

# VT RTA-268A/B/C Manual

**An Ultra Low-Noise Real Time Acoustic Analyzer, Sound Level Meter, Distortion Analyzer, Polarity Tester, Time Delay Finder ...**



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

## TABLE OF CONTENTS

<b>1 INSTALLATION AND QUICK START GUIDE .....</b>	<b>3</b>
1.1 PACKAGE CONTENTS .....	3
1.1.1 Standard Package .....	3
1.1.2 Optional Items .....	4
1.2 HARDWARE CONNECTION .....	5
1.3 HARDWARE DRIVER INSTALLATION .....	6
1.4 MULTI-INSTRUMENT SOFTWARE INSTALLATION AND CONFIGURATION .....	6
1.4.1 Install Multi-Instrument .....	6
1.4.2 Start Multi-Instrument .....	6
1.4.3 Configure Multi-Instrument .....	8
1.5 INPUT OF SOUND LEVEL CALIBRATION DATA .....	13
1.5.1 0dB Reference Vr .....	14
1.6 MICROPHONE FREQUENCY COMPENSATION .....	15
1.7 TWENTY MOST FREQUENTLY USED MEASUREMENT SETTINGS .....	17
1.8 MORE ACOUSTIC ANALYSIS FUNCTIONS AVAILABLE IN MULTI-INSTRUMENT PRO OR ABOVE .....	21
1.9 LIST OF AUDIO TEST WAV FILES (AudioTestCD.zip) .....	21
1.10 OPERATION NOTES .....	23
1.11 UAC-232 CONTROL PANEL .....	23
1.12 UAC-232 MIX CONTROL .....	24
1.13 TIME DELAY MEASUREMENT .....	25
1.13.1 Two-channel Time Delay Measurement using Cross Correlation .....	25
1.13.2 Single-channel Time Delay Measurement using Auto Correlation .....	27
1.13.3 Single-Channel Time Delay Measurement using Pulse Emission .....	27
1.14 AUDIO AND ULTRASOUND ANALYSIS .....	27
<b>2 SPECIFICATIONS .....</b>	<b>29</b>
2.1 VT RTA-268A/B/C HARDWARE CONFIGURATION .....	29
2.2 OVERALL VT RTA-268A/B/C SPECIFICATIONS .....	29
2.3 UAC-232 AUDIO INTERFACE SPECIFICATIONS .....	29
2.4 MEASUREMENT MICROPHONE M215 SPECIFICATIONS (FOR VT RTA-268A) .....	33
2.5 MEASUREMENT MICROPHONE M215L SPECIFICATIONS (FOR VT RTA-268B) .....	34
2.6 MEASUREMENT MICROPHONE M50 SPECIFICATIONS (FOR VT RTA-268C) .....	35
2.7 MULTI-INSTRUMENT SOFTWARE SPECIFICATIONS .....	35
<b>3 MULTI-INSTRUMENT SOFTWARE LICENSE INFORMATION .....</b>	<b>45</b>
3.1 LICENSE TYPES .....	45
3.2 LICENSE UPGRADE FROM ONE LEVEL TO ANOTHER .....	45
3.3 SOFTWARE UPGRADE IN THE SAME LICENSE LEVEL .....	45
<b>4 WARRANTY .....</b>	<b>45</b>
<b>5 DISCLAIMER .....</b>	<b>46</b>

# 1 Installation and Quick Start Guide

VT RTA-268A/B/C is a series of PC-based ultra-low-noise huge-dynamic-range real-time acoustic analyzers, powered by the comprehensive virtual instrument software, Multi-Instrument®. The USB audio interface used in the analyzers has two ADCs working in tandem to create a single output data stream in 32-bit float format. One “low gain” ADC is optimized for high-level signals, and the other “high gain” ADC is optimized for low-level signals. This state of the art dual-ADC architecture provides an unmatched dynamic range compared with the conventional single ADC approach. With a proper microphone, it is now possible to capture sound across the entire human hearing range without sacrificing fidelity at low signal levels or encountering clipping at high signal levels, and this can be achieved without the hassle of manual analog gain adjustment.

VT RTA-268A/B/C has built-in DACs, so it can also be used as a stereo signal generator at the same time.

## 1.1 Package Contents

### 1.1.1 Standard Package

A standard VT RTA-268A/B/C Package contains the following items:

- 1) XLR Measurement microphone and its accessories (a windshield, a microphone clip and stand)



*XLR Measurement Microphone*



*Windshield*



*Microphone Clip and Stand*

- 2) XLR-to-USB Audio Interface



- 3) USB Cable (1 m)



## 4) XLR Cable (1.5 m)



## 5) CD (containing the copy-protected Multi-Instrument software)



The latest software can always be downloaded from [www.virtins.com/MIsetup.exe](http://www.virtins.com/MIsetup.exe).

## 6) USB hardkey (containing a Multi-Instrument Pro license)



## 7) Audio Test WAV files (containing a list of audio test tones and noises)

They can be downloaded at: [www.virtins.com/AudioTestCD.zip](http://www.virtins.com/AudioTestCD.zip) and then burnt into a CD or saved into a USB memory stick. They can also be generated directly from the Signal Generator of Multi-Instrument.

## 8) Carrying Case



## 9) Unique Sound Level Calibration Data (in CD)

## 10) Unique Microphone Calibration Data (in CD, only applicable to RTA-268C)

### 1.1.2 Optional Items

- 1) An additional XLR measurement microphone and its accessories
- 2) P48-to-IEPE converter (so that IEPE sensors can be connected to the XLR input)

- 3) IEPE sensors
- 4) Software license upgrade

## 1.2 Hardware Connection

### Sound Measurement

Connect the XLR measurement microphone to the input jack 1 of the audio interface.



Connect the corresponding ends of the USB cable to the XLR-to-USB Audio interface and a PC's USB port. When this is complete, the blue LED at the top right corner of the front panel lights up, indicating that it is receiving power.



Then switch on the 48V phantom power of input jack 1. This is required by the condenser measurement microphone used here, meanwhile, the Guitar/Bass switch for HiZ support MUST be turned off.

### Software Activation

The USB hardkey needs to be plugged into any USB port of the PC in order to activate the Multi-Instrument software. Otherwise the software will run under the 21-day fully functional

free trial mode before the trial period expires.

### **Test Signal Generation**

You can generate the test signals from the Signal Generator of Multi-Instrument. The XLR-to-USB audio interface has two ¼-inch TRS jacks on its back panel, respectively for left and right balanced line-level outputs, and a ¼-inch TRS stereo headphone jack on its front panel. The test signals output from these jacks can be fed into the audio system under test.

Alternatively, you can play the test signals from an audio CD or a memory stick.

## **1.3 Hardware Driver Installation**

The device driver installer RTA268ABC\_DriverInstaller.exe is located in the Drivers\VTRTA268ABC directory in the CD. When you install the Multi-Instrument software, a copy of the device driver installer will also be installed in the software installation directory\Drivers\VTRTA268ABC. Run it to install the driver. After driver installation, the device will be supported by both Sound Card MME and ASIO drivers in Multi-Instrument.

## **1.4 Multi-Instrument Software Installation and Configuration**

Multi-Instrument is a powerful multi-function virtual instrument software. 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 DSOs and so on. It consists of multiple test instruments such Oscilloscope, Spectrum Analyzer, and Multimeter, etc.

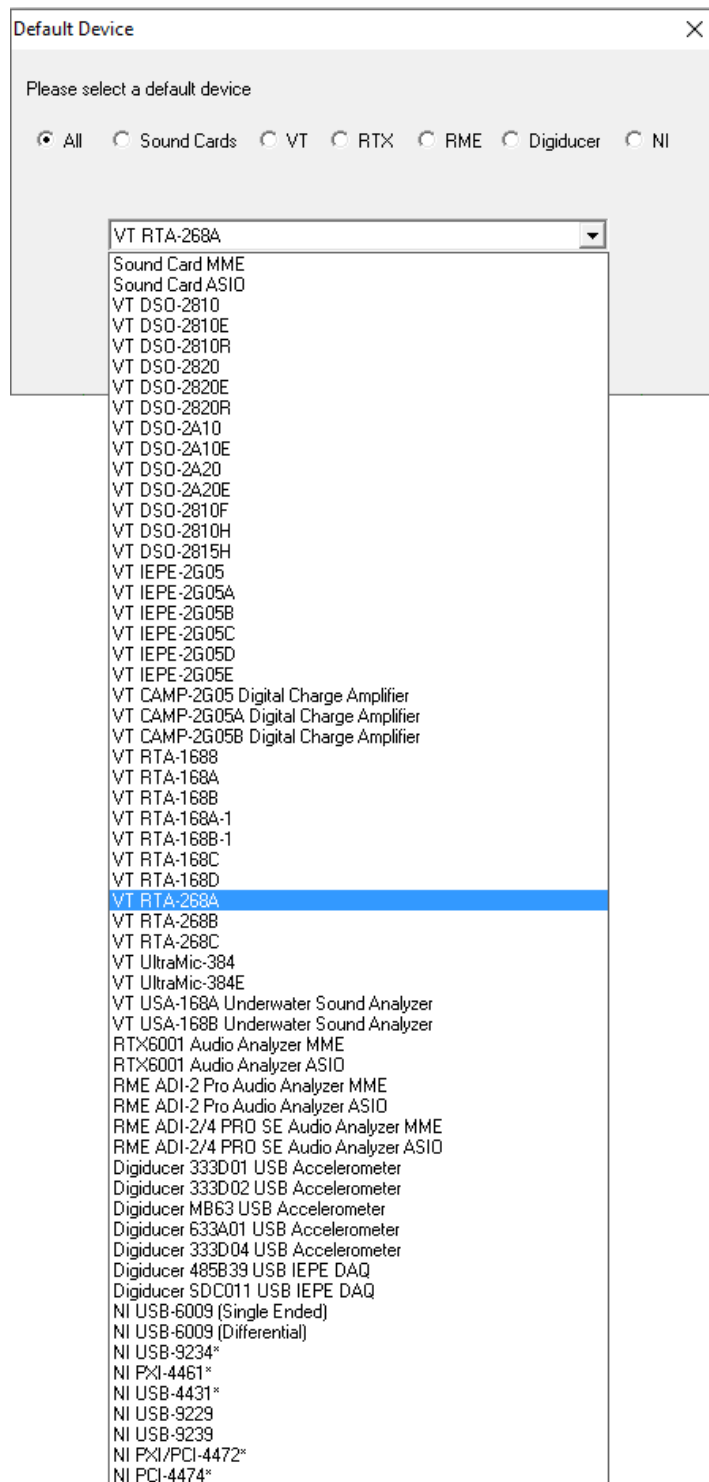
### **1.4.1 Install Multi-Instrument**

Insert the Multi-Instrument installation CD into your computer's CD-ROM drive and follow the instruction on the screen to install the Multi-Instrument software. The installation file can also be downloaded from [www.virtins.com/MIsetup.exe](http://www.virtins.com/MIsetup.exe).

### **1.4.2 Start Multi-Instrument**

To start Multi-Instrument, on the Windows desktop, select [Start]>[All Programs]>[Multi-Instrument]>[VIRTINS Multi-Instrument], or simply double click the MI icon.

If the software is started for the very first time after installation, it will prompt the user to select a default device (see figure below). Select VT RTA-268A/B/C according to the hardware device to be used.



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. Otherwise, you will have to enter the unique calibration data that come with the product package again.**

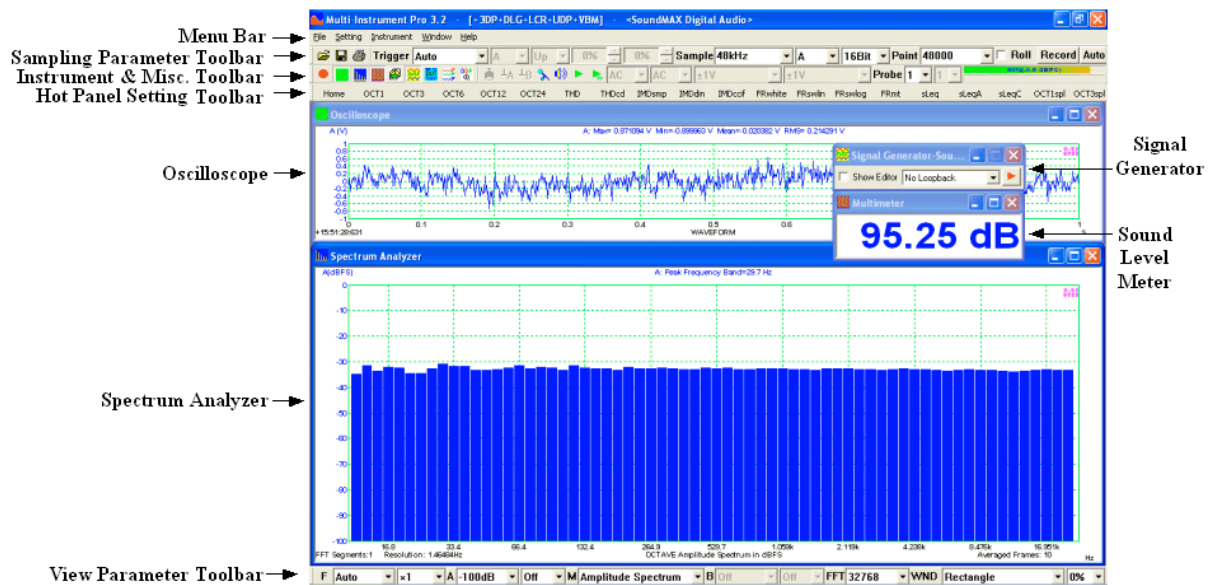


In case the driver has not yet been installed, the software will prompt the user to install the driver. Restarting program is required after that.

After the default device is selected, the software will prompt the user to select a default color scheme (Skin). The default skin can also be changed later via [Setting]>[Display].



The main window of the software will open after the above skin selection. The following figure shows a typical screen layout (obtained by clicking the “OCT3” button in the Hot Panel Setting Toolbar after the launch of the software). Please refer to the software manual for detailed functions of the software. The software manual can be accessed via [Start]>[All Programs]>[Multi-Instrument]>[VIRTINS Multi-Instrument Manual] (in PDF format) or [VIRTINS Multi-Instrument Help] (in HTML format) on Windows Start menu, or [Help]>[Software Manual] or F1 inside the software.



### 1.4.3 Configure Multi-Instrument

In Multi-Instrument, the menu items are enabled / disabled based on context. Many menu items are disabled when the Oscilloscope or the Signal Generator is running. To do the configuration, stop the oscilloscope first by pressing the green button at the upper left corner of the screen (see figure below). The button will turn red once the Oscilloscope is stopped.



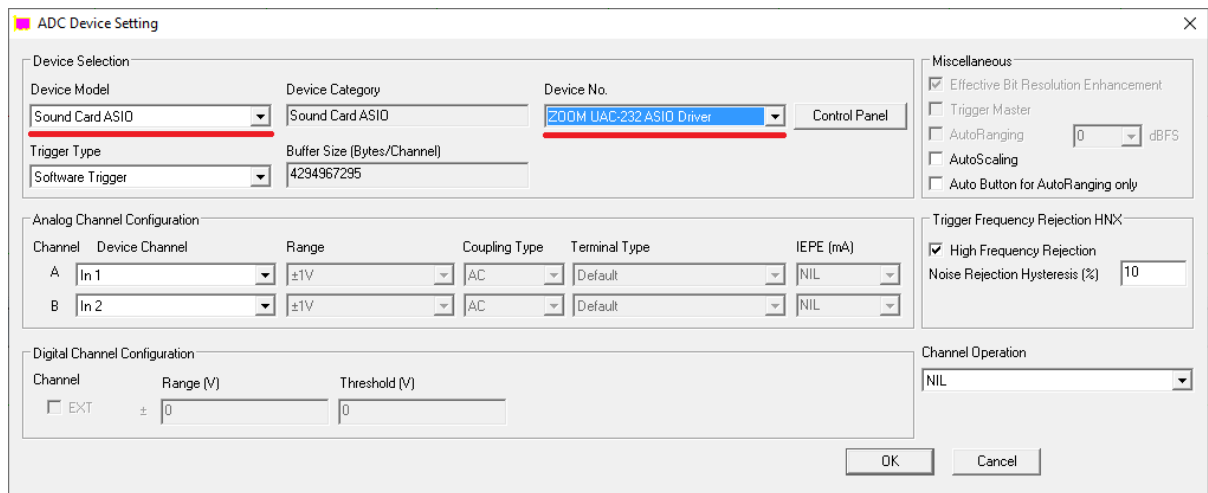


### 1.4.3.1 Configure Sound Recording Devices for Multi-Instrument

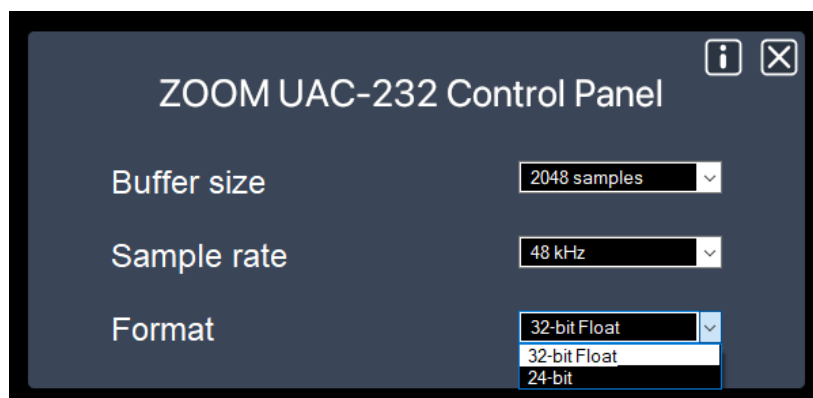
The audio interface can be used with either MME or ASIO driver. ASIO driver is recommended as it uses the sampling rate selected in Multi-Instrument directly and bypasses the possible sampling rate conversion and audio signal enhancement by Windows, which may otherwise alter the original samples and lead to measurement inaccuracies.

#### 1.4.3.1.1 Using ASIO Driver

Go to [Setting]>[ADC Device], and select “Sound Card ASIO” in the “Device Model” field. Then choose “ZOOM UAC-232 ASIO Driver” in the “Device No.” field (see figure below). This is to configure the audio interface as the sound recording device for the software.



Click the “Control Panel” button to bring up the control panel for the audio interface (see figure below).



#### 1.4.3.1.1.1 Buffer Size

There is generally no need to change the default Buffer Size for test and measurement applications. A large buffer size is preferred, though.

#### *1.4.3.1.1.2 Sampling Rate*

Under ASIO, the Sampling Rate will be set directly in Multi-Instrument and thus changing it here will have no effect. Instead, the Sampling Rate here will passively follow what is set in Multi-Instrument.

On the other hand, if MME is used instead, you can only change the actual sampling rate of the audio interface here. The change here will be reflected in the sampling rate displayed in the Sound Recording Tab under the Windows Control Panel. When the sampling rate set in Multi-Instrument differs from the one here, Windows will conduct sampling rate conversion automatically.

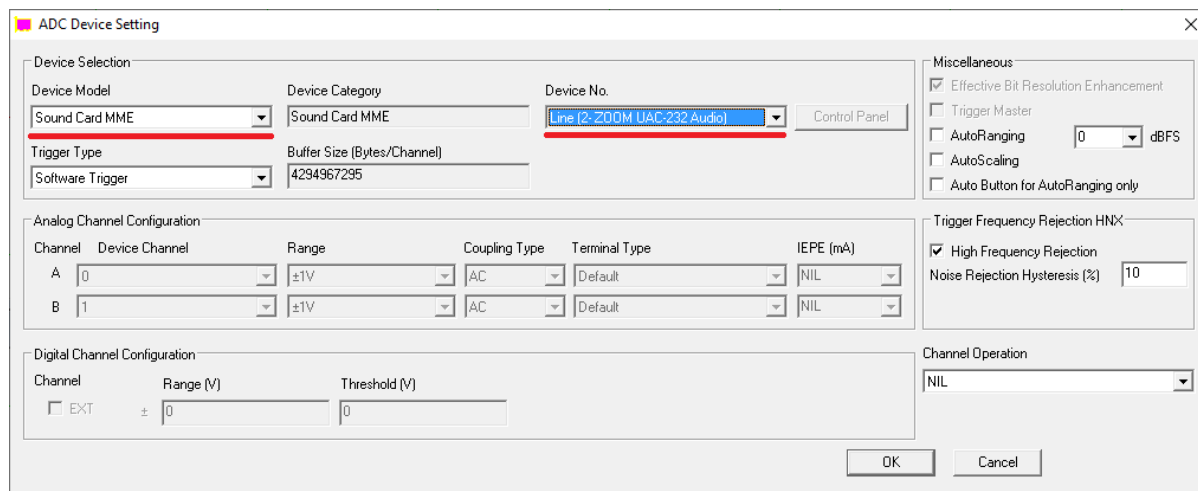
#### *1.4.3.1.1.3 Format*

Two formats are supported by the audio interface: 24-bit (Integer) or 32-bit Float. One unique feature of the audio interface is its support for 32-bit float format. This format provides much finer resolution than the 24-bit (Integer) format when the signal is weak, thanks to the audio interface's state of the art dual-ADC architecture, with one optimized for weak signals and the other optimized for strong ones. Both formats have the same resolution when the signal is strong. It is thus recommended to use the 32-bit float format for sound recording (ADC). However, currently its ASIO driver does not support 32-bit float format for signal output (DAC). Hence, MME should be used for sound playback (DAC) instead. When the 32-bit float format is used, the Sampling Bit Resolution in Multi-Instrument must be set to 32-bit, otherwise, an error message will pop up. When the ADC is running in the 32-bit float mode, the "32-Bit FLOAT" sign on the front panel of the audio interface will be lit up, and "Float32" will be displayed at the middle bottom of the Oscilloscope window in Multi-Instrument.

Now, if you start the oscilloscope by pressing the red button at the upper left corner of the screen, and then talk before the measurement microphone, you should be able to see your "voices" in the Oscilloscope and Spectrum Analyzer.

#### *1.4.3.1.2 Using MME Driver*

Go to [Setting]>[ADC Device], and select "Sound Card MME" in the "Device Model" field. Then choose "Line (ZOOM UAC-232 Audio)" in the "Device No." field (see figure below). This is to configure the audio interface as the sound recording device for the software.



**ADC Device Setting**

**Device Selection**

Device Model: Sound Card MME | Device Category: Sound Card MME | Device No.: Line (2-ZOOM UAC-232 Audio) | Control Panel

Trigger Type: Software Trigger | Buffer Size (Bytes/Channel): 4294967295

**Analog Channel Configuration**

Channel	Device Channel	Range	Coupling Type	Terminal Type	IEPE (mV)
A	0	±1V	AC	Default	NIL
B	1	±1V	AC	Default	NIL

**Digital Channel Configuration**

Channel: EXT | Range (V): ±0 | Threshold (V): 0

**Miscellaneous**

☒ Effective Bit Resolution Enhancement

☐ Trigger Master

☐ AutoRanging: 0 dBFS

☐ AutoScaling

☐ Auto Button for AutoRanging only

**Trigger Frequency Rejection HNZ**

☒ High Frequency Rejection

Noise Rejection Hysteresis (%): 10

**Channel Operation**

NIL

OK Cancel

Note that the displayed name of the audio interface may vary a bit under different Windows versions or with different USB ports.

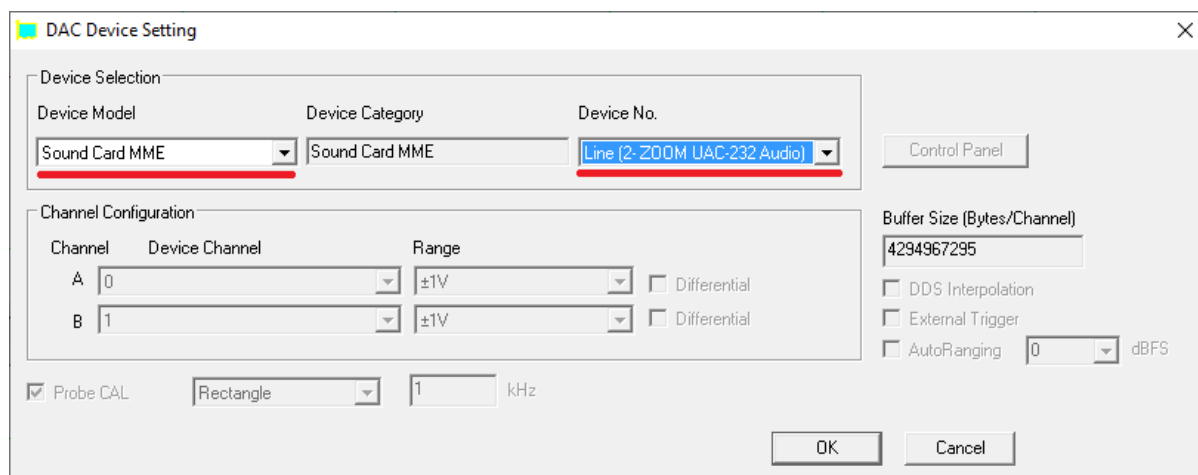
Now, if you start the oscilloscope by pressing the red button at the upper left corner of the screen, and then talk before the measurement microphone, you should be able to see your “voices” in the Oscilloscope and Spectrum Analyzer.

Note that MME does not support 32-bit float format.

#### 1.4.3.2 Configure Sound Playback Devices for Multi-Instrument

The audio interface has left and right balanced line-level outputs and a stereo headphone output. You can use the Signal Generator in Multi-Instrument to generate the test signals through them.

If ASIO with 32-bit float format (recommended) is used for sound recording (ADC), MME must be chosen for sound playback (DAC). To do that, go to [Setting]>[DAC Device], and select “Sound Card MME” in the “Device Model” field and “Line (ZOOM UAC-232 Audio)” in the “Device No.” field (see figure below).



**DAC Device Setting**

**Device Selection**

Device Model: Sound Card MME | Device Category: Sound Card MME | Device No.: Line (2-ZOOM UAC-232 Audio) | Control Panel

**Channel Configuration**

Channel	Device Channel	Range	Differential
A	0	±1V	<input type="checkbox"/>
B	1	±1V	<input type="checkbox"/>

☒ Probe CAL | Rectangle | 1 kHz

Buffer Size (Bytes/Channel): 4294967295

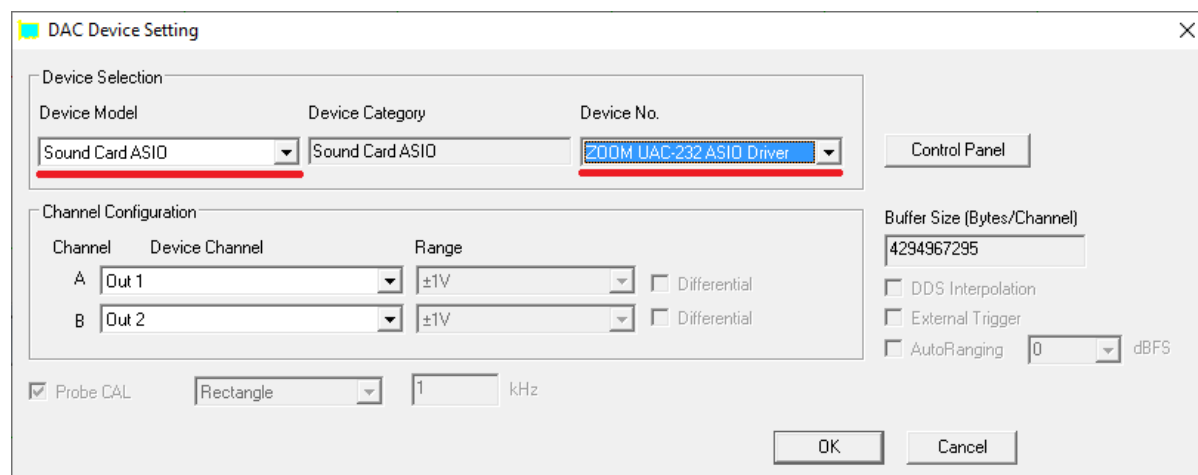
☐ DDS Interpolation

☐ External Trigger

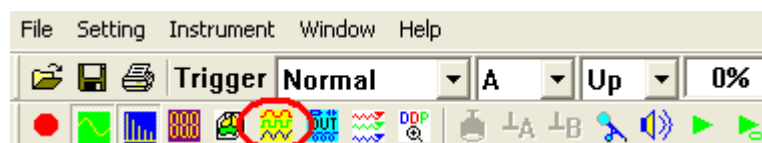
☐ AutoRanging: 0 dBFS

OK Cancel

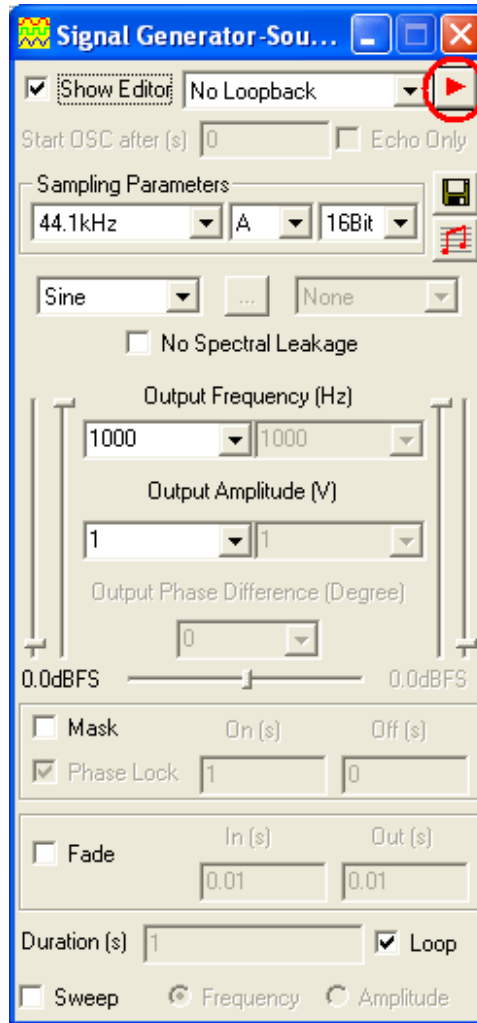
If ASIO with 24-bit format is used for sound recording (ADC), ASIO must be chosen for sound playback (DAC). To do that, go to [Setting]>[DAC Device], and select “Sound Card ASIO” in the “Device Model” field and “ZOOM UAC-232 ASIO Driver” in the “Device No.” field (see figure below).



If MME is used for sound recording (ADC), then MME must be chosen for sound playback (DAC).



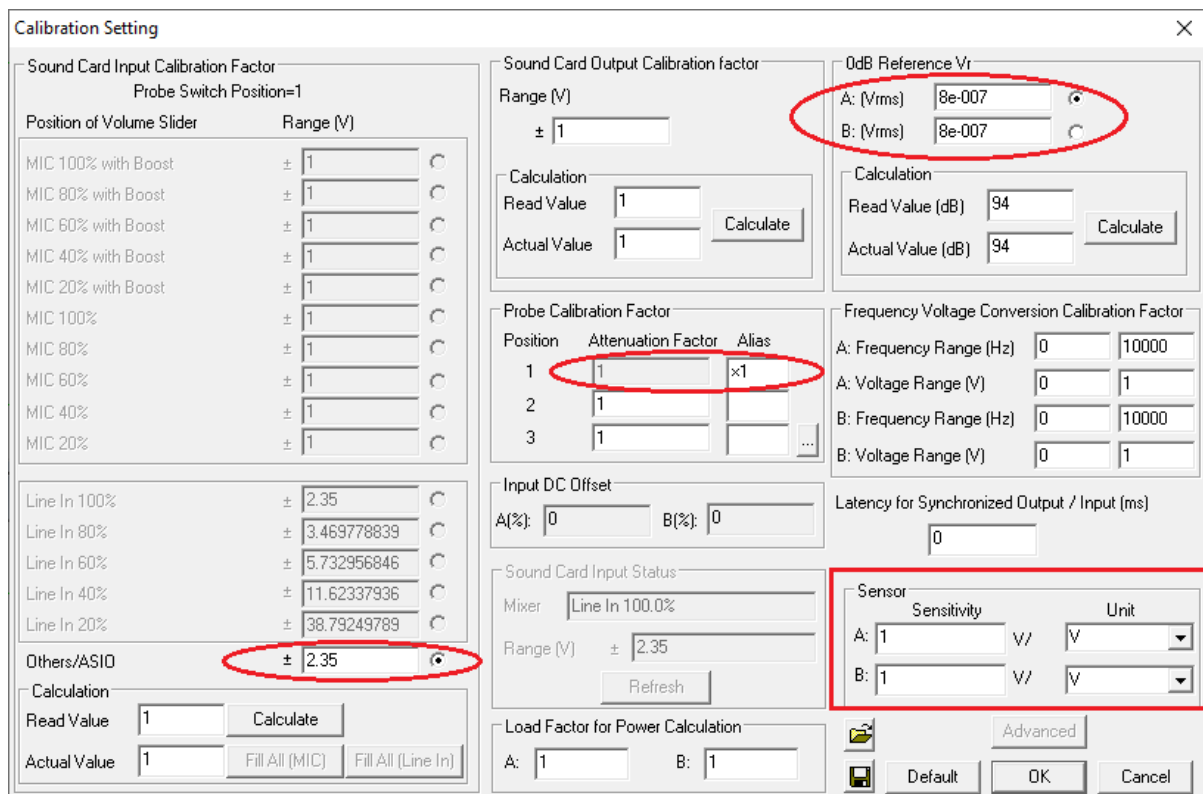
Now, if you press the Signal Generator button (see figure above), the Signal Generator panel will be opened (see figure below). Press the red triangle button at the upper right corner of the Signal Generator panel, you should hear a 1kHz test tone from the speaker or earphone connected to the audio interface. Press it again to stop the sound.



## 1.5 Input of Sound Level Calibration Data

Sound Level Calibration is not required for those relative measurements such as frequency response, THD, THD+N, IMD, etc.. It is required only if you want to measure the absolute sound level.

Please refer to the calibration data sheet provided in the product package and enter the calibration data in the respective highlighted fields in the Calibration Setting dialog box (see figures below). The Calibration Setting dialog box can be opened via [Setting]>[Calibration] in Multi-Instrument. Note that the “0dB Reference Vr” calibration data in the following figure are examples only. You should enter the actual calibration data provided inside the product package.



**Calibration Setting**

Sound Card Input Calibration Factor  
Probe Switch Position=1

Position of Volume Slider      Range (V)

MIC 100% with Boost	± 1	<input type="radio"/>
MIC 80% with Boost	± 1	<input type="radio"/>
MIC 60% with Boost	± 1	<input type="radio"/>
MIC 40% with Boost	± 1	<input type="radio"/>
MIC 20% with Boost	± 1	<input type="radio"/>
MIC 100%	± 1	<input type="radio"/>
MIC 80%	± 1	<input type="radio"/>
MIC 60%	± 1	<input type="radio"/>
MIC 40%	± 1	<input type="radio"/>
MIC 20%	± 1	<input type="radio"/>

Line In 100%      ± 2.35      ☐

Line In 80%      ± 3.469778839      ☐

Line In 60%      ± 5.732956846      ☐

Line In 40%      ± 11.62337936      ☐

Line In 20%      ± 38.79249789      ☐

Others/ASIO      ± 2.35      ☒

Calculation  
Read Value: 1      Calculate  
Actual Value: 1      Fill All (MIC)      Fill All (Line In)

Sound Card Output Calibration factor  
Range (V)      ± 1  
Calculation  
Read Value: 1      Calculate  
Actual Value: 1

0dB Reference Vr  
A: (Vrms)      8e-007      ☒  
B: (Vrms)      8e-007      ☐  
Calculation  
Read Value (dB): 94      Calculate  
Actual Value (dB): 94

Probe Calibration Factor  
Position      Attenuation Factor      Alias  
1      1      ×1  
2      1  
3      1

Frequency Voltage Conversion Calibration Factor  
A: Frequency Range (Hz)      0      10000  
A: Voltage Range (V)      0      1  
B: Frequency Range (Hz)      0      10000  
B: Voltage Range (V)      0      1

Input DC Offset  
A(%): 0      B(%): 0

Sound Card Input Status  
Mixer: Line In 100.0%  
Range (V)      ± 2.35      Refresh

Latency for Synchronized Output / Input (ms)      0

Sensor  
Sensitivity      Unit  
A: 1      V/V      V  
B: 1      V/V      V

Load Factor for Power Calculation  
A: 1      B: 1

Advanced  
Default      OK      Cancel

The audio interface has an ultra-low noise level and offers a huge dynamic range of about 135dB, owing to its state-of-the-art dual-ADC architecture. Consequently, the absence of analog gain adjustment knobs on the audio interface panel is a deliberate design choice, as the exceptional dynamic range eliminates the need for such adjustments.

The audio interface also comes with a piece of utility software called UAC-232 Mix Control (see Section 1.12), where the digital gain of the input channels 1 & 2 can be adjusted from 0dB to 60dB. **Please note that the calibration data are valid only when the digital gains are set at 0dB.** The digital gain should always be kept at 0dB in order not to invalidate the calibration. Under the 32-bit float mode, the quality of the acquired data will not be affected by the digital gain settings.

### 1.5.1 0dB Reference Vr

The “0dB Reference Vr” in the Calibration Setting dialog box is used to finally calibrate the input voltage to dBSPL. The input voltage here should be considered as a relative value as the actual input is not a voltage but a sound pressure. The “0dB Reference Vr” is the parameter to be recalibrated if a sound level recalibration is necessary. To do the recalibration, simply enter the actual sound level value into the “Actual Value” edit box and the measured sound level value into the “Read Value” edit box, and then press the “Calculate” button ONCE. It should be noted that the Sensor Sensitivity is kept at 1V/V in this method.

In case you want to display the measured raw data in Pa rather than Volt, then you should enter the “actual” Sensor Sensitivity in V/Pa, which can be calculated using the following formula:

$$\text{Sensitivity} = [\text{"0dB Reference Vr"} \text{ (in Volt)}] / [\text{Standard 0dB Reference (in Pa)}]$$

where the Standard 0dB Reference for sound pressure level in air is 20  $\mu\text{Pa}$  (i.e.  $2\text{e-}005$  Pa) and that in water is 1  $\mu\text{Pa}$  (i.e.  $1\text{e-}006$  Pa). If the "actual" Sensor Sensitivity is used, then the standard 0dB reference should be entered into the "0dB Reference Vr" edit box. For example, the following two methods are equivalent.

Method 1: [0dB Reference Vr] =  $8\text{e-}007$  (V) and [Sensor Sensitivity] = 1 V/V

Method 2: [0dB Reference Vr] =  $2.0\text{e-}005$  (Pa) and [Sensor Sensitivity] = 0.04 V/Pa

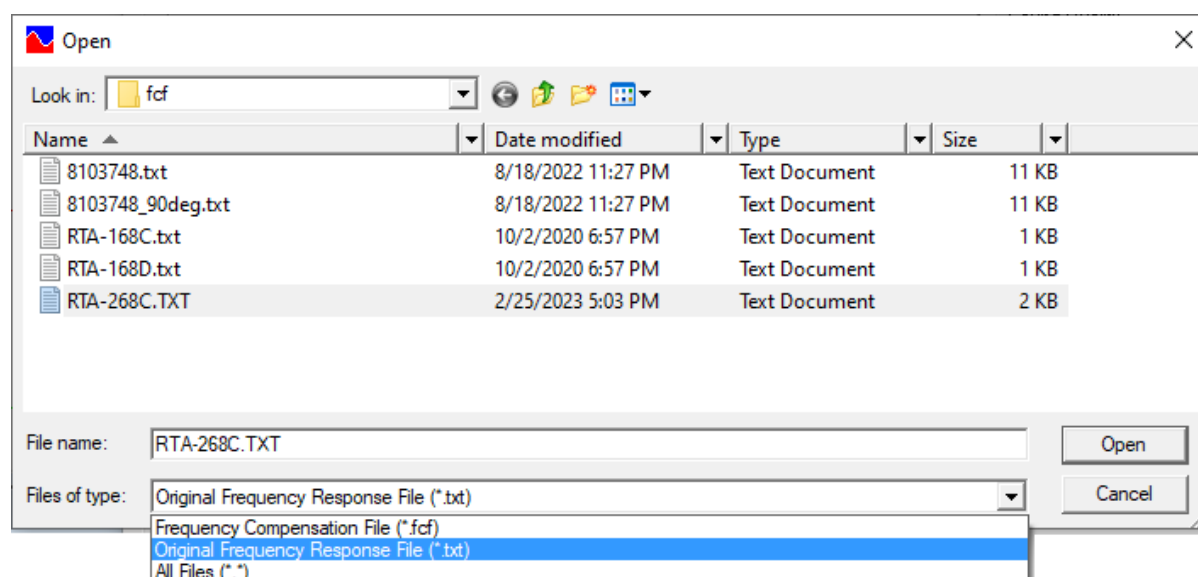
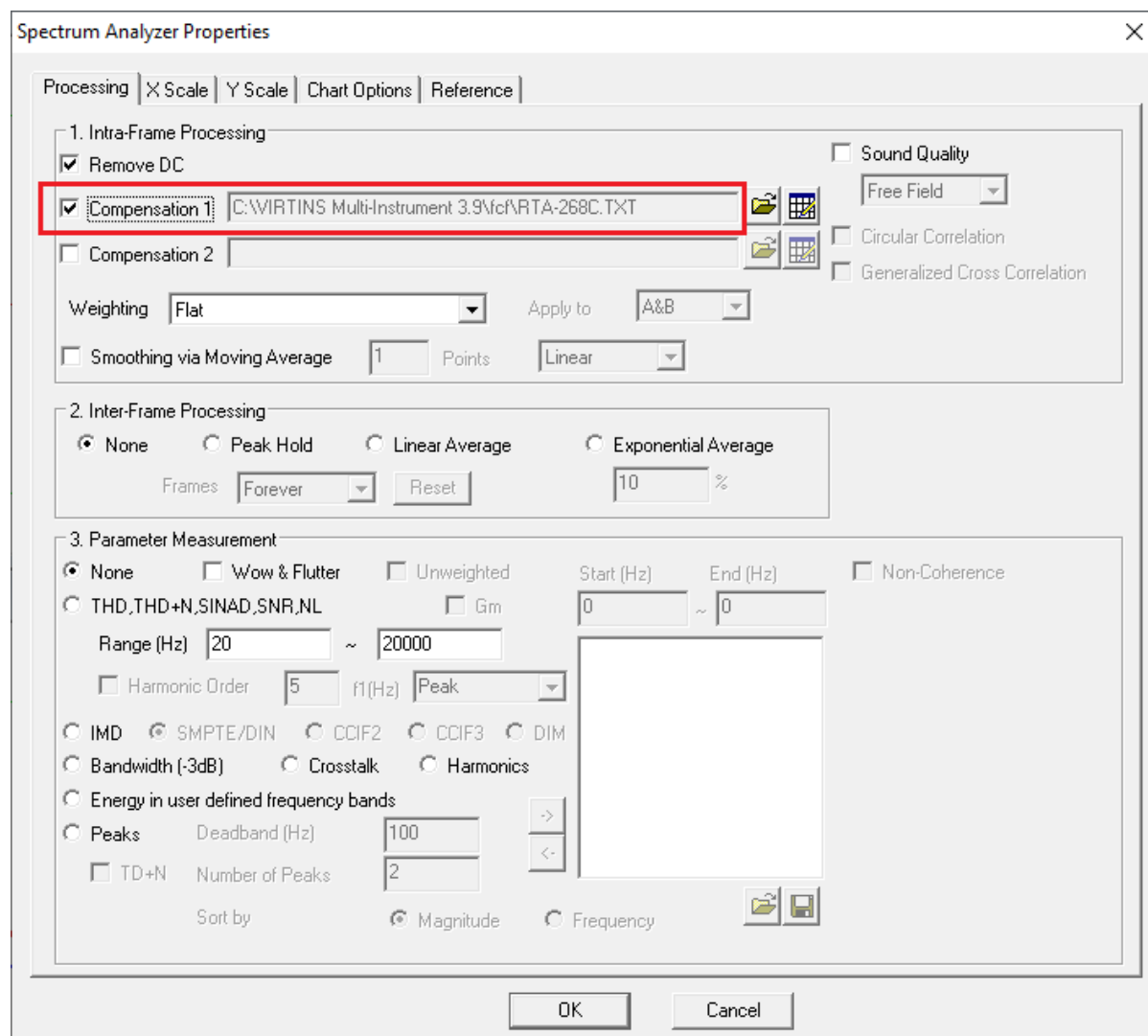
## 1.6 Microphone Frequency Compensation

The measurement microphones in RTA-268 series have extremely flat frequency responses in the audio frequency range. Generally, there is no need to compensate for the microphone frequency response. However, a higher degree of accuracy can be achieved if the microphone comes with a frequency response file or a frequency compensation file. The difference between these two files is the sign of the gain values (in dB). For example, a gain of 3dB in a frequency response file should have a gain of -3dB in its equivalent frequency compensation file.

The XLR measurement microphone in RTA-268C comes with an individually calibrated Original Frequency Response File RTA-268C.txt. "Original" means that it has exactly the same format and contents as the calibration file provided by the microphone manufacturer. **This file is located in the CD directory "\Microphone Frequency Compensation File". Please copy this file to the "\fcr" directory under the installation directory of Multi-instrument in the hard disk (if the file already exists, overwrite it), so that you can use it later.** This file is also used by the default panel settings configured in the Hot Panel Setting Toolbar (The third toolbar from the top), except the buttons labelled "Ultrasound", "TDac", "TDcc" and "TDgcc".

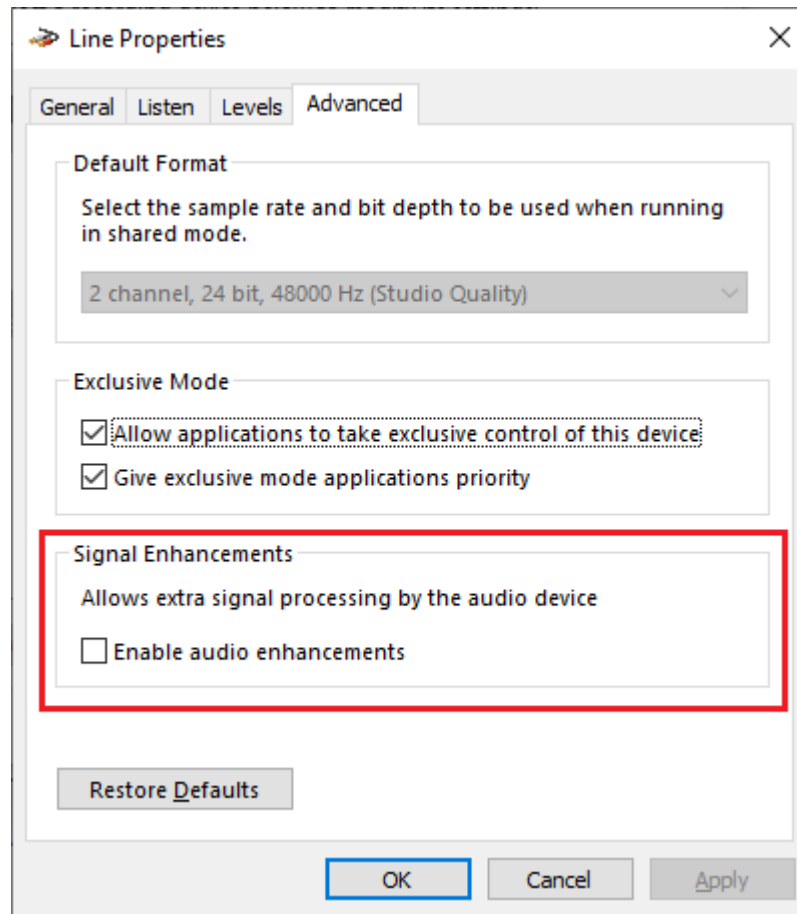
You can load the Original Frequency Response File by right clicking anywhere within the Spectrum Analyzer window, selecting [Spectrum Analyzer Processing]> "Intra-Frame Processing"> "Compensation 1", changing the "Files of type" from "Frequency Compensation File (\*.fcr)" to "Original Frequency Response File (\*.txt)" in the pop-up File Open dialog box, and loading that file (see figures below). If you want to apply it to the twenty most frequently used panel settings in the Hot Panel Setting Toolbar, you will need to configure and save the corresponding panel setting files one by one. Luckily, this has already been done for the default panel settings.





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. One of the possible problems caused by these features is the unwanted alteration of the frequency response of the setup. Using ASIO driver instead of MME driver can avoid these problems.



## 1.7 Twenty Most Frequently Used Measurement Settings

Multi-Instrument bundled with VT RTA-268A/B/C comes with many pre-configured panel setting files. This saves you time in configuring various parameters for some frequently performed measurements by yourself. You can load these panel setting files via [Setting]>[Load Panel Setting]. Furthermore, 20 most frequently used panel setting files are pre-configured in the Hot Panel Setting Toolbar (The third toolbar from the top). You can load one of them by a single mouse click. These 20 measurement settings are:

### (1) Home: Default Setting

The factory default panel setting. It is equivalent to the [File]>[New] command.

### (2) OCT3spl: 1/3 Octave Analysis (Avg. 10) in SPL

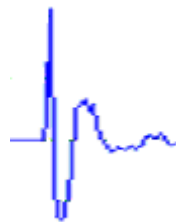
Pink noise will be generated by pressing the start button of the Signal Generator. The pink noise will be injected into the DUT (Device Under Test), and the response of the DUT will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. A flat curve in the Spectrum Analyzer would indicate a flat magnitude frequency response of the DUT. The equivalent continuous sound level in dB will be displayed in the Multimeter window.

### (3) OCT3ppn: 1/3 Octave Analysis

Same as OCT3spl, but periodic pink noise instead of ordinary (non-periodic) pink noise is used as the stimulus. One advantage of using periodic pink noise is that its spectrum is ideally flat under octave analysis even without inter-frame averaging. Also, the Y axis is displayed in dBFS instead of dB SPL in Spectrum Analyzer.

### (4) Polarity: Speaker, Microphone, Amplifier Polarity Tester with Crest Factor Check

A polarity test signal will be generated by pressing the start button of the Signal Generator. Point the microphone close to the center of the speaker under test along its axis, and observe the polarity of the captured pulse in the oscilloscope window. If the pulse goes positive initially, then the polarity of the speaker under test is positive, and vice versa.



Positive



Negative

### (5)

#### RT60pn: Reverberation Analysis via Interrupted Noise (1/1 Octave Band) (applicable to VT RTA-268A/B)

Pink noise will be generated for 5 seconds only by pressing the start button of the Signal Generator. It will be projected into the enclosure under test. After about 4.5 seconds, the Oscilloscope will be started automatically by the software under the “Sync. No Loopback” mode of the Signal Generator. This is to capture the last 0.5 second of the emitted pink noise and 4.5 seconds of its subsequent decay process. **The data analysis may take quite a long time. Please be patient!** The measured RT20 value in each 1/1 octave band as well as its overall value will be displayed. RT30 can also be displayed if needed. Please refer to the sections for Reverberation in the software manual for details.

#### Ultrasound: Audio and Ultrasound Analysis (applicable to VT RTA-268C)

Unlike other pre-configured panel settings in this toolbar, the microphone frequency response file RTA-268C.txt is not configured in the Spectrum Analyzer in order to allow for a much wider frequency range, which is theoretically from 0 to 96 kHz.

### (6) RT60imp: Reverberation & Speech Intelligibility Analysis via Impulse Response (1/1 Octave Band)

Reverberation and Speech intelligibility analysis will be performed based on Impulse Response Schroeder Integration. The impulse response can be obtained through direct methods (such as a gunshot) or indirect methods (such as transfer function measurement). **The data analysis may take quite a long time. Please be patient!** Please refer to the sections for Reverberation and Speech Intelligibility in the software manual for details.

### (7) THDcd: 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 should be used if different sound cards are used for signal input and output. It should also be used if the 1 kHz test tone is played from an audio CD or a memory stick.

(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 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) 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.

(12) 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 inter-frame averaging.

(13) FRswLin: Magnitude Frequency Response (Frequency Sweep, Linear)

A 0.68266667-second 20Hz-to-20kHz linear frequency swept sine wave will be generated by pressing the start button of the Signal Generator. The 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. Note that you will need to adjust the trigger level so that the Oscilloscope will be triggered just upon the start of the sweep.

(14) FRswLog: Magnitude Frequency Response (Frequency Sweep, Log)

A 0.68266667-second 20Hz-to-20kHz logarithmic frequency swept sine wave will be generated by pressing the start button of the Signal Generator. The 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. Note that you will need to adjust the trigger level so that the Oscilloscope will be triggered just upon the start of the sweep.

(15) FRmt: Magnitude Frequency Response (MultiTone, 31 1/3 Octave Bands)

A multitone consists of 31 1/3 octave band center frequencies from 20 Hz to 20kHz will be generated by pressing the start button of the Signal Generator. The 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.

(16)sLeq: Short Equivalent Continuous Sound Level (Short Leq, 125ms)

This panel setting will display the short equivalent continuous sound level. The value is time-averaged over continuous 125 ms.

(17)sLeqA: Short Equivalent Continuous Sound Level (Short Leq, 125ms) (A-Weighted)

This panel setting will display the A-weighted short equivalent continuous sound level. The value is time-averaged over continuous 125 ms.

(18)TDac: Single-Channel Time Delay Measurement via Auto Correlation

White noise will be generated from Output Ch.A by pressing the start button of the Signal Generator. The superimposed direct and reflected signals from Input Ch.A will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. The time delay between them will be measured by locating the second highest auto correlation peak along the time axis. The corresponding auto correlation coefficient will also be displayed. A positive coefficient indicates that the superimposed two signals are positively correlated and vice versa. A deadband of 0.25ms is configured in [Spectrum Analyzer Processing] by default to enforce a minimum interval between the first and second highest auto correlation peaks in order to avoid false peak identification.

(19)TDcc: Dual-Channel Time Delay Measurement via Cross Correction (oB->iB)

White noise will be generated from Output Ch.B by pressing the start button of the Signal Generator. The signals from Input Ch.A and Ch.B will be captured and analyzed by the Oscilloscope and Spectrum Analyzer. The time delay between them will be measured by locating the highest cross correlation peak along the time axis. The corresponding cross correlation coefficient will also be displayed. A positive coefficient indicates the two signals are positively correlated and vice versa. The signal fed to Input Ch.B can also be a loopback signal from Output Ch.B. If that is the case, it is assumed that the loopback is done through UAC-232 Mix Control without any external connection. The Output Ch.A is set to NONE to cut off the loopback from Output Ch.A to Input Ch.A. The test signal will be output from Output Ch.B only. This loopback method introduces a system lag that is a function of sampling rate (e.g. 625 $\mu$ s@48kHz, 250 $\mu$ s@192kHz). It can be removed from the measured time delay using UDDP (User Defined Data Point). This corrected time delay will also be shown.

(20)TDgcc: Dual-Channel Time Delay Measurement via Generalized Cross Correction (oB->iB)

This panel setting is similar to TDcc, except that [Spectrum Analyzer Processing]> “Intra-Frame Processing” > “Generalized Cross Correlation” checkbox is ticked. Generalized cross correlation can be used with more general test signals such as live speech or music, while cross correlation can only be used with test signals with a flat spectrum such as white noise and MLS.

The sampling rate for VT RTA-268A/B is configured to be 48kHz for all the 20 panel settings above, while that for VT RTA-268C is configured to be 192kHz. Note that under ASIO, changing sampling rate may cause reordering of the device list under MME. If that happens, you just need to reconfigure the Device No. under MME.

## 1.8 More Acoustic Analysis Functions Available in Multi-Instrument Pro or Above

Multi-Instrument is very versatile and supports a comprehensive set of acoustic measurements. To highlight a few not mentioned in the previous sections: Reverberation (including RT60), Speech Intelligibility (including STI), ANSI / CEA-2010 Subwoofer Peak SPL Test, Cumulative Spectral Decay, Speaker Rub & Buzz, Sound Quality (loudness, loudness level, sharpness, articulation index, etc.), etc. Some of these functions are only available in Multi-Instrument Pro or above. Please refer to Multi-Instrument software manual for more details.

## 1.9 List of Audio Test WAV Files (AudioTestCD.zip)

Track	Description	Length (second)	Recommended Panel setting to be used
1	Pink Noise in Phase (-1dBFS)	298	OCT1 ~ OCT24 OCT1spl, OCT3spl
2	Pink Noise out of Phase (-1dBFS)	298	
3	White Noise in Phase (-1dBFS)	298	FRwhite
4	White Noise out of Phase (-1dBFS)	298	
5	Log Sweep 20Hz~20kHz (-1dBFS)	60	
6	Log Sweep 20Hz~20kHz out of Phase (-1dBFS)	60	
7	Linear Sweep 20Hz~20kHz (-1dBFS)	60	
8	Linear Sweep 20Hz~20kHz out of Phase (-1dBFS)	60	
9	Log Sweep 20Hz~200Hz (-1dBFS)	60	
10	Log Sweep 20Hz~200Hz out of Phase (-1dBFS)	60	
11	Linear Sweep 20Hz~200Hz (-1dBFS)	60	
12	Linear Sweep 20Hz~200Hz out of Phase (-1dBFS)	60	
13	Log Sweep 2kHz~20kHz (-1dBFS)	60	
14	Log Sweep 2kHz~20kHz out of Phase (-1dBFS)	60	
15	Linear Sweep 2kHz~20kHz (-1dBFS)	60	
16	Linear Sweep 2kHz~20kHz out of Phase (-1dBFS)	60	
17	Log Sweep 20Hz~20kHz (-1dBFS)	0.683	FRswlog
18	Log Sweep 20Hz~20kHz out of Phase (-1dBFS)	2.73	
19	Linear Sweep 20Hz~20kHz (-1dBFS)	0.683	FRswlin
20	Linear Sweep 20Hz~20kHz out of Phase (-1dBFS)	2.73	
21	MultiTone32Octave1/3 in Phase (-1dBFS)	60	FRmt
22	MultiTone32Octave1/3 out of Phase (-1dBFS)	60	



23	1kHz (0dBFS)	60	
24	1kHz (-1dBFS)	60	THDcd
25	1kHz (-3dBFS)	60	
26	1kHz (-10dBFS)	60	
27	1kHz (-60dBFS)	60	
28	60Hz and 7kHz mixed at 4: 1 (-1dBFS)	60	IMDsmg
29	250Hz and 8kHz mixed at 4: 1 (-1dBFS)	60	IMDdin
30	19kHz and 20kHz mixed at 1: 1 (-1dBFS)	60	IMDccif
31	1kHz Square (-1dBFS)	60	
32	Digital Silence	60	
33	LeftRight (English)	60	
34	Left Right Center Surround (English)	60	
35	LeftRight (Chinese)	60	
36	Left Right Center Surround (Chinese)	60	
37	12.5 Hz (-1dBFS)	60	1/3 Octave Band 1
38	16 Hz (-1dBFS)	60	1/3 Octave Band 2
39	20 Hz (-1dBFS)	60	1/3 Octave Band 3
40	25 Hz (-1dBFS)	60	1/3 Octave Band 4
41	31.5 Hz (-1dBFS)	60	1/3 Octave Band 5
42	40 Hz (-1dBFS)	60	1/3 Octave Band 6
43	50 Hz (-1dBFS)	60	1/3 Octave Band 7
44	63 Hz (-1dBFS)	60	1/3 Octave Band 8
45	80 Hz (-1dBFS)	60	1/3 Octave Band 9
46	100 Hz (-1dBFS)	60	1/3 Octave Band 10
47	125 Hz (-1dBFS)	60	1/3 Octave Band 11
48	160 Hz (-1dBFS)	60	1/3 Octave Band 12
49	200 Hz (-1dBFS)	60	1/3 Octave Band 13
50	250 Hz (-1dBFS)	60	1/3 Octave Band 14
51	315 Hz (-1dBFS)	60	1/3 Octave Band 15
52	400 Hz (-1dBFS)	60	1/3 Octave Band 16
53	500 Hz (-1dBFS)	60	1/3 Octave Band 17
54	630 Hz (-1dBFS)	60	1/3 Octave Band 18
55	800 Hz (-1dBFS)	60	1/3 Octave Band 19
56	1250 Hz (-1dBFS)	60	1/3 Octave Band 21
57	1600 Hz (-1dBFS)	60	1/3 Octave Band 22
58	2000 Hz (-1dBFS)	60	1/3 Octave Band 23
59	2500 Hz (-1dBFS)	60	1/3 Octave Band 24
60	3150 Hz (-1dBFS)	60	1/3 Octave Band 25
61	4000 Hz (-1dBFS)	60	1/3 Octave Band 26
62	5000 Hz (-1dBFS)	60	1/3 Octave Band 27
63	6300 Hz (-1dBFS)	60	1/3 Octave Band 28
64	8000 Hz (-1dBFS)	60	1/3 Octave Band 29
65	10000 Hz (-1dBFS)	60	1/3 Octave Band 30
66	12500 Hz (-1dBFS)	60	1/3 Octave Band 31
67	16000 Hz (-1dBFS)	60	1/3 Octave Band 32
68	20000 Hz (-1dBFS)	60	1/3 Octave Band 33
69	999.9481201 Hz (-1dBFS)	60	THD
70	One 0.5ms Inverted Saw Tooth Pulse Every 100ms (-1dBFS)	1	Polarity Test

Note that you can compare the sound level of the in-phase signal with that of the out-of-phase signal to judge whether you need to swap the polarity of the speakers in one channel.



## 1.10 Operation Notes

- The sound level of the test signal should be at least 30 dB higher than that of the background noise in order to achieve sufficient signal-to-noise ratio in the measurement.
- The volume of the test signal should be adjusted such that the input peak level (displayed at the upper right corner of the screen) is in the range of 10%~95% (preferably around 85%). This is to ensure sufficient measurement accuracy and at the same time avoid distortion caused by input saturation. **When the input peak level reaches 100% (the bar turns fully red), the results cannot be trusted. However, please note that under the 32-bit float mode, this is not true when the digital gain is greater than 0dB.**

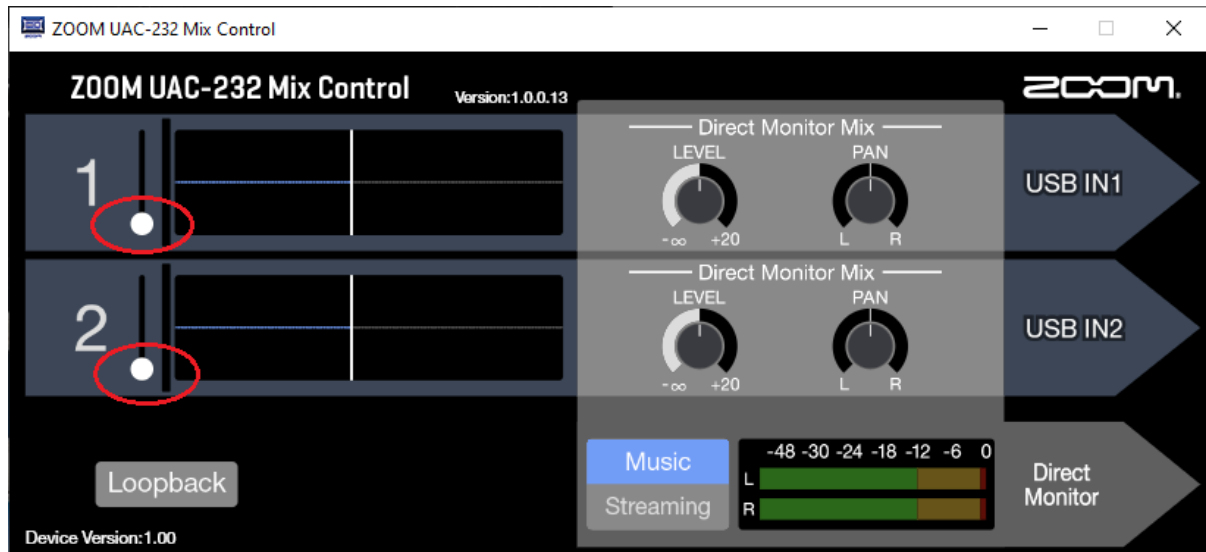


- You can change the number of frames for averaging by right clicking anywhere within the Spectrum Analyzer window and selecting [Spectrum Analyzer Processing]> “Inter-Frame Processing”> “Linear Average” and choose a number in the “Frames” field.
- You can change the frequency weighting by right clicking anywhere within the Spectrum Analyzer window and selecting [Spectrum Analyzer Processing]> “Intra-Frame Processing” and choose a weighting profile in the “weighting” field.
- You can save a panel setting file via [Setting]>[Save Current Panel Setting]
- You can load a panel setting file via [Setting]>[Load Panel Setting].
- You can configure your own Hot Panel Setting Toolbar via [Setting]>[Configure Hot Panel Setting Toolbar].
- You can lock the panel setting via [Help]>[Lock Panel Setting].
- You can unlock the panel setting via [Help]>[Unlock Panel Setting].
- You can set or change the password for unlocking the panel setting via [Setting]>[Change Password].

## 1.11 UAC-232 Control Panel

UAC-232 Control Panel is installed during driver installation. Normally, there is no need to open it as the default settings will work perfectly. Please refer to Section 1.4.3.1.1 for details. Note that all the settings on the control panel are stored in the driver on the computer rather than the audio interface hardware.

## 1.12 UAC-232 Mix Control



UAC-232 Mix Control is an application designed to control the internal mixer settings of UAC-232. It can be installed manually if needed. Its installation file RTA268ABC\_Mix\_Control.msi is located in the Drivers\VTRTA268ABC directory in the CD. When you install the Multi-Instrument software, a copy of it will also be installed in the software installation directory\Drivers\VTRTA268ABC. Run it to manually install the utility program if necessary.

Normally, there is no need to use it as the default settings from Virtins Technology will work perfectly. As mentioned previously, the digital gain settings for input channels 1&2 should be kept at 0dB in order not to invalidate the entered calibration data. Under the 32-bit float mode, the recording quality will not be degraded by the digital gain settings. Please note that the digital gain settings for XLR condenser microphones (with 48V phantom power on), XLR dynamic microphones (with 48V phantom power off), TRS line-level input (with HiZ off), and TS Guitar/Bass input (with HiZ on) are kept separately in the hardware. The audio interface can recognize the input device connected automatically and set the corresponding digital gain. **For the measurement microphone used here, 48V phantom power must be switched on and HiZ must be kept off, this will not only ensure the microphone to function correctly, but also set its correct digital gain settings (i.e. 0dB)**

The Loopback button is switched off by default. If it is turned on, the signals input through the input jacks 1 & 2 and the stereo signals output from the computer will be mixed and then input to the computer again (looped back). This function has its very particular use in time delay measurement, which will be introduced later.

Music mode will keep the two input channels separate while Streaming mode will merge them into a mono channel. Music mode is selected by default. It should be kept in that mode generally for test and measurement applications.

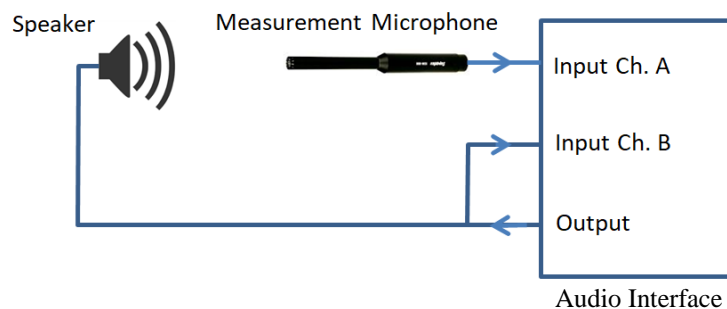
Note that all the settings on UAC-232 Mix Control are stored in the audio interface hardware rather than the software on the computer.

## 1.13 Time Delay Measurement

Multi-Instrument supports a few methods for time delay measurement. Only a subset of them is introduced here.

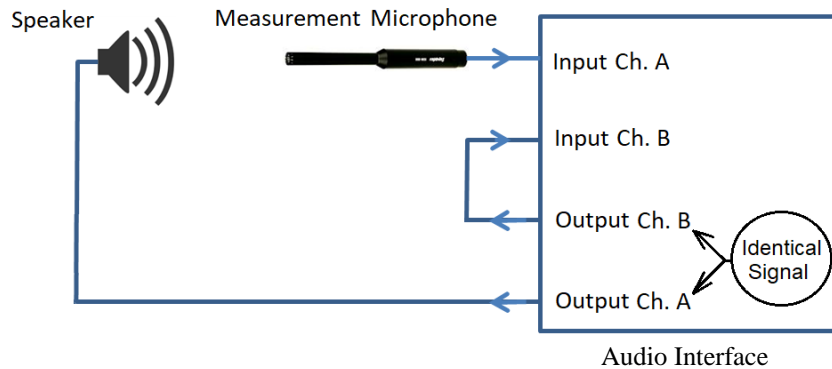
### 1.13.1 Two-channel Time Delay Measurement using Cross Correlation

Cross correlation can be used to measure the time delay between the signals in two channels which are originated from the same source. Very often, the signal in one channel is obtained by connecting directly to the output channel that generates the signal. This is called a loopback. It is used as a time reference.

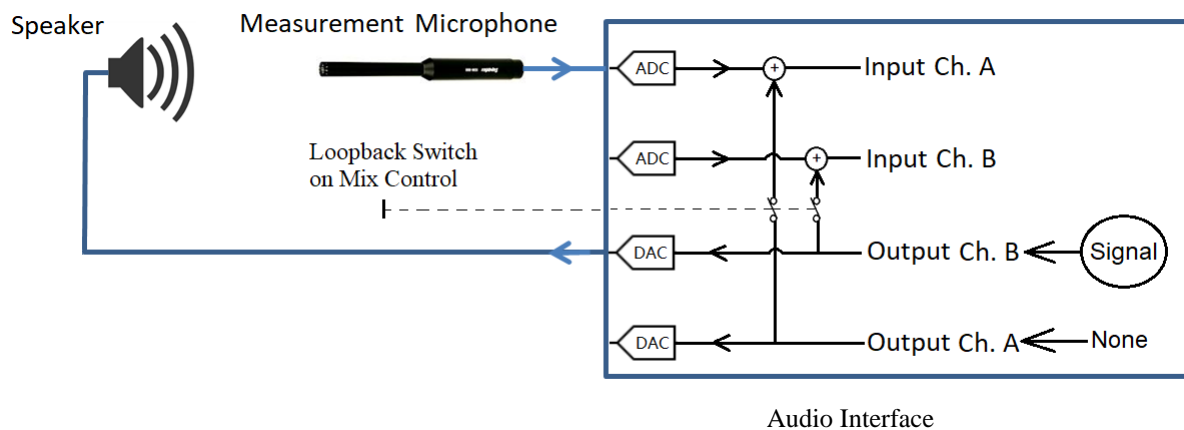


The figure above shows the connection diagram for two-channel time delay measurement with a loopback. The signal output by the signal generator will go through two different paths after the external signal splitter: one is looped back directly to Input Ch. B, the other goes through the speaker, air and microphone before entering Input Ch. A. As the sound travels much slower than the electric signal, the signal received in Input Ch. A will be delayed considerably compared with that in Input Ch. B. This time delay can be measured using the (Linear) Cross Correlation function of the Spectrum Analyzer in Multi-Instrument. It is reflected by the location of the cross correlation peak on the time axis. A positive time delay value means that the signal in Input Ch. A is delayed and vice versa. The correlation peak can be in either positive direction or negative direction. A positive peak shows that the two signals are positively correlated and vice versa. The stimulus used must be a wide-band signal with a “white” (i.e. flat) spectrum, such as white noise, MLS or unit impulse. This is to ensure only a single prominent correlation peak located right at the point of the time delay on the time axis. In the situations where the signal is not generated intentionally by the testing equipment (e.g. live speech or music) or when there are strong background noises and reverberation, (Linear) Generalized Cross Correlation should be used instead. Generalized Cross Correlation effectively whitens the signal before performing cross correlation and as a result, the signal-to-noise ratio of the measurement is greatly improved. The Generalized Cross Correlation flag can be set via [Spectrum Analyzer Processing]>“Intra-Frame Processing”>“Generalized Cross Correlation”. More information on the Cross Correlation function can be found in the software manual.

If there are two simultaneous output channels carrying the same test signal, then there is no need to split the output signal in the above connection diagram. Simply wire Output Ch.A to the speaker, and loopback Output Ch.B to Input Ch.B (see figure below). An external cable is still needed for the loopback connection. The signal generated in Output Ch.A and Output Ch.B must be the same for this connection method.



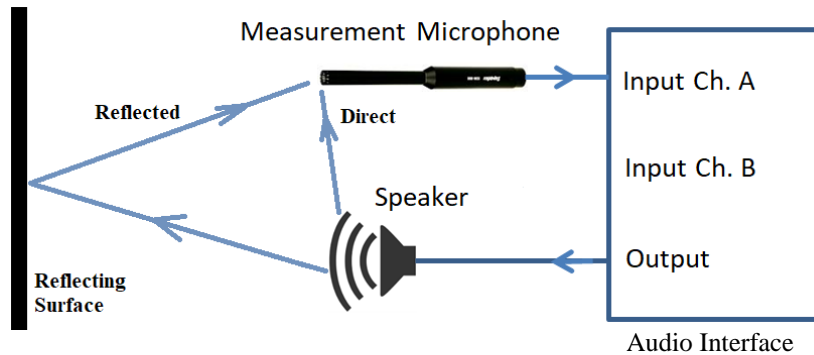
The easiest way to do the loopback is actually to use the loopback switch on UAC-232 Mix Control, as shown in the following diagram. No external splitter and loopback cable are required. The only issue is that it provides a loopback for both channels. With Multi-Instrument, this issue can be solved through stereo output by selecting “A&B” as the output channels, setting the signal in Output Ch.A to “None”, and generating the test signal only through Output Ch. B. This effectively breaks the loopback from Output Ch.A to Input Ch.A, leaving only Output Ch.B and Input Ch.B connected internally. In this case, the speaker should be connected to Output Ch.B, via either the balanced line-level TRS output jack or the headphone jack. The amplitude of the loopback signal can be controlled only through the Signal Generator in Multi-Instrument, as well as the Windows Volume Control if MME is used for DAC. The output volume knobs on the audio interface panel and the input digital gain settings will not affect it. It is possible to generate the test signal (e.g. music) using an external program, such as Windows Media Player or YouTube, instead of the Signal Generator of Multi-Instrument. In this case, be sure to put the volume of Output Ch.A to 0% using the stereo balance control under Windows Sound Playback Control Panel.



In the diagram above, the signal from the input jack always lags the loopback signal by at least a fixed amount of time due to the additional DAC and ADC processes. This system introduced time lag varies with the sampling rate, as shown below. It should be subtracted from the measured time delay. This can be done using User Defined Data Point (UDDP) in Multi-Instrument, in order to display directly the time delay or time delay derived variables, such as distance or direction of sound source.

Sampling Rate	44.1kHz	48kHz	88.2kHz	96kHz	176.4kHz	192kHz
Lag (μs)	680	625	420	385	272	250

### 1.13.2 Single-channel Time Delay Measurement using Auto Correlation



The above figure shows the connection diagram for single channel time delay measurement. It measures the time delay in two signals superimposed in one channel. No loopback is needed. It can be used when two channel time delay measurement is not possible or convenient. The time delay is measured by identifying the location of the second auto correlation peak on the time axis. The test signals that can be used in this method are similar to those used in the cross correlation methods described previously. The drawback of this method is that the signal-to-noise ratio is lower. Please refer to the section for Auto Correlation function in the software manual for details.

### 1.13.3 Single-Channel Time Delay Measurement using Pulse Emission

This is a conventional and widely used approach. Unlike the aforementioned correlation methods that typically involve continuous stimulation of the system under test, the pulse emission method requires brief pulse emissions to ensure easy separation of the echo from the emission along the time axis. The former is very robust and immune to noises due to the large amount of energy pumped into the system. However, the correlation methods require the waveforms of the two signals, between which the time delay is to be measured, are more or less similar. If this condition cannot be satisfied, then the pulse emission method can be used instead.

Multi-Instrument boasts an intelligent algorithm that automatically measures the time delay between emission and echo or multiple echoes, even in very harsh environment such as deep oil wells. It requires a license of Multi-Instrument Full Package. Please refer to the Echo section of the software manual for details.

## 1.14 Audio and Ultrasound Analysis

The highest sampling rate supported by the audio interface is 192 kHz and its bandwidth under this sampling rate is from 9Hz to 92kHz. Thus it has the potential to measure sound beyond the audio frequency range. The microphones that come with RTA-268A/B/C have frequency response specifications: 20Hz~31.5kHz, 20Hz~20kHz, and 3Hz~50kHz, respectively. Outside the specified ranges, the frequency response may not be that flat.

Again, ASIO driver is recommended and the sampling rate 192 kHz should be used for infrasound and ultrasound measurements.

## 2 Specifications

### 2.1 VT RTA-268A/B/C Hardware Configuration

	RTA-268A	RTA-268B	RTA-268C
Measurement Microphone	M215 	M215L 	M50 
XLR-to-USB Audio Interface	UAC-232 		

### 2.2 Overall VT RTA-268A/B/C Specifications

	RTA-268A	RTA-268B	RTA-268C
Frequency Range	20Hz~31.5kHz	20Hz~20kHz	9Hz~50kHz
Sound Level Measurement Range	20dB~126 dB	22dB~138dB	22dB~128dB
Sampling Rate	44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz, 192 kHz		
Bit Depth	32-bit Float / 24-bit		
Number of Input Channels	2		
Number of Output Channels	2		
Frequency Accuracy	0.001%		
Frequency Weighting	Flat, A, B, C, Z, ITU-R 468		
Time Weighting	Linear, Exponential (Equivalent continuous sound level ( <i>Leq</i> ) fully complies with IEC61672)		
Sound Level Measurement Accuracy	± 0.3 dB at 94dB, 1kHz after calibration		
Sound Level Calibration	Calibrated using Type 1 Sound Level Calibrator		
Octave Analysis	1/1, 1/3, 1/6, 1/12, 1/24, 1/48, 1/96 (Complies with IEC61260)		
Other Functions	THD, THD+N, SINAD, SNR, SMPTE IMD, DIN IMD, CCIF2 IMD, Frequency Response, RT60, etc.  Much more functions are described in the software manual.		
USB Interface	USB 2.0		
USB Audio	USB Audio Class 2.0 (driver installation required)		
Power Supply	USB bus powered (5V)		
System Requirements	Windows 10/11, 64 bit. Minimum Screen Resolution: 1024 × 600		
Calibration	Individually calibrated.		

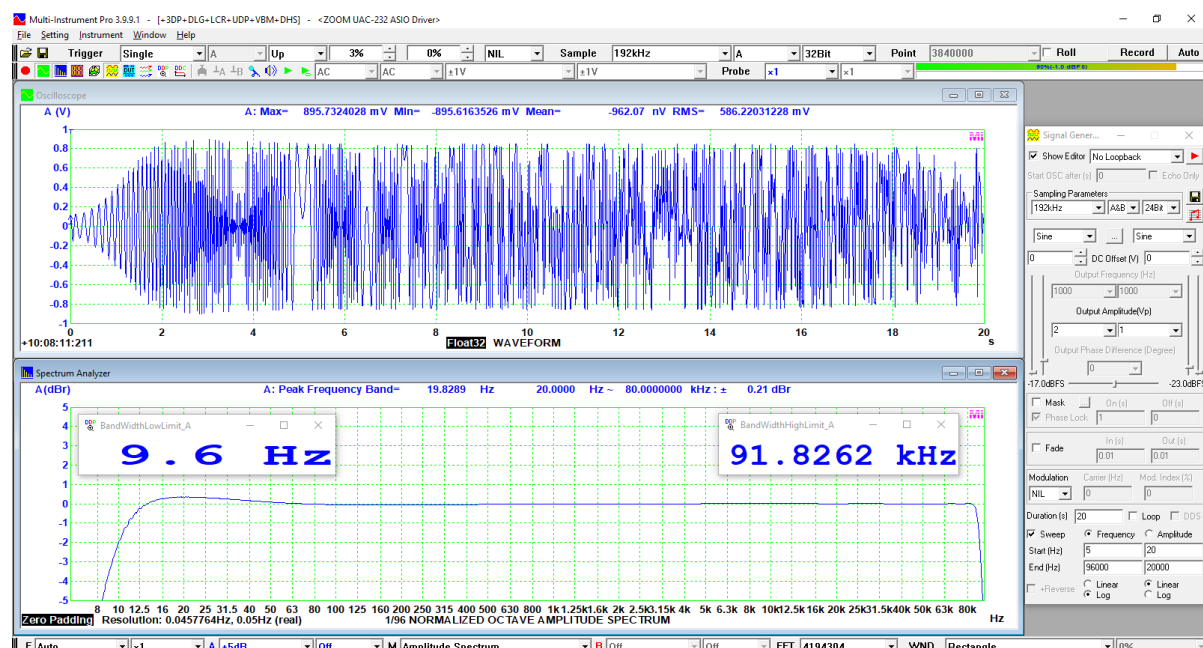
### 2.3 UAC-232 Audio Interface Specifications

	Input Impedance	Max. Voltage	Dynamic Range	XLR Phantom Power
Left (Input 1): XLR/TRS/TS(HiZ) Combo Jack	XLR: 2.7 kΩ	XLR: 2.35Vp	XLR: 134 dB	48V
Right (Input 2): XLR/TRS Combo Jack	TRS: 4 kΩ / 1 MΩ (when HiZ on)	TRS: 23.0Vp / 4.30Vp (when HiZ on)	TRS: 129 dB TS(HiZ): 123 dB	

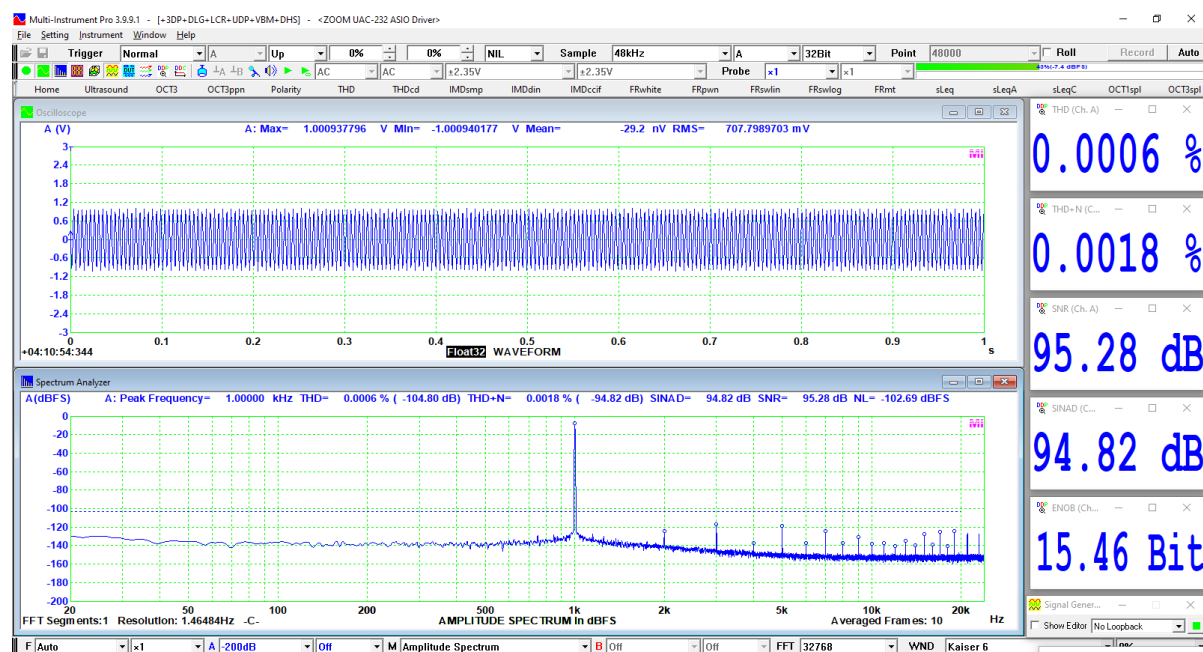


	Output Impedance	Max. Voltage
Left (Output 1): TRS (balanced) Jack	100Ω	9.0Vp (adjustable through volume knob)
Right (Output 2): TRS (balanced) Jack		
Headphone	12Ω	4.4Vp (adjustable through volume knob)

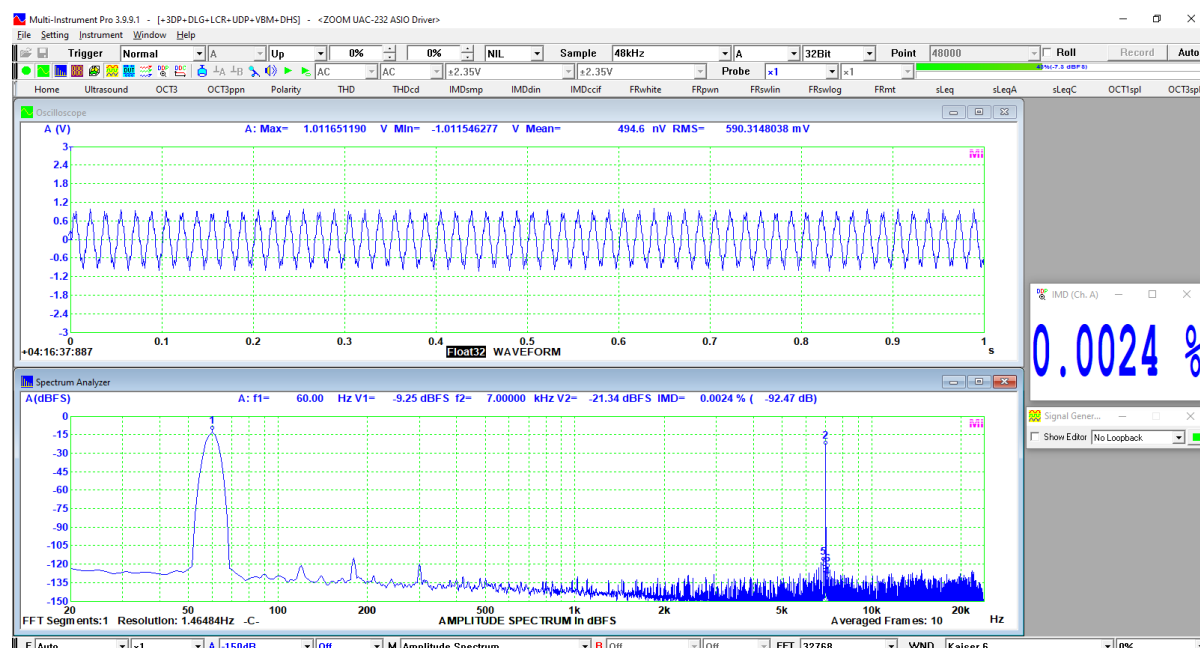
MIDI	IN/OUT (5-pin DIN connectors)
Input Frequency Response	<p>44.1 kHz: 20Hz~20kHz, +0.1 dB / -0.2 dB -3dB: 9Hz~21.8kHz</p> <p>48 kHz: 20Hz~20kHz, ±0.21dB -3dB: 9Hz~23.8kHz</p> <p>96 kHz: 20Hz~40kHz, ±0.22dB -3dB: 9Hz~47.6kHz</p> <p>192 kHz: 20Hz~80kHz, +0.1 dB / -0.2 dB -3dB: 9Hz~91.8kHz</p>
TRS Output Frequency Response	<p>44.1 kHz: 20Hz~20kHz, +0.0 dB / -0.1 dB -3dB: 1.9Hz~21.1kHz</p> <p>48 kHz: 20Hz~20kHz, ±0.1dB -3dB: 1.9Hz~22.8kHz</p> <p>96 kHz: 20Hz~40kHz, ±0.1dB -3dB: 2.1Hz~45.6kHz</p> <p>192 kHz: 20Hz~80kHz, +0.0 dB / -1.0 dB -3dB: 2.1Hz~93.2kHz</p>
Headphone Output Frequency Response	<p>44.1 kHz: 20Hz~20kHz, +0.0 dB / -0.5 dB -3dB: 1.5Hz~20.9kHz</p> <p>48 kHz: 20Hz~20kHz, ±0.3dB -3dB: 1.5Hz~22.8kHz</p> <p>96 kHz: 20Hz~40kHz, ±0.1dB -3dB: 1.6Hz~45.6kHz</p> <p>192 kHz: 20Hz~80kHz, +0.0 dB / -1.0 dB -3dB: 1.7Hz~95.2kHz</p>
USB Connector	Type C
Power Consumption	Max. 2.5W
Dimension	164 mm (W) × 105.5 mm (D) × 65 mm (H)
Weight	355g



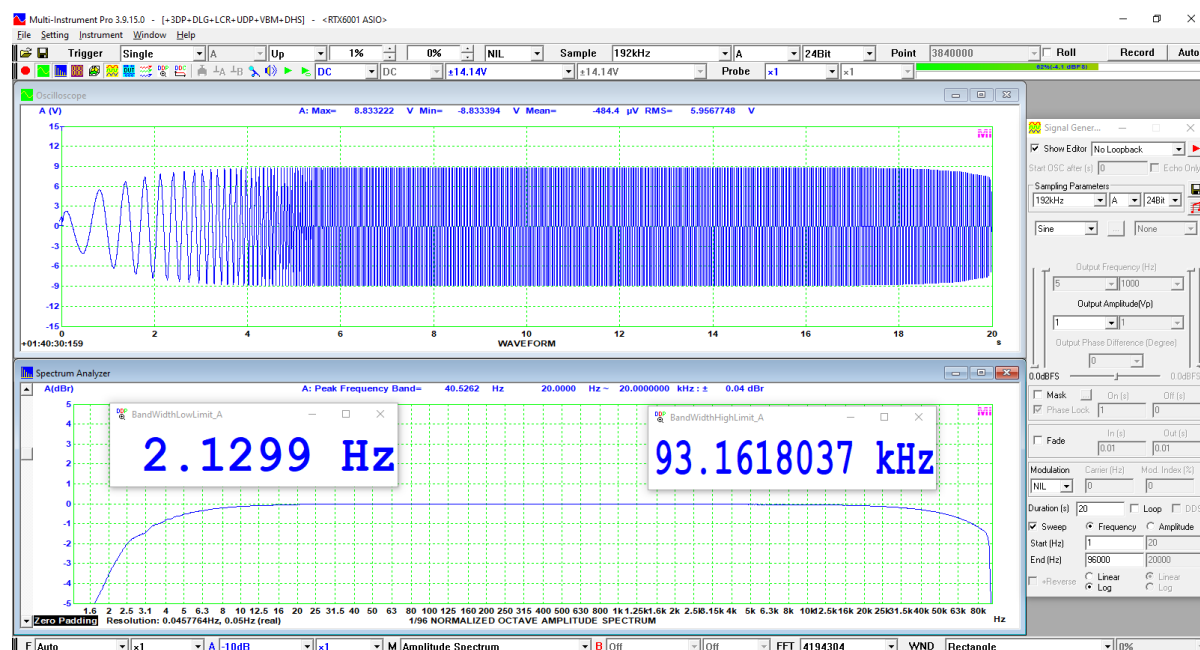
**Typical Magnitude Frequency Response of XLR Input @ Sampling Rate 192kHz**



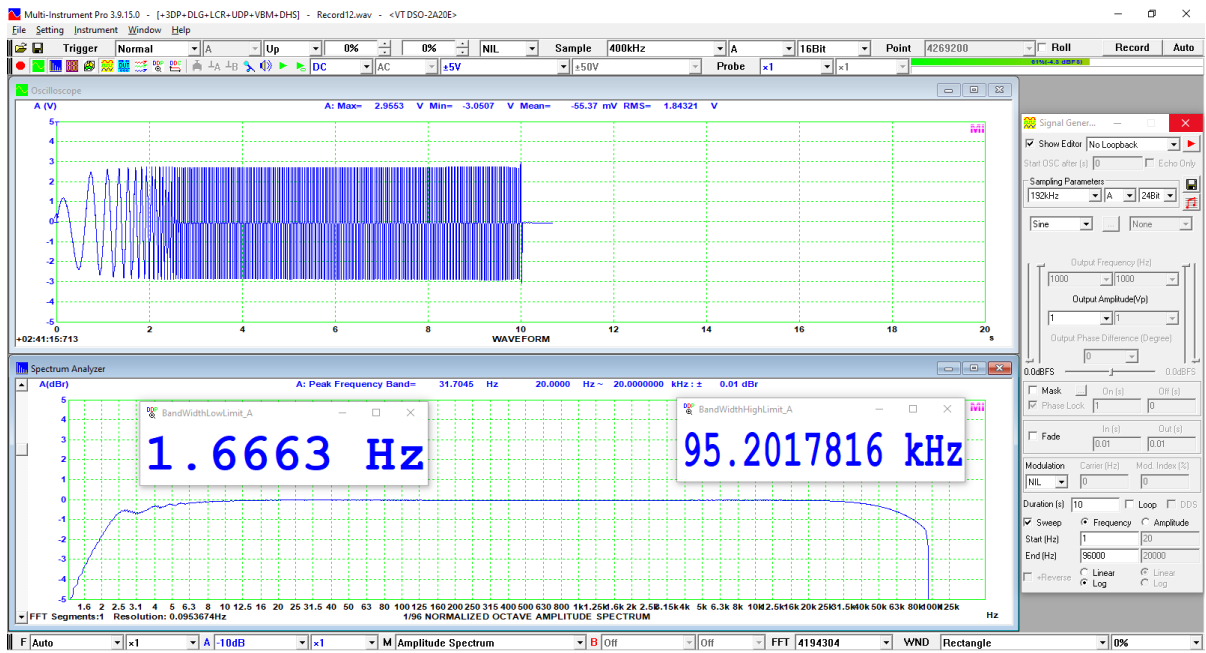
**THD, THD+N, SNR, SNAD, ENOB of XLR Input @ 1kHz, 1Vp**



**SMPTE IMD of XLR Input @ 1Vp**



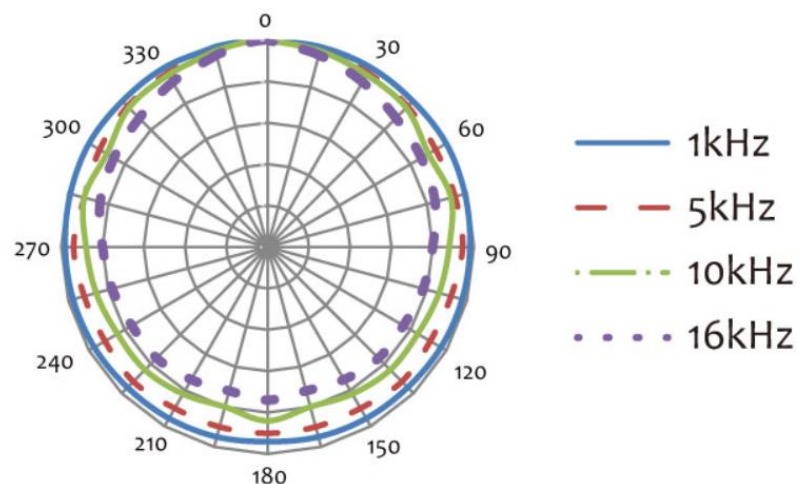
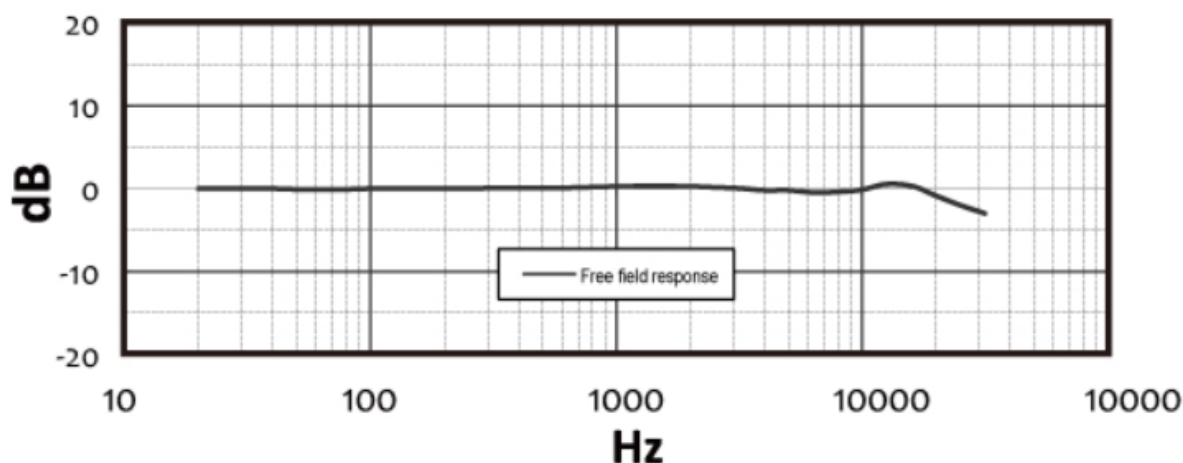
**Typical Magnitude Frequency Response of Balanced Line Output @ Sampling Rate 192kHz**



**Typical Magnitude Frequency Response of Stereo Headphone Output @ Sampling Rate 192kHz**

**2.4 Measurement Microphone M215 Specifications (for VT RTA-268A)**

Transducer Type	½” low-noise pre-polarized condenser
Polar Pattern	Omni-directional
Sensitivity (±2dB)	40 mV/Pa
Frequency Response	20Hz ~ 31.5kHz
Power Supply	48V Phantom Power
Max. SPL (<1% THD)	135dB
Equivalent Noise Level	<18 dBA
Output Impedance	<75 Ω
Temperature Coefficient (dB/°C)	0.009
Humidity Coefficient (dB/%RH)	0.001
Pressure Coefficient (dB/kpa)	0.008
Length	155mm
Weight	120g
Connector	3-pin XLR
Compliance	IEC 61094 and IEC 61672 Class 1

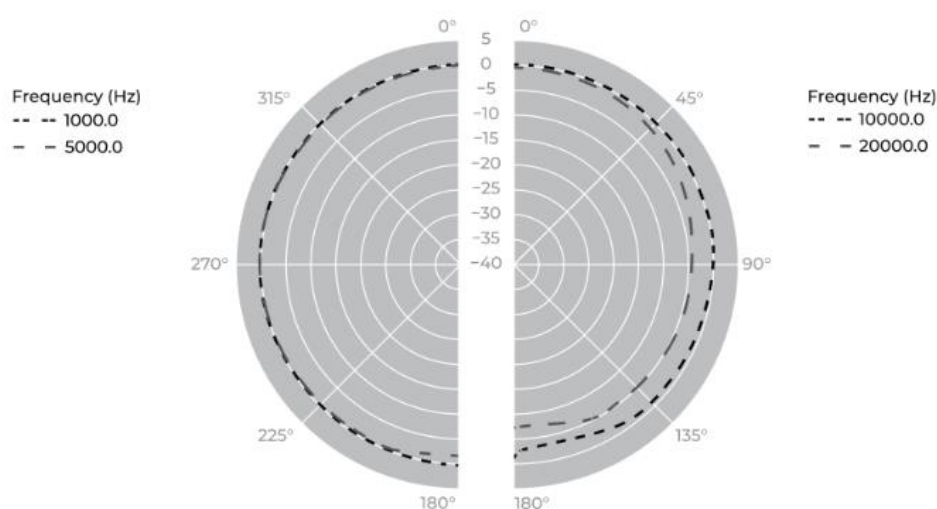
**Typical Polar Pattern****Typical Frequency Response**

## 2.5 Measurement Microphone M215L Specifications (for VT RTA-268B)

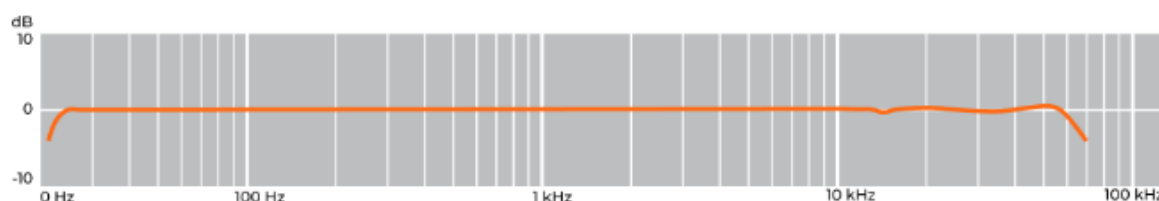
Transducer Type	½" low-noise pre-polarized condenser
Polar Pattern	Omni-directional
Sensitivity (±2dB)	10 mV/Pa
Frequency Response	20Hz ~ 20kHz
Power Supply	48V Phantom Power
Max. SPL (<1% THD)	145dB
Equivalent Noise Level	<20 dBA
Output Impedance	<75 Ω
Temperature Coefficient (dB/°C)	0.009
Humidity Coefficient (dB/%RH)	0.001
Pressure Coefficient (dB/kpa)	0.008
Length	155mm
Weight	120g
Connector	3-pin XLR
Compliance	IEC 61094 and IEC 61672 Class 1

## 2.6 Measurement Microphone M50 Specifications (for VT RTA-268C)

Transducer Type	Small Diaphragm low-noise pre-polarized condenser
Polar Pattern	Omni-directional
Sensitivity (Typical)	34 mV/Pa
Frequency Response	3Hz ~ 50kHz
Power Supply	24~48V Phantom Power, 10mA
Max. SPL	140dB
Equivalent Noise Level	<20 dBA
Output Impedance	65Ω
Minimum Output Load	600Ω
Dimension	229×22mm
Weight	225g
Connector	3-pin XLR



**Typical Polar Pattern**



**Typical Frequency Response**

*Note: This is only a typical frequency response chart. The measurement microphone M50 is calibrated individually and the calibration data is included in the product package.*

## 2.7 Multi-Instrument Software Specifications

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.



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
<b>General Functions</b>							
ADC / DAC 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	✓	✓	✓	✓	✓	✓

		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					√	√

		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	√	√	√ (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		√		√	√	√

		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		√		√	√	√
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						√
	Total Non-Coherent Distortion + Noise						√
	GedLee Metric						√
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			√	√	√	√
	Pink Noise			√	√	√	√
	MultiTones			√	√	√	√
	Arbitrary Waveform			√	√	√	√
	MLS			√	√	√	√
	DTMF			√	√	√	√
	Musical Scale			√	√	√	√
	Wave File				√	√	√

		Sound Card Oscilloscope	Sound Card Spectrum Analyzer	Sound Card Signal Generator	Multi-Instrument Lite	Multi-Instrument Standard	Multi-Instrument Pro
	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				√	√	√
<b>Multimeter</b>							
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					√	√
	Jitter Statistics						√
Settings	Counter Trigger Hysteresis				√	√	√
	Counter Trigger Level				√	√	√
	Frequency Divider				√	√	√
<b>DDP (Derived Data Point) Viewer</b>							

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

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
<b>Spectrum 3D Plot</b>							
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						
<b>Data Logger</b>							
Real Time Logging							
Load Historical Log File							
Three logging methods (Fastest, Time Interval, Update Threshold)							
262 derived data points available for logging							
Up to $8 \times 8 = 64$ variables can be logged simultaneously							
<b>LCR Meter</b>							
High Impedance Measurement							
Low Impedance Measurement							
Up to 8 X-Y Plots (Linear/Log)							
<b>Device Test Plan</b>							
25 Instructions							
Create/Edit/Lock/Execute/L oad/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							

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



## **3 Multi-Instrument Software License Information**

### **3.1 License Types**

The License of Multi-Instrument software has six levels and six add-on modules/functions. The six levels are: Sound Card Oscilloscope, Sound Card Spectrum Analyzer, Sound Card Signal Generator, Multi-Instrument Lite, Multi-Instrument Standard, Multi-Instrument Pro. The six add-on modules/functions are: Spectrum 3D Plot, Data Logger, LCR Meter, Device Test Plan, Vibrometer, and Dedicated Hardware Support.

The license contained in the standard VT RTA-268A/B/C package is a hardkey activated Multi-Instrument Pro license, without any add-on modules/functions. No softkey (activation code) is provided. The software will run under the licensed mode as long as the USB hardkey (dongle) is connected to your computer before you launch the Multi-Instrument software.

Note: If the software is launched without the USB hardkey connected to the computer, it will enter into 21-day fully functional trial mode, unless the software is activated by a softkey (activation code), which is NOT included in the standard VT RTA-268A/B/C package and should be purchased separately as a brand-new license if needed.

### **3.2 License Upgrade from One Level to Another**

You can purchase an upgrade of the license, e.g. from Multi-instrument Pro to Multi-Instrument Pro + Data Logger, at any time if necessary. After you purchase the upgrade, a small upgrade package file will be sent to you via email. You can then use it to upgrade the license information inside the USB hardkey by selecting [Start]>[All Programs]>[Multi-Instrument]>[VIRTINS Hardware Upgrading Tool] on your Windows desktop.

### **3.3 Software Upgrade in the Same License Level**

Software upgrade in the same license level (if the hardkey is still supported by the new version), e.g. from Multi-Instrument 3.0 Pro to Multi-Instrument 3.1 Pro, is always FREE. You just need to download the new version from our website and install it on any computer.

Thus, please do check frequently with our website to see if a new version or build is available.

## **4 Warranty**

Virtins Technology guarantees this product against defective materials and manufacturing defects for a period of 12 months. During this period of warranty, a replacement of the faulty part will be shipped to the buyer's address free of charge upon receiving and verifying the

returned faulty part. The Warranty is only applicable to the original buyer and shall not be transferable. The warranty shall exclude malfunctions or damages resulting from acts of God, fire, civil unrest and/or accidents, and defects from using wrong electrical supply/voltage and/or consequential damage by negligence and/or abuse, as well as use other than in accordance with the instructions for operation. The Warranty shall immediately cease and become void if the hardware is found to have been tampered, modified, repaired by any unauthorised person(s). Decisions by Virtins Technology on all questions relating to complaints as to defects either of workmanship or materials shall be deemed conclusive and the buyer shall agree to abide by such decisions.

## **5 Disclaimer**

This document has been carefully prepared and checked. No responsibility can be assumed for inaccuracies. Virtins Technology reserves the right to make changes without prior notice to any products herein to improve functionality, reliability or other design aspects. Virtins Technology does not assume any liability for losses arising out of the use of any product described herein; neither does its use convey any license under its patent rights or the rights of others. Virtins Technology does not guarantee the compatibility or fitness for purpose of any product listed herein. Virtins Technology's products herein are not authorized for use as components in life support services or systems. Virtins Technology should be informed of any such intended use to determine suitability of the products.