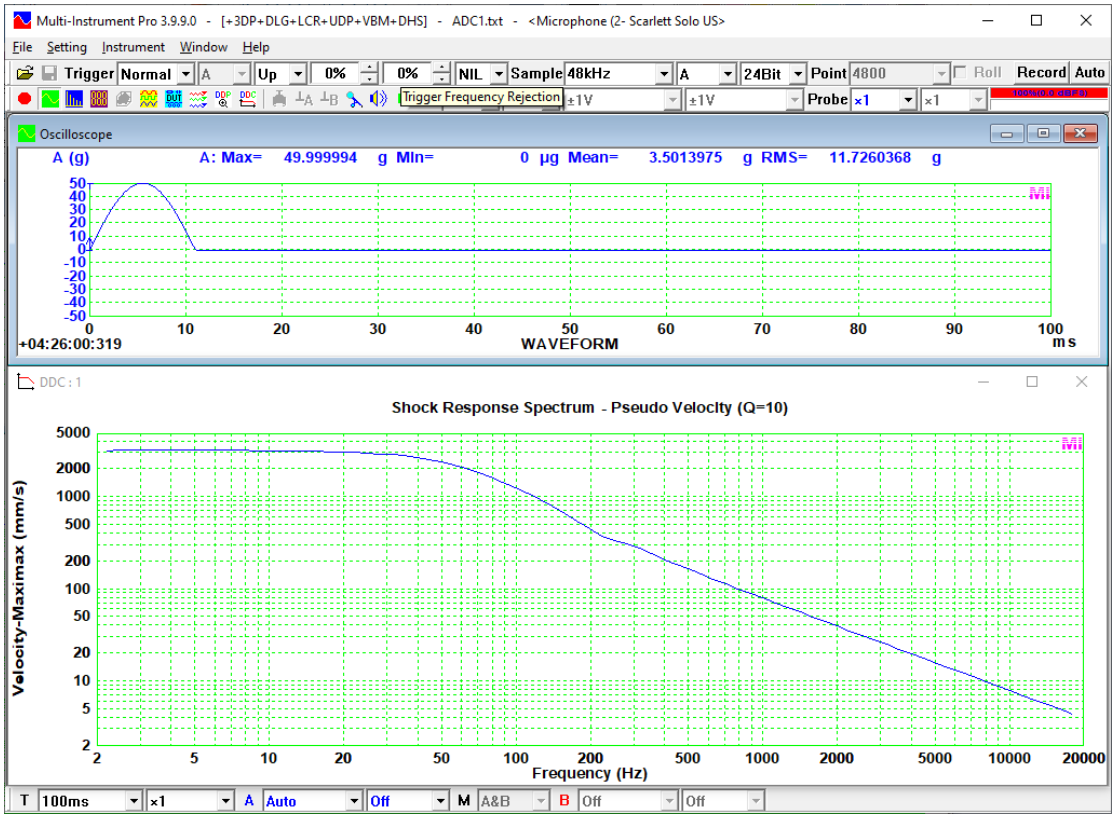


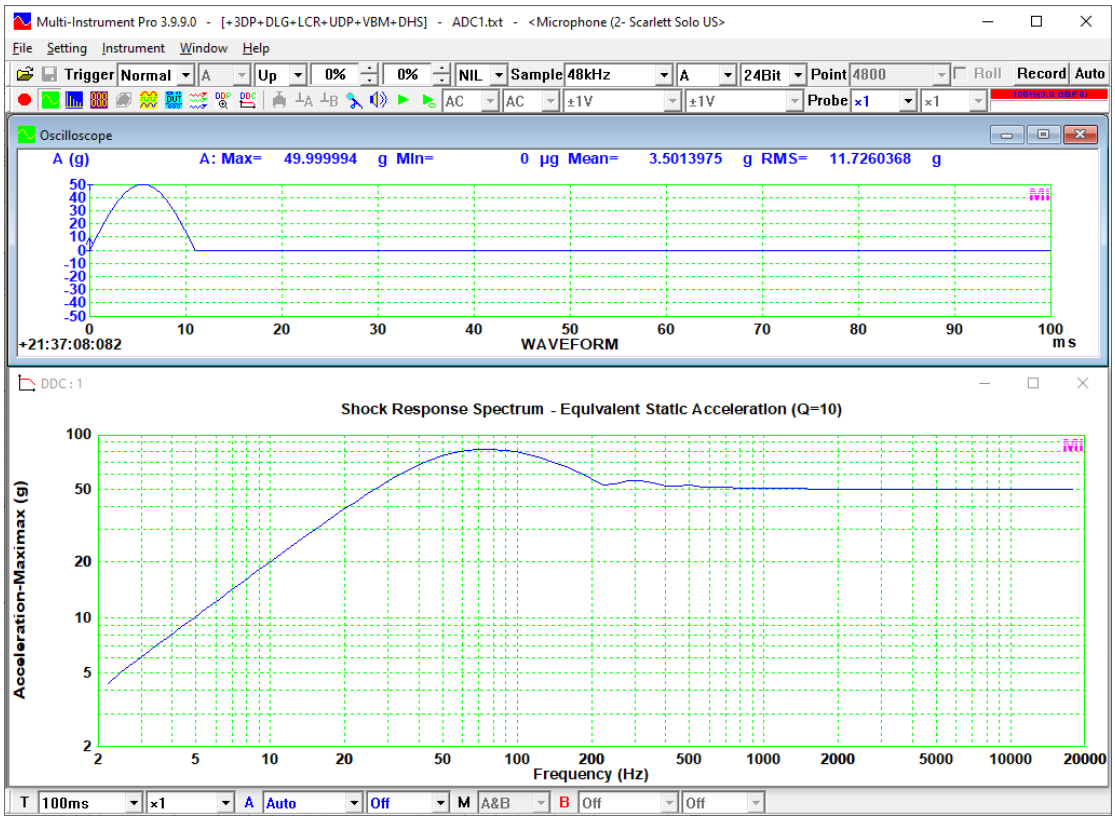
Relative Velocity Shock Response Spectrum



Relative Displacement Shock Response Spectrum



Pseudo Velocity Shock Response Spectrum



Equivalent Static Acceleration Shock Response Spectrum

This DDC is derived from one frame of Oscilloscope data.

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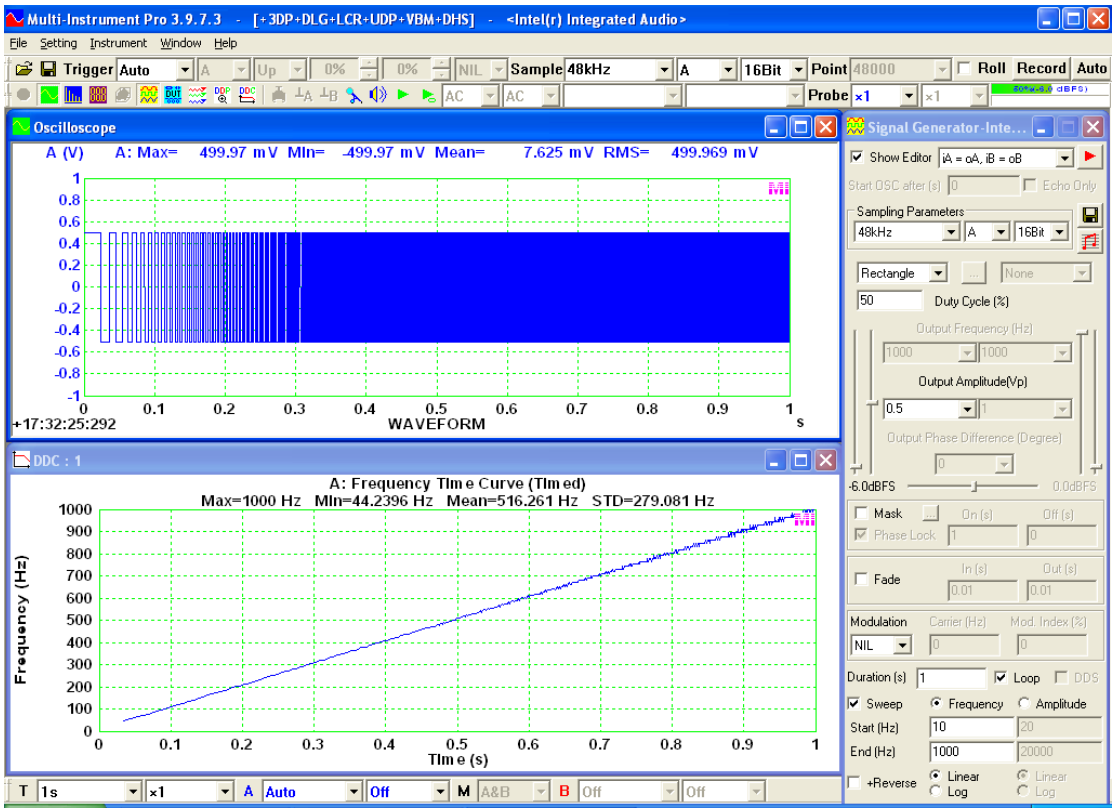
For more information on mechanical shock testing, please refer to *Shock Testing with Multi-Instrument - Classical Shock Pulses and Shock Response Spectrum*.

Download link:
https://www.virtins.com/doc/Shock-Testing-with-Multi-Instrument-Classical-Shock-Pulses-and-Shock-Response-Spectrum.pdf

11.3.9 Frequency Time Curve (Timed)

Frequency Time Curve here shows the time counted (short-time averaged) frequency variation with time. It is obtained using the frequency counting method introduced previously in the Multimeter chapter. If the Multimeter window is not opened, then the counter trigger level, trigger hysteresis and frequency divider are 0%, 10% and 1 respectively. Otherwise, these parameters are determined by the respective parameters in Multimeter View Parameter Toolbar. The following figure shows the frequency time curve of a frequency linearly swept square wave signal.

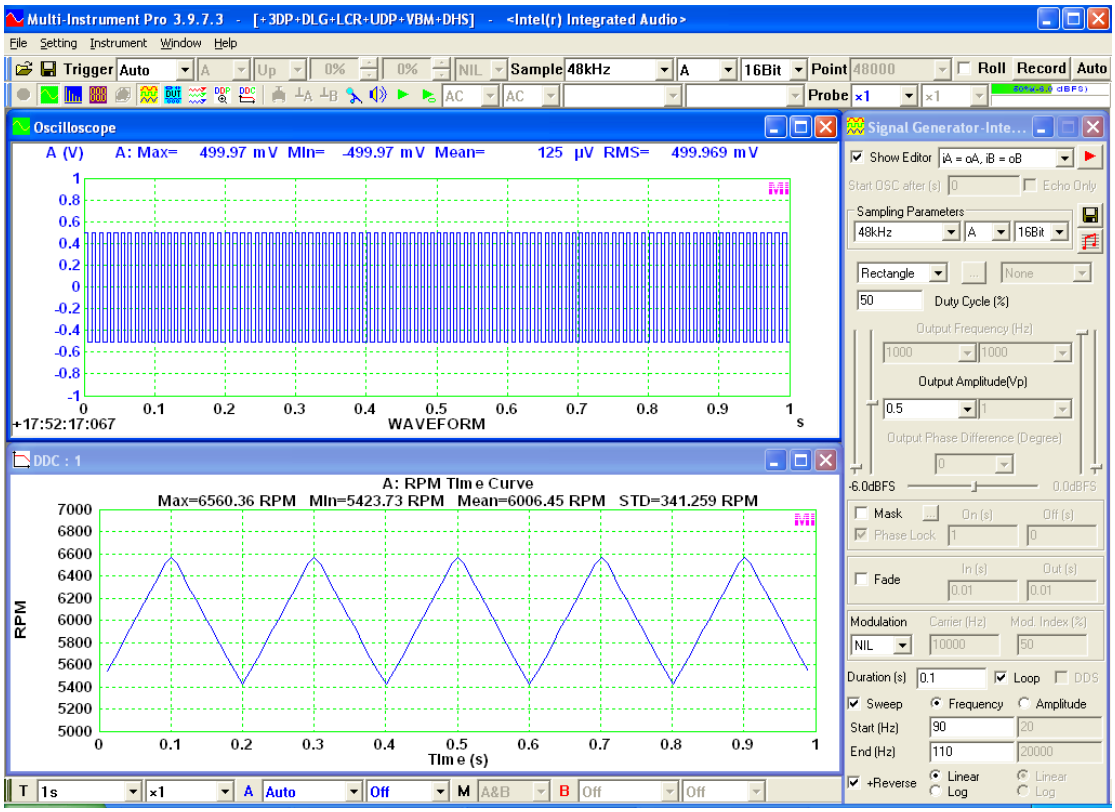


This DDC is derived from one frame of Oscilloscope data.

11.3.10 RPM Time Curve

RPM Time Curve shows the rotational speed (in RPM) variation with time. It is obtained using the frequency counting method introduced previously in the Multimeter chapter. If the Multimeter window is not opened, then the counter trigger level, trigger hysteresis and frequency divider are 0%, 10% and 1 respectively.

Otherwise, these parameters are determined by the respective parameters in Multimeter View Parameter Toolbar. The following figure shows the RPM time curve of a rotational speed with a reciprocating deviation.



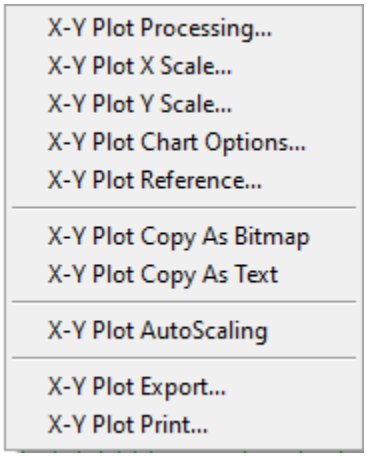
This DDC is derived from one frame of Oscilloscope data.

11.3.11 Spectrum Analysis on Selection

This DDC displays the spectrum of the selected portion of the waveform in the Oscilloscope. The FFT size will be automatically set to a value equal to or greater than the selected waveform length. If the selected waveform length exceeds the maximum allowed FFT size, the data will be truncated. It is possible to change the window function and Y scale of the spectrum analysis. Peak frequency and its magnitude will also be displayed.

11.4 Context Menu

DDC does not have its own menu and view parameter toolbar. Its functions are accessed via its context menu. The context menu (see figure below) pops up when you right click anywhere within the DDC window.



The above menu items and their associated functions are the same as those of the X-Y Plot in the Device Test Plan, except [X-Y Plot Processing], which is only applicable to DDC. Its function has been described previously.

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12 Samples and Templates

Different categories of samples and templates are provided in different subdirectories under the main installation directory of the software. Although these samples and templates are provided mainly for sound card based systems, they can be applied to other hardware systems with or without modifications.

Subdirectory	File Name	Description
\dtp	..	Default directory for device test plans
	AudioParameter_SR44100_A.dtp	Single channel audio parameter test plan using a sampling rate of 44100 Hz. It can be used to conduct loop back test for sound card itself.
	AudioParameter_SR44100_AB.dtp	Dual channel audio parameter test plan using a sampling rate of 44100 Hz. It can be used to conduct loop back test for sound card itself.
	AudioParameter_SR48000_A.dtp	Single channel audio parameter test plan using a sampling rate of 48000 Hz. It can be used to conduct loop back test for sound card itself.
	AudioParameter_SR48000_AB.dtp	Dual channel audio parameter test plan using a sampling rate of 48000 Hz. It can be used to conduct loop back test for sound card itself.
	HighLowLimit.dtp	Magnitude response against high and low limits.
	SteppedSineFrequencyResponseMeasurement_Demo.dtp	Frequency response measurement demo using stepped sine. Note that for demo purpose, a 5513 Hz second order Butterworth low pass digital filter is applied to Channel A by the software, and Channel A and Channel B should be fed with the same signal in order to obtain the frequency response of the low pass filter.
	SteppedSineFrequencyResponseMeasurement.dtp	Frequency response measurement test plan using stepped sine.
	THD+N_THD_SNR_Magnitude_vs_Frequency.dtp	THD+N, THD, SNR, Magnitude vs Frequency Plots measured using frequency stepped sine signals.
	THD+N_THD_vs_Magnitude_Power.dtp	THD+N, THD, vs Magnitude, Power Plots measured using amplitude stepped sine signals.
	SMPTE_IMD_vs_Magnitude_Power.dtp	SMPTE IMD vs Magnitude, Power Plots measured using amplitude stepped dual-sinewave signals.
	Crosstalk_vs_Frequency.dtp	Crosstalk vs Frequency Plot measured using frequency stepped sine signal.
	HappyBirthDay.dtp	“Happy Birth Day” music.
	MagnitudeFrequencyResponseBySweepAtOneThirdOctave.dtp	Magnitude frequency response measurement using frequency sweep at one third octave center frequencies
	UDTdefault.dtp	Empty test plan loaded as the default user defined plan.
	Zscdefault.dtp	Default sound card input impedance test plan.

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	LCRdefault.dtp	Default LCR test plan.
	LCR_50-10000Hz.dtp	LCR test plan with test frequencies ranging from 50Hz to 10000Hz.
	LCR_50Hz.dtp	LCR test plan with test frequency at 50 Hz.
	LCR_100Hz.dtp	LCR test plan with test frequency at 100 Hz.
	LCR_200Hz.dtp	LCR test plan with test frequency at 200 Hz.
	LCR_500Hz.dtp	LCR test plan with test frequency at 500 Hz.
	LCR_2000Hz.dtp	LCR test plan with test frequency at 2000 Hz.
	LCR_5000Hz.dtp	LCR test plan with test frequency at 5000 Hz.
	LCR_10000Hz.dtp	LCR test plan with test frequency at 10000 Hz.
\export	..	Default directory for data export
\fcf	..	Default directory for frequency compensation files.
	A_Weighting.fcf	A weighting frequency compensation file.
	B_Weighting.fcf	B weighting frequency compensation file.
	C_Weighting.fcf	C weighting frequency compensation file.
	ITU-R468_Weighting.fcf	ITU-R468 weighting frequency compensation file.
	RecordingRIAA.fcf	Recording RIAA frequency compensation file
	ReproducingRIAA.fcf	Reproducing RIAA frequency compensation file
	ITU_R_ARM_Weighting.fcf	ITU-R ARM frequency compensation file
	Z_Weighting_10Hz-20kHz.fcf	10Hz-20kHz Z weighting frequency compensation file
	BandWidthLimiting_20Hz-20kHz.fcf	20Hz-20kHz band width limiting frequency compensation file
\frf	..	Default directory for frequency response files.
	BandPass_1k-2kHz.frf	Frequency response file for 1 k~2 kHz band pass digital filter
	BandStop_20-80Hz.frf	Frequency response file for 20~80 Hz band stop digital filter
	HighPass_1kHz.frf	Frequency response file for 1 kHz high pass digital filter
	LowPass_1kHz.frf	Frequency response file for 1 kHz low pass digital filter
	RecordingRIAA.frf	Frequency response file for recording RIAA
	ReproducingRIAA.frf	Frequency response file for reproducing RIAA
	SpeechMale.frf	Frequency response file for male speech transmission index measurement
	SpeechFemale.frf	Frequency response file for female speech transmission index measurement
\iir	..	Default directory for IIR coefficient files.
	BandPass_5513-11025Hz_SR44100Hz_Butterworth_IIR_Order12.iir	IIR coefficient file for the 12th order Butterworth 5513~11025 Hz band pass digital IIR filter, sampled at 44100 Hz.
	BandPass_5513-11025Hz_SR44100Hz_Chebyshev_IIR_Order12.iir	IIR coefficient file for the 12th order Chebyshev 5513~11025 Hz band pass digital IIR filter, sampled at 44100 Hz.
	LowPass_5513Hz_SR44100Hz_Butterworth_IIR_Order2	IIR coefficient file for the 2th order Butterworth 5513 Hz low pass digital IIR filter, sampled at 44100 Hz.

	LowPass_5513Hz_SR44100Hz_Butterworth_IIR_Order12	IIR coefficient file for the 12th order Butterworth 5513 Hz low pass digital IIR filter, sampled at 44100 Hz.
	LowPass_5513Hz_SR44100Hz_Chebyshev_IIR_Order12	IIR coefficient file for the 12th order Chebyshev 5513 Hz low pass digital IIR filter, sampled at 44100 Hz.
	LowPass_5513Hz_SR44100Hz_Parks-McClellan_FIR_Order34.iir	IIR coefficient file for the 34th order Parks-McClellan 5513 Hz low pass digital FIR filter, sampled at 44100 Hz.
	HighPass_5513Hz_SR44100Hz_Butterworth_IIR_Order12	IIR coefficient file for the 12th order Butterworth 5513 Hz high pass digital IIR filter, sampled at 44100 Hz.
	HighPass_5513Hz_SR44100Hz_Chebyshev_IIR_Order12.iir	IIR coefficient file for the 12th order Chebyshev 5513 Hz high pass digital IIR filter, sampled at 44100 Hz.
	C_Weighting_SR44100Hz_IIR_Order4.iir	IIR coefficient file for the 4th order C weighting digital IIR filter, sampled at 44100 Hz.
	C_Weighting_SR48000Hz_IIR_Order4.iir	IIR coefficient file for the 4th order C weighting digital IIR filter, sampled at 48000 Hz.
	C_Weighting_SR96000Hz_IIR_Order4.iir	IIR coefficient file for the 4th order C weighting digital IIR filter, sampled at 96000 Hz.
	C_Weighting_SR192000Hz_IIR_Order4.iir	IIR coefficient file for the 4th order C weighting digital IIR filter, sampled at 192000 Hz.
	MovingAverageTap2.iir	IIR coefficient file for 2-Tap Moving Average FIR filter
	MovingAverageTap4.iir	IIR coefficient file for 4-Tap Moving Average FIR filter
	MovingAverageTap8.iir	IIR coefficient file for 8-Tap Moving Average FIR filter
	MovingAverageTap16.iir	IIR coefficient file for 16-Tap Moving Average FIR filter
	MovingAverageTap32.iir	IIR coefficient file for 32-Tap Moving Average FIR filter
	MovingAverageTap64.iir	IIR coefficient file for 64-Tap Moving Average FIR filter
	MovingAverageTap128.iir	IIR coefficient file for 128-Tap Moving Average FIR filter
	MovingAverageTap256.iir	IIR coefficient file for 256-Tap Moving Average FIR filter
	RecordingRIAA_SR44100Hz_IIR_Order2.iir	IIR coefficient file for Recording RIAA at Sampling Rate 44100Hz
	RecordingRIAA_SR48000Hz_IIR_Order2.iir	IIR coefficient file for Recording RIAA at Sampling Rate 48000Hz
	RecordingRIAA_SR88000Hz_IIR_Order2.iir	IIR coefficient file for Recording RIAA at Sampling Rate 88000Hz
	ReproducingRIAA_SR44100Hz_IIR_Order2.iir	IIR coefficient file for Reproducing RIAA at Sampling Rate 44100Hz
	ReproducingRIAA_SR48000Hz_IIR_Order2.iir	IIR coefficient file for Reproducing RIAA at Sampling Rate 48000Hz
	ReproducingRIAA_SR88000Hz_IIR_Order2.iir	IIR coefficient file for Reproducing RIAA at Sampling Rate 88000Hz
	ReproducingRIAA_SR96000Hz_IIR_Order2.iir	IIR coefficient file for Reproducing RIAA at Sampling Rate 96000Hz
	ReproducingRIAA_SR192000Hz_IIR_Order2.iir	IIR coefficient file for Reproducing RIAA at Sampling Rate 192000Hz

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\log	..	Log file directory for Data Logger
\psf	..	Default directory for panel setting files.
	MagnitudeFrequencyResponse_LinearSweptSine(20~20kHz)_SingleChannel.psf	Panel setting file for magnitude frequency response measurement using 20~20kHz frequency linearly swept sine.
	MagnitudeFrequencyResponse_LogSweptSine(20~20kHz)_SingleChannel.psf	Panel setting file for magnitude frequency response measurement using 20~20kHz frequency logarithmically swept sine.
	MagnitudeFrequencyResponse_MultiTone_OCT3_SingleChannel.psf	Panel setting file for magnitude frequency response measurement using MultiTones aligned to 1/3 octave bands.
	MagnitudeFrequencyResponse_PeriodicWhiteNoise.psf	Panel setting file for magnitude frequency response measurement using periodic white noise.
	MagnitudeFrequencyResponse_SweptSine(20~22050Hz,300s)_PeakHold.psf	Panel setting file for magnitude frequency response measurement using 300-second 20~22050Hz swept sine together with peak hold function in Spectrum Analyzer.
	MagnitudeFrequencyResponse_WhiteNoise(InterframeAverage).psf	Panel setting file for magnitude frequency response measurement using white noise together with inter-frame average function in Spectrum Analyzer.
	MagnitudeFrequencyResponse_WhiteNoise(IntraframeAverage).psf	Panel setting file for magnitude frequency response measurement using white noise together with intra-frame average function in Spectrum Analyzer.
	SteppedSineFrequencyResponseMeasurement.psf	Panel setting file for frequency response measurement using stepped sine. To be used with the corresponding test plan.
	SteppedSineFrequencyResponseMeasurement_Demo.psf	Panel setting file for frequency response measurement demo using stepped sine. To be used with the corresponding test plan.
	Zscdefault.psf	Panel setting file for sound card input impedance measurement. To be used with the corresponding test plan.
	LCRdefault.psf	Panel setting file for LCR measurement. To be used with the corresponding test plan.
	ViewWindowFunction_24Bit_1024Pt.psf	Panel setting file viewing the characteristics of various Window functions.
	NoiseLevel.psf	Panel setting file for noise level measurement when there is no input signal.
	NoiseLevel_A-Weighting.psf	Panel setting file for A weighted noise level measurement when there is no input signal.
	THD_FFT16384_SR44100.psf	Panel setting file for THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 44100 Hz.
	THD_FFT16384_SR44100_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 44100 Hz.
	THD_FFT16384_SR48000.psf	Panel setting file for THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 48000 Hz.
	THD_FFT16384_SR48000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 48000 Hz.
	THD_FFT16384_SR48000_FifthOrder.psf	Panel setting file for THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 48000 Hz. THD is calculated to the fifth order.

	THD_FFT16384_SR96000.psf	Panel setting file for THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 96000 Hz.
	THD_FFT16384_SR96000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 96000 Hz.
	THD_FFT16384_SR192000.psf	Panel setting file for THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 192000 Hz.
	THD_FFT16384_SR192000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 16384 and Sampling Frequency = 192000 Hz.
	THD_FFT16384_FifthOrder.psf	Panel setting file for THD, THD+N measurement with FFT size = 16384. THD is calculated to the fifth order.
	THD_FFT16384_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 16384.
	THD_FFT16384.psf	Panel setting file for THD, THD+N measurement with FFT size = 16384.
	THD_FFT32768_SR44100.psf	Panel setting file for THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 44100 Hz.
	THD_FFT32768_SR44100_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 44100 Hz.
	THD_FFT32768_SR48000.psf	Panel setting file for THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 48000 Hz.
	THD_FFT32768_SR48000_SingleChannel.psf	Panel setting file for THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 48000 Hz. Single Channel only.
	THD_FFT32768_SR48000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 48000 Hz.
	THD_FFT32768_SR48000_A-Weighting_SingleChannel.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 48000 Hz. Single Channel only.
	THD_FFT32768_SR48000_FifthOrder.psf	Panel setting file for THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 48000 Hz. THD is calculated to the fifth order.
	THD_FFT32768_SR96000.psf	Panel setting file for THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 96000 Hz.
	THD_FFT32768_SR96000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 96000 Hz.
	THD_FFT32768_SR192000.psf	Panel setting file for THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 192000 Hz.
	THD_FFT32768_SR192000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 32768 and Sampling Frequency = 192000 Hz.

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	THD_FFT32768_FifthOrder.psf	Panel setting file for THD, THD+N measurement with FFT size = 32768. THD is calculated to the fifth order.
	THD_FFT32768_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 32768.
	THD_FFT32768.psf	Panel setting file for THD, THD+N measurement with FFT size = 32768.
	\Pro\THD_FFT65536_SR44100.psf	Panel setting file for THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 44100 Hz.
	\Pro\THD_FFT65536_SR44100_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 44100 Hz.
	\Pro\THD_FFT65536_SR48000.psf	Panel setting file for THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 48000 Hz.
	\Pro\THD_FFT65536_SR48000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 48000 Hz.
	\Pro\THD_FFT65536_SR48000_FifthOrder.psf	Panel setting file for THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 48000 Hz. THD is calculated to the fifth order.
	\Pro\THD_FFT65536_SR96000.psf	Panel setting file for THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 96000 Hz.
	\Pro\THD_FFT65536_SR96000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 96000 Hz.
	\Pro\THD_FFT65536_SR192000.psf	Panel setting file for THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 192000 Hz.
	\Pro\THD_FFT65536_SR192000_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 65536 and Sampling Frequency = 192000 Hz.
	\Pro\THD_FFT65536_FifthOrder.psf	Panel setting file for THD, THD+N measurement with FFT size = 65536. THD is calculated to the fifth order.
	\Pro\THD_FFT65536_A-Weighting.psf	Panel setting file for A weighted THD, THD+N measurement with FFT size = 65536.
	\Pro\THD_FFT65536.psf	Panel setting file for THD, THD+N measurement with FFT size = 65536.
	IMD_CCIF2_19kHz(1)_20kHz(1)_FFT32768.psf	Panel setting file for IMD CCF2 (19 kHz + 20 kHz, 1:1) measurement with FFT size = 32768
	IMD_CCIF3_13kHz(1)_14kHz(1)_FFT32768.psf	Panel setting file for IMD CCF3 (13 kHz + 14 kHz, 1:1) measurement with FFT size = 32768
	IMD_CCIF3_14kHz(1)_15kHz(1)_FFT32768.psf	Panel setting file for IMD CCF3 (14 kHz + 15 kHz, 1:1) measurement with FFT size = 32768
	IMD_CCIF3_15kHz(1)_16kHz(1)_FFT32768.psf	Panel setting file for IMD CCF3 (15 kHz + 16 kHz, 1:1) measurement with FFT size = 32768

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	IMD_DIN_250Hz(4)_8kHz(1)_FFT32768.psf	Panel setting file for IMD DIN (250 Hz + 8 kHz, 4:1) measurement with FFT size = 32768
	IMD_SMPTE_60Hz(4)_7kHz(1)_FFT32768.psf	Panel setting file for IMD SMPTE (60 Hz + 7 kHz, 4:1) measurement with FFT size = 32768
	CrosstalkAB_FFT32768_SR44100.psf	Panel setting file for Crosstalk A->B measurement with FFT size = 32768 and Sampling Frequency = 44100 Hz
	CrosstalkAB_FFT32768_SR48000.psf	Panel setting file for Crosstalk A->B measurement with FFT size = 32768 and Sampling Frequency = 48000 Hz
	CrosstalkAB_FFT32768_SR96000.psf	Panel setting file for Crosstalk A->B measurement with FFT size = 32768 and Sampling Frequency = 96000 Hz
	CrosstalkAB_FFT32768_SR192000.psf	Panel setting file for Crosstalk A->B measurement with FFT size = 32768 and Sampling Frequency = 192000 Hz
	CrosstalkBA_FFT32768_SR44100.psf	Panel setting file for Crosstalk B->A measurement with FFT size = 32768 and Sampling Frequency = 44100 Hz
	CrosstalkBA_FFT32768_SR48000.psf	Panel setting file for Crosstalk B->A measurement with FFT size = 32768 and Sampling Frequency = 48000 Hz
	CrosstalkBA_FFT32768_SR96000.psf	Panel setting file for Crosstalk B->A measurement with FFT size = 32768 and Sampling Frequency = 96000 Hz
	CrosstalkBA_FFT32768_SR192000.psf	Panel setting file for Crosstalk B->A measurement with FFT size = 32768 and Sampling Frequency = 192000 Hz
	Octave1.psf	1/1 Octave Analysis with pink noise
	Octave3.psf	1/3 Octave Analysis with pink noise
	Octave6.psf	1/6 Octave Analysis with pink noise
	Octave12.psf	1/12 Octave Analysis with pink noise
	Octave24.psf	1/24 Octave Analysis with pink noise
	Octave3_PeriodicPinkNoise.psf	1/3 Octave Analysis with periodic pink noise
	\Pro\Octave48.psf	1/48 Octave Analysis
	\Pro\Octave96.psf	1/96 Octave Analysis
	\Pro\BodePlot_WhiteNoise.psf	Bode Plot (transfer function or frequency response) measurement using white noise, with stimulus fed into Channel B and response from DUT fed into Channel A.
	\Pro\BodePlot_LinearSweptSine(20~20kHz).psf	Bode Plot (transfer function or frequency response) measurement using frequency linearly swept sine, with stimulus fed into Channel B and response from DUT fed into Channel A.
	\Pro\BodePlot_LogSweptSine(20~20kHz).psf	Bode Plot (transfer function or frequency response) measurement using frequency logarithmically swept sine, with stimulus fed into Channel B and response from DUT fed into Channel A.
	\Pro\BodePlot_PeriodicWhiteNoise.psf	Bode Plot (transfer function or frequency response) measurement using periodic white noise, with stimulus fed into Channel B and response from DUT fed into Channel A.

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	\Pro\BodePlot_PeriodicPinkNoise.psf	Bode Plot (transfer function or frequency response) measurement using periodic pink noise, with stimulus fed into Channel B and response from DUT fed into Channel A.
	THD+N_THD_SNR_Magnitude_vs_Frequency.psf	Open the device test plan: THD+N_THD_SNR_Magnitude_vs_Frequency.dtp.
	THD+N_THD_vs_Magnitude_Power.psf	Open the device test plan: THD+N_THD_vs_Magnitude_Power.dtp.
	SMPTE_IMD_vs_Magnitude_Power.psf	Open the device test plan: SMPTE_IMD_vs_Magnitude_Power.dtp.
	AudioParameter_SR48000_A.psf	Open the device test plan: AudioParameter_SR48000_A.dtp.
	Polarity_Tester.psf	Speaker, microphone and amplifier polarity tester
	Polarity_Tester_WithCrestFactorCheck.psf	Speaker, microphone and amplifier polarity tester with crest factor check to confirm the presence of test tone
	PowerFactory_FFT32768_SR48000.psf	Power factor calculation by measuring the voltage and current simultaneously
	\pro\UnweightedPeakWowAndFlutter_10s.psf	Unweighted Peak Wow and Flutter Measurement (Average frequency measured by Spectrum Analyzer)
	\pro\WeightedPeakWowAndFlutter_10s.psf	Weighted Peak Wow and Flutter Measurement (Average frequency measured by Spectrum Analyzer)
	\pro\UnweightedPeakWowAndFlutterFrequenceCounter_10s.psf	Unweighted Peak Wow and Flutter Measurement (Average frequency measured by Frequency Counter)
	\pro\WeightedPeakWowAndFlutterFrequenceCounter_10s.psf	Weighted Peak Wow and Flutter Measurement (Average frequency measured by Frequency Counter)
	\pro\Loudness.psf	Loudness , Loudness Level and Sharpness Measurement
	VTDSO\Impedance_MultiTones_FFT32768_2810E.psf	VT DSO-2810E/2820E/2A10E/2A20E impedance measurement with 32768-point FFT MultiTone stimulus
	VTDSO\Impedance_MultiTones_Octave24_2810E.psf	VT DSO-2810E/2820E/2A10E/2A20E impedance measurement with 1/24 octave MultiTone stimulus
	VTDSO\Impedance_LogSweptSine_Octave96_2810E.psf	VT DSO-2810E/2820E/2A10E/2A20E impedance measurement with frequency logarithmically swept sine stimulus
	\Pro\Impedance_MultiTones_FFT65536_SoundCard.psf	Sound Card impedance measurement with 65536-point FFT MultiTone stimulus
	\Pro\Impedance_MultiTones_Octave24_SoundCard.psf	Sound Card impedance measurement with 1/24 octave MultiTone stimulus
	Pro\Impedance_LogSweptSine_Octave96_SoundCard.psf	Sound Card impedance measurement with frequency logarithmically swept sine stimulus
	\Pro\GedLeeMetric_FFT32768_SR48000_SingleChannel.psf	GedLee Metric measurement in Ch. A with FFT size = 32768 and Sampling Frequency = 48000 Hz
	\Pro\GedLeeMetric_FFT32768_SR48000.psf	GedLee Metric measurement in Ch. A & B with FFT size = 32768 and Sampling Frequency = 48000 Hz
	\Pro\Harmonic_FFT32768_SR48000_SingleChannel	Harmonic analysis for Ch. A with FFT size = 32768 and Sampling Frequency = 48000 Hz

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	\Pro\Harmonic_FFT32768_SR48000	Harmonic analysis for Ch. A&B with FFT size = 32768 and Sampling Frequency = 48000 Hz
	\Pro\MagnitudeFrequencyResponse_LogSweptSine_Octave96.psf	Panel setting file for magnitude frequency response measurement using 10-second 5~21000Hz logarithmic swept sine + 1/96 octave analysis.
	THD_FFT32768_ResidualComparison.psf	1kHz sinewave compared with its duplicate with fundamental attenuated by 120 dB via FIR
	J-Test_16Bit.psf	16-bit J-Test
	J-Test_24Bit.psf	24-bit J-Test
	RT60InterruptedNoise5s.pdf	Reverberation time measurement using interrupted noise method, Sampling Duration =5s
	DIM30_FFT131072_SR192000.psf	DIM30 measurement with FFT size=131072 and Sampling Frequency = 192 kHz
	DIM100_FFT262144_SR384000.psf	DIM100 measurement with FFT size = 262144 and Sampling Frequency=384 kHz
	\Pro\CEA-2010-20Hz.psf	20Hz subwoofer max. output test according to CEA-2010
	\Pro\CEA-2010-25Hz.psf	25Hz subwoofer max. output test according to CEA-2010
	\Pro\CEA-2010-31.5Hz.psf	31.5Hz subwoofer max. output test according to CEA-2010
	\Pro\CEA-2010-40Hz.psf	40Hz subwoofer max. output test according to CEA-2010
	\Pro\CEA-2010-50Hz.psf	50Hz subwoofer max. output test according to CEA-2010
	\Pro\CEA-2010-63Hz.psf	63Hz subwoofer max. output test according to CEA-2010
	\Pro\ImpulseResponse_Acoustic_LogSweptSine(10~24kHz)_SingleChannel.psf	Acoustic impulse response measurement with frequency logarithmically swept sine using single-channel method
	\Pro\ImpulseResponse_Acoustic_MLS_CrossCorrelation_SingleChannel.psf	Acoustic impulse response measurement by cross-correlation of MLS using single-channel method
	\Pro\ImpulseResponse_Acoustic_PeriodicPinkNoise_SingleChannel.psf	Acoustic impulse response measurement with periodic pink noise using single-channel method
	\Pro\ImpulseResponse_Acoustic_PeriodicWhiteNoise_SingleChannel.psf	Acoustic impulse response measurement with periodic white noise using single-channel method
	\Pro\ImpulseResponse_Acoustic_PeriodicWhiteNoise_CrossCorrelation_SingleChannel.psf	Acoustic impulse response measurement by cross-correlation of periodic white noise using single-channel method
	\Pro\ImpulseResponse_Acoustic_WhiteNoise_CrossCorrelation_SingleChannel.psf	Acoustic impulse response measurement by cross-correlation of white noise using single-channel method
	\Pro\ImpulseResponse_LogSweptSine(10~24kHz)_SingleChannel.psf	Impulse response measurement with frequency logarithmically swept sine using single-channel method
	ImpulseResponse_MLS_CrossCorrelation_SingleChannel.psf	Impulse response measurement by cross-correlation of MLS using single-channel method
	\Pro\ImpulseResponse_PeriodicPinkNoise_SingleChannel.psf	Impulse response measurement with periodic pink noise using single-channel method

	\Pro\ImpulseResponse_PeriodicWhiteNoise_SingleChannel.psf	Impulse response measurement with periodic white noise using single-channel method
	ImpulseResponse_PeriodicWhiteNoise_CrossCorrelation_SingleChannel.psf	Impulse response measurement by cross-correlation of periodic white noise using single-channel method
	ImpulseResponse_WhiteNoise_CrossCorrelation_SingleChannel.psf	Impulse response measurement by cross-correlation of white noise using single-channel method
	psf\pro\RT60InterruptedNoise5s.psf	Reverberation Time RT60 measurement using interrupted noise method
	psf\pro\RT60-STIImpulseResponse5s.psf	Reverberation Time RT60 and Speech Transmission Index STI measurement using impulse response method
	Pdf\VTDSO\StepResponseSR100MHz.psf	Step Response Analysis
\export	..	Default directory for output files
	DacFs200kHz2vMultiTonesAlignedToAdc50kHzFFT32768.txt	VT DSO-2810E/2820E/2A10E/2A20E impedance measurement reference Vo with 32768-point FFT MultiTone stimulus
	DacFs200kHz2vMultiTonesAlignedToAdc50kHzOctave24.txt	VT DSO-2810E/2820E/2A10E/2A20E impedance measurement reference Vo with 1/24 octave MultiTone stimulus
	DacFs200kHz1(6)vLogSweptSineAdc50kHzOctave96.txt	VT DSO-2810E/2820E/2A10E/2A20E impedance measurement reference Vo with frequency logarithmically swept sine stimulus
\ref	..	Default directory for reference files
	HighLimitRef.ref	High limit reference file
	LowLimitRef.ref	Low limit reference file
	RecordingRIAA.ref	Recording RIAA reference file
	ReproducingRIAA.ref	Reproducing RIAA reference file
	CEA-2010-20Hz.ref	20Hz subwoofer max. output test reference file according to CEA-2010
	CEA-2010-25Hz.ref	25Hz subwoofer max. output test reference file according to CEA-2010
	CEA-2010-31.5Hz.ref	31.5Hz subwoofer max. output test reference file according to CEA-2010
	CEA-2010-40Hz.ref	40Hz subwoofer max. output test reference file according to CEA-2010
	CEA-2010-50Hz.ref	50Hz subwoofer max. output test reference file according to CEA-2010
	CEA-2010-63Hz.ref	63Hz subwoofer max. output test reference file according to CEA-2010
\tcf	..	Default directory for tone configuration files
	IMD_CCIF2_19kHz(1)_20kHz(1).tcf	Tone configuration file for IMD CCIF2 (19 kHz + 20 kHz, 1:1) measurement.
	IMD_CCIF3_13kHz(1)_14kHz(1).tcf	Tone configuration file for IMD CCIF3 (13 kHz + 14 kHz, 1:1) measurement.
	IMD_CCIF3_14kHz(1)_15kHz(1).tcf	Tone configuration file for IMD CCIF3 (14 kHz + 15 kHz, 1:1) measurement.
	IMD_CCIF3_15kHz(1)_16kHz(1).tcf	Tone configuration file for IMD CCIF3 (15 kHz + 16 kHz, 1:1) measurement.
	IMD_DIN_250Hz(4)_8kHz(1).tcf	Tone configuration file for IMD DIN (250 Hz + 8 kHz, 4:1) measurement.
	IMD_SMPTE_60Hz(4)_7kHz(1).tcf	Tone configuration file for IMD SMPTE (60 Hz + 7 kHz, 4:1) measurement.

	OCT1_MultiTone_11.tcf	Tone configuration file for 1/1 octave bands for frequency response measurement
	OCT3_MultiTone_32.tcf	Tone configuration file for 1/3 octave bands for frequency response measurement
	OCT6_MultiTone_63.tcf	Tone configuration file for 1/6 octave bands for frequency response measurement
	OCT12_MultiTone_125.tcf	Tone configuration file for 1/12 octave bands for frequency response measurement
	DIM30_SharplyBandLimited.tcf	DIM30 test signal configuration with a 30kHz ideal low-pass filter
	DIM30_SinglePole_LowpassFiltered.tcf	DIM30 test signal configuration with a 30kHz first-order low-pass filter
	DIM100_SharplyBandLimited.tcf	DIM100 test signal configuration with a 100kHz ideal low-pass filter
	DIM100_SinglePole_LowpassFiltered.tcf	DIM100 test signal configuration with a 100kHz first-order low-pass filter
\wav	..	Default directory for WAV files as well as TXT files of raw data
	TextFileImport_SingleChannel.txt	A sample TXT file containing raw data of a single channel for File Import command.
	TextFileImport_DualChannel.txt	A sample TXT file containing raw data of two channels for File Import command.
	\window	This directory contains the 24-bit 1024-point WAV files for 60 window functions. These wave files can be used to evaluate the characteristics of these window functions.
	DacFs200kHz2vMultiTonesAlignedToAdc50kHzFFT32768.wav	VT DSO-2810E/2820E/2A10E/2A20E 32768-point FFT MultiTone stimulus for impedance measurement
	DacFs48kHzMultiTonesAlignedToAdc48kHzFFT65536.wav	Sound Card 65536-point FFT MultiTone stimulus for impedance measurement
\wfl	..	Default directory for waveform library
	heartbeat.wfl	A sample arbitrary waveform library file
	sine.wfl	Sine-shaped waveform library file
	sinepulse.wfl	Sine-pulse-shaped waveform library file
	square.wfl	Square-shaped waveform library file
	triangle.wfl	Triangle-shaped waveform library file
	sawtooth.wfl	Sawtooth-shaped waveform library file
	InvertedSawtooth.wfl	Inverted Sawtooth-shaped waveform library file
	sinc1.wfl	SINC (contains more than 10 peaks) waveform library
	sinc2.wfl	SINC (contains about 10 peaks) waveform library
	J-Test_16Bit.bpl	16 bit J-Test BitPerfect library
	J-Test_24Bit.bpl	24 bit J-Test BitPerfect library
	SineAtOneFourthFs_16Bit.bpl	16 bit, ¼ sampling frequency sine wave BitPerfect library
	SineAtOneFourthFs_24Bit.bpl	24 bit, ¼ sampling frequency sine wave BitPerfect library
	DC_16Bit.bpl	16 bit Full-Scale (positive) DC signal
	DC_24Bit.bpl	24 bit Full-Scale (positive) DC signal

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13 User Customization and Software Development

Multi-Instrument provides the following user customizable features:

1. You can save a frequently used panel setting as a Panel Setting File via [Setting]>[Save Current Panel Setting], and load it via [Setting]>[Load Panel Setting] when you need it.

2. You can save your default pane setting via [Setting]>[Save Current Panel Setting as Default]. It will be loaded automatically the next time after software startup. If you do not want unauthorized users to change this default setting, tick "Lock Panel Setting after startup" in the Display Setting dialog box which is opened via [Setting]>[Display], and set a password for unlocking panel setting via [Setting]>[Change Password].

   If you tick "Save Current Panel Setting on exit" in the Display Setting dialog box, then the software will save the panel setting as default on exit, and load it the next time after software startup.

3. You can configure your frequently use Panel Setting Files in the Hot Panel Setting Toolbar, so that they can be loaded at one single mouse click. If you do not want unauthorized users to change these pre-configured settings, tick "Lock Panel Setting after startup" in the Display Setting dialog box which is opened via [Setting]>[Display], and set a password for unlocking panel setting via [Setting]>[Change Password]. In this way, the unauthorized users are only allowed to use these pre-configured panel settings.

4. You can configure, save and load your own device test plan and LCR meter. If you want to load a device test plan automatically at software startup, open the Device Test Plan panel, load the device test plan and press "Save Current Panel Setting as Default" in the Device Test Plan panel.

5. You can hide the toolbars via [Setting]>[Display] and tick the respective checkboxes.

Multi-Instrument provides the following software development features:

1. Multi-Instrument can work as an ActiveX automation server so that an external program can access the data and functions that Multi-Instrument exposes. You can integrate Multi-Instrument into your own software seamlessly via the ActiveX automation server interfaces exposed by Multi-Instrument.

   Please refer to: *Multi-Instrument Automation Server Interfaces*

   Download link:
   https://www.virtins.com/Multi-Instrument-Automation-Server-Interfaces.pdf

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The above document and the sample automation client programs in Visual C++, Visual Basic, Visual C# and Python can be found in the AutomationAPIs directory of the software.

2. You can use the vtDAQ and vtDAO interface DLLs supplied in this software to allow your own back-end software to interface to sound cards, NI DAQmx cards, VT DSOs, VT RTAs, VT IEPE, VT CAMP, etc.. You can also develop your own vtDAQ and vtDAO compatible DLLs to allow Multi-Instrument to interface to your own hardware.

   Please refer to: *vtDAQ and vtDAO_Interfaces*

   Download link:
   https://www.virtins.com/vtDAQ-and-vtDAO-Interfaces.pdf

   The above document and the sample DAQ and DAO back-end programs in Visual C++, Visual C# and LabVIEW can be found in the DAQDAOAPIs directory of the software.

3. Virtins Technology's Signal Processing and Analysis (vtSPA) Application Programming Interfaces (APIs) provides a suite of generic APIs for data processing and analysis. It contains some unique features / algorithms originated and only available from Virtins Technology.

   Please refer to: *Signal Processing and Analysis (vtSPA) Interfaces*

   Download link:
   https://www.virtins.com/Signal-Processing-and-Analysis-APIs.pdf

   The above document and the sample programs in Visual C++ and Visual C# can be found in the DAQDAOAPIs directory of the software.

Furthermore, Multi-Instrument is well prepared to be rebranded for OEM services. Its look and feel can be readily changed through configuration without even reprogramming. Contact Virtins Technology if interested.

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14 References

1. vtDAQ and vtDAO Interfaces

   Download link:
   https://www.virtins.com/vtDAQ-and-vtDAO-Interfaces.pdf

2. Multi-Instrument Automation Server Interfaces

   Download link:
   https://www.virtins.com/Multi-Instrument-Automation-Server-Interfaces.pdf

3. VIRTINS Sound Card Oscilloscope Probe Manual

   Download link:
   https://www.virtins.com/Virtins-Sound-Card-Oscilloscope-Probe-Manual.pdf

4. Basic Oscilloscope, Spectrum Analyzer, Multimeter and Signal Generator Functional Tests using Multi-Instrument and Pocket Multi-Instrument with Sound Cards

   Download link:
   https://www.virtins.com/doc/D1001/Basic_Oscilloscope_Spectrum_Analyzer_Multimeter_and_Signal_Generator_Functional_Tests_using_Multi-Instrument_and_Pocket_Multi-Instrument_with_Sound_Cards_D1001.pdf

5. FFT Basics and Case Study using Multi-Instrument

   Download link:
   https://www.virtins.com/doc/D1002/FFT_Basics_and_Case_Study_using_Multi-Instrument_D1002.pdf

6. Evaluation of Various Window Function using Multi-Instrument

   Download link:
   https://www.virtins.com/doc/D1003/Evaluation_of_Various_Window_Functions_using_Multi-Instrument_D1003.pdf

7. EMU Tracker Pre Test Report using Multi-Instrument

   Download link:
   https://www.virtins.com/doc/D1004/EMU_Tracker_Pre_Report_D1004.pdf

8. M-Audio Mobile TrackerPre Test Report using Multi-Instrument

   Download link:
   https://www.virtins.com/doc/D1005/M-Audio_MobilePre_Report_D1005.pdf

9. VT XLR-to-USB Pre Test Report using Multi-Instrument

   Download link:

https://www.virtins.com/doc/D1006/VT_XLR-to-USB_Pre_Report_D1006.pdf

10. EMU 0204 Test Report using Multi-Instrument

    Download link:
    https://www.virtins.com/doc/D1007/EMU_0204_Report.pdf

11. Signal Processing and Analysis (vtSPA) Intefaces

    Download link:
    https://www.virtins.com/Signal-Processing-and-Analysis-APIs.pdf

12. Polarity Check for Speakers, Microphones and Amplifiers using Multi-Instrument

    Download link:
    https://www.virtins.com/doc/Polarity-Check-for-Speakers-Microphones-and-Amplifiers-using-Multi-Instrument.pdf

13. Focusrite Scarlett Solo Test Report using Multi-Instrument

    Download link:
    https://www.virtins.com/doc/Focusrite-Scarlett-Solo-Test-Report-using-Multi-Instrument.pdf

14. Wow and Flutter Measurement using Multi-Instrument

    Download link:
    https://www.virtins.com/doc/Wow-and-Flutter-Measurement-using-Multi-Instrument.pdf

15. Measurement of Total Harmonic Distortion and Its Related Parameters using Multi-Instrument

    Download link:
    https://www.virtins.com/doc/Measurement-of-Total-Harmonic-Distortion-and-Its-Related-Parameters-using-Multi-Instrument.pdf

16. Measurements of Various Intermodulation Distortions (IMD, TD+N, DIM) using Multi-Instrument

    Download link:
    https://www.virtins.com/doc/Measurements-of-Various-Intermodulation-Distortions-IMD-TD+N-DIM-using-Multi-Instrument.pdf

17. Shock Testing with Multi-Instrument - Classical Shock Pulses and Shock Response Spectrum

    Download link:

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https://www.virtins.com/doc/Shock-Testing-with-Multi-Instrument-Classical-Shock-Pulses-and-Shock-Response-Spectrum.pdf

18. Frequency Weightings for Sound and Vibration Perceived by Humans using Multi-Instrument

    Download link:
    https://www.virtins.com/doc/Frequency-Weightings-for-Sound-and-Vibration-Perceived-by-Humans-using-Multi-Instrument.pdf

19. Software Methods for Non-Linear Distortion Compensation and Noise Reduction in Measurements

    Download link:
    https://www.virtins.com/doc/Software-Methods-for-Non-Linear-Distortion-Compensation-And-Noise-Reduction-In-Measurements.pdf

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15 FAQ

1. For a sound card based system, is there any test lead or probe associated with the software? How to make the connection to external devices?

The dedicated sound card oscilloscope probe can be purchased separately from us. However, if you want to make the probe by yourself, please refer to Section "Input & Output Connection" in the first chapter.

2. For a sound card based system, how to calibrate the input channel and output channel?

The simplest way is to use a multimeter. Please refer to Section "Calibration" in the Oscilloscope chapter for detail. If VIRTINS Sound Card Oscilloscope Probes are used, then you also need to refer to the manual of the probe.

3. Can I use all instruments such as Oscilloscope, Spectrum Analyzer and Signal Generator simultaneously?

Yes. For example, you can use the Signal Generator to generate a test signal to a Device Under Test (DUT), then capture and analyze the response from that device via the Oscilloscope and Spectrum Analyzer simultaneously.

4. Does the software support external USB sound cards?

Yes. It supports any Windows compatible sound cards. You can select the sound card via [Setting]>[ADC Device] or [DAC Device].

5. Does the software support multiple sound cards in one computer?

Yes. Each instance of the software supports the selection of one sound card for input and one for output. The selection is done via [Setting]>[ADC Device] or [DAC Device]. If you run multiple instances of the software, you can use multiple sound cards or multiple channels of one sound card simultaneously.

6. Can I see the generated signal in the Oscilloscope in real time?

Yes, you can. Different types of loopback modes are provided in the Signal Generator. Please refer to Section "Loopback Mode" in the Signal Generator chapter for detail.

7. Can I analyze data that are not directly acquired from the ADC device?

Yes. You can use the software to analyze any WAV data or TXT data as long as the required format is followed. Please refer to Sections "Open" and "Import" in the Oscilloscope chapter for detail.

8. Can it measure DC signal with a sound card?

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It depends on the sound card used. Most of sound card inputs are AC-coupled. There is normally a capacitor in series at the input, which blocks DC and very-low-frequency components.

9. Can it output DC signal with a sound card?

Generally No. Almost all sound cards are AC-coupled at their output.

10. How to make the software accessible by different limited user accounts and administrator accounts under Windows NT/2000/XP/2003/Vista/7/8/8.1/10/11?

The software must be installed and then unlocked by an administrator. It can be accessed by any users including restricted users afterwards.

11. "MMSYSERR_INVALHANDLE" and then "Fail to start DAQ!" are displayed after lunching the software?

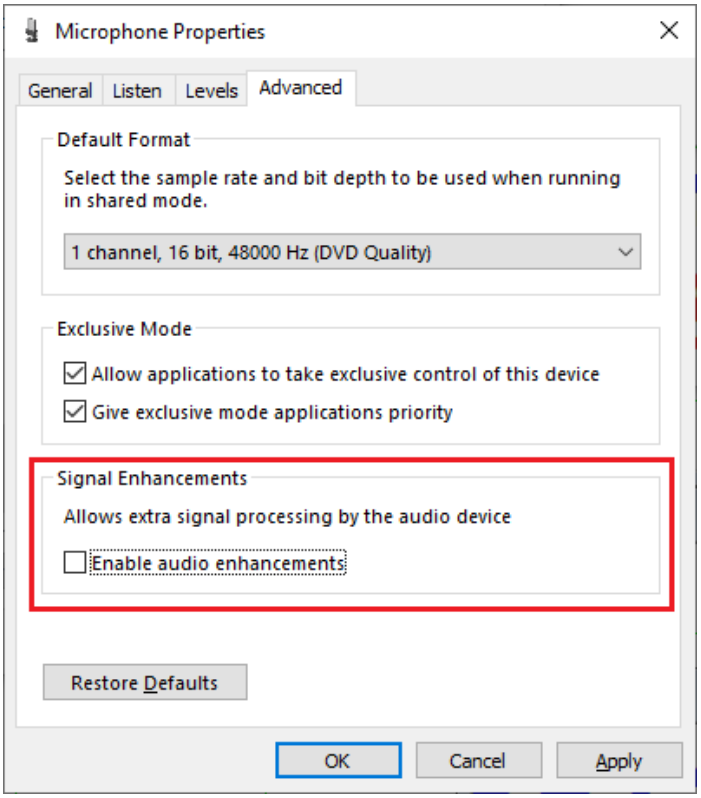
This is most likely because your computer microphone is selected as the ADC Device and Sound Card MME driver is used, but your Windows microphone privacy setting prevents the software from accessing your microphone. You can enable it via [Windows Start Button] -> [Setting] -> [Privacy] -> [Microphone].

12. "WAVERR_BADFORMAT" pops up followed by "Fail to start DAQ!" if Signal Generator is already running or "Fail to Start DAO" if Oscilloscope is already running?

This is most likely because you are using the same sound card as both the ADC and DAC devices, and Sound Card MME driver is used, but the sampling rates configured under the "Advanced" tab of Windows Recording and Playback Controls differ.

13. Should I disable the default "Enable Audio Enhancements" option under the "Advanced" tab in Windows Recording Control for the audio device used by the software?

Yes, it MUST be disabled. Some Windows versions / editions come with some audio signal enhancement features which are enabled by default. These features must be disabled through the Sound Recording Control under Windows Control Panel to prevent them from altering the originally sampled data, as shown below. This is particularly important when Sound Card MME driver is used in the software.



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