



CIM Audio Test Software

User Guide

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1 Version history

Description	Version	Date	Initials
Release for CATS 6.4.x	1.0.0	2017-09-18	TLA
Release for CATS 6.4.1	1.0.1	2019-05-06	TLA
Release for CATS 7.0.0	2.0.0	2019-12-06	TLA
Release for CATS 7.1.0	2.1.0	2020-09-02	TLA
Release for CATS 8.0.0	2.2.0	2023-09-26	TLA
Release for CATS 9.0.0	2.3.0	2026-02-25	TLA

2 Introduction

CIM Audio Test (CATS) is a suite of custom step type for NI TestStand the enables the user to perform a wide range of electro-acoustic tests and measurements.

This document describes the use of CATS, which uses a plug-in architecture to provide a flexible and expandable set of measurements. It also results in more user-friendly configuration panels by exposing tailored user interfaces for specific measurement.

A plug-in includes a generator and an analyser module that are merged into one display panel.

The step types are described in the following chapters:

- Audio Test
- LimiTEST
- LiveView
- Calibration Tool
- Calibration manager

2.1 Abbreviations

Abbreviation	Description
CATS	CIM Audio Test Suite
NI	National Instruments
TS	National Instruments TestStand
NI-MAX	National Instruments Measurement and Automation Explorer
FFT	Fast Fourier Transform
DUT	Device Under Test
AGC	Automatic Gain Control
Platform	The hardware mode in which to run CATS; CATS has platforms for each type of supported hardware, as well as a simulation platform for running without hardware
Plug-in	CATS application that performs measurements and analyzes data collected from those measurements

2.2 References

Ref.	Name	Location
[1]		

2.3 Definitions

Definition	Description
dBspl	Sound pressure level in dB re. 20 μ Pa
Pa	Pascal, derived SI-unit for pressure = N/m ²

3 Getting started

This section describes the steps to follow to install and configure CIM Audio test (CATS).

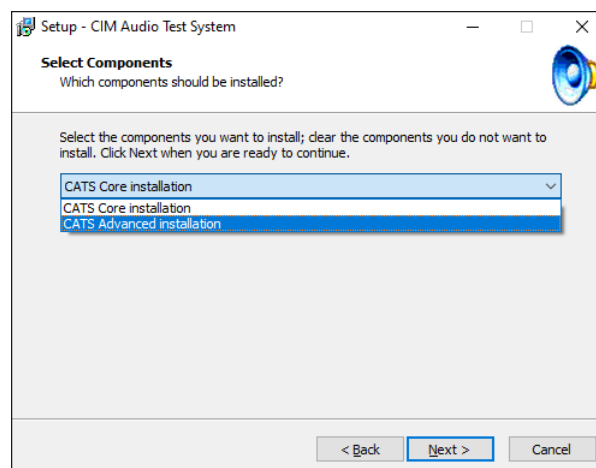
3.1 Installation

Before installing CATS, you should complete the installation of the NI software and hardware that you will be using.

- NI TestStand 2023 Q3 or later.
- NI LabVIEW Run-time Engine 2023 Q3 (if not installed by TestStand).
- NI-DAQmx 2023 Q3 or later (optional if not using NI hardware platforms).
- Install the NI hardware or create simulated device in NI-MAX.

If you have more than one version of TestStand installed, you must use the TestStand version selector to select which version to use with CIM Audio Test.

Now run the CIM Audio Test installer and follow the instructions. Make sure to select the correct component option that match your license:



3.1.1 Activating your license

When you start to use the CATS configuration tool and step types, you will be prompted to activate the licenses. Depending on the sequence of actions you will be prompted for Core license or the Advanced license.

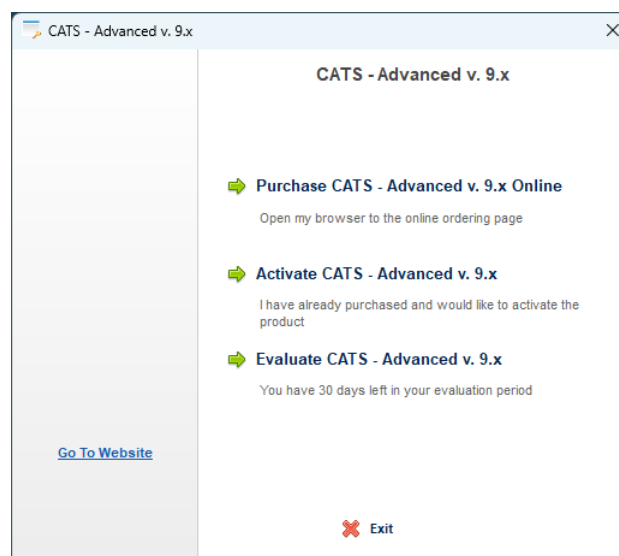


Figure 1: License activation dialogs for Core and Advanced features.

If you select the 'Evaluate' option at the bottom, you can use all features of the software for a trial period of 30 days. You will however be prompted again whenever a plug-in or module is loaded.

Select the 'Purchase' option at the top to go to the CIM web shop and buy the licenses you need.

If you have already received the licence ID and activation code, select the 'Activate' option in the middle. Then select the activation method:

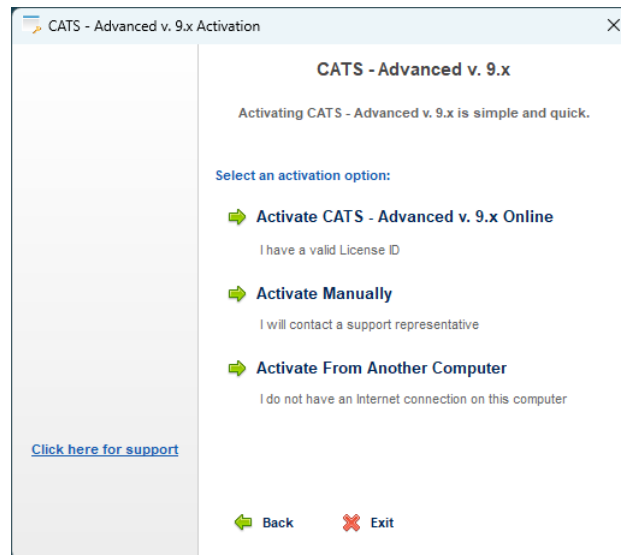


Figure 2: License activation options.

Typically, if the computer is connected to the internet, you will select to activate online.

Alternative you can use another computer to complete the online or use the manual method to get two activation codes via email.

3.2 Hardware configuration

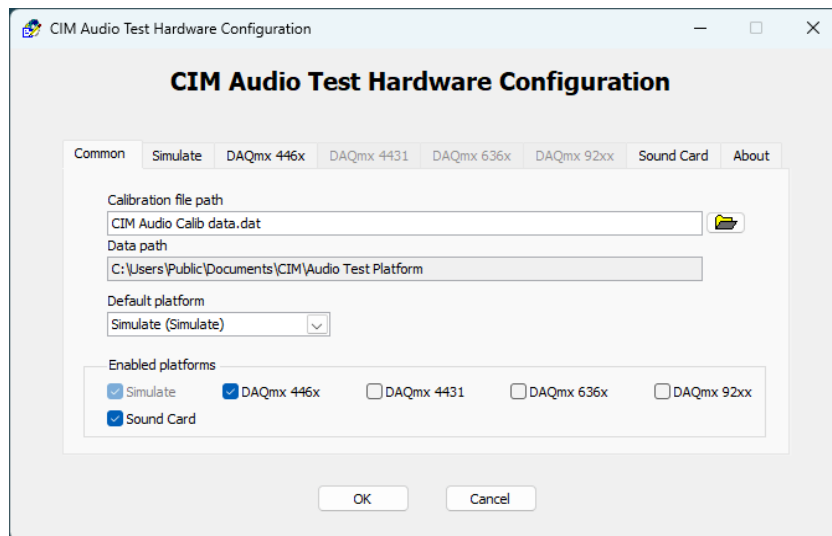
After completing the installation, you must configure which devices to use with CIM Audio Test.

From the Windows Start menu select:

All Programs | CIM | CIM Audio Test | CIM Audio Test Configuration Tool

Select each tab to configure the available platforms.

3.2.1 Common page

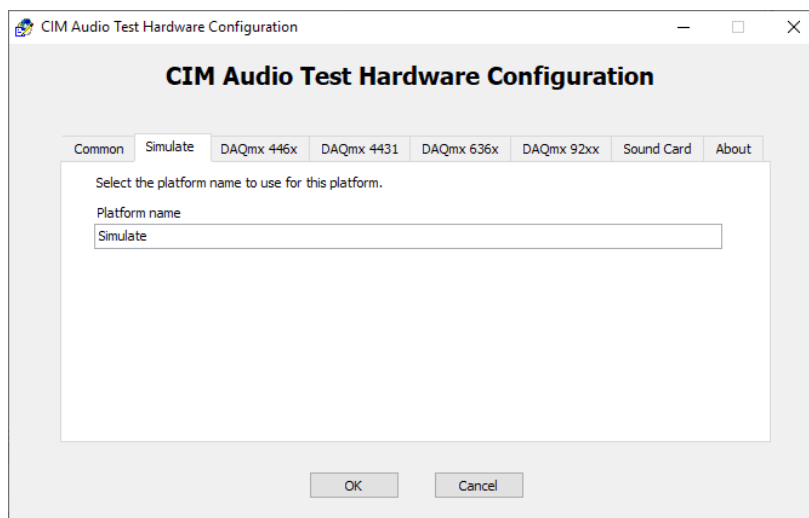


The common page allows you to specify some general settings:

- Select a calibration file for use in the calibration tool. If calibration is not needed just leave unchanged.
- Select which of the installed hardware platforms to use by default in the test steps.
- Select which of the installed hardware platform to enable. Only enabled platforms will be available in the test steps. When a platform is disabled, its configuration page is greyed out.

3.2.2 Simulate page

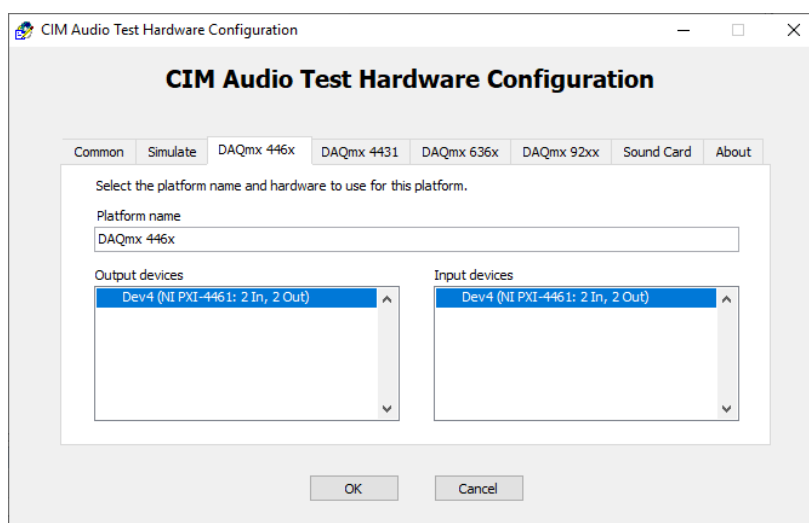
The Simulate platform allows you to perform simulated measurement for demonstration or educational purpose.



Only the display name for the platform can be configured.

3.2.3 DAQmx pages

All the DAQmx have the same layout and options:



Use the controls to specify a name and the devices to use for the platform, as described in the following table.

Control	Description
Platform name	Specifies a name for the platform.
Output devices	Specifies the output device(s) to use for this platform, selected from a list of all applicable PXI and PCI devices in the system. You can select multiple devices by holding <Ctrl> while clicking the devices.
Input devices	Specifies the input device(s) to use for this platform, selected from a list of all applicable PXI and PCI devices in the system. You can select multiple devices by holding <Ctrl> while clicking the devices.

Only devices that are supported by the CIM Audio Test platform are shown in the list boxes. Normally the same device appears in both lists because it provides both input and output.

NOTES:

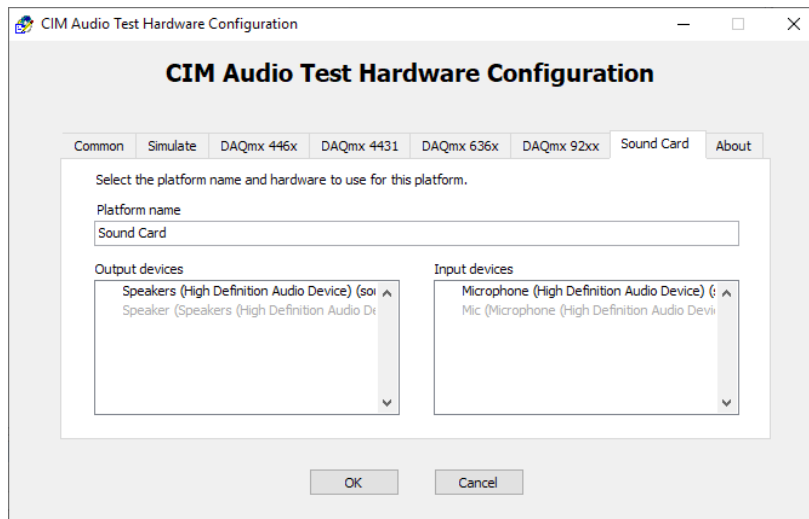
- If you select multiple devices in either the **Output devices** or **Input devices** control, you must select devices of the same platform within each control. Selecting both PXI and PCI devices within the control results in an error when you run CATS.
- When selecting multiple devices (DAQmx 446x platform only) in the **Output devices** or **Input devices** control, the topmost selected device in the list is the default master device. If you would like to specify a different device as the master, you must edit the registry.
- If a device that you previously selected in the Output devices or Input devices control no longer exists in Measurement & Automation Explorer (MAX), the device appears in grey in the list. You must either deselect the device from the list or reconnect the device and configure it in MAX.

3.2.4 Sound Card page

Standard Windows sound cards, including devices like headsets that acts like sound card, can be used for audio output and input.

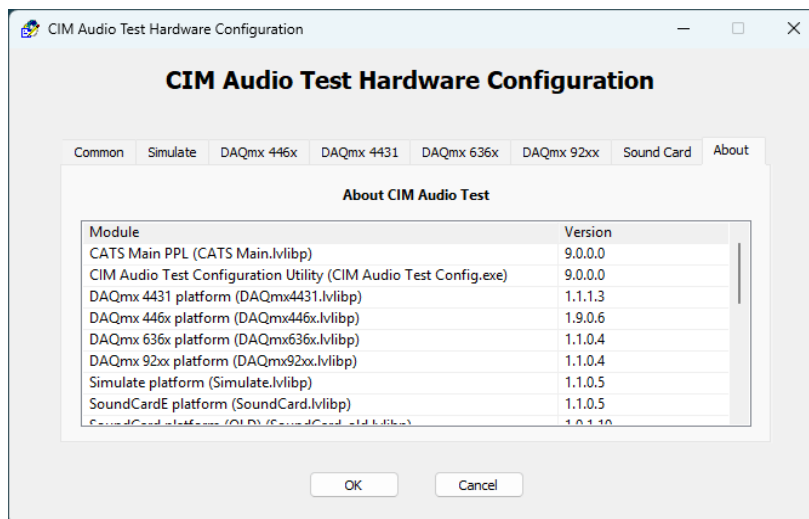
In contrast to DAQmx devices, sound card input and output sections typically appears with different names in each list.

You can select a specific device by clicking it in the list, or type part of the name. In the example, 'Speaker' was typed as output device name, which match the name shown in parenthesis.



3.2.5 About page

This page list the name and version numbers of the installed modules that make up CIM Audio Test.

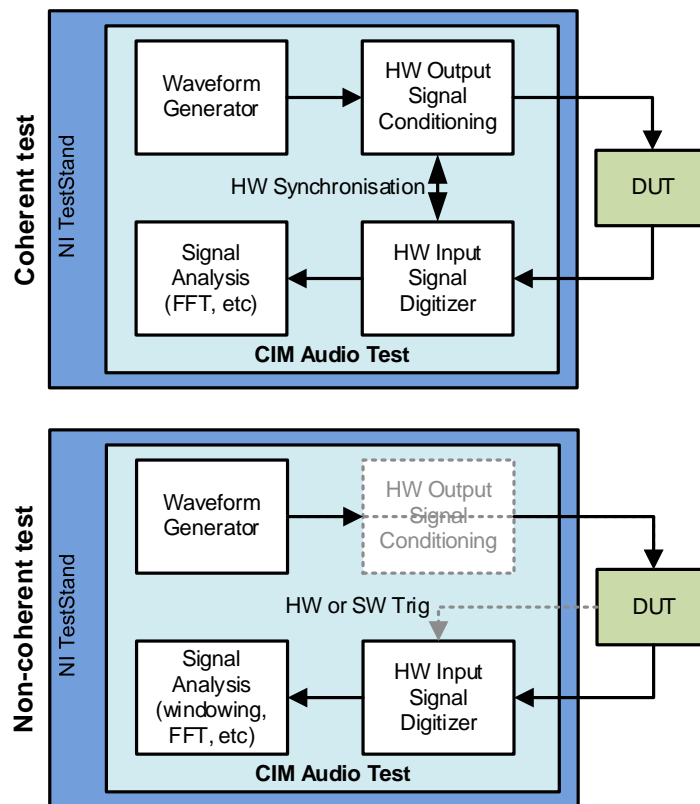


3.3 Operating principles

CATS can operate based on coherent as well as non-coherent sampling principles. The optimum test can be obtained using a coherent sampling. In coherent test, a system under test operates in a repeatable manner, where a best trade-off between speed and accuracy can be obtained. It also means that phase and delay measurements can be assured in a practical manner.

A system, where the signal generation (output to DUT) is synchronized with the input (output from the DUT) is coherent only if all frequency and timing functions are related in exactly integer ratios. It means that endpoints of an acquired sample must wrap smoothly

from the last sample to the first sample. If this is not the case, a discontinuity will be present when wrapping the endpoints and the signal is non-coherent.



In coherent testing, the signal generation is synchronized to the signal acquisition. If the signal from the DUT is independent of the CATS clock, a windowing function must typically be employed.

In a non-coherent case, the energy of the individual tones spreads to the surrounding spectral bins and hence leads to a certain corruption of the signal levels of other tones. It means that the frequency spectrum does not provide a fully correct analysis. In some cases the corruption can lead to quite misleading results.

In many test applications for CATS it is often not practically feasible to apply coherent testing. Reasons can typically be that the waveform generation is not synchronized with that of the signal acquisition. In such situations, signal generation and signal acquisition are not obtaining their test frequencies from a common master clock. Hence, the test system resources are not synchronized to each other and the system itself is consequently not coherent by nature.

If the signal under test is non-coherent, we must employ techniques for ensuring that disadvantages due to the non-coherency nature are suppressed diligently to an acceptable level. Such techniques typically include the use of windowing functions, and CATS supports various windowing functions for enabling professional test – also in the non-coherent test situation.

A coherent test may typically be obtained e.g. when testing hearing aids and loudspeakers, where CATS generates the signals to the DUT and acquires the responses from the DUT. In contrast, a non-coherent test typically applies to test of many consumer products such as CDs, DVDs, headsets, etc. In such test cases, the signal generation is normally independent of CATS. The signal generation can be from a CD or DVD disk or other means of generating the signal. It means that the signal generation is “free-running” with respect to the sample clock of CATS, since it is not feasible to obtain exactly the same frequency from two free-

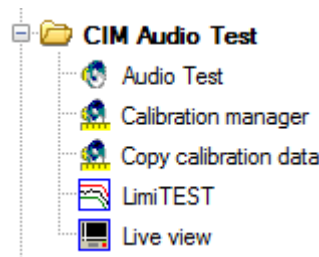
running master clocks. For such test cases, CATS alternatively offers software and hardware triggering, combined with the use of windowing functions.

4 User Interfaces

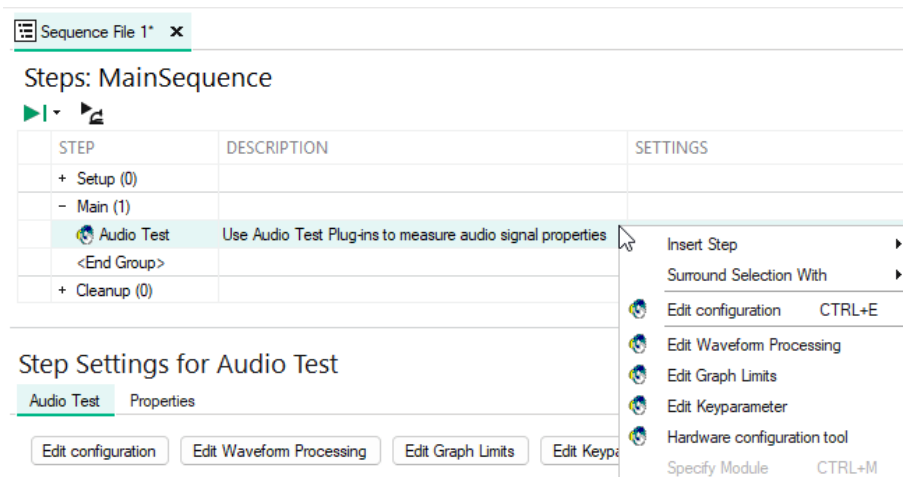
This section gives a brief overview of the step type user interfaces in CATS. The main step types are:

- **Audio Test** configuration panel where you configure the test signal and measurement.
- **LimiTEST** includes three types of post processing and evaluation of measurement results.
- **LiveView** enables display of measurement results in a pop-up window to the test operator.
- **Calibration** provides tools to perform end-to-end electro-acoustical calibration procedure for absolute referenced measurements.

Each of the test step types are available from the TestStand insertion palette along with the built-in step types.



Drag the desired step type into the test sequence and open its configuration panel from the step settings tab or the context menu.



The Audio Test step type also includes the function of the LimiTEST step. This means you can perform signal generation, analysis and advanced evaluation in one step.

4.1 Update step type files

When you use CATS with a different TestStand version than it was released for, TestStand will prompt you about a type conflict with 'NI_LimitMeasurement' in the following type palette files:

- CIM_Audio_Test.ini
- CIM_LimitTest.ini
- CIM_LiveView.ini

In the dialog select 'Use Currently Loaded Type' and click 'OK':

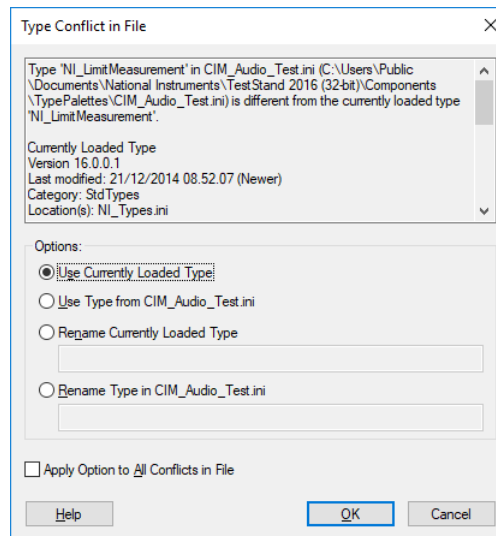


Figure 3: Example of TestStand type conflict dialog.

The reason for this is that CATS rely on build in TestStand types that has changed in newer version of TestStand.

When you close TestStand, you will be prompted to save the changes in the above CIM .ini files. Make sure to save those to avoid the type conflict dialogs next time you use CATS.

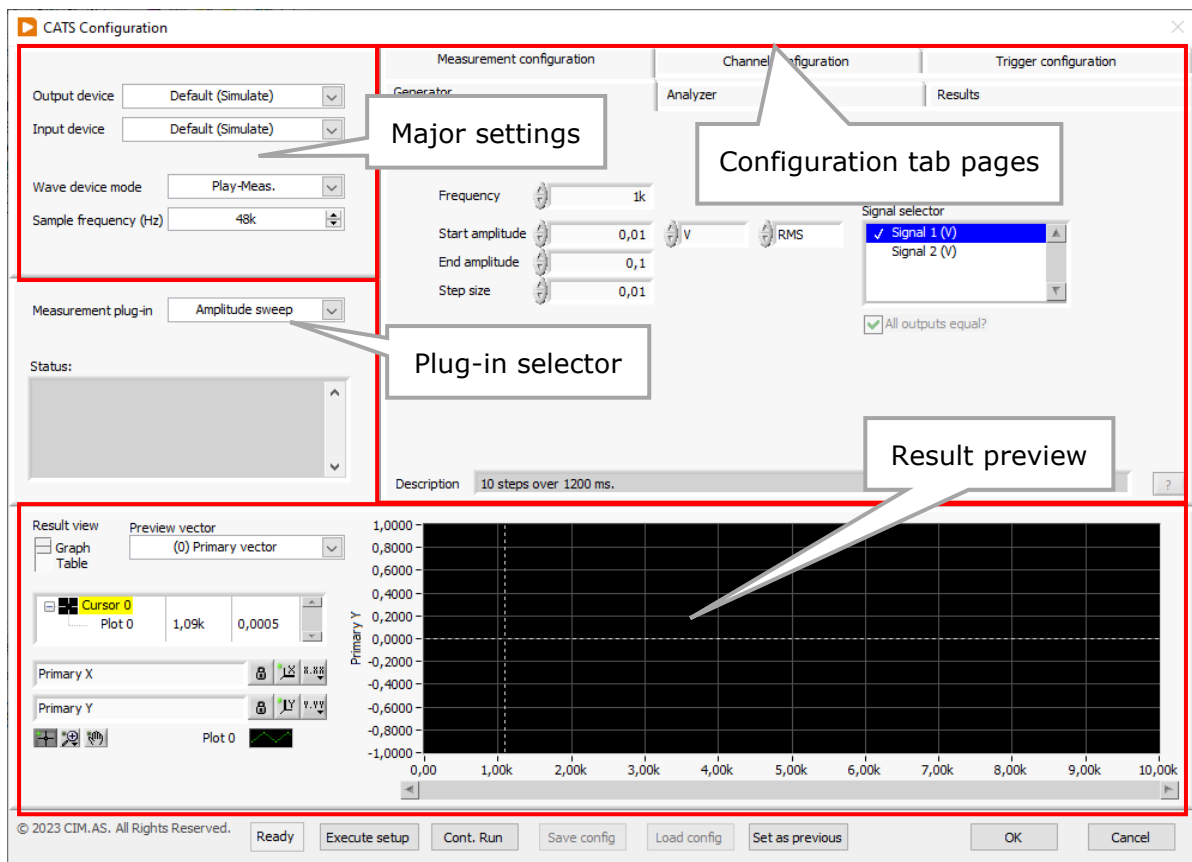
5 Audio Test configuration panel

This configuration panel is used to set up all kinds of test signal and measurement using any of the available hardware platforms.

Signal plug-ins provide all the signal generation and measurement functions. Please refer to section 0 to get a description of the operation for each plug-in.

5.1 Panel overview

The panel is a modular container for a selectable signal plug-in that provides all the signal generation and analysis settings. The remains of the configuration panel provide generic configurations for channel settings, trigger settings and result preview.



5.2 Major settings

The major settings control how the data plug-in operates. These settings are common to all plug-ins.

5.2.1 Select Hardware Platform

The 'Output device' and 'Input device' controls select which hardware platform is currently in use for output and input respectively.

In a typical installation only one platform is installed, and the same device is used for both input and output. In this case CATS will make the input and output operate in a clock and trigger synchronized mode, if possible, to enable coherent testing.

Selecting the 'Default' option will use the default platform that is specified in the hardware configuration tool. This means that if the test step sequence is opened on a system with a different default platform it will use the default of that system.

Opening a test step that use a specific platform on a different system will result in an error message if that platform is not available.

For more advanced installations the platform can be selected independently for input and output device. In this case CATS is not able to synchronize devices automatically.

5.2.2 Wave Device Mode

This control selects how the signal play and measurement operations are handled:

- **Stop**
Is typically used to reset and end a continuous output operation. However, this is rarely used because the wave device mode of a new measurement simply replaces any active operation.
- **Play-Meas.**
Is the default mode for coherent testing where the created waveform signal is sent once to the output and the response is simultaneously acquired at the input.
- **Play only.**
This mode outputs the signal repeatedly even after the test step ends. The input acquisition is not started in this mode. This mode makes it possible to create a continuous tone while using other test functions to perform measurements.
- **Play cont.**
This mode outputs the signal repeatedly even after the test step ends. The response at the input is acquired once simultaneously.
- **Meas.**
This mode is used to measure the input of an external signal, without starting signal output.

When using the Play-Meas mode some plug-ins may automatically append a bit of silence before and after the specified signal, along with a synchronisation signal. The synchronisation signal is used by the plug-in to correct the acquired signal for the transmission delays, through output circuit, loudspeaker, DUT and microphone.

5.2.3 Sample Frequency

This control specifies the sample frequency for the waveform device. Both output and input use the same sample frequency.

The default value is 48 kS/s but the value may be coerced to the valid range as limited by the selected platforms.

5.3 Signal plug-in

This control selects which data plug-in to use in the current test step. The drop-down box will automatically show all compatible data plug-ins that are installed in the plug-ins folder.

The configuration panel of the selected plug-in is shown in the 'Measurement configuration' tab.

Each measurement plug-in consists of three tabs: a generator tab, an analyzer tab and a result tab. These tabs are visible when the Measurement configuration is selected. These tabs configure the measurement timing and analysis parameters that are specific to the plug-in.

NOTE: The duration of the output signal and acquired signal can easily add up to a very long signal. Be careful not to specify long measurement and delay times and at the same time high number of tone steps. Long signals may cause CATS to hang or cause an out of memory error.

5.4 Status line

The status line shows messages about the configuration.

A red background color indicates an error, while a yellow background indicates a warning. On the configuration tabs the associated control is also colored.

When more than one status message is available, an index will appear to allow the user to browse through the messages.

5.5 Measurement Configuration

This tab contains the configuration panel of the selected plug-in. The individual plug-ins are designed for different types of measurements and consequently number of controls are different.

In general, each plug-in includes both signal generator and analyzer function. Typically, a plug-in will calculate multiple result vectors on one or more channels.

Please refer to section 0 for a description of each plug-in.

5.6 Channel Configuration

This tab contains the configuration for all output and input channels. The plug-in shell supports multiple output and input channels depending on the selected measurement plug-in and platform.

The notation is that the data plug-in requests a number of output and input signals that must be assigned to a channel on the hardware.

Please note that one channel can only be assigned for one signal. Also the channels must be assigned in increasing order:

- OK to assign: Signal 1 ⇔ Channel 1, Signal 2 ⇔ Channel 3...
- NOT OK to assign: Signal 1 ⇔ Channel 2, Signal 2 ⇔ Channel 1...

Other configurations may result in error or unpredictable result.

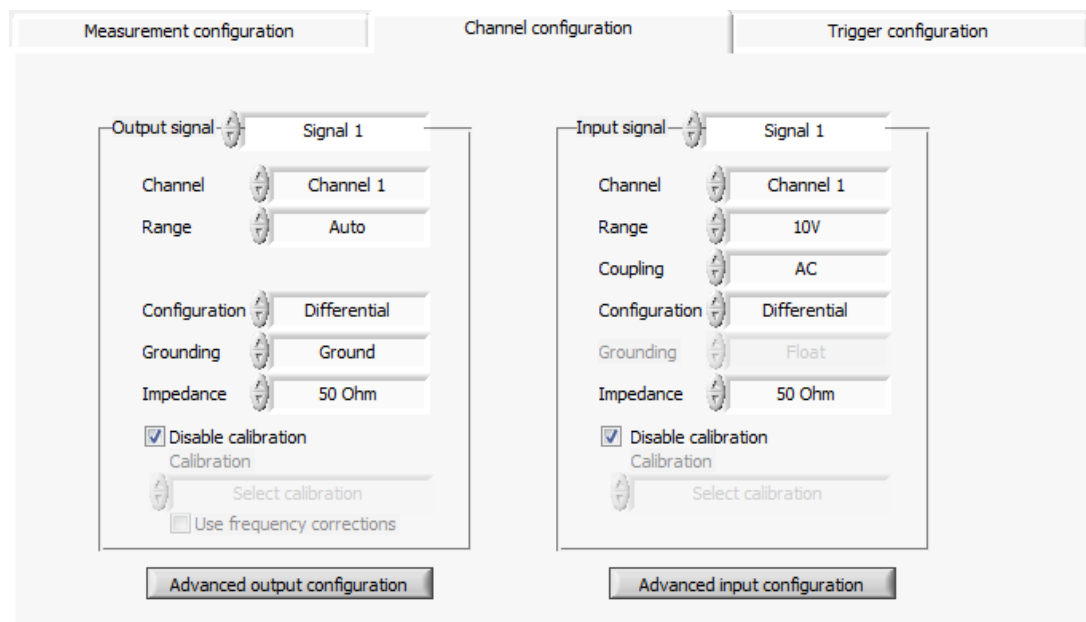


Figure 4: Channel configuration

The available range and coupling options etc. depends on the selected hardware platform. Range values are shown in the value and unit as calculated from the hardware voltage range using the selected calibration value and unit selected on the plug-in configuration panel. The hardware voltage range value is shown in the tool tip when placing the mouse cursor over the range control.

Each channel is can be individually assigned to a calibration data item. When calibration is disabled, as it is mostly the case in electrical measurement, the measurement values are

5.6.1 Advanced Configuration

Some platforms provide additional settings, like IEPE power supply on NI-446x.

Click the advanced configuration button to open the platform specific configuration panel.

The 'Simulate' platform provides advanced configuration to allow users to change how the simulated signal is processed.

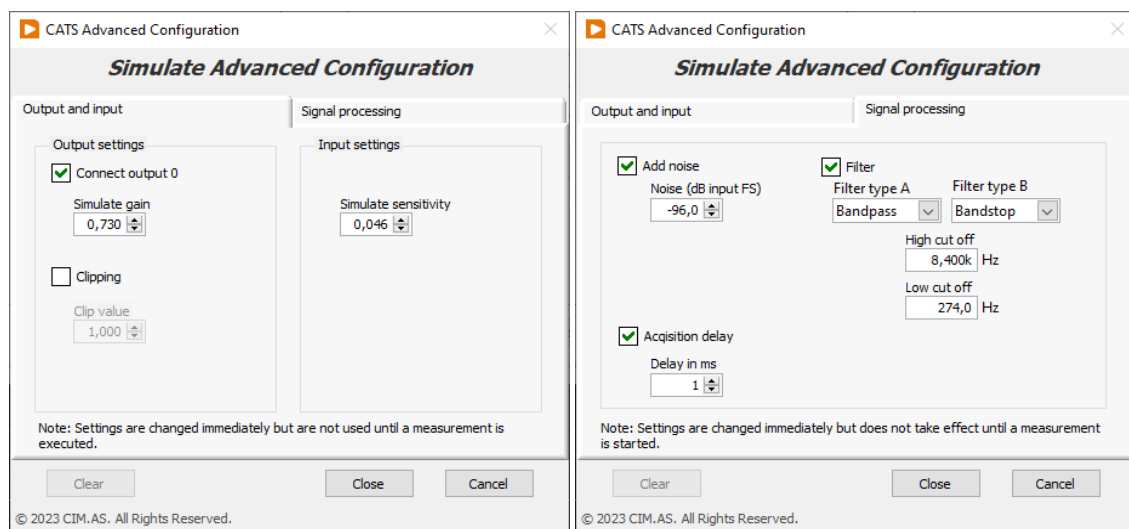


Figure 5: Advanced Configuration - Simulate

In general, the advanced configuration panel for a platform covers both input and output, so the same window appears when the same platform is selected.

5.7 Trigger Configuration

The settings on this tab control how the waveform acquisition is triggered. Trigger capabilities depend on the selected input device.

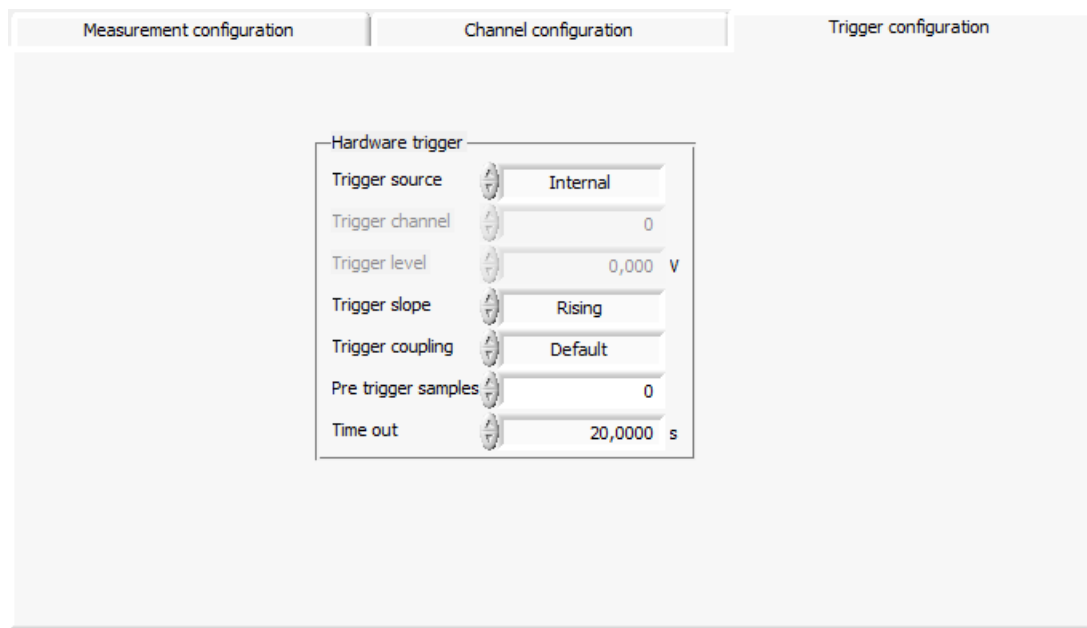


Figure 6: Trigger settings

In a typical hardware configuration using a single DAQ card (fx. NI-4461) with both analog input and output, the selected input channel will trigger the output channels.

5.7.1 Trigger Source

Selects the source for the trigger to start waveform output and input operation.

- **Internal**
This is the default option that will immediately trigger both input and output.
- **Analog input**
Use the analog signal level on the specified input channel
- **External trigger**
Use the external trigger input connector.
- **PFI pin**
Use the digital level on the specified PFI input line.
- **RTSI pin**
Use the digital level on the specified RTSI input line.

The available options and exact operation depend on the selected hardware platform. When possible, the platform driver attempts to trigger output and input operation synchronously.

5.8 Result Preview

The bottom frame of the setup panel is reserved for result preview. Result data is present after using the 'Execute setup' function.

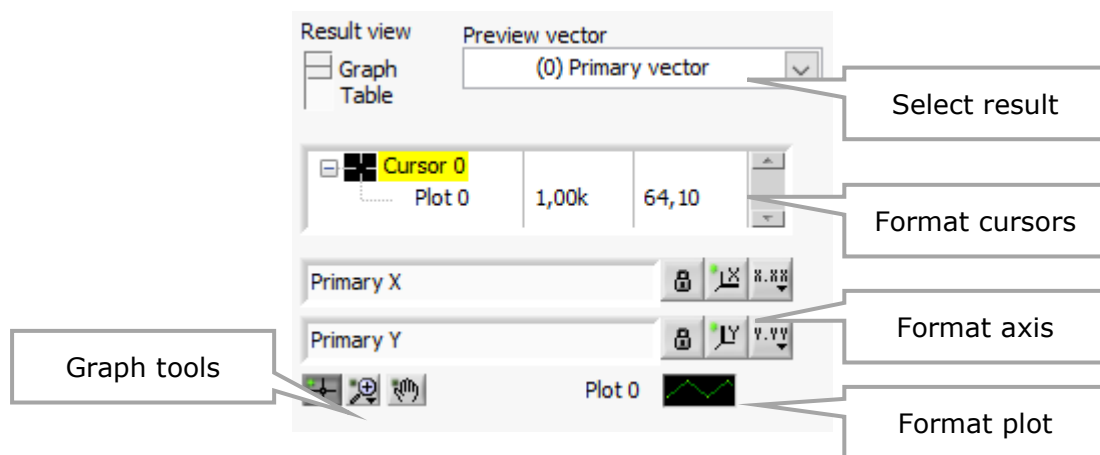


Figure 7: Result preview settings

All result vectors are available after executing the setup and can be reviewed one at a time, using the preview vector selector control.

5.8.1 Result View

The plug-in shell provides a graphical and a tabular view that can be toggled using the Result view switch.

NOTE: Turn off auto-scaling when displaying long time records in the graph. Auto-scaling large data requires a great amount of system resources and causes long response times.

5.8.2 Preview Vector




Most data plug-ins return an array of vector results, where one vector may contain a time signal, a FFT spectrum or other two-dimensional data. The available results of the current plug-in are automatically listed in the drop-down box.

This control selects which vector to view in the graph or table. Because all vectors are returned as one result, the user can easily browse through all of the results without executing the measurement over again.





5.8.3 Adjust Graph and Cursor Display

When using the graph for result preview, its display properties can be modified to suit the selected result vector. The formatting is quite comprehensive, but the following paragraphs will list some of the commonly used options.

Axis Formatting Controls

Control	Description
	Lock or unlock auto-scaling on the axis. When auto-scaling is locked the axis is scaled automatically whenever the graph data is updated.
	Click this button to auto-scale the axis.
	Click this button and select formatting options for the axis, including numeric formats, linear or logarithmic scaling and tick marks.

Plot Formatting Controls

Control	Description
	Selects the cursor move mode, to allow dragging the cursor.
	Selects one of the available zooming modes.
	Selects the plot move mode, to allow dragging the plots on the graph.
	Click the plot template to select plot style, color, point style, interpolation and many other properties to format the graph.

5.9 Other Controls and Indicators

Function Buttons

The function buttons along the bottom of the window have the following meanings:

- **Execute setup**
Use this function to test the settings directly. This is equivalent to running the function as a test step in the test sequencer.
- **Cont. Run**
This function is similar to Execute setup, but this function executes the setup repeatedly.
- **Save setup**
This function is reserved for future use.
- **Load setup**
This function is reserved for future use.
- **Set as previous**
Use this function to re-load all settings, as they were the last time the configuration panel was closed using the OK button.
- **OK**
When selecting OK the set-up is stored before returning to the test sequencer.
- **Cancel**
Selecting Cancel will reset the settings to the values, as they were when the front panel was opened, before returning to test sequencer.

The execution status will light up while measurement is running, including signal generation and result calculation.

The result is displayed in preview graph. This is often the most direct way to check that the settings are correct.

5.10 Configuring Multi-Channel Plug-ins

Most plug-ins support multiple channels, while some support only one channel input and output.

A typical plug-in can output the same signal on the available output channels and analyzing signals from the available input channels.

Understanding Signals and Channels

To have a simple yet flexible user interface on the plug-ins operate on signals.

Each signal is tied to a hardware channel as configured on the channel configuration tab explained in Channel Configuration.

The plug-in shows a list of signals that it can generate or analyze respectively. The number of available signals is limited by the number of channels that the currently selected platform provides, or the number of signal that the plug-in can handle limits the number of signals.

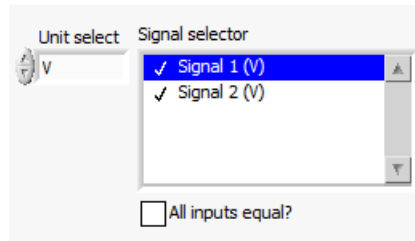


Figure 8: Signal selection list and unit selector

Checkmarks indicate that the signal is enabled; while the highlight indicates which signal the unit selector applies to.

Checkmarks are set or cleared by double clicking on the signal item in the list. Signals can only be selected in a continuous sequence and must start at the first signal.

For input signals the units can be selected individually by clicking the signal to highlight it, then selecting the unit.

5.11 Understanding Result Vectors

When a plug-in can operate on a selectable number of signals the organization of the result vectors must be equally dynamic.

In general, a result is either a single signal calculations or dual signal calculation. For example, a RMS vector calculation is based in a single signal, while a gain vector is the proportion of two signals.

A plug-in typically calculates several single signal vectors and a few dual signal vectors based on some of the single signal vectors.

When multiple signals are selected the result vectors are organized so that the single signal vectors for 'Signal 1' are followed by the single signal vectors for 'Signal 2', and so forth. Thereafter follows the dual signal vectors for signal 1 and 2, 2 and 3 etc.

In the example below the vector list of a plug-in that returns N single signal vectors and M dual vectors after analyzing S signals.

Index	Analysis	
0	Primary vector	
$N \cdot i + 1$	Signal $i+1$ Vector 1	Single signal vectors
...		
$N \cdot i + N$	Signal $i+1$ Vector N	
$N \cdot S + M \cdot j + 1$	Signal $j+1$ vs. Signal j Vector 1	Dual signal vectors
...		
$N \cdot S + M \cdot j + M$	Signal $j+1$ vs. Signal j Vector M	

$i \in \{0; S-1\}$ and $j \in \{0; S-2\}$

If only one signal is analyzed, the dual signal vectors are not calculated. The following figure illustrates five single vectors (N) and two dual vectors (M) on two signals (S).

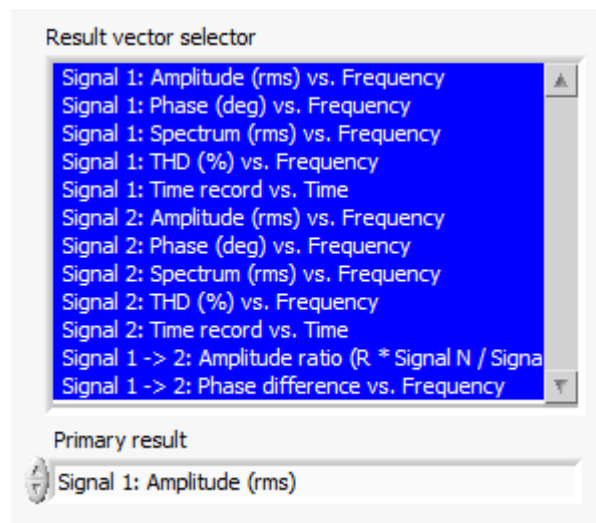


Figure 9: Result vector list and selector

When analyzing multiple signals, the number of vector and amount of data can grow rapidly. This will affect the performance of the test sequencer and it is advised to select only relevant vectors.

The vector list shows result vectors based on the current settings. Click items using Shift or Ctrl to select a range or add items to the selection.

6 Signal plug-ins

This chapter describes the details of each signal plug-in.

6.1.1.1 THD Calculation Settings

Many plug-ins provide THD measurements on each signal segment. The different methods for THD calculation as shown by the formulas below.

$$THD = \sqrt{\frac{\sum Harmonics^2}{Detected^2}}$$

$$THD_{IEC} = \sqrt{\frac{\sum Harmonics^2}{Detected^2 + \sum Harmonics^2}}$$

$$THD + N = \frac{RMS(Residual)}{RMS(Detected)} = \frac{1}{SINAD}$$

Equation 1: THD calculation methods

All measurements are using the highest detected tone near the specified frequency as the fundamental tone.

Note that the THD values are normally plotted at the output frequency rather than the measured frequency of the detected fundamental. This is ensuring a more consistent result and limit testing. You can use the measured frequency result vector to test that the detected frequency matches the output frequency.

In frequency sweeps, when any of the selected harmonics for the test tone is above the Nyquist frequency, the THD value is not calculated and return 'NaN' (shown as 'IND' in TestStand).

6.2 Amplitude sweep plug-in

This plug-in can generate and analyze a sequence of pure tones with increasing or decreasing amplitude. The tone segments are of equal length and same frequency.

It generates the same output signal on one or more channels and analyzes input on one or more channels.

The signal is built from a sequence of pure tone blocks with specified Frequency and varying amplitude.

6.2.1 General Description

The frequency of the output signal may be adjusted to make each block contain an integer number of cycles. This is to ensure that the amplitude shift is continuous.

The duration of each tone block is the sum of Pre-delay, Post-Delay and Measure time. Only the signal within the measurement time is used for analysis. Delay times are used as buffer against time delays between input and output channel, and for settling time for the device under test.

To ensure that the transitions from one tone to the next are done at zero-crossings the actual frequency is adjusted to be coherent within the constraints of the measure time and sample rate. As a result, the pre- and post-delay times are also adjusted to be within the specified value the measure time. The initial delay may also be adjusted but is not limited by measure time.

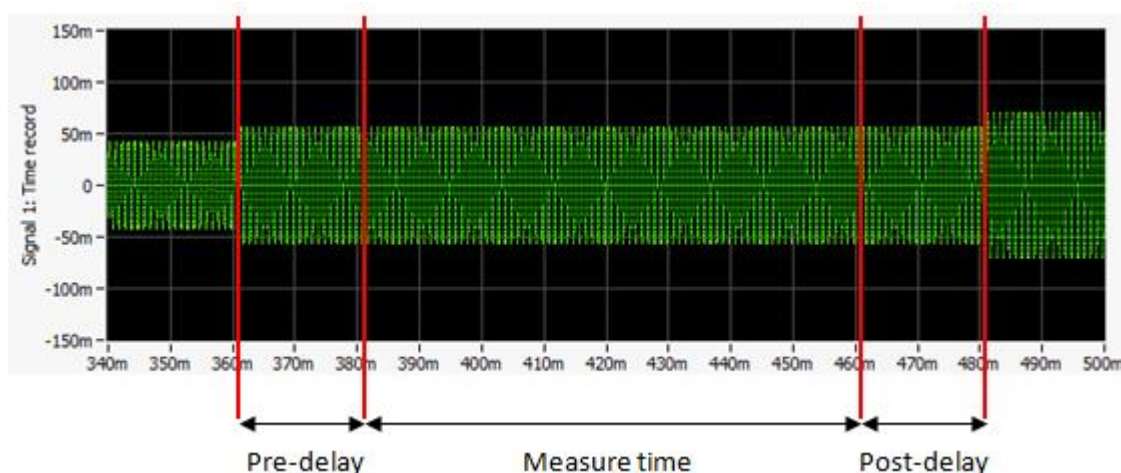


Figure 10: Example of tone block time signal produced by Amplitude sweep plug-in

The amplitude range is specified by Start- and End amplitude, in the selected unit. Step size is also specified in the selected unit and determines the number of steps.

6.2.2 Generator Settings

The generator settings define the content of the output signal.

So even if the plug-in is used in measure only mode the generator settings are used as a specification for number to tone steps, levels and frequency.

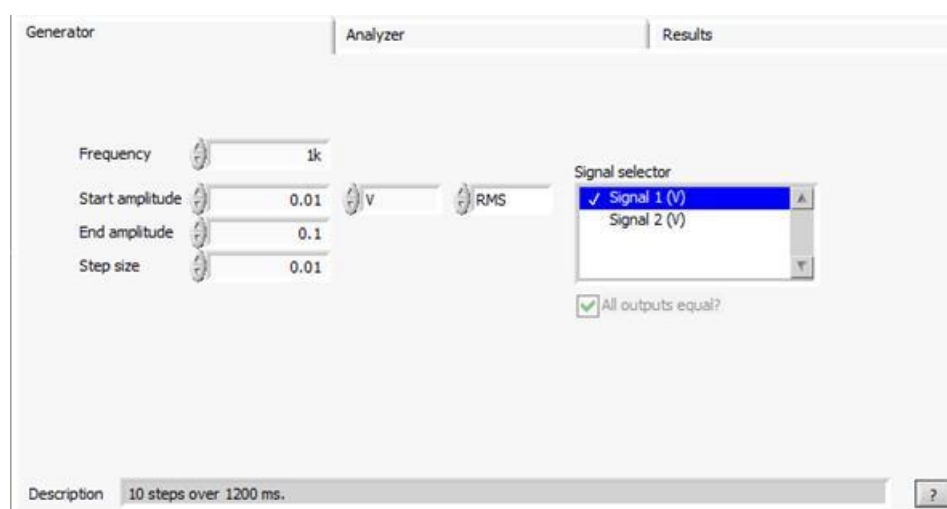


Figure 11: Amplitude sweep plug-in generator configuration panel

The generator supports multiple channels, but with the restriction that all channels output the same signal. This implies that all channels must use same unit.

Control	Unit	Range	Description
Frequency	Hz	0 - +inf	Test signal frequency for all tone blocks. May be automatically adjusted to fit an integer number of cycles into a tone block.
Start amplitude	Unit*	0 - +inf	Amplitude of the first tone block, specified in the selected unit.

Control	Unit	Range	Description
End amplitude	Unit*	0 - +inf	Target amplitude of the last tone block, specified in the selected unit. Depending on the Step size the last tone block may or may not exactly match this amplitude.
Step size	Unit*	±inf	Amplitude step from one tone block to the next, specified in the selected unit.
* You can specify the unit of measurement on each channel for this control.			

6.2.3 Analyzer Settings

The analyzer settings define the measurement timing for each tone segment. Together with the generator settings this defines the complete test signal and acquisition.

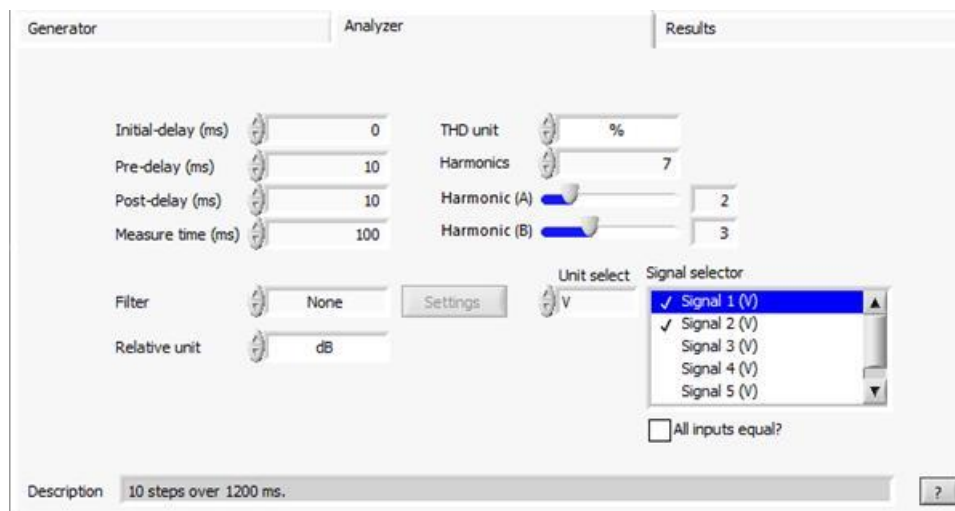


Figure 12: Amplitude sweep plug-in analyzer configuration panel

The analyzer supports multiple signals that can have selectable units.

Control	Unit	Range	Description
Initial-delay	ms	0 - +inf	The duration of initial signal before the first measurement time.
Pre-delay	ms	0 - +inf	A pre-delay time before the actual measurement time, within each tone block.
Post-delay	ms	0 - +inf	A post-delay time after the actual measurement time, within each tone block.
Measure time	ms	0 - +inf	The actual measurement time within each tone block.

Control	Unit	Range	Description
THD unit	—	None % dB	Select the unit for THD and THD+noise measurements. None returns the unit free ratio of fundamental tone to harmonics or harmonics and noise respectively.
Harmonics	—	1 - +inf	Specifies the number of harmonics to include in the THD measurement.
Harmonic (A)	—	2 - (Harmonics - 1)	Specifies the selected harmonic to use for nth harmonic calculations.
Harmonic (B)	—	Harmonic (A) - Harmonics	Specifies the selected harmonic to use for nth harmonic calculations.
Filter	—	None Butterw. A-filter ITU-R 468	Selects a digital software filter on the input signal. This can be used to filter out noise outside the multi-tone range.
Relative unit	—	None % dB	Select the unit for relative amplitude measurements.

Input Filtering

Also to improve measurements the plug-in provides a filter that is applied to the entire acquired signal, before extracting the FFT block. This allows the Initial delay to serve as settling time for the filter. The following filter functions are available:

- None
- Butterworth
- A-filter
- ITU-R 468

When using the Butterworth filter its default settings are: 6'th order High Pass Filter with lower cut-off at 125 Hz. Filter settings can be changed using the 'Settings' button.

6.2.4 Result Settings

On the result settings tab a list of all available analyzer results are shown. All highlighted results are returned to the test sequencer in the listed order.

To reduce the amount of data returned to the test sequencer, NI recommends deselecting result vectors that are not relevant to the current measurement.

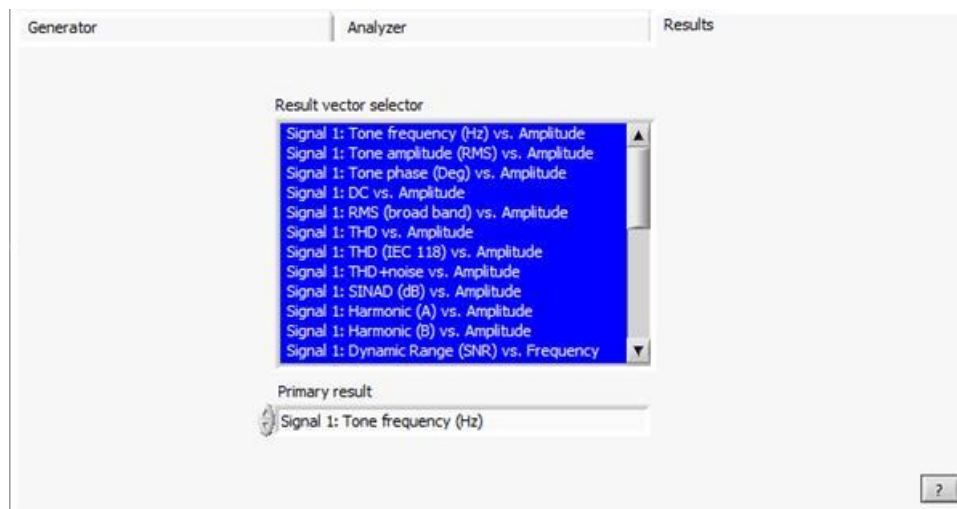


Figure 13: Amplitude sweep plug-in result configuration panel

The following analyzes are returned as vector results. The vector at index 0 is selected by the *Primary result* control.

Index	Analysis	Unit	Description
0	Primary y-values vs. Primary x-values	As primary y	Copy of the result vector you chose as the primary result.
1	Tone frequency (Hz) vs. output amplitude	Hz	Measured frequency of each tone segment.
2	Tone amplitude (RMS) vs. output amplitude	Unit*	Measured narrow-band amplitude (FFT) of each tone segment.
3	Tone phase (Deg.) vs. output amplitude	Deg	The phase of each tone segment.
4	DC vs. output amplitude	Unit*	Estimate of the amplitude of each tone segment in DC.
5	RMS vs. output amplitude	Unit*	Measured broadband RMS amplitude of each tone segment.
6	THD vs. output amplitude	THD unit	Measured total harmonic distortion in the selected unit for each tone segment.
7	THD (IEC 118) vs. Frequency (detected)	THD Unit	THD value after IEC method and frequency of highest detected tone in the measured signal.
8	THD+noise vs. output amplitude	THD unit	Measured total harmonic distortion and noise in the selected unit for each tone segment.

Index	Analysis	Unit	Description
9	SINAD vs. output amplitude	dB	Measured SINAD value in dB of each tone segment. This is reciprocal of THD+noise.
10	Harmonic (A) vs. Frequency	Unit*	Measured selected harmonic and frequency of the highest detected tone in the measured signal.
11	Harmonic (B) vs. Frequency	Unit*	Measured selected harmonic and frequency of the highest detected tone in the measured signal.
12	Dynamic Range (SNR) vs. output amplitude	dB	Measured dynamic range based on the current sweep. This is also known as signal-to-noise ratio in the presence of signal.
13	Time record vs. Time	Unit*†	The acquired raw data scaled to units and calibration
14	Signal N / Signal N+1 vs. output amplitude	Relative unit	Relative tone amplitudes (vector 2) of two input channels for each tone segment.
15	Phase Signal N – Signal N+1 difference vs. output amplitude	Deg.	Phase difference between two input channels for each tone segment.
<p>* You can specify the unit of measurement on each channel for this control.</p> <p>† If the selected unit is dBspl, the result is returned in Pa.</p>			

Note that the definition of RMS value at index 5 includes any DC component in the signal. To calculate the RMS value excluding the DC, use the following formula:

$$\text{RMS(AC)} = \sqrt{\text{RMS}^2 - \text{DC}^2}$$

Equation 2: Correcting a broadband RMS value for DC component

Dynamic Range Measurement

Dynamic Range (DR) is the recommended method for obtaining the so-called 'Signal-to-Noise in the presence of signal' value.

Other SNR methods measure a specific signal level to the noise level with no signal. This may be appropriate for analog systems but may give unreal value for digital equipment that often mutes the output when no input signal is present.

The Dynamic Range is measured using the following steps:

- Locate the point where THD is 1%.
- The full-scale level is defined at the output level 0.5 dB lower than the 1% THD point. If the THD value does not exceed 1% the last level is used.

- Get the SINAD value at the output level 60 dB below the full-scale level.
- Dynamic Range is the above SINAD value plus 60 dB.

For the dynamic range measurement to give a correct result it must be configured to span from the full scale of the device under test to at least 60 dB below that point.

Even though linear interpolation is used to estimate relevant values, the sweep must be configured for a relatively fine step size to locate the full-scale value correctly.

6.3 Bluetooth Headset plug-in

This plug-in can generate and analyze a multi-tone signal. The test signal and analysis can be adapted wide range of measurements, using the rich set of controls.

It generates output on one channel and analyzes input on one channel.

6.3.1 General Description

This plug-in is designed for testing Bluetooth headsets etc. using a multi-tone signal.

One specialty of measuring audio over a Bluetooth connection is that it will from time to time suffer from packet loss that may result in distortion or dropouts. For this reason the analysis includes input filtering and averaging.

The analysis of measurements includes RLR, Receive Loudness Rating and SLR, Send Loudness Rating.

Still the plug-in is very flexible and can be used for generic measurements as well.

Special Plug-In Units

This table shows how the special plug-in units are translated to standard CATS units for calibration purposes and summarize the result units.

Generator units	V	dBPa	dBspl	dBmV	Pa
CATS units	V	dBspl	dBspl	dBspl	Pa
Time record	V	Pa	Pa	Pa	Pa
Power spectrum	V_{rms}^2	Pa_{rms}^2	Pa_{rms}^2	Pa_{rms}^2	Pa_{rms}^2
Amplitude response	V_{rms}	$dBPa_{rms}$	$dBspl_{rms}$	$dBmV_{rms}$	Pa_{rms}
1/3-oct. response	V_{rms}	$dBPa_{rms}$	$dBspl_{rms}$	$dBmV_{rms}$	Pa_{rms}
Sensitivity - send - receive	dB Pa/V	dB V/Pa	dB V/Pa	dB Pa/V	dB V/Pa

6.3.2 Generator Settings

This section explains the operation of the generator and effect of the main controls.

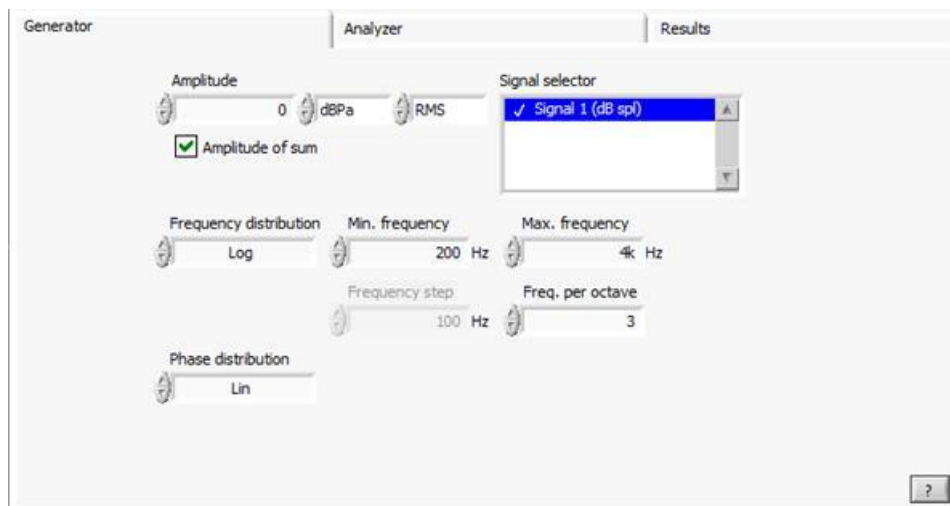


Figure 14: BT-Headset generator configuration panel

Amplitude Settings

The output amplitude is specified in the selected unit that may be limited by the selected measurement type. The output signal is a sum of individual tones over the specified frequency range. The tone frequencies are adjusted to fit an integer number of cycles within the measurement time (coherent tone).

When tones are added the peak value of the sum signal is much higher than that of a single tone.

Selecting the Amplitude of sum control will scale the sum signal to match the specified amplitude.

Alternatively, if the Amplitude of sum is unselected, the amplitude value specifies the amplitude of the individual tones. In this case the specified amplitude must be much lower than the selected range on the output channel – typically the amplitude must be set to allow a crest factor of 15 – 25 dB.

Signal Timing

The timing of the signal is controlled by the Measure time, Priming delay and Measure delay controls. The picture below shows how the values are related to the test signal.

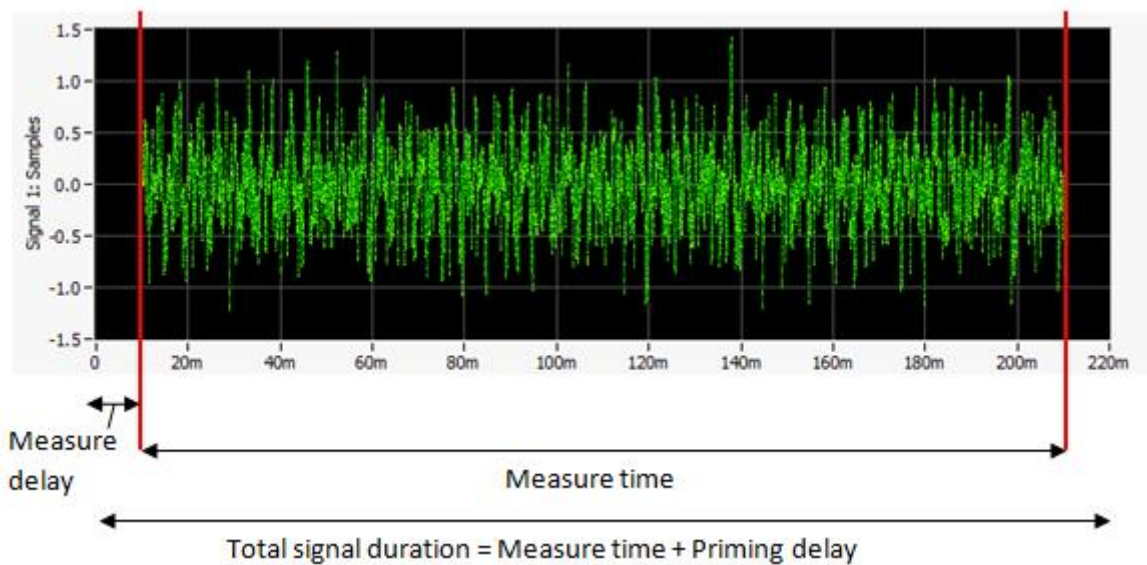


Figure 15: BT-Headset signal and measurement timing

NOTE: The duration of the output signal in the above example is 220 ms but only the analyzed part of the signal is shown in the graph.

Frequency Settings

The number of tones and their frequency values are specified by the controls Min. frequency, Max. frequency and Frequency distribution. The distribution can be either linear or logarithmic and enables Frequency step or Freq. per octave respectively.

Each frequency value is adjusted internally to produce an integer number of cycles within the measurement time. This allows coherent measurement of the response, where all of the test signal is located in single bins in the spectrum.

Wave Device Mode

This indicator reflects the setting of Wave device mode control on the plug-in shell and shows how the test signal generated.

In Play-meas. mode the test signal is created as a single waveform buffer of total duration equal to the sum of Measure time and Priming delay.

In Play only and Play cont. mode the test signal is a waveform buffer of duration Measure time that is playing in a looped mode to produce a continuous signal.

For all cases the test signal can be measured in a coherent way because all tones contain integer numbers of cycle within the FFT block size.

6.3.2.1 Generator Controls Summary

Control	Unit	Range	Description
Amplitude	Unit*	$\pm\text{inf}$	Amplitude of each tone component in the output signal, using the selected unit. Note that the addition of all the tone components will increase the crest factor of the signal and the amplitude of the sum signal is much higher than individual tones.

Control	Unit	Range	Description
Unit	—	V dBV dBmV Pa dBP _a dBspl	Selects the unit of amplitude or result values for BT-Headset plug-in: V – Volt dBV – $20 \log (X/1 \text{ V})$, 0 dBV = 1V dBmV – $20 \log (X/1 \text{ mV})$, 60 dBmV = 1 V Pa – Pascal dBP _a – $20 \log (X/1 \text{ Pa})$, 0 dBP _a = 1 Pa dBspl – $20 \log (X/20 \text{ uPa})$, 94 dBspl = 1 Pa The special dB units are converted internally to/from standard CATS units.
Unit type	—	RMS Peak Pk-toPk	Selects the type of unit: RMS, Peak, Pk-to-Pk.
Amplitude of sum	—	T/F	Selects if the amplitude specifies the amplitude of the individual tones or their sum. Please note that when adding the individual tones the crest factor of the resulting multi-tone signal can be 15 – 20 dB depending on number of tones etc.
Phase distribution	—	Lin Random	Selects how the tone phases are distributed when forming the multi-tone signal: Lin, Random.
Frequency distribution	—	Lin Log	Selects frequency distribution for the multi-tone signal. Lin, Log
Min. frequency	Hz	0.. <i>n</i>	Minimum frequency of the multi-tone signal
Max. frequency	Hz	0.. <i>n</i>	Maximum frequency of the multi-tone signal
Frequency step	Hz	0.. <i>n</i>	Specifies the distance in Hz between frequencies in Lin distribution.
Freq. per octave	—	1.. <i>n</i>	Specifies the number of tones per octave in Log distribution.
* You can specify the unit of measurement on each channel for this control.			

6.3.3 Analyzer Settings

This section explains the operation of main analyzer functions.

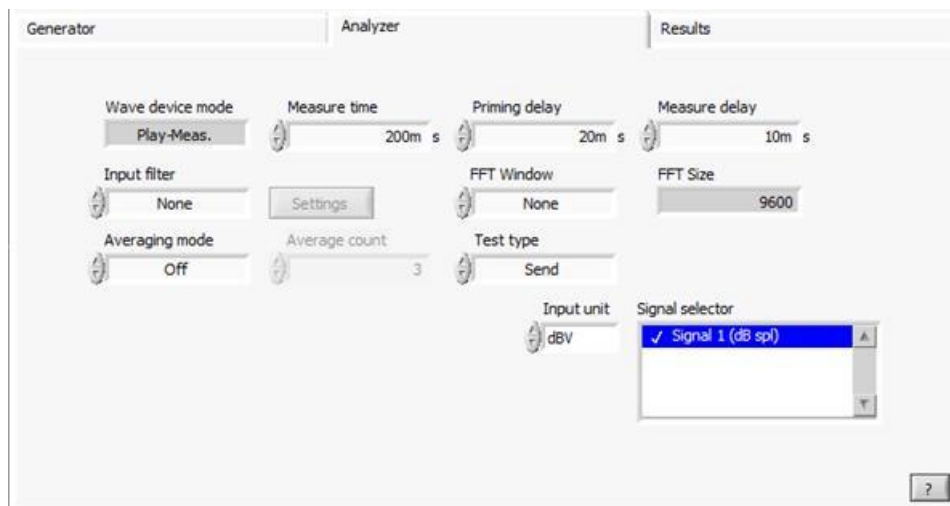


Figure 16: BT-Headset analyzer configuration panel

FFT Settings

When measuring coherent signals the FFT window control is normally set to None because the test signal is continuous over the FFT block size. If this is not possible, due to excessive noise etc. a suitable FFT window must be selected.

The FFT size is calculated automatically from Measurement time and sample frequency on the plug-in shell.

Averaging Modes

The plug-in provides four averaging modes to help improve measurement of noisy signals and reduce the effect of dropouts in Bluetooth link etc.

The Average count value sets the number of FFT spectra to average before further analysis. When averaging is enabled the test signal is automatically prolonged to match the extra measurement time.

- **Off** – no averaging
- **Combined** – calculates the linear average of N medians based on N spectra.
- **Median** – calculates the median on N spectra.
- **Lin average** – calculates a linear average on N spectra.
- **Exp. average** – calculates the exponential average of N spectra.

Input Filtering

Also to improve measurements the plug-in provides a filter that is applied to the entire acquired signal, before extracting the FFT block. This allows the value specified in the Priming delay control to serve as settling time for the filter. The following filter functions are available:

- None
- Butterworth
- A-filter
- ITU-R 468

When using the Butterworth filter its default settings are: 6'th order High Pass Filter with lower cut-off at 125 Hz. Filter settings can be changed using the Settings button.

Test Types

The Test type control is used to select the calculation mode for Loudness Rating. Send and Receive refers to the common notion headset measurement and implies a limited selection of units for input and output signal.

- **Send** – Send Loudness Rating – SLR
- **Receive** – Receive Loudness Rating – RLR
- **Generic** – LR is not calculated. Selectable unit for non LR measurements

6.3.3.1 Analyzer Controls Summary

Control	Unit	Range	Description
Measure time	s	0..10	Duration of multi-tone signal in seconds. This value limits the minimum frequency and specifies the FFT size.
Priming delay	s	0..10	Duration of multi-tone priming signal in seconds. Use this to add some signal to compensate for delays in the signal path, like anti-alias filter delay, travel delay etc.
Measure delay	s	0..10	Measurement delay in seconds, from start of priming signal to start of FFT block.
Input filter	—	None Butterw. A-filter ITU-R 468	Selects a digital software filter on the input signal. This can be used to filter out noise outside the multi-tone range. Filter settings may be review or change using "mIxHM BT-Headset Filter param GLOBAL.vi" in the plug-in .llb.
FFT Window	—	0..9	Selects windowing function to apply to each FFT block. None , Hanning, Hamming, Blackman-Harris, Exact Blackman, Blackman, Flat Top, 4 Term B-Harris , 7 Term B-Harris, Low Sidelobe
Averaging mode	—	0..4	Selects averaging mode for FFT spectra. Off, Combined, Median, Lin. Average, Exp. average
Average count	—	1.. <i>n</i>	Specifies the number of FFT spectra to use for the selected average mode. When averaging is off the value is ignored.
Test type	—	0..2	Selects calculation mode for Loudness Rating: Send – Send Loudness Rating – SLR Receive – Receive Loudness Rating – RLR Generic – Selectable unit for non LR measurements

Control	Unit	Range	Description
Input unit	—	V dBV dBmV Pa dBPa dBspl	<p>Selects the unit of amplitude or result values for BT-Headset plug-in:</p> <p>V – Volt</p> <p>dBV – $20 \log (X/1 \text{ V})$, 0 dBV = 1V</p> <p>dBmV – $20 \log (X/1 \text{ mV})$, 60 dBmV = 1 V</p> <p>Pa – Pascal</p> <p>dBPa – $20 \log (X/1 \text{ Pa})$, 0 dBPa = 1 Pa</p> <p>dBspl – $20 \log (X/20 \text{ uPa})$, 94 dBspl = 1 Pa</p> <p>The special dB units are converted internally to/from standard CATS units.</p>

6.3.4 Result Settings

The vector at index 0 is selected by the Primary result control. The following analyzes are returned as vector results.

Index	Analysis	Unit	Description
0	Primary y-values vs. Primary x-values	As primary y	Copy of the result vector you chose as the primary result.
1	Samples vs. Time, s	Input unit*	Last acquired time record.
2	RMS values vs. index	Input unit	RMS value of each FFT block in an averaged measurement.
3	Averaged power spectrum vs. Frequency, Hz	Input unit	Resulting average power spectrum.
4	Amplitude vs. Frequency, Hz	Input unit	Amplitude values of averaged power spectrum at the generator frequencies.
5	Gain response, dB vs. Frequency, Hz	dB	Input amplitude relative to output amplitude in dB.
6	Sensitivity, dB vs. Frequency, Hz	dB	Sensitivity values in the 14 bands 200 – 4k calculated from the amplitude vector.
7	Loudness Ratings vs. Index	dB	Selected type of Loudness Rating values calculated from the sensitivity vector.
8	Clearance of primary result vs. Index	Input unit	Vertical clearance from the primary result to upper and lower limit on the test step.
9	Offset for clearance vs. Index	Input unit	Offset value that must be added to the primary result vector to achieve maximum clearance within the limits on the test step.

Index	Analysis	Unit	Description
* If the input unit is dB the result is returned in the basic unit V or Pa.			

NOTE: The Clearance (8) and Offset (9) vectors cannot be calculated in the current version when using this plug-in on TestStand.

RMS Values Vector

This vector is a bit special as it contains a series of RMS values, one for each FFT block in an averaged measurement, arranged as shown below:

Index	0	1	2	3	...
X(i)	Variance coefficient	1	2	3	...
Y(i)	Total RMS	RMS(first FFT)	RMS(second FFT)	RMS(third FFT)	...

This vector can be used to measure the noise level. By returning the RMS values for each FFT block the test programmer has the opportunity to check that the value of each block is equal, and not influenced by external noise events or Bluetooth drop-outs.

The variance coefficient of the RMS block is included as an indication of the reproducibility, and calculated using the formula:

$$\text{Variance coefficient} = \frac{\sigma^2}{\mu} = \frac{1}{\mu} \sum_{i=0}^{n-1} \frac{(RMS_i - \mu)^2}{n-1}$$

where μ is the mean and σ is the standard deviation of the RMS values.

Loudness Rating

Loudness Rating (LR) is calculated as SLR or RLR according to the selected test type. LR calculation is based on the measured Sensitivity values using the formula and values in ITU-T P.79 (09/1999) section 5.

6.4 Coherent sweep plug-in

This plug-in can generate and analyze a sequence of coherent tones. The test signal can be configured with respect to timing, frequency range and resolution.

It can generate a test signal on one or more channels and analyze input on one or more channels. Due to timing constraints signal timing applies to all signals and only the output amplitudes can be specified on each output signal.

6.4.1 General Description

The output signal consists of a sequence of tone segments, all with the same amplitude. CATS uses a close connection between the generator and analyzer to analyze each segment into a number of result vectors, thus optimizing test time.

You specify the timing of the tone segments primarily in the Min cycles control, which specifies the minimum number of cycles that must be generated for each tone measurement segment.

Because you specify the measurement time in terms of test signal cycles, lower frequencies require longer measurement times and higher frequencies require shorter measurement times. However, all frequency measurements achieve the same relative accuracy.

You can limit measurement time with the Max measure time control. When you use this control, CATS adjust the tone frequency to be coherent with the measurement time.

The duration of each tone block is the sum of the values in the Pre-delay, Post-Delay, and Max measure time controls. Only the signal within the measurement time is used for analysis. Delay times are used as buffer against time delays between input and output channel, and for settling time for the device under test.

To ensure that the transitions from one tone to the next are done at zero-crossings the actual frequency is adjusted to be coherent within the constraints of the measure time and sample rate. As a result, the pre- and post-delay times are also adjusted to be within the specified value the measure time. The initial delay may also be adjusted but is not limited by measure time.

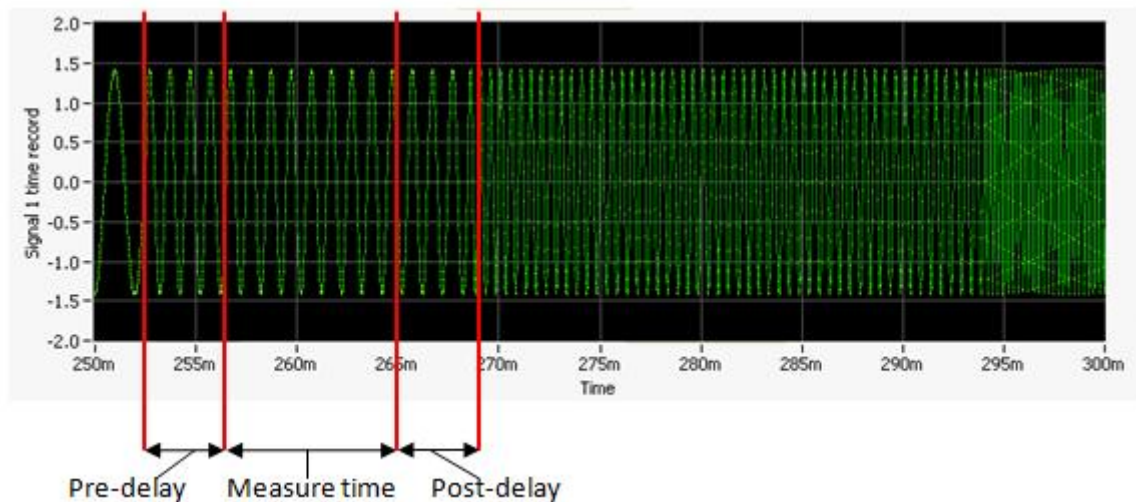


Figure 17: Coherent sweep test signal timing

You can specify the frequency range using the Start freq. and End freq. controls and using Step type to select linear or logarithmic steps. You can also specify a list of frequencies in a table. The frequencies are generated in the order they appear in the table. You can type in the values using SI notation and local system decimal point. If the frequency values are imported from a file, the values must be in decimal point format.

The tone sequence is wrapped in a synchronization signal with silence before and after. The plug-in analyzer uses the signal to locate the tone segments very accurately, even when the test signal is passed through a test device that might introduce a delay.

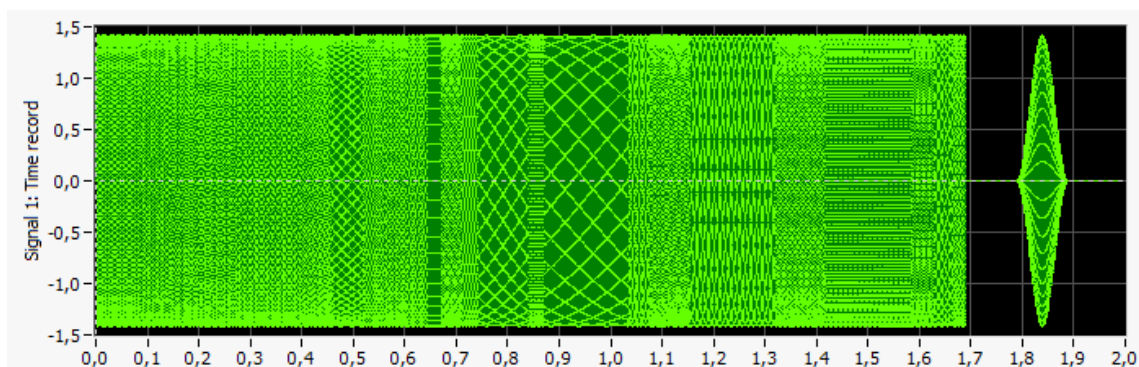


Figure 18: Coherent sweep time record

6.4.2 Generator Setting

The generator settings specify the content of the output signal and the number of tone segments.

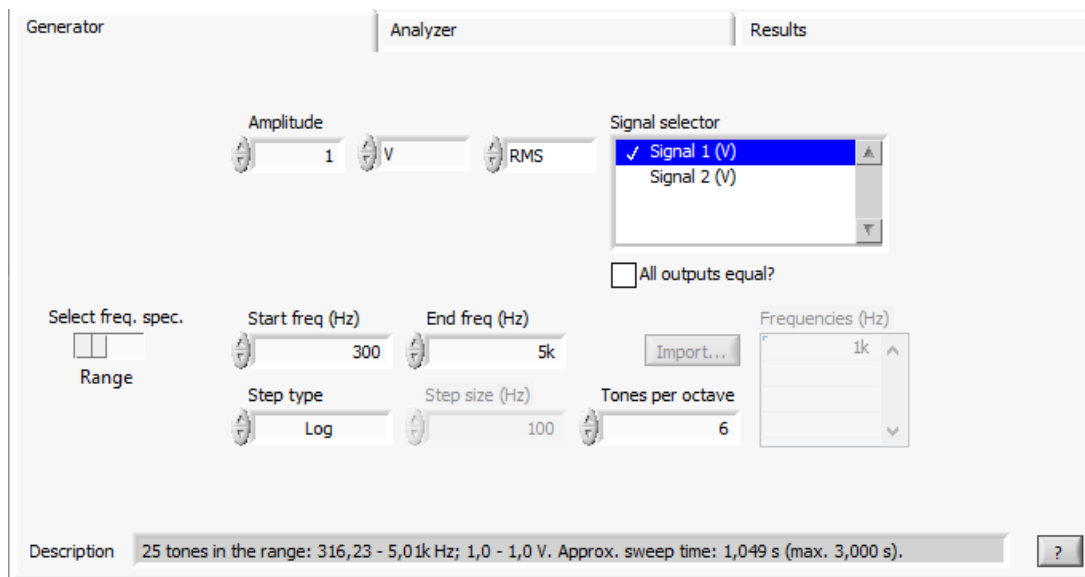


Figure 19: Coherent sweep generator configuration panel

The following table includes information about controls for configuring the plug-in generator settings.

Control	Unit	Range	Description
Amplitude	Unit*	—	Specifies the output amplitude in the selected unit.
Select freq. spec.	—	Range Table	Specifies whether to define a frequency range or a table of frequencies.
Start freq.	Hz	0.. <i>n</i>	First frequency in sequence of tones.
End freq.	Hz	0.. <i>n</i>	Last frequency in the sequence of tones.
Step type	—	Log Lin	Specifies whether the frequency distribution is linear or logarithmic.
Step size	Hz	—	Specifies the linear frequency step in Hz.
Tones per octave	—	0.. <i>n</i>	Sets the logarithmic frequency distribution as tones per octave.
Import	—	—	Imports a list of frequencies from a file that you specify. The file must define the frequencies using a decimal point format.
Frequencies	Hz	—	Contains a list of frequencies to generate. You can specify frequencies in the table manually or import frequencies from a file using the Import control.
* You can specify the unit of measurement on each channel for this control.			

6.4.3 Analyzer Settings

The analyzer settings specify the measurement timing for each tone segment. The analyzer and generator settings together specify the complete test signal and acquisition.

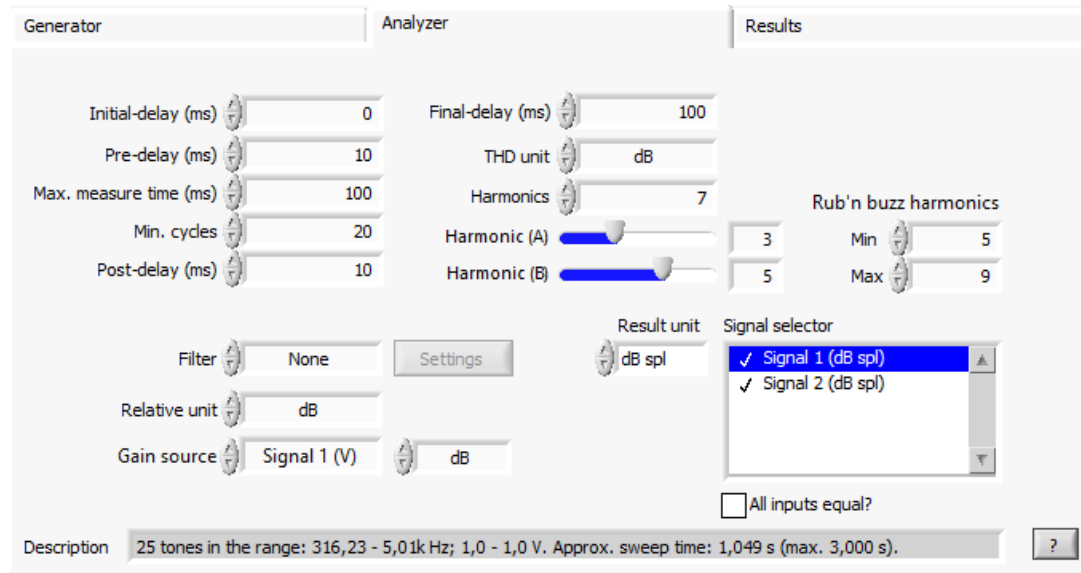


Figure 20: Coherent sweep analyzer configuration panel

The controls for setting test signal and analysis properties are listed below.

Control	Unit	Range	Description
Initial-delay	ms	0.. n	The amount of initial signal before the first amplitude step.
Pre-delay	ms	0.. n	Each tone segment is prolonged by a pre-delay time before the actual measurement time.
Max. measure time	ms	0.. n	Sets a maximum time to limit the measurement time of each segment.
Min. cycles	—	1.. n	Sets the minimum number of tone cycles in each measurement segment.
Post delay	ms	0.. n	Each tone segment is prolonged by a post-delay time after the actual measurement time.
Final delay	ms	0.. n	Sets a delay (silence) after the synchronize signal. Use this to prevent sync signal to be cut off when DUT or input has long delay.
THD unit	—	None % dB	Selects the unit for THD, THD+noise and Rub'n buzz measurements.
Harmonics	—	1.. n	Specifies the number of harmonics to include in the THD measurements.

Control	Unit	Range	Description
Harmonic (A)	—	2..(Harmonics – 1)	Specifies the selected harmonic to use for nth harmonic calculations.
Harmonic (B)	—	Harmonic (A).. Harmonics	Specifies the selected harmonic to use for nth harmonic calculations.
Rub'n buzz harmonic min.	—	2..n	Specifies the minimum harmonic to include in the rub'n buzz calculation.
Rub'n buzz harmonic max.	—	Min.+1..n	Specifies the maximum harmonic to include in the rub'n buzz calculation.
Filter	—	None Butterw. A-filter ITU-R 468	Selects a digital software filter on the input signal. This can be used to filter out noise outside the multi-tone range.
Relative unit	—	None % dB	Select the unit for relative amplitude measurements (between input channels).
Gain source	—	None <outputs>	Select which output signal to use for input/output gain calculation on the selected input signal.
Gain unit	—	None % dB	Select the unit for input/output gain calculation on the selected input signal.

Input Filtering

Also, to improve measurements the plug-in provides a filter that is applied to the entire acquired signal, before extracting the FFT block. This allows the value specified in the Initial delay control to serve as settling time for the filter. The following filter functions are available:

- None
- Butterworth
- A-filter
- ITU-R 468

When using the Butterworth filter its default settings are: 6'th order High Pass Filter with lower cut-off at 125 Hz. Filter settings can be changed using the Settings button.

6.4.4 Result Settings

The Results tab displays all available result vectors in the Result vector selector. Select vectors in the Result vector selector to include them in the analysis.

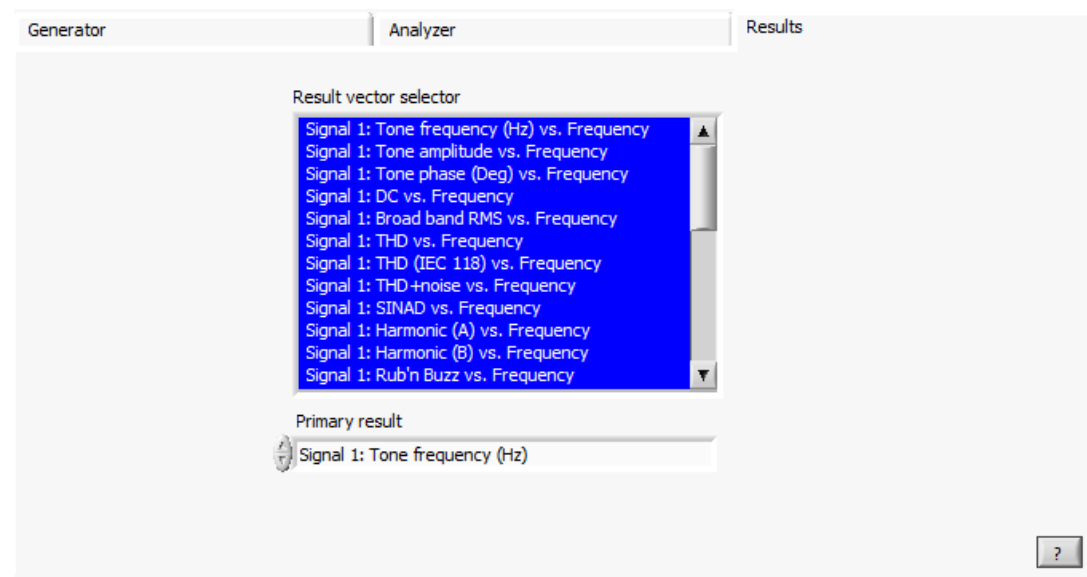


Figure 21: Coherent sweep result settings

Index	Analysis	Unit	Description
0	Primary y-values vs. Primary x-values	As primary y	Copy of the result vector you chose as the primary result.
1	Tone frequency (Hz) vs. Output frequency	Hz	Measured frequency of each tone segment.
2	Tone amplitude (RMS) vs. Output frequency	Unit*	Measured amplitude value of each tone segment.
3	Tone phase (Deg) vs. Output frequency	Deg.	Measured phase of each tone segment.
4	DC vs. Output frequency	Unit*	DC estimate of each tone segment.
5	Broad band RMS vs. Output frequency	Unit*	Measured broad band RMS value of each tone segment.
6	THD vs. Output frequency	THD Unit	Total Harmonic distortion of each tone segment.
7	THD (IEC 118) vs. Output frequency	THD Unit	THD value after IEC method and frequency of highest detected tone in the measured signal.
8	THD+noise vs. Output frequency	THD Unit	Total Harmonic distortion and noise of each tone segment.
9	SINAD vs. Output frequency	THD Unit	Measured SINAD value of each tone segment. This is reciprocal of THD+noise.

Index	Analysis	Unit	Description
10	Harmonic (A) vs. Output frequency	Unit*	Measured selected harmonic and frequency of the highest detected tone in the measured signal.
11	Harmonic (B) vs. Output frequency	Unit*	Measured selected harmonic and frequency of the highest detected tone in the measured signal.
12	Rub'n buzz vs. Output frequency	THD Unit	Measured rub'n buzz value of each tone segment.
13	Gain vs. Output frequency	Gain unit*	Measured gain from the specified output signal to the present input signal for each signal segment
14	Time record vs. Time	Unit*†	The acquired raw data scaled to units and calibration
15	Amplitude of Signal N / Signal N+1 vs. Frequency	Relative unit	Relative tone amplitudes (vector 2) of two input channels for each tone segment.
16	Phase Signal N – Signal N+1 vs. Frequency	Deg.	Phase difference between the two input channels for each tone segment.
<p>* You can specify the unit of measurement on each channel for this control.</p> <p>† If the output unit is dBspl the result unit is changed to Pa.</p>			

6.4.4.1 Rub'n buzz calculation

The rub'n buzz value is basically calculated like THD (see section 6.1.1.1), but using only the harmonics selected for rub'n buzz:

$$Rub'n\ buzz = \sqrt{\frac{\sum Rub'n\ buzz\ harmonics^2}{Detected\ tone^2}}$$

Note that when any of the selected harmonics for the test tone is above the Nyquist frequency, the rub'n buzz value is not calculated and return 'NaN' (shown as 'IND' in TestStand).

6.4.4.2 Gain calculation

The gain calculation from output to input on the selected signal is based on the RMS value of each tone segment.

Depending on the specified units, the gain value is calculated as:

$$Gain = \frac{A_{in}}{A_{out}} \text{ for linear unit, or}$$

$$Gain = A_{in} - A_{out} \text{ for dB units}$$

All special cases for mixed units are handled automatically, including calibration values.

6.4.4.3 Special Usage Hints

This section lists some useful hints and caveats about the use of this plug-in.

- When using this plug-in for THD measurements the measurement time, you must set a sufficient measurement duration in Min cycles and Max measure time to ensure a detailed resolution of the FFT spectrum.
- To optimize sweep time, adjust parameter settings.
- Use the frequency table to create sweeps with special frequency spacing in different ranges.

6.5 Dual Tone plug-in

This plug-in can generate and analyze a sequence of coherent dual tones. The test signal can be configured with respect to timing, frequency range and resolution.

It can generate a test signal on one or more channels and analyze input on one or more channels. Due to timing constraints signal timing applies to all signals and only the output amplitudes can be specified on each output signal.

6.5.1 General Description

The output signal of this plug-in consists of a sequence of dual tone segments. A dual tone segment is the sum of a primary tone and a secondary tone. Each segment contains a combination of new tone frequencies according to the settings. The amplitudes of the two tones can be set individually but remains constant over the segments.

CATS uses a close connection between the generator and analyzer to analyze each segment into a number of result vectors, thus optimizing test time.

You specify the timing of the tone segments primarily in the **Min cycles** control, which specifies the minimum number of cycles that must be generated for each tone measurement segment.

Because you specify the measurement time in terms of test signal cycles, lower frequencies require longer measurement times and higher frequencies require shorter measurement times. However, all frequency measurements achieve the same relative accuracy.

You can limit measurement time with the **Max measure time** control. When using this control, CATS adjusts the tone frequency to be coherent with the measurement time.

The frequency range applies to primary tone and is specified directly by **Start freq** and **End freq** where **Step type** selects linear or logarithmic steps. The frequency of the secondary tone is determined by the selected **Dual tone mode** and **Dual tone freq**.

Alternatively, a list of frequencies can be specified in the table. The frequencies will be generated in the order they appear in the table. You can type in the values using SI notation and local system decimal point. If the frequency values are imported from a file, the file must use decimal point format.

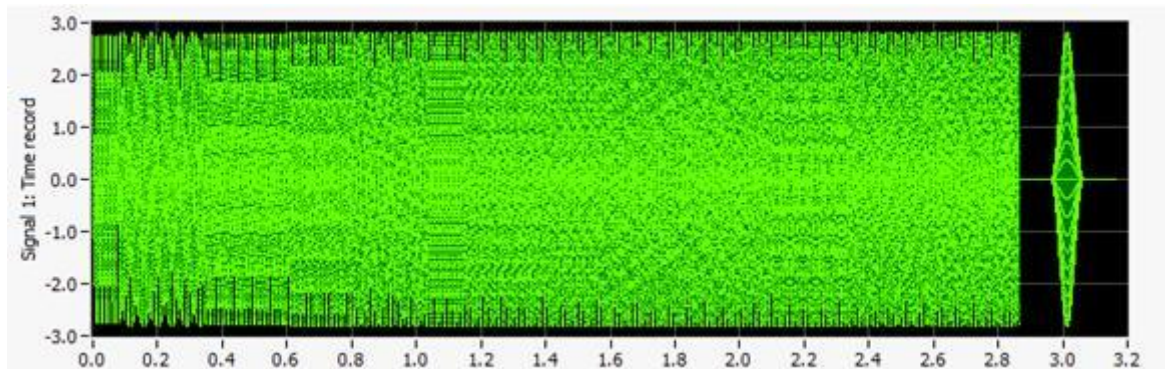


Figure 22: Dual tone time record

The tone sequence is wrapped in a synchronization signal with silence before and after. The plug-in analyzer uses the signal to locate the tone segments very accurately, even when the test signal is passed through a test device that might introduce a delay.

6.5.2 Generator Settings

The amplitudes of the primary and secondary tone are set individually but share the same unit.

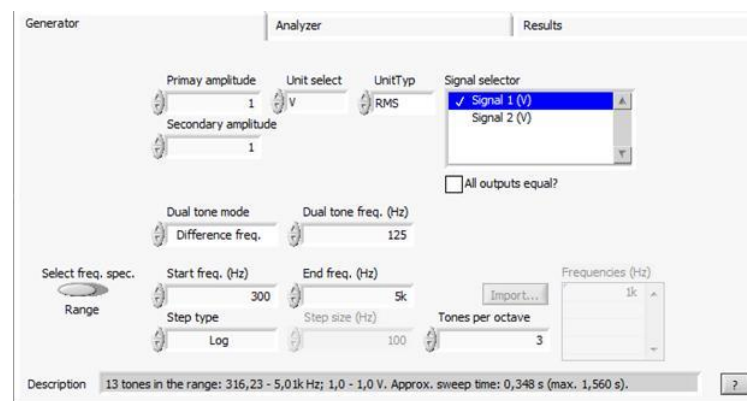


Figure 23: Dual tone generator configuration panel

Control	Unit	Range	Description
Primary amplitude	Unit*	0.. <i>n</i>	Specifies the primary amplitude to use.
Secondary amplitude	Unit*	0.. <i>n</i>	Specifies the secondary amplitude to use.
Dual tone mode	—	Difference Excitation	Select how the secondary tone relates to the primary frequency: Difference freq. – a frequency offset from the primary tone. Excitation freq. – a fixed frequency that does not change with the primary tone.
Dual tone freq.	Hz	0.. <i>n</i>	The frequency value for the secondary tone.
Select freq. spec.	—	Range Table	Specifies whether to define a frequency range or a table of frequencies.

Control	Unit	Range	Description
Start freq.	Hz	0.. n	First frequency of the primary tone in the sequence of tones.
End freq.	Hz	0.. n	Last frequency of the primary tone in the sequence of tones.
Step type	—	Log Lin	Specifies whether the frequency distribution is linear or logarithmic.
Step size	Hz	—	Specifies the linear frequency step in Hz.
Tones per octave	—	0.. n	Specifies the logarithmic frequency distribution as tones per octave.
Import	—	—	Imports a list of frequencies from a file that you specify. The file must define the frequencies using a decimal point format.
Frequencies	Hz	—	Contains a list of frequencies to generate. You can specify frequencies in the table manually or import frequencies from a file using the Import control.
* You can specify the unit of measurement on each channel for this control.			

6.5.3 Analyzer Settings

These controls set the test signal duration and analysis properties.

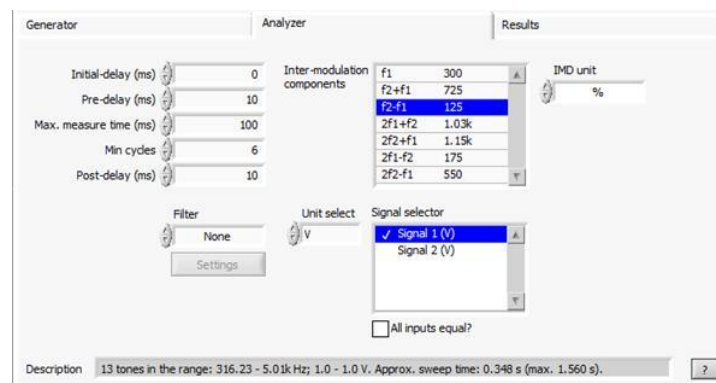


Figure 24: Dual tone analyzer configuration panel

NOTE: When working with dual tone signals it is the minimum frequency component that rules the number of cycles and measurement time.

Control	Unit	Range	Description
Initial-delay	ms	0.. n	The amount of initial signal before the first tone segment.
Pre-delay	ms	0.. n	Each tone segment is prolonged a pre-delay time before the actual measurement time.
Max. measure time	ms	0.. n	Sets a maximum time to limit the measurement time.

Control	Unit	Range	Description
Min. cycles	—	1.. n	Sets the minimum number of tone cycles in each measurement segment.
Post delay	ms	0.. n	Each tone segment is prolonged a post – delay time after the actual measurement time.
Inter-modulation parameters	—	—	Select the inter-modulation products that are used for calculating the Inter-Modulation Distortion (IMD).
IMD unit	—	None % dB	Selects unit for the IMD value.
Filter	—	None Butterw. A-filter ITU-R 468	Selects a digital software filter on the input signal. This can be used to filter out noise outside the dual-tone range.

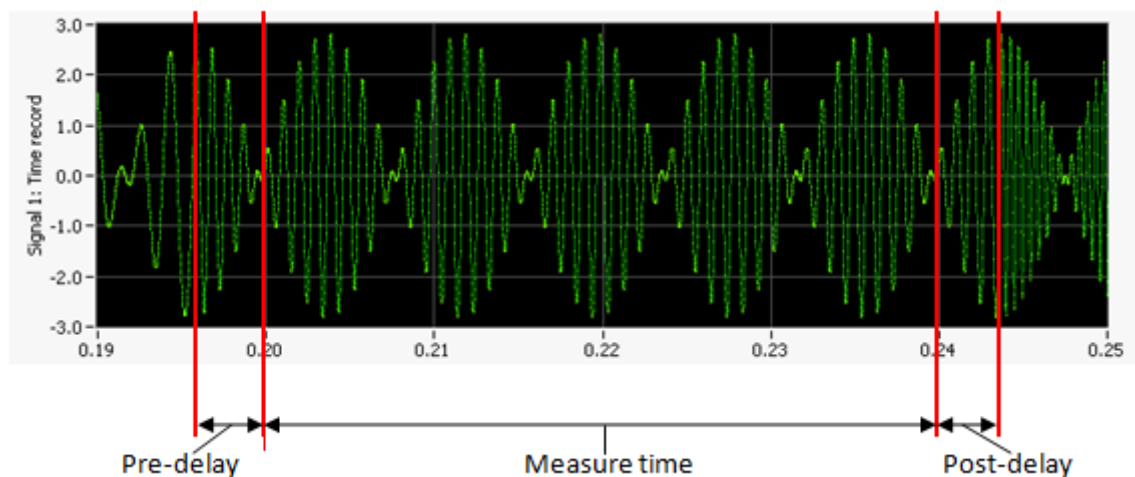


Figure 25: Dual tone signal timing

6.5.4 Result Settings

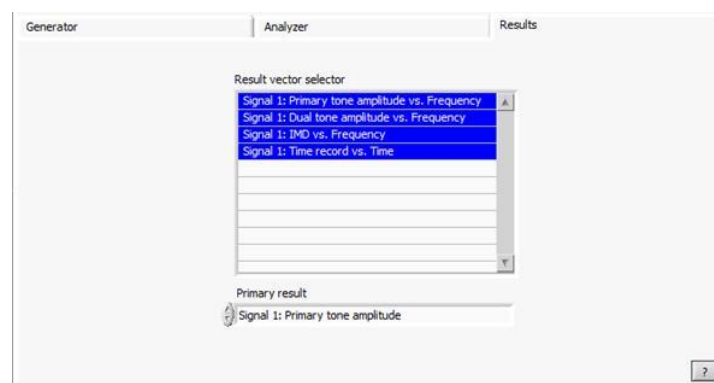


Figure 26: Dual tone result configuration panel

Index	Analysis	Unit	Description
0	Primary y-values vs. Primary x-values	As primary y	Copy of the result vector you chose as the primary result.
1	Primary tone amplitude vs. Frequency	Unit*	Measured amplitude of the primary tone alone.
2	Dual tone amplitude vs. Frequency	Unit*	Measured amplitude of the secondary tone alone.
3	IMD vs. Frequency	IMD unit	Inter-Modulation Distortion value of each segment, based on the selected IMD components.
4	Time record vs. Time	Unit*†	Time record of the input signal.
<p>* You can specify the unit of measurement on each channel for this control.</p> <p>† If the output unit is dBspl the result unit is changed to Pa.</p>			

NOTE: Amplitude measurements requires that the input signal is coherent with output signal because the value is extracted from a single bin in the spectrum.

6.5.4.1 IMD Calculation Method

The inter-modulation distortion values are calculated according to IEC 118-0 section 7.12.2, where the IMD is expressed as the level of the selected distortion product referred to the level of the dual tone.

For compliance with IEC-118 the user must configure appropriate signal parameters and select the desired IMD component.

$$IMD = \sqrt{\frac{(First\ selected\ IMD\ component)^2}{(Dual\ tone\ component)^2}}$$

6.6 Multiple Pure Tone plug-in

This plug-in can generate and analyze a sequence of pure tones of equal length, with linear or logarithmic spacing, or a sequence of pure tones from a list of specific frequencies.

It generates an output signal on one or more channels and analyzes input on one or more channels. Due to timing constraints signal timing applies to all signals and only the output amplitudes can be specified on each output signal.

6.6.1 General Description

The combined signal of multiple pure tones is a train of blocks. Each block contains a pure tone signal of length given by the Measure time.

Each block is extended with a pre-delay and post-delay section to give a buffer against time delays between input and output channels. The delays can also be used to introduce settling time for a device under test.

The MPT generator envelopes each block so that the amplitude rises from zero to the specified amplitude over a specified block length. Likewise, the signal tapers from full amplitude to zero at the end of the block. The envelope follows a cosine window.

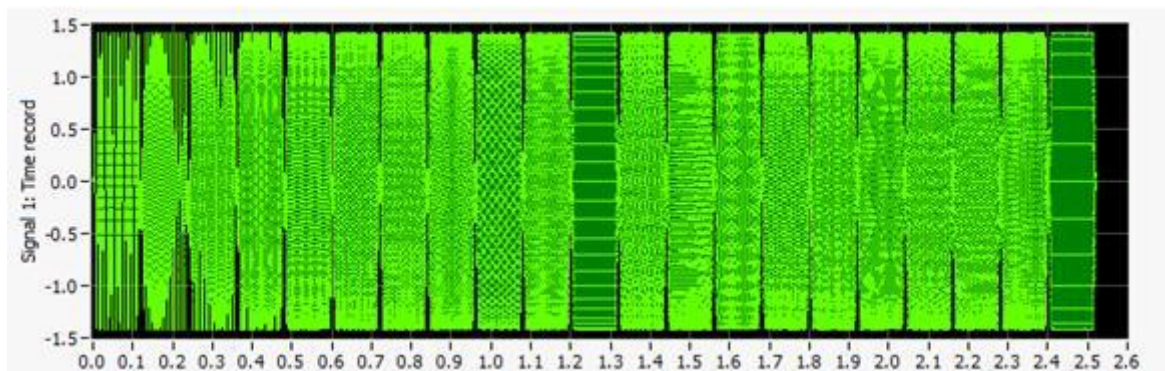


Figure 27: Multiple pure tone test signal

The pre- and post-delay segments should be longer than the tapering, to avoid that the rise- and fall-sections become included in the analysis.

6.6.2 Generator Settings

The frequency parameters specify the frequency range and step resolution to determine the number of tone segments. Frequency step distribution type can be either logarithmic or linear.

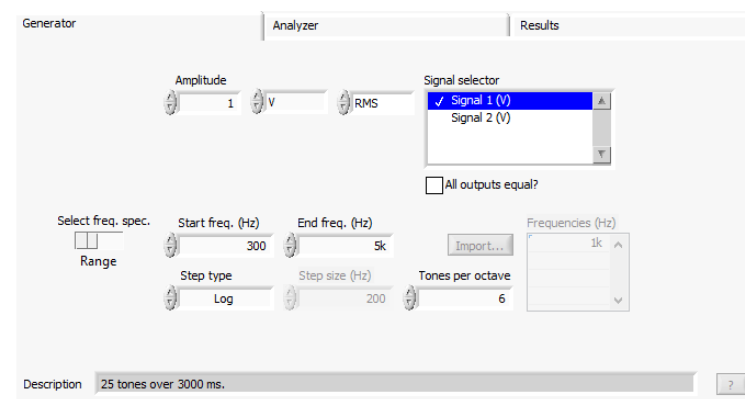


Figure 28: Multiple pure tone generator configuration panel

By setting a start freq. that is higher than the end freq. the plug-in will create a decreasing sweep signal.

Control	Unit	Range	Description
Amplitude	Unit*	0 - +inf	The output amplitude in the selected unit on all selected channels.
Select freq. spec.	—	Range Table	Specifies whether to define a frequency range or a table of frequencies.
Start freq	Hz	0 - +inf	The start frequency for calculating discrete frequencies.
End freq	Hz	0 - +inf	The end frequency for calculating discrete frequencies.
Step type	—	Lin Log	Specifies whether the frequency distribution is linear or logarithmic.

Control	Unit	Range	Description
Step size	Hz	0 - +inf	Specifies the step size for linear frequency step intervals.
Tones per octave	—	0 - +inf	Specifies the number of tones per octave for logarithmic frequency step intervals.
Import	—	—	Imports a list of frequencies from a file that you specify. The file must define the frequencies using a decimal point format.
Frequencies	Hz	—	Contains a list of frequencies to generate. You can specify frequencies in the table manually or import frequencies from a file using the Import control.
* You can specify the unit of measurement on each channel for this control.			

6.6.3 Analyzer Settings

Together with frequency steps specified on the Generator tab, the timing parameters for each segment determines the complete test signal and acquisition.

Specify the measure time for each frequency step in the Measure time control. The total tone segment includes the pre- and post delay that is by default controlled automatically by the Tapering time. To assure that the tapering does not affect the measurement, the delay values are set to be 1.5 times the tapering value.

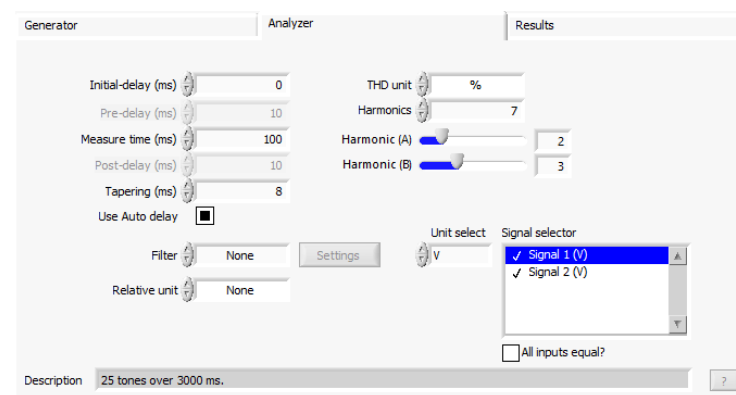


Figure 29: Multiple pure tone analyzer configuration panel

To manually specify pre- and post-delay values, disable Use Auto delay. When a delay value is set shorter than 1.5 times the tapering value it is colored yellow to warn about including the tapering into the measurement.

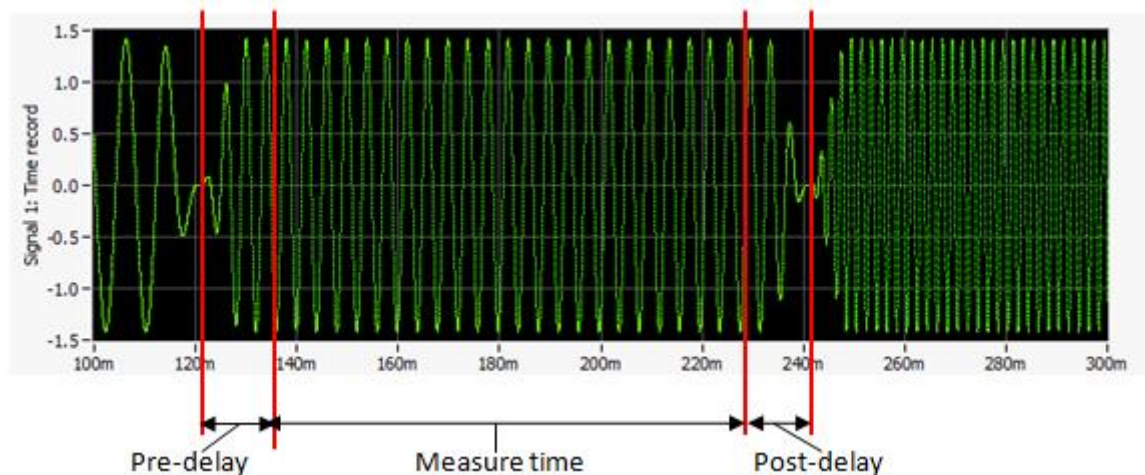


Figure 30: Time record of a Multiple Pure Tone segment

Use the unit selector to select a unit for the selected input signal.

Control	Unit	Range	Description
Initial-delay	ms	0 - +inf	The amount of initial signal before the first amplitude step.
Pre-delay	ms	0 - +inf	A pre-delay time before the actual Measure time , within each frequency block. Should at least be equal to Tapering .
Measure time	ms	0 - +inf	The actual measure time within each frequency block that will be analyzed
Post-delay	ms	0 - +inf	A post-delay time after the actual Measure time , within each frequency block. Should at least be equal to Tapering .
Tapering	ms	0 - +inf	Set the amount of tapering at each end of a tone segment.
Use Auto delay	—	On / Off	Select this option to set pre- and post delays automatically.
THD unit	—	None % dB	Select the unit for THD and THD+noise measurements. None returns the unit free ratio of fundamental tone to harmonics and harmonics and noise respectively.
Harmonics	—	1 - + inf	Specifies the number of harmonics to include in the THD measurements.
Harmonic (A)	—	2 - (Harmonics - 1)	Specifies the selected harmonic to use for nth harmonic calculations.
Harmonic (B)	—	Harmonic (A) - Harmonics	Specifies the selected harmonic to use for nth harmonic calculations.

Control	Unit	Range	Description
Filter	—	None Butterw. A-filter ITU-R 468	Selects a digital software filter on the input signal. This can be used to filter out noise outside the multi-tone range.
Relative unit	—	None % dB	Select the unit for relative amplitude measurements.

Input Filtering

Also, to improve measurements the plug-in provides a filter that is applied to the entire acquired signal, before extracting the FFT block. This allows the Initial delay to serve as settling time for the filter. The following filter functions are available:

- None
- Butterworth
- A-filter
- ITU-R 468

When using the Butterworth filter its default settings are: 6'th order High Pass Filter with lower cut-off at 125 Hz. Filter settings can be changed using the 'Settings' button.

6.6.4 Result Settings

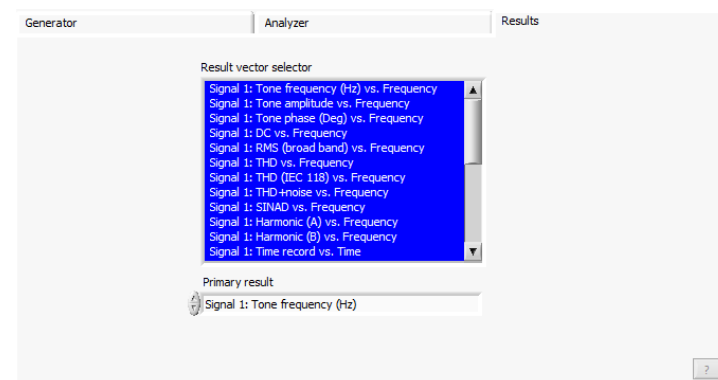


Figure 31: Multiple Pure Tone results configuration panel

The following analyzes are returned as vector results. The vector at index 0 is selected by the Primary result control.

Index	Analysis	Unit	Description
0	Primary y values vs. primary x values	As primary y	Copy of the result vector you chose as the primary result.
1	Tone frequency (Hz) vs. output frequency	Hz	Measured frequency of each tone block.
2	Tone amplitude vs. output frequency	Unit*	Measured narrow-band amplitude (FFT) of each tone block.
3	Tone phase (Deg) vs. output frequency	Deg.	The tone phase of each tone block.

Index	Analysis	Unit	Description
4	DC vs. Output frequency	Unit*	DC estimate for each tone block.
5	RMS (broad band) vs. Output frequency	Unit*	Broadband RMS of each tone block.
6	THD vs. output frequency	THD unit	THD of each frequency block (including highest harmonic).
7	THD (IEC 118) vs. Output frequency	THD Unit	THD value after IEC method and frequency of highest detected tone in the measured signal.
8	THD + noise vs. Output frequency	THD unit	THD + noise for each frequency block.
9	SINAD vs. Output frequency	THD unit	Measured SINAD value of each tone segment. This is reciprocal of THD+noise.
10	Harmonic (A) vs. Output frequency	Unit*	Measured selected harmonic and frequency of the highest detected tone in the measured signal.
11	Harmonic (B) vs. Output frequency	Unit*	Measured selected harmonic and frequency of the highest detected tone in the measured signal.
12	Time record vs. Time	Unit*†	The acquired raw data scaled to units and calibration.
13	Amplitude ratio (Signal N / Signal N+1) vs. Output frequency	Relative unit	Relative tone amplitudes (vector 2) of two input channels for each tone segment.
14	Phase difference vs. Output frequency	Deg.	Phase difference between two input channels for each tone block.
<p>* You can specify the unit of measurement on each channel for this control.</p> <p>† If the output unit is dB SPL the result unit is changed to Pa.</p>			

6.6.5 Special Usage Hints

This section lists some useful hints and caveats to the use of this plug-in.

- Frequencies are generated to exactly the specified value, without rounding for coherence.
- Output level can be calibrated with frequency correction.
- High power output at each frequency, because all of the signal energy is concentrated at a single frequency.
- All frequencies (low and high) are allocated the same block length.

- Relatively long test signal because each frequency is represented as a block in the combined signal.
- Phase difference requires that the two input channels are sample simultaneously.

6.7 Multi-tone plug-in

This plug-in can generate and analyze a coherent multi-tone burst. The frequency resolution and hence the measurement time can be specified.

It generates an output signal on one or more channels and analyzes input on one or more channels. Due to timing constraints signal timing applies to all signals and only the output amplitudes can be specified on each output signal.

6.7.1 General Description

A multi-tone signal consists of a number of super-imposed sine waveforms. Each waveform has an integer number of cycles, which results in a coherent signal.

The following figure illustrates a signal in the frequency domain containing 25 tones, each sharply defined in the spectrum. Each tone in the signal has an amplitude of 0.1 volts RMS.

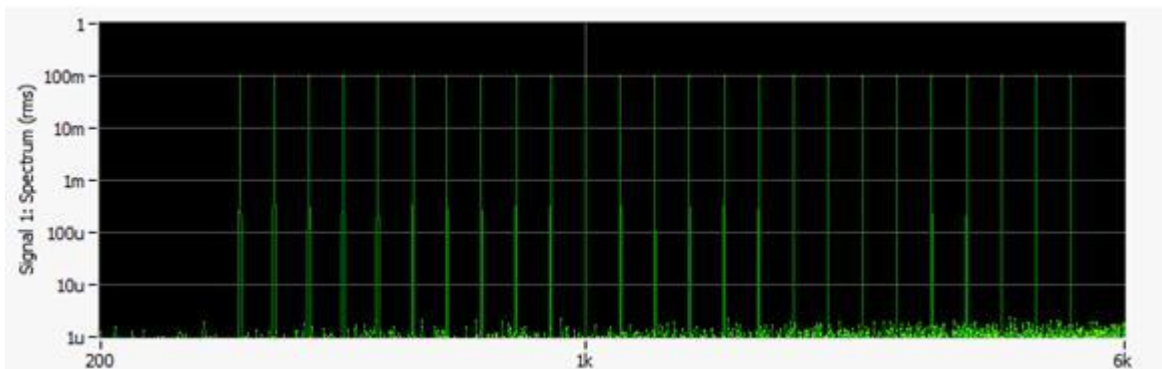


Figure 32: FFT spectrum of a multi-tone signal

Creating all waveforms with frequencies that are a multiple of a common factor can ensure an integer number of cycles. This factor is the frequency resolution.

The frequency resolution also determines the length of the multi-tone signal, because the shortest signal for a given resolution is its reciprocal.

For example, if the frequency resolution is 2 Hz, the shortest signal can be $1/2 \text{ Hz} = 0.5$ seconds, which can contain exactly one 2 Hz cycle, and any number of waveforms that have frequencies that are multiples of 2.

The amplitude of the multi-tone signal is higher than the amplitude of each waveform because they are super-imposed.

The time signal in the following figure consists of 25 tones, each with an amplitude of 0.1 volts, but the amplitude of the multi-tone signal is more than 10 times that.

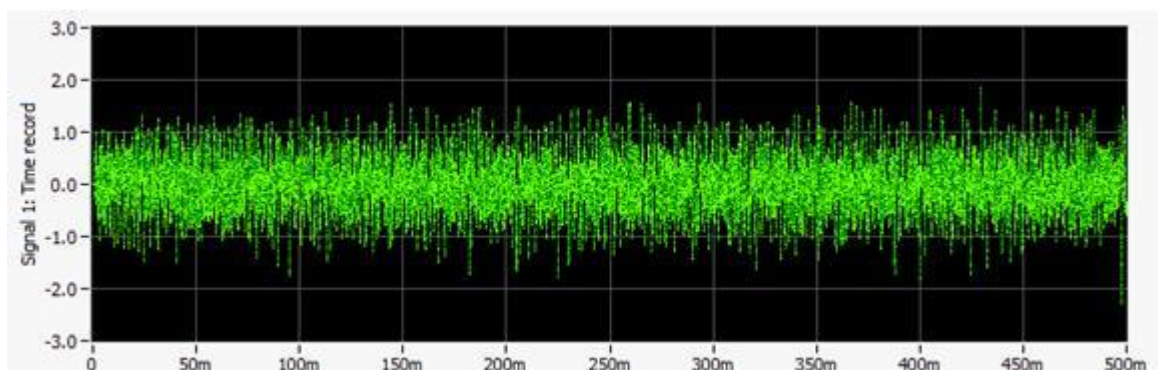


Figure 33: Multi-tone signal containing 25 tones, each with an amplitude of 0.1 Volts and linear phase shift between tones

6.7.2 Generator Settings

The generator can generate the signal specified in the Generator tab on the selected output channels.

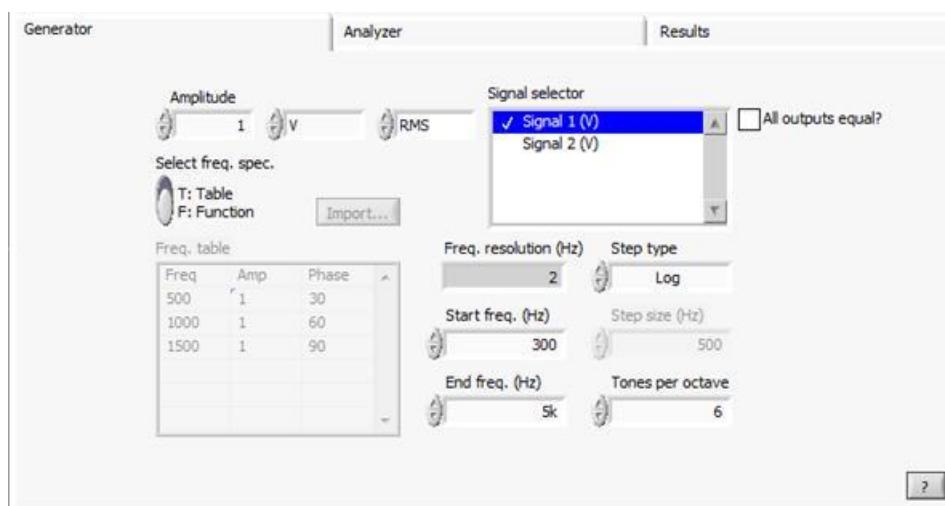


Figure 34: Generator controls

The amplitude for each tone is specified in units chosen in the selector (V, A, dB SPL, A/m, Pa) in RMS, Peak, or Peak to Peak levels.

Control	Unit	Range	Description
Start freq	Hz	0 - +inf	The start frequency for calculating discrete frequencies.
End freq	Hz	0 - +inf	The end frequency for calculating discrete frequencies
Step type	—	Lin / Log	Linear or logarithmic frequency step intervals
Step size	Hz	0 - +inf	The step size for linear frequency step intervals
Tones per octave	—	0 - +inf	The number of tones per octave for logarithmic frequency step intervals

Control	Unit	Range	Description
Table / Function	—	Table / Function	Switch for selecting calculated frequencies (Function) or frequencies taken from a table (Table)

Freq. resolution has been changed to an indicator that shows the resolution of the FFT spectrum that forms the basis of the measurements. The frequency resolution is determined by the measurement time on the analyzer tab and the sample rate.

A linear phase shift is applied between calculated frequencies.

6.7.2.1 Controls for Frequency Table

The multi-tone signal frequencies can be specified by parameters or listed in a table.

If you specify frequencies from a table, the frequencies are still rounded to the nearest multiple of the value in the Freq resolution control.

As illustrated in the following figure, the table contains three columns: Frequency in Hz, Amplitude in units specified under the Generator, and Phase in degrees. You can specify different values in the frequency table for each signal.

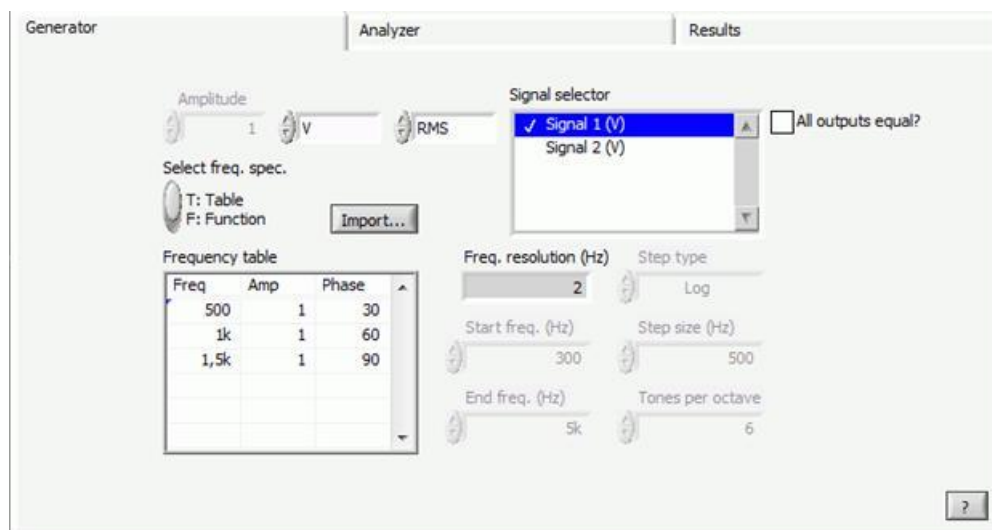


Figure 35: Signal description controls for frequency selections from a table

Click **Import** to populate the table with data from a file. The file must be tab-delimited with three columns and must use decimal points. To clear the table, click **Empty table**.

6.7.3 Analyzer Settings

The analyzer settings include parameters to control measurement timing and result values.

Measure time specifies the measurement time and consequently the FFT block size.

Priming delay expands the test signal to allow for some settling and delays in the signal path. This means that the complete test signal will be the length of the multi-tone signal ($1/\text{freq. resolution}$) plus the priming delay.

Measure delay starts the analysis after the acquisition to avoid settling effects. The Measure delay should be shorter than the priming delay to avoid analyzing after the multi-tone ends.

Filter selects a digital software filter on the input signal. This can be used to filter out noise outside the multi-tone range.

FFT Window selects a time-domain windowing function to apply to the time signal in the measurement time segment.

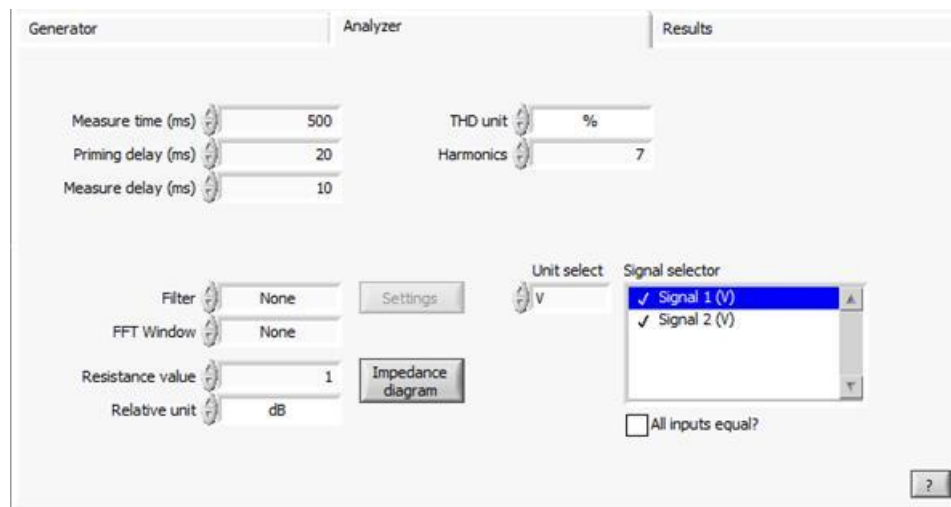


Figure 36: Analyzer control

Relative unit selects the unit for the amplitude ratio measurement. Please refer to the following section for information on setting up an impedance measurement.

Use the unit selector to select individual unit for the selected input signal.

6.7.3.1 Measuring Impedance

The impedance diagram is displayed when the button Impedance diagram is clicked. This shows a basic wiring diagram for measuring impedance z using two input channels and one output channel.

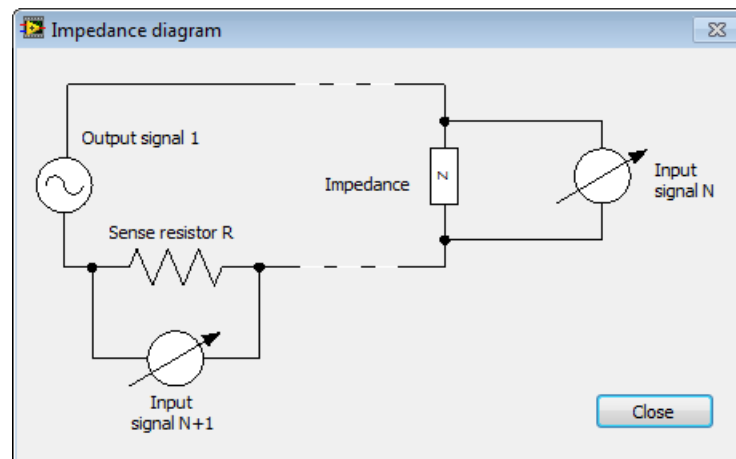


Figure 37: Basic wiring diagram for measuring impedance z

The value of the sense resistor in ohms can be entered into the control "Resistance value" and select 'None' for relative unit, to give the correct units in the impedance result (index 6).

If a phase difference analysis is required and the phase shows discontinuity around ± 180 degrees, you may need to swap the terminals on one of the input channels.

6.7.4 Result Settings

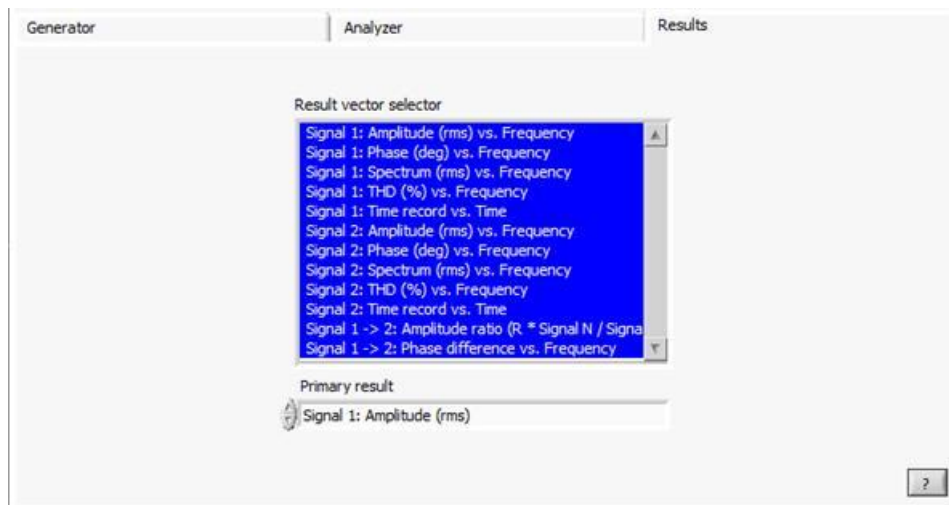


Figure 38: Multi-tone results configuration

The following analyzes are returned as vector results. The vector at index 0 is selected by the *Primary result* control.

Index	Analysis	Unit	Description
0	Primary y values vs. primary x values	As primary y	Copy of the result vector you chose as the primary result.
1	Amplitude (rms) vs. frequency	As output	The amplitude in the spectrum at each frequency (narrow-band).
2	Phase	Deg	The phase of each frequency.
3	Spectrum (rms) vs. frequency	Unit*	The full amplitude spectrum.
4	THD vs. frequency	THD unit	THD of each frequency (including highest harmonic). Care must be taken in choosing frequencies to avoid interference between harmonics and fundamentals.
5	Time record vs. time	Unit*†	The acquired raw data scaled to units and calibration
6	Amplitude ratio ($R \cdot \text{Signal } N / \text{Signal } N+1$) vs. frequency.	Relative unit	Relative tone amplitudes (vector 1) of two input channels for each tone segment.
7	Phase difference between Signal N and Signal N+1 vs. frequency	Degrees	The phase difference between Signal N and Signal N+1.

* You can specify the unit of measurement on each channel for this control.

† If the output unit is dBSPL the result unit is changed to Pa.

6.7.4.1 Special Usage Hints

This section lists some useful hints and caveats to the use of this plug-in.

- Short signal and fast analysis
- The resulting amplitude of the combined signal limits the total number of tones. More tone gives higher crest factor.
- Impedance measurement requires two input channels.
- Phase difference requires that the two input channels are sampled simultaneously.
- For THD measurement, the frequencies must be chosen carefully to prevent that harmonic components interfere.
- Output level can be calibrated with frequency correction.
- Frequencies are rounded to multiples of the frequency resolution.

6.8 Polarity plug-in

Polarity measurements are typically done on loudspeaker units to check that it is wired correctly. When correctly wired the speaker unit or driver should produce a positive sound pressure when a positive voltage is applied to the positive terminal.

Channel Configuration

The Polarity plug-in generates the output signal on a single output channel and acquires the response on a single input channel.

Single channel operation is most suitable because each speaker unit must be measured individually to prevent one speaker from disturbing the other.

The plug-in does not require absolute signal level values to detect the polarity. Calibration can therefore be omitted.

6.8.1 New Version Compatibility

The current version of the plug-in operates in a different way than the previous version.

To improve measurement stability the type of excitation signal has been changed and a new detection method has been implemented.

One significant difference is that this version repeats a number of test pulses at the same frequency and uses both filtering and time domain averaging to minimize effect of background noises.

Backward Compatibility

In order to support test sequences build with the previous versions of the plug-in, this version will use the method of the previous version when it receives an older version of the setup parameters.

NOTE: This operation is only supported temporarily, and it is strongly recommended to switch to the new method.

6.8.2 Generator Settings

The generators settings are used specify series of identical pulses. The amplitude and unit values specify the amplitude as is it was a continuous sine wave, e.g. amplitude of 100 mV_{rms} will produce a pulse with a peak value of 141 mV. The true rms value of the pulse train is less than the specified value because of the relatively long delay between the individual pulses.

The switch 'Using original setup' indicates if the plug-in is running in the backward compatibility mode (on position). In this case the control is enabled and can be switched to the new method.

When the switch is in the off position the plug-in uses the new method and the switch is disabled, and it is not possible revert to using the old method.

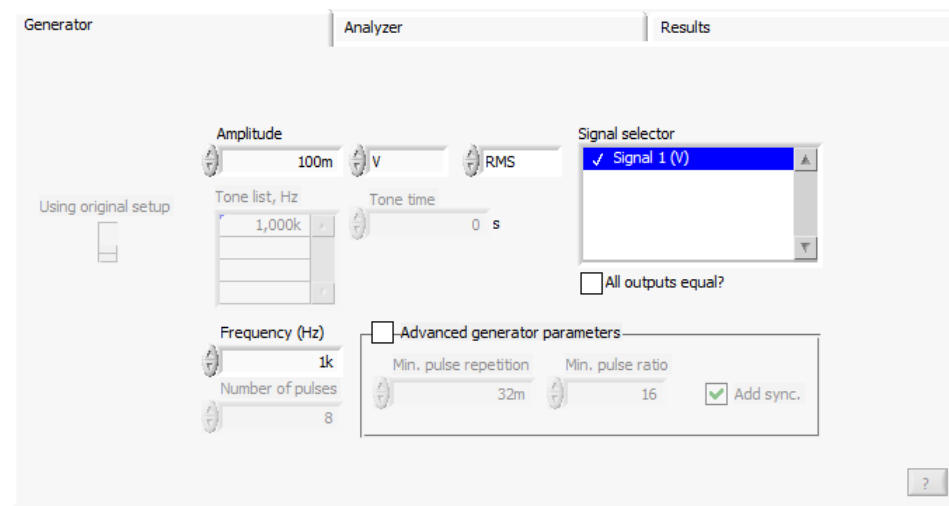


Figure 39: Generator settings for the Polarity plug-in

Also, the center frequency of the pulse is specified for the continuous sine wave, but due to the inherent shaping of the pulse the actual frequency is slightly higher.

The 'Number of pulses' value is disabled because it is calculated automatically from the value specified on the analyzer tab. Only when the wave device mode is set to play only this control is enabled.

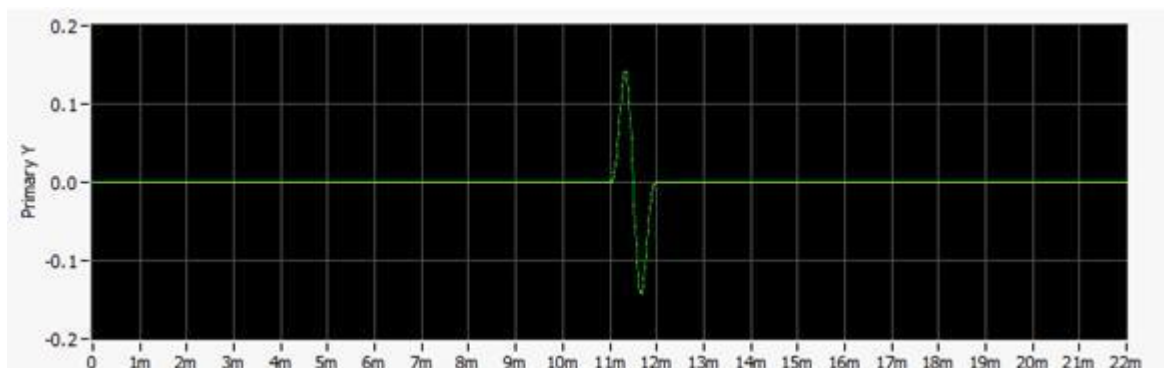


Figure 40: Single pulse (part of the test signal output signal)

Advanced generator parameters include values to adjust the timing of the pulse train. 'Min. pulse repetition' specifies the minimum time in seconds between pulses, while the 'Min pulse ratio' specifies the minimum pulse cycles between pulses.

When the 'Add sync.' Check box is selected, a synchronization tone is appended to the pulse train in order to improve pulse detection even in systems with long delays, e.g. Bluetooth.

Note that the advanced parameters are in effect, even when the controls are greyed out. You can enable the controls by selecting the 'Advanced generator parameters' check box.

With the frequency set to 1 kHz the cycle time is 1 ms. The actual pulse repetition time is then determined by maximum of 32 ms repetition time and 16 ms (16 x 1 ms) pulse ratio.

6.8.3 Analyzer Settings

The settings on this tab specify the conditions for the measurement.

'Initial delay' and 'Post delay' defines the duration of a zero signal at the beginning and end of the pulse train respectively. 'Final delay' specifies the duration of a zero-signal following the sync tone.

'Number of pulses' specifies how many pulses to include in the averaging before detecting the polarity. The acquired pulses are averaged in the time domain to minimize the effect of background noises.

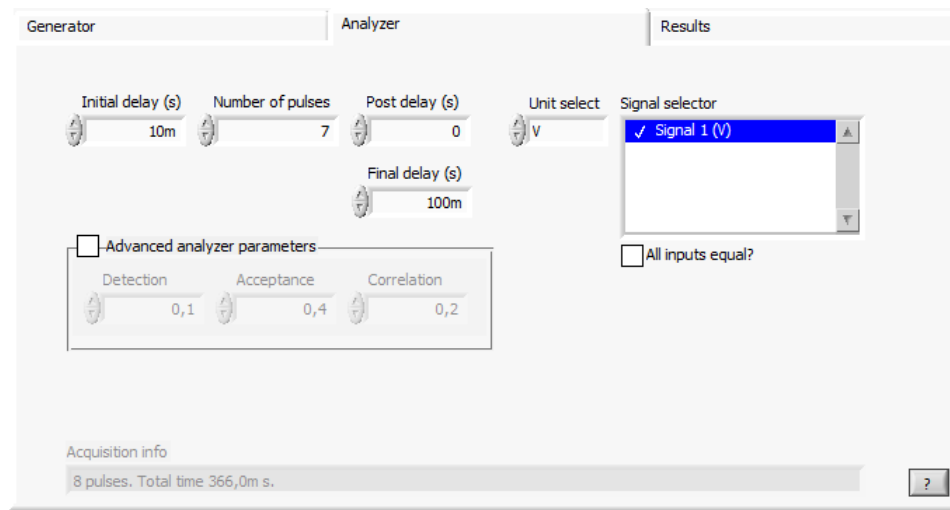


Figure 41: Analyzer settings for the Polarity plug-in

Typically, the measurement is done in the 'Play meas.' mode where the generated pulse train is automatically created with one pulse more than requested for the measurement. Analysis will automatically skip the first pulse that may be affected by settling effects in the speaker under test. Using only pulses in the settled part of the pulse train will improve the averaging and provide more stable results.

The 'Advanced analyzer parameters' include values that control the detection of the first deflection in the averaged pulse. Note that the advanced parameters are in effect, even when the controls are greyed out. You can enable the controls by selecting the 'Advanced analyzer parameters' check box.

6.8.4 Results Settings

The result list allows selecting which vector to return for the test sequencer. The main result vector is either the 'Polarity (f)' or 'Polarity (i)' that returns the same y-value but includes the actual pulse frequency or index value for x-value, respectively.

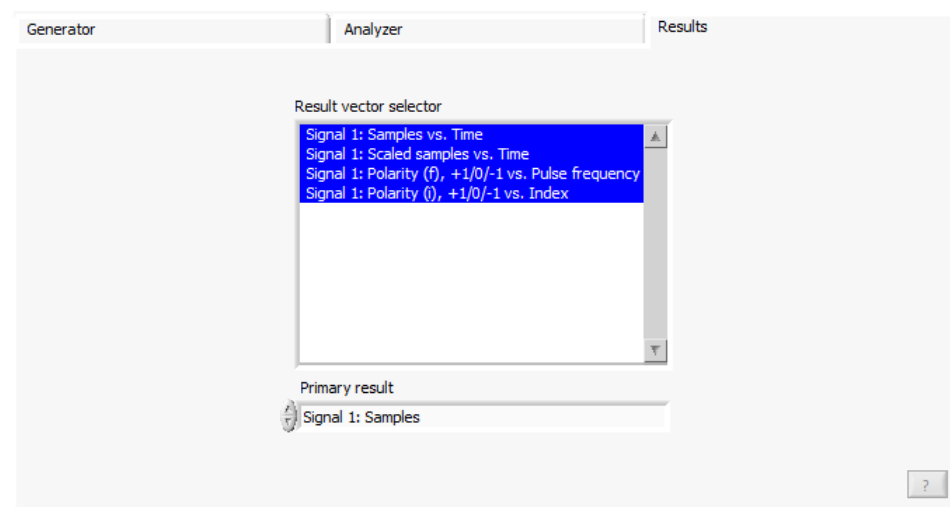


Figure 42: Polarity plug-in result selection

Other vectors are provided primarily for reviewing the test conditions during test program development.

Depending on the type of loudspeaker the averaged of the acquired response could look like this:

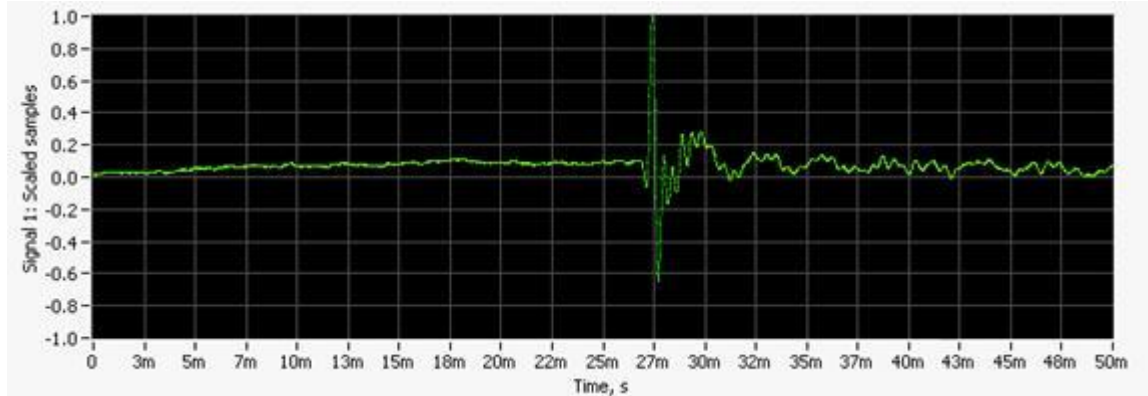


Figure 43: Acquired input signal

Because loudspeakers differ in mechanical and electrical properties it is necessary to select test tone frequencies that are reasonable transmitted by the speaker. As a rule of thumb test frequencies from one octave above resonance and up are recommended.

Please refer to Selecting Test Frequencies for details about selecting test frequencies.

Whenever the generator settings are changed the acquisition info line is updates to show total signal length.

The plug-in returns four vectors, as described in the table. Any of the results can be selected as the primary vector for easy evaluation in the test sequencer.

Vector	Description
1: Samples	The acquired time record samples in the selected input unit.
2: Scaled samples	Time record where each pulse block is averaged and scaled for the numeric peak to be within +/-1.
3: Polarity (f), +1/0/-1	Detected polarity value of the scaled pulse. The value is +1 for pos., -1 for neg. or 0 for uncertain. The polarity values are plotted at the actual pulse frequency on the x-axis.
4: Polarity (i), +1/0/-1	Detected polarity value of the scaled pulse. plotted at 1 on the x-axis. The value is +1 for pos., -1 for neg. or 0 for uncertain.

Normally the 'Polarity (i)' vector is selected as primary vector because it gives an averaged detection over all of the pulses and a simple value that can easily be evaluated with a point limit or key parameter in the test sequencer.

Selecting the 'Return primary only' allows the user to skip returning unused vectors to the test sequencer. This option makes it easier to set limits and view or log results in the sequencer.

6.8.4.1 Polarity (f) vectors

This vector contains only one point that is the polarity value of the averaged pulses.

- Positive => +1
- Uncertain => 0
- Negative => -1

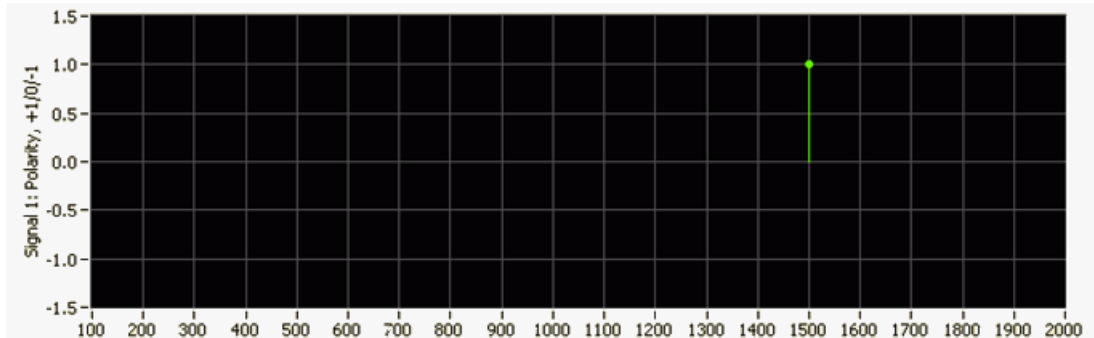


Figure 44: Polarity (f) vector

The value is plotted at the actual frequencies of the test pulse. This frequency is typically higher than the specified pulse frequency due to the effects of the internal pulse shaping.

6.8.4.2 Polarity (i) vectors

This vector contains only one point that is the polarity value of the averaged pulses.

- Positive => +1
- Uncertain => 0
- Negative => -1

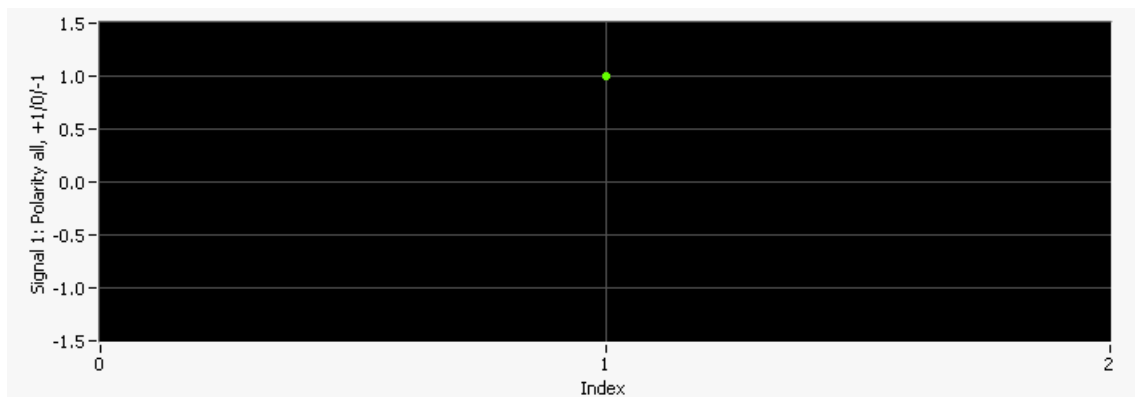


Figure 45: Polarity (i) vector

The x-value is always 1, making it easy to evaluate with a point limit or key parameter.

6.8.5 Application Hints

This section includes some practical hints for using the plug-in for typical applications.

6.8.5.1 Polarity Measurement System

Before starting real measurements, it is important to know the polarity of the test system itself.

In most cases the polarity of the electrical signal path is positive – producing a positive output voltage when a positive input voltage is applied. If the output path is inverting the generated signal amplitude can be set to a negative value (except when using dBspl unit) to compensate.

For acoustic measurements however the type of microphone determines the polarity of the signal path.

Pre-polarized microphones typically create a positive output voltage for a positive pressure. This is because the build-in electrets creates a negative polarization voltage.

Conversely for a microphone using a positive external polarization voltage, a positive pressure creates a negative output voltage and vice versa.

6.8.5.2 Selecting Test Frequencies

To measure the polarity of a speaker correctly the test frequency must be set to match the speaker's mechanical properties.

When developing the test program for a speaker polarity test it is therefore recommended to first test the speaker manually over a wider range of frequencies and then select the best frequency for the actual automated test.

As an example an electro-dynamic speaker impedance and phase curve looks like this: a resonance, a resistive region and an inductive region.

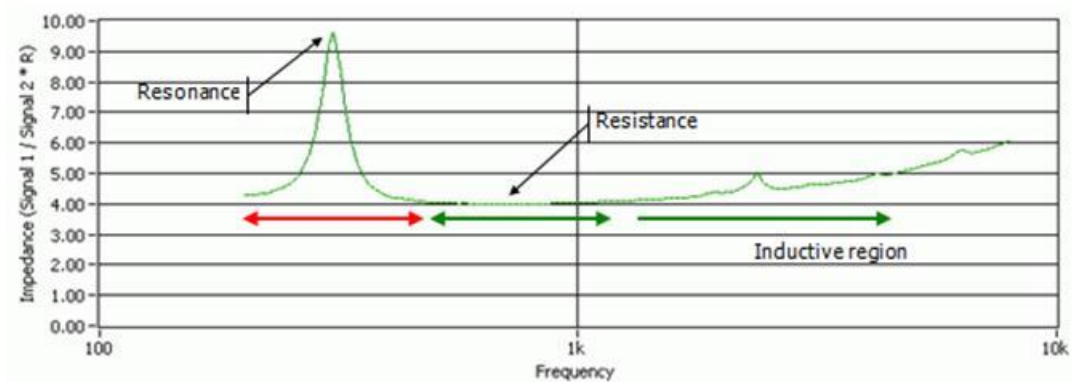


Figure 46: Typical speaker impedance magnitude graph

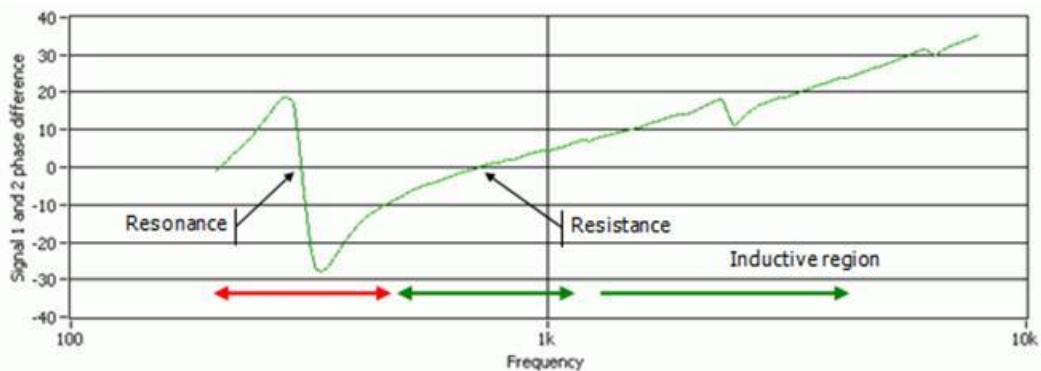


Figure 47: Typical speaker impedance phase graph

In the above example the resonance is at 300 Hz and the resistive region is centered at 730 Hz. These graphs are included here for reference - different types of speakers will have different impedance graphs.

Normally it is preferable to measure polarity above from the resonance frequency where the speaker output reproduces the input better.

The impedance can be measured with the Multi-tone plug-in using a sense resistor as shown on the diagram on its configuration panel.

However doing an impedance measurement is optional, and in many cases it is sufficient to look at the scaled samples to determine the suitable test frequencies.

The response of the speaker can be examined manually by trying out a few test frequencies and use the "Execute setup" button to do the measurement. Select the "Scaled samples" for result preview:

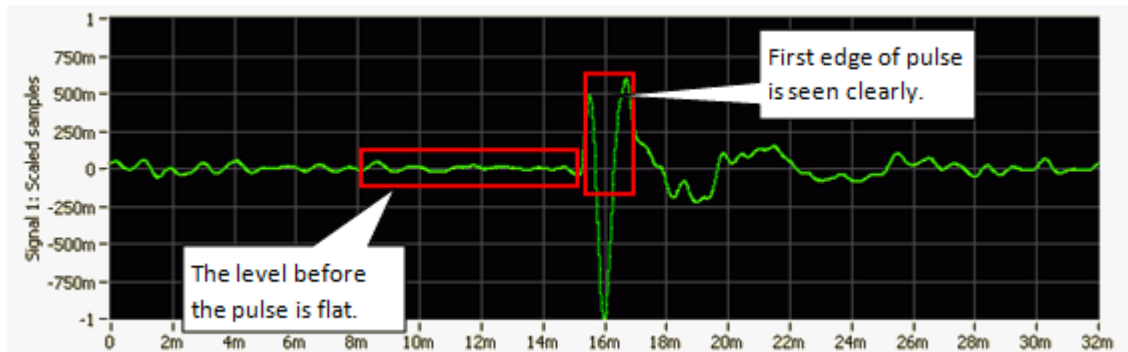


Figure 48: Identifying a good test frequency

The first edge of the pulse must be seen clearly relative to the preceding level. The polarity detection algorithm will accept the pulse when the absolute numeric value is above 0.4 (Acceptance limit) and the level in front of the pulse is flat within 0.1 (Detection limit).

If the level before the pulse is not flat it may be an indication that reflections of one pulse has not died out before the next pulse starts. Increasing the delay between pulses or minimizing reflections by adding damping material to nearby surfaces can solve this.

6.9 Single-tone plug-in

This plug-in can generate a single tone or output a waveform file. The analysis provides a number of single tone parameters based on a single acquired waveform segment.

6.9.1 General Description

The main purpose of this plug-in is to provide users with a simple way to set up common audio measurements that requires only a single measurement at a fixed frequency and level.

The Single tone generator envelopes the signal block so that the amplitude rises from zero to the specified amplitude over a specified block length. Likewise, the signal tapers from full amplitude to zero at the end of the block. The envelope follows a cosine window.

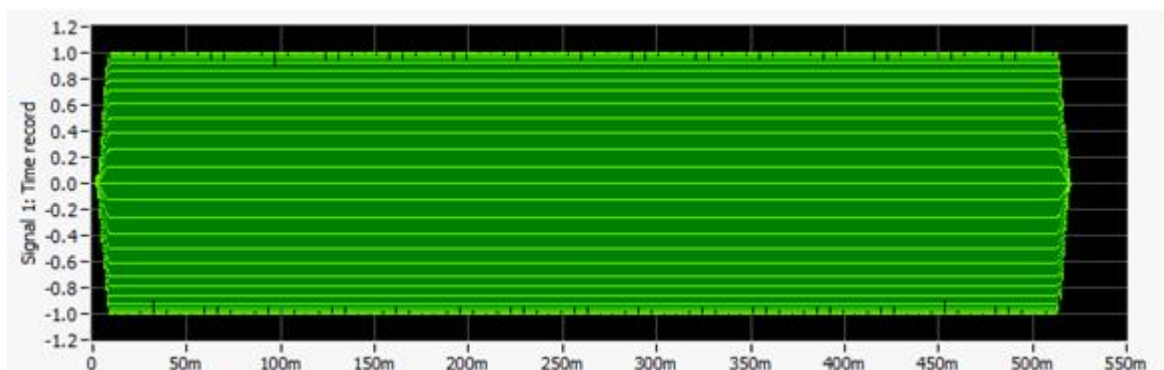


Figure 49: Single tone generator time signal

On the measurement specification tab the measurement time and delays can be specified. Using these parameters the test signal and measurement timing can be adjusted to suit internal delays and settling times in the device under test.

The Single tone plug-in also allows generating output from a standard .wav file.

6.9.2 Generator Settings

The generator supports two modes of operation:

- **Analog mode** – can generate one or more channels of waveform output (PCM) or output a standard wav-file with one or more channels.
- **Digital mode** – besides generating PCM and wav-file as in the analog mode an AC3 encoded audio file can be selected from a library of test files that covers a number of sample rates, bit-rates and channel layouts. This mode also enables the option to set status bits for the SPDIF data stream.

NOTE: The AC3 encoded audio is no longer supported.

Where the PCM signal generation is identical for the two modes, the real difference is when it comes to encoded audio file output that is supported by the digital mode only. In the following sections the generator settings for each mode is detailed.

6.9.2.1 Analog and Digital Mode PCM Generator

In analog mode the frequency and amplitude of the sine waveform test signal is specified. Note that the default unit is set V_{peak} rather than V_{rms} that is typical for most other plug-ins. The reason is that the focus of this plug-in is the digital audio test where the signal level is often set with reference to the full scale which CATS defines as 1 V_{peak} .

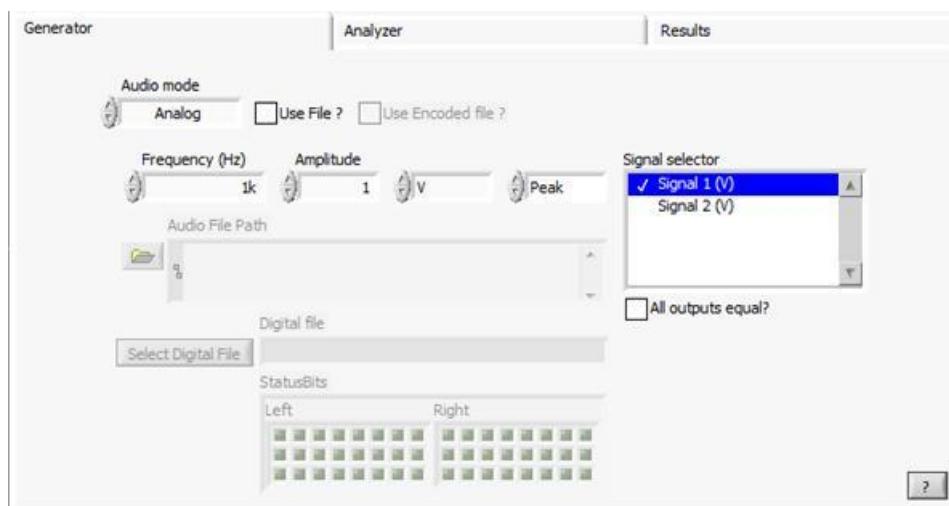


Figure 50: Single tone generator plug-in configuration panel

The test signal is specified individually for each enabled signal in the signal selector list box. The number of available signals in the list box is determined by the selected hardware platform.

Timing properties for the generated test signal is specified on the Analyzer tab as described later.

6.9.2.2 Analog and Digital Mode Wave File Option

When enabling the Use File option, the Audio File Path control is enabled and the Frequency control is disabled. Browse for a standard wave file to output when the step executes.

Make sure that the sample rate of the file and the sample rate specified in CATS are equal. If the sample rates do not match the file is output at the rate specified by CATS and a warning is shown on the setup panel.

If the wave file contains more than one channel they are mapped to the selected signals in a sequential manner.

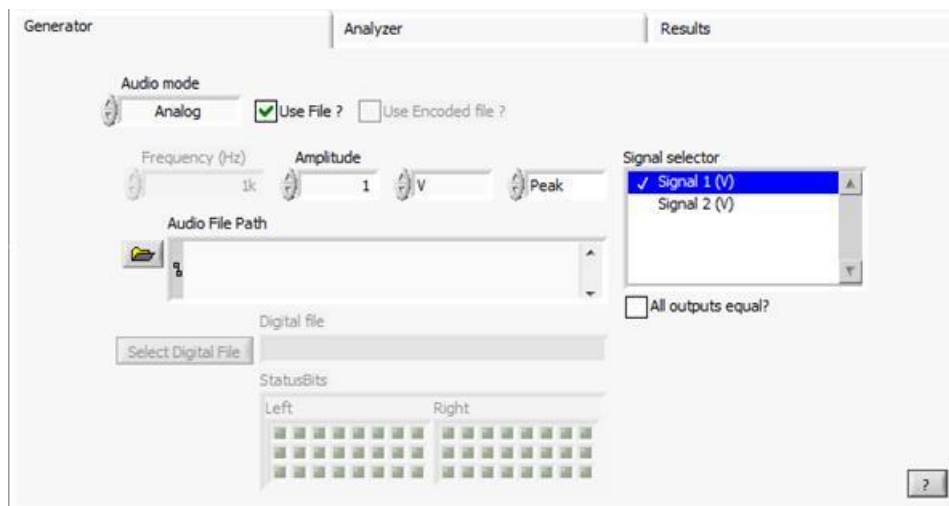


Figure 51: Single tone analog generator panel - wave-file

When using the wave file output option, the actual level of the test signal is defined by a combination of the signal level in the wave file and specified amplitude settings.

Because the magnitude of the numeric values in the wave file varies with the bit depth representation in the file, it is more convenient to normalize the full-scale value of the file as 1 V_{peak}. The amplitude and unit type then scales this value to its final output level.

This scheme may seem a bit complicated but ensures that the wave file can be output at a calibrated level.

$$\text{Output amplitude} = \text{Amplitude setting} \times \text{File amplitude}$$

6.9.3 Analyzer Settings

The analyzer settings specify the timing properties for the signal acquisition and for the generator in PCM mode as detailed in Timing Parameters.

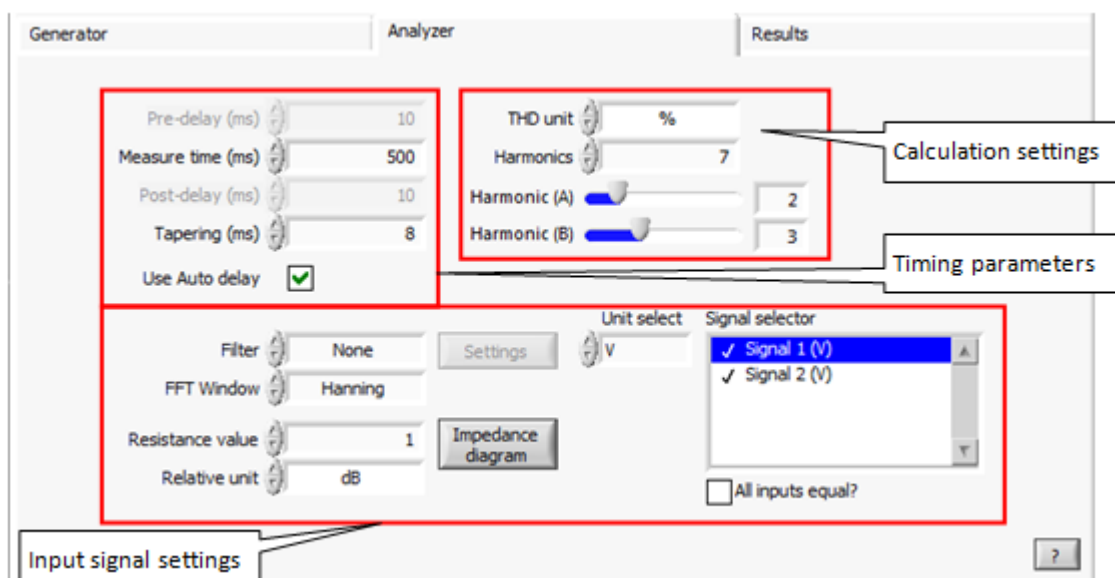


Figure 52: Single tone analyzer configuration panel

Besides timing properties, the Analyzer tab also includes filter and unit settings, etc. for the input signals that are described in Input Signal Settings.

6.9.3.1 Timing Parameters

The Single-tone plug-in analyzes a single block of acquired test signal. As seen from the figure below the total acquisition is the sum of Pre-delay, Measure time and Post-delay.

By default the Use Auto Delay option is selected to set the pre- and post-delays automatically according to the Tapering value. Tapering controls the time for fade-in and fade-out at each end of the generated PCM signal.

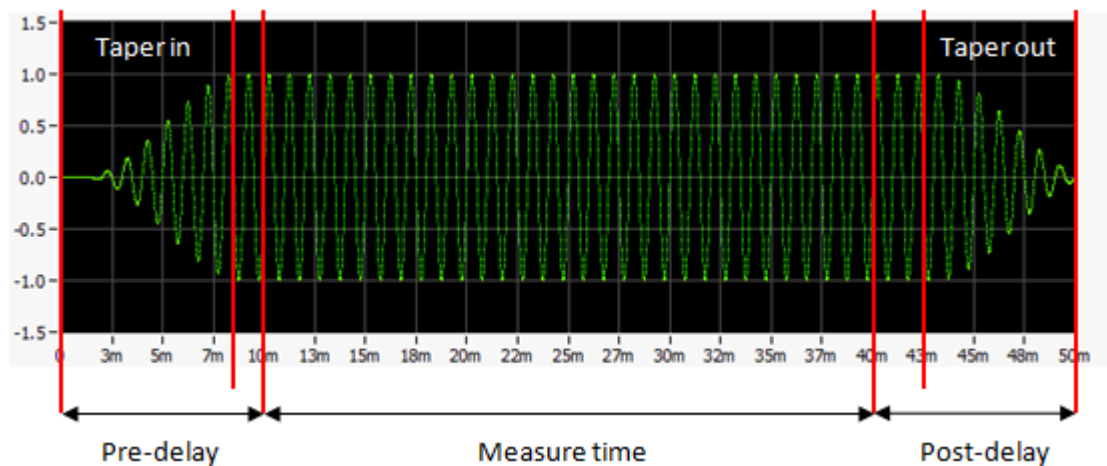


Figure 53: Example of time signal with measure time of 30 ms, pre- and post-delays of 10 ms and tapering of 8 ms

When the generator is set to output a .wav file the tapering value has no effect and the auto delay option can be turned off to enable setting the delay values independently according to the selected .wav file.

6.9.3.2 Calculation Settings

For this plug-in the THD unit and Harmonics can be specified. The parameters apply to all types of THD measurements like THD, THD+noise and SINAD.

6.9.3.3 Input Signal Setting

For each input signal in the Signal selector list box the Unit select, Filter and FFT Window can be specified.

The filter is applied to the entire acquired signal, before extracting the FFT block. This allows the Pre-delay to serve as settling time for the filter. The following filter functions are available:

- None
- Butterworth
- A-filter
- ITU-R 468

When using the Butterworth filter its default settings are: 6'th order High Pass Filter with lower cut-off at 125 Hz. Filter settings can be changed using the 'Settings' button.

If calibration is enabled on the channel, the unit must be the same as the calibration unit to avoid an error message.

Selecting the 'All inputs equal' option makes it easy to configure all signals in the same way.

Finally, the parameters Resistance value and Relative unit are used by the two channel measurements like impedance and gain. The parameters can be specified for each signal pair, Signal 1 and 2, 2 and 3, etc.

6.9.4 Result Settings

Select which result vectors to return to the test sequencer. Select only relevant results to reduce memory load on the test sequencer and improve performance.

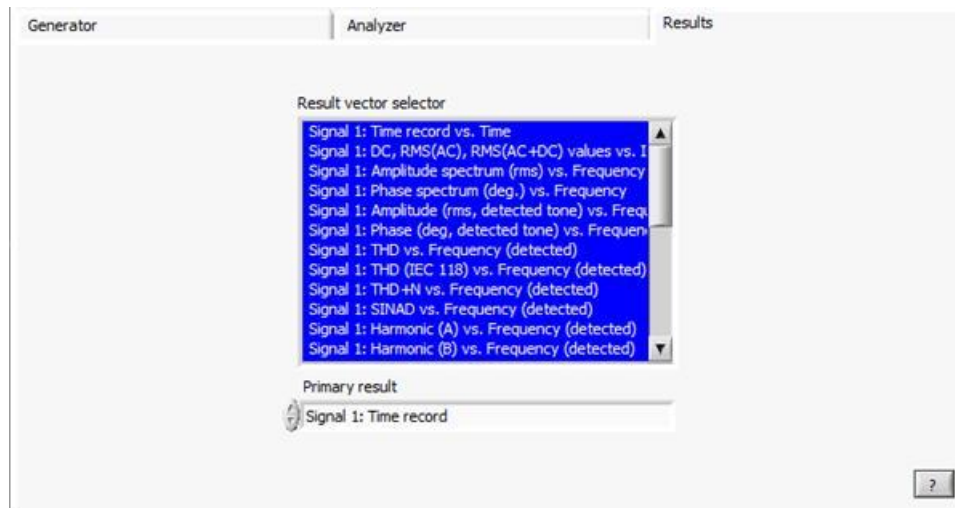


Figure 54: Single tone result configuration panel

The following analyzes are returned as vector results. The vector at index 0 is selected by the *Primary result* control.

Index	Analysis	Unit	Description
0	Primary y values vs. primary x values	As primary y	Copy of the result vector you chose as the primary result.
1	Time record vs. Time	Unit*†	Time record (optionally calibrated) on the input channel.
2	DC, RMS(AC), RMS(AC+DC) values vs. Index	Unit*	Measured DC and broad-band RMS values at index 0, 1, and 2 respectively.
3	Amplitude spectrum (rms) vs. Frequency	Unit*	The full amplitude spectrum of the measured signal.
4	Phase spectrum (deg.) vs. Frequency	Degrees	The full phase spectrum of the measured signal.
5	Amplitude (rms, detected tone) vs. Frequency (detected)	Unit*	Amplitude and frequency of highest detected tone in the measured signal.
6	Phase (deg, detected tone) vs. Frequency (detected)	Degrees	Phase and frequency of highest detected tone in the measured signal.
7	THD vs. Frequency (detected)	THD Unit	THD value and frequency of highest detected tone in the measured signal.

Index	Analysis	Unit	Description
8	THD (IEC 118) vs. Frequency (detected)	THD Unit	THD value after IEC method and frequency of highest detected tone in the measured signal.
9	THD+N vs. Frequency (detected)	THD Unit	THD+Noise and frequency of highest detected tone in the measured signal. This is reciprocal of signal-in-noise-and-distortion (SINAD).
10	SINAD vs. Frequency (detected)	THD unit	SINAD and frequency of highest detected tone in the measured signal. This is reciprocal of THD+N.
11	Harmonic (A) vs. Frequency	Unit*	Measured selected harmonic and frequency of the highest detected tone in the measured signal.
12	Harmonic (B) vs. Frequency	Unit*	Measured selected harmonic and frequency of the highest detected tone in the measured signal.
13	Status bits (0..191) vs. Index	—	Acquired status bits. This is only valid when using the Digital Audio platform.
14	Amplitude ratio ($R * \text{Signal } N / \text{Signal } N+1$) vs. frequency.	Relative unit	Relative tone amplitudes (vector 3) of two input channels.
15	Phase difference between Signal N and Signal N+1 vs. frequency	Degrees	The phase difference between Signal N and Signal N+1.
<p>* You can specify the unit of measurement on each channel for this control.</p> <p>† If the output unit is dB SPL the result unit is changed to Pa.</p>			

6.9.4.1 Application hints

When using the wave file output make sure that the pre- and post-delay values are set to extract the desired part of the test signal. If any settling or other signal change is included in the measurement time it may have considerable impact on the measured spectrum and THD values.

6.10 Step Response plug-in

This plug-in can generate and analyze a pure tone signal the change instantly from a low level to a higher level and back. The test signal can be configured with respect to frequency, levels and timing.

It can generate output on one channel and analyze input on one channel.

6.10.1 General Description

The purpose of this plug-in is to measure the attack and release time of an Automatic Gain Control (AGC) device fx. in a hearing aid.

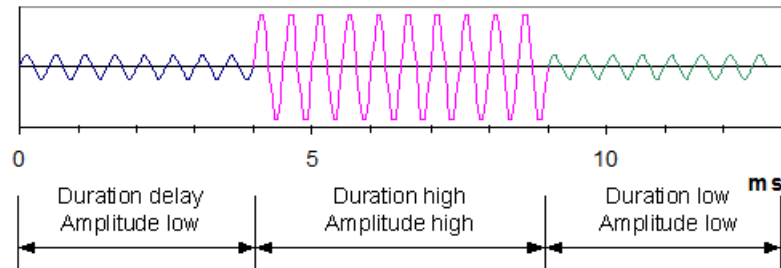


Figure 55: Step response signal specifications

This measurement function produces a pure tone signal that is stepped from low- to high-amplitude and back at specific times.

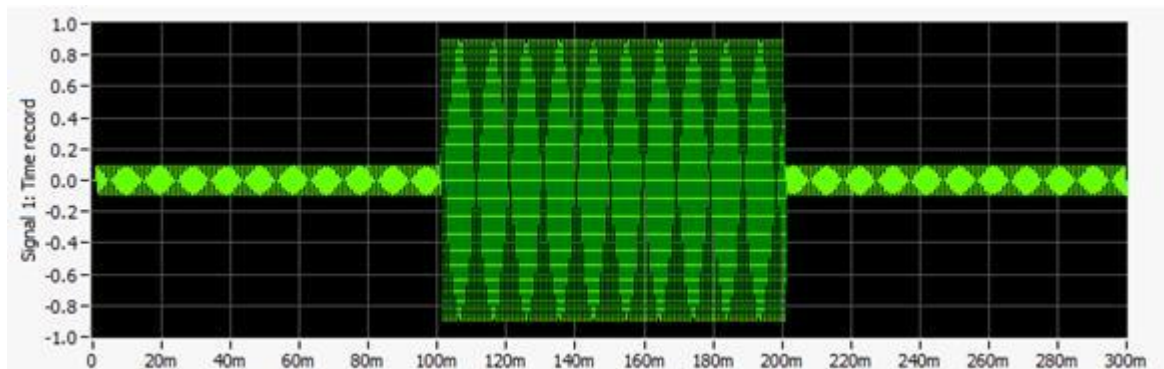


Figure 56: Step response test signal

This type of signal is typically used to measure attack and release times of an AGC circuit.

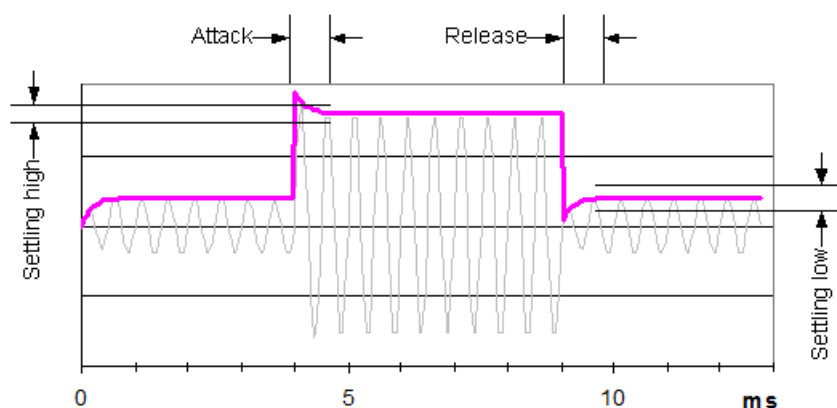


Figure 57: Attack and release time definition

Attack and release times are measured on an envelope curve, as the time from when the signal level change, until the output level has settled within the specified limits.

6.10.1.1 Enhanced Analyzer Method

The current version of the plug-in uses a new method to form the envelope and also a new method for analyzing the envelope. The new method is more robust than the previous version as it does not rely on a pilot tone to locate the time of the level shift in the test signal.

For backward compatibility the original method is still supported in this version.

6.10.2 Generator Settings

These controls set the test signal and analysis properties.

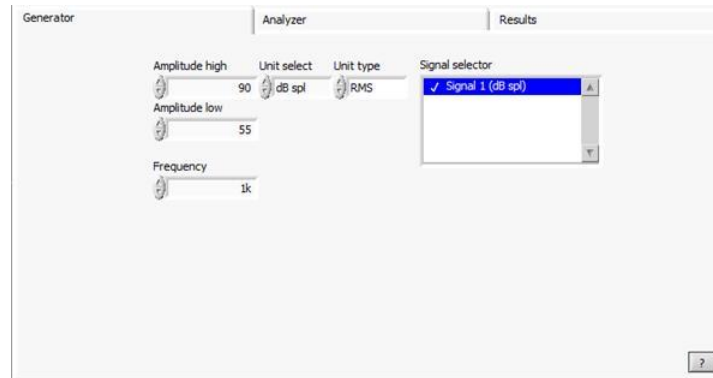


Figure 58: Step response generator configuration panel

Control	Unit	Range	Description
Amplitude high	Unit*	—	Amplitude for the high level of the step in the selected unit.
Amplitude low	Unit*	—	Amplitude for the low level of the step in the selected unit.
Frequency	Hz	0 - +inf	Frequency of the test signal tone (carrier frequency).
* You can specify the unit of measurement on each channel for this control.			

6.10.3 Analyzer Settings

The settings on this tab specify the timing not only for the analyzer but also for the generator. Also defines the setting limits that are used for calculating the attack and release time values.

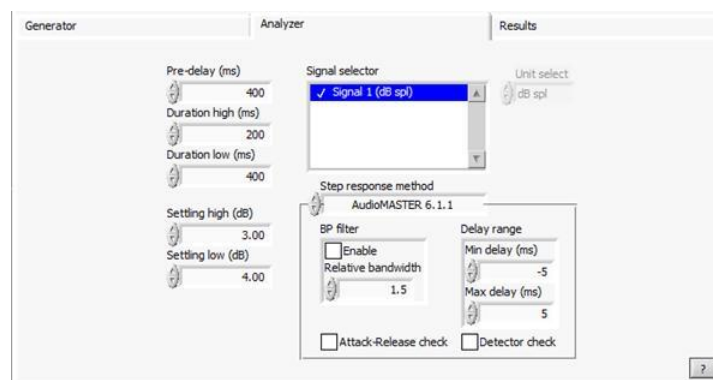


Figure 59: Step response analyzer configuration panel

The input unit is linked to the output unit and cannot be changed.

Control	Unit	Range	Description
Pre-delay	ms	—	Time delay for the signal to be on the low level, before it is stepped to the high level.
Duration high	ms	—	Duration of the high level signal.
Duration low	ms	—	Duration of the low level signal.
Settling high	dB	—	Settling criteria for the step up, attack time.
Settling low	dB	—	Settling criteria for the step down, release time.
Step response method	—	0..1	Selects Step response measurement method. The following parameters only have effect when option 1 is selected. 0 : '<= AudioMASTER 6.1' – original method 1 : 'AudioMASTER 6.1.1' – new method
BP filter	—	—	Settings for band pass filter to apply before calculating the envelope. See below for details.
Enable	—	T/F	Enables the band pass filter.
Relative bandwidth	None	—	Specifies the relative bandwidth of the filter.
Attack-Release check	—	T/F	Enables a check for valid attack and release measurements. An error is returned if the settling level for attack or release time is not crossed
Delay range	—	—	Specifies the accept range for detecting the rising edge of the test signal. The values must be set to include delays introduced by generation and acquisition hardware and the DUT itself. The values are relative to the specified signal timing.
Min delay	ms	—	Minimum acceptable edge delay from nominal value.
Max delay	ms	—	Maximum acceptable edge delay from nominal value.
Detector check	—	T/F	When enabled an error is returned if the rising edge is not found within the delay range.

Band Pass Filter Settings for Envelope

The new analyzer first removes any DC from the acquired signal and then applies the optional 4'th order Butterworth band pass filter before calculating the envelope and detecting the rising edge of the step response. The filter cut-off frequencies are:

$$f_{lower} = \frac{f_c}{n} \text{ and } f_{upper} = f_c \cdot n \text{ where } n = \frac{BW_r + \sqrt{BW_r^2 + 4}}{2}, f_c \text{ is the test frequency.}$$

Equation 3: Band pass filter cut-off frequencies as function of 'Relative band width' and f_c

By default, the band pass filter is disabled, but it can be enabled to make the envelope less sensitive to noise that are not near the test frequency. Enabling the filter may affect the envelope level or shape if the DUT produces significant levels of distortion.

6.10.4 Result Settings

From the list of available results, select which vectors to return to the sequencer. Then select one vector to be the primary vector.

By default, only the 'Step time' and 'Envelope' vectors are selected, as these are the most relevant results.

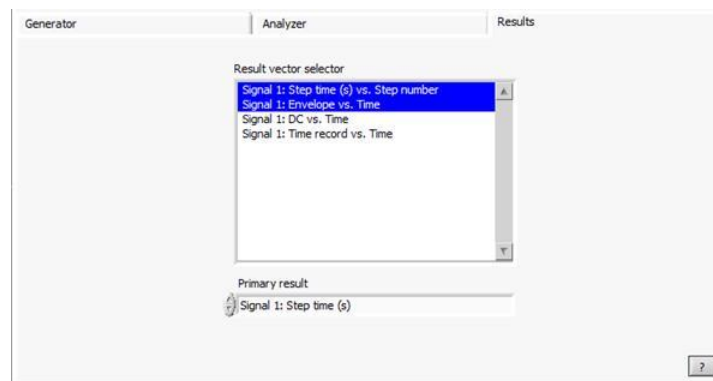


Figure 60: Step response result selection panel

Index	Analysis	Unit	Description
0	Primary y-values vs. Primary x-values	As primary y	Copy of the result vector you chose as the primary result.
1	Step time (s) vs. Step number	s	Settling time for each signal level change. Attack time is at index 1 and Release time is at index 2.
2	Envelope vs. Time	As output	Returns the signal RMS peak envelope of the time record.
3	DC vs. Time	As output	DC estimate of the time record.
4	Time record vs. Time	As output*	Time record of the input signal.
* If output unit is dBspl the result unit is changed to Pa.			

Note that when 'Step response method' is set to 'AudioMASTER 6.1.1' vector 3, 'DC' is empty as any DC is automatically removed before creating the envelope. Also, the attack and/or release times in vector 1, 'Step time' may return the value -1 to indicate that the envelope did not rise above the high settling limit or below the low settling limit.

6.11 Waveform Analyzer plug-in

This plug-in can generate a uniform white noise signal, or an arbitrary signal described by a formula. Analysis functions include FFT and fractional octave band analysis with selectable frequency range and resolution.

It generates output on one or more channels and analyzes input on one or more channels.

6.11.1 General Description

The waveform analyzer is used for acquiring signals for general analysis. Such signals are typically noise signals and arbitrary waveforms. The module does not require any information about the structure of the signal.

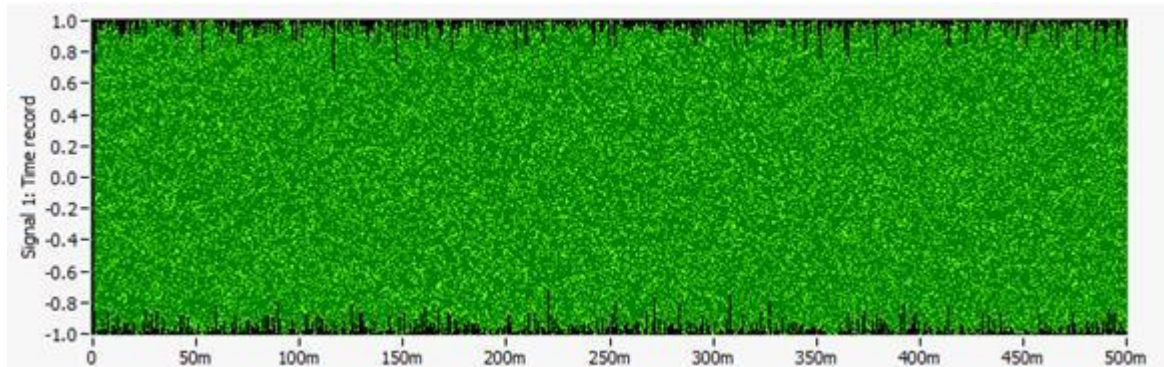


Figure 61: Waveform analyzer noise signal

It also has an arbitrary function generator, which can construct waveforms as functions of time. The following figure illustrates the spectrum of such a function (the glide tone sweep defined in the formula field, with an amplitude of 1 volt).

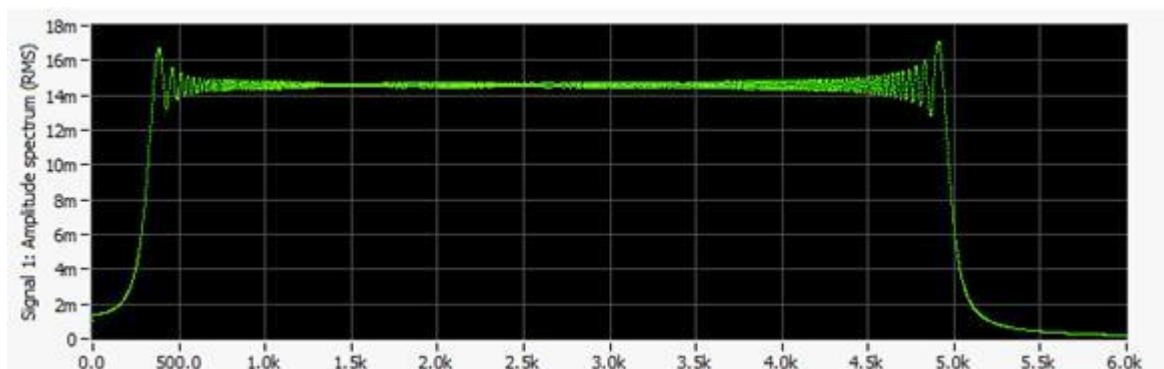


Figure 62: Amplitude spectrum of a chirp signal (500 to 5000 Hz) without any windowing function

6.11.2 Generator Settings

The generator can generate two types of waveforms. The first is a uniform white noise waveform, the setting for which is illustrated in the following figure.

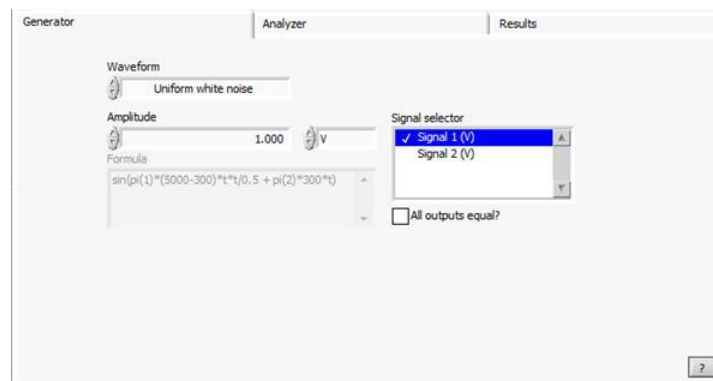


Figure 63: Uniform white noise selection with amplitude control

The second type is a waveform described by a formula, the setting for which is illustrated in the following figure.

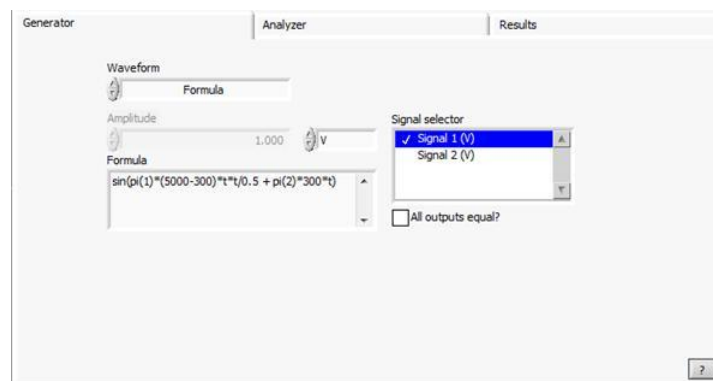


Figure 64: Waveform generation by formula (amplitude as a function of time), in this case a chirp over 0.5 seconds from 300 to 5000 Hz and with an amplitude of 1 Volt

The generation and acquisition time is specified on the Analyzer tab.

The formula must define the amplitude as a function of time t or sample number n . The following tables give the variables, operators, and functions that are valid for defining the function. Refer to Special Usage Hints for examples of waveform functions.

Variable	Description
n	Number of samples generated so far
t	Number of elapsed seconds
fs	Sampling frequency

Table 1: Variables for creating waveforms as functions of time (or sample number)

Operator	Description
$**$	exponentiation
$+$, $-$, $!$, \sim	unary plus, unary negation, logical not, bit complement
$*$, $/$, $\%$	multiplication, division, modulus (remainder)
$+$ and $-$	addition and subtraction

Operator	Description
>> and <<	arithmetic shift right and shift left
>, <, >=, and <=	greater, less, greater or equal, and less or equal
!= and ==	inequality and equality
&	bit and
^	bit exclusive or
	bit or
&&	logical and
	logical or
? :	conditional evaluation

Table 2: Valid operators in the formula

Function	Description
abs(x)	Returns the absolute value of x.
acos(x)	Computes the inverse cosine of x in radians.
Acosh(x)	Computes the inverse hyperbolic cosine of x.
asin(x)	Computes the inverse sine of x in radians.
Asinh(x)	Computes the inverse hyperbolic sine of x.
atan(x)	Computes the inverse tangent of x in radians.
Atan2(x,y)	Computes the arctangent of y/x in radians.
Atanh(x)	Computes the inverse hyperbolic tangent of x.
ceil(x)	Rounds x to the next higher integer (smallest integer $\geq x$).
cos(x)	Computes the cosine of x, where x is in radians.
Cosh(x)	Computes the hyperbolic cosine of x.
cot(x)	Computes the cotangent of x ($1/\tan(x)$), where x is in radians.
Csc(x)	Computes the cosecant of x ($1/\sin(x)$), where x is in radians.
Exp(x)	Computes the value of e raised to the x power.
Expm1(x)	Computes one less than the value of e raised to the x power ((ex) - 1).
Floor(x)	Truncates x to the next lower integer (largest integer $\leq x$).
getexp(x)	Returns the exponent of x.
getman(x)	Returns the mantissa of x.
int(x)	Rounds x to the nearest integer.

Function	Description
Intrz(x)	Rounds x to the nearest integer between x and zero.
Ln(x)	Computes the natural logarithm of x (to the base of e).
Inp1(x)	Computes the natural logarithm of $(x + 1)$.
Log(x)	Computes the logarithm of x (to the base of 10).
Log2(x)	Computes the logarithm of x (to the base of 2).
Max(x,y)	Compares x and y and returns the larger value.
Min(x,y)	Compares x and y and returns the smaller value.
Mod(x,y)	Computes the remainder of x/y , when the quotient is rounded toward $-\infty$.
Pow(x,y)	Computes x raised to the y power.
Rand()	Produces a floating-point number between 0 and 1 exclusively.
Rem(x,y)	Computes the remainder of x/y , when the quotient is rounded to the nearest integer.
Sec(x)	Computes the secant of x , where x is in radians ($1/\cos(x)$).
Sign(x)	Returns 1 if x is greater than 0, returns 0 if x is equal to 0, and returns -1 if x is less than 0.
Sin(x)	Computes the sine of x , where x is in radians.
Sinc(x)	Computes the sine of x divided by x ($\sin(x)/x$), where x is in radians.
Sinh(x)	Computes the hyperbolic sine of x .
sizeofDim(ary,di)	Returns the size of the dimension di specified for the array ary .
Sqrt(x)	Computes the square root of x .
tan(x)	Computes the tangent of x , where x is in radians.
Tanh(x)	Computes the hyperbolic tangent of x .

Table 3: Valid functions in the formula

6.11.3 Analyzer Settings

The analyzer specifies the timing that applies to both generation and acquisition. The total time is the sum of Pre-delay, Measure time and Post-delay.

The filter is applied to the entire acquired signal, before extracting the FFT block. This allows the Pre-delay to serve as settling time for the filter. The following filter functions are available:

- None
- Butterworth
- A-filter
- ITU-R 468

When using the Butterworth filter its default settings are: 6'th order High Pass Filter with lower cut-off at 125 Hz. Filter settings can be changed using the 'Settings' button.

The selected time domain window is applied only to the measurement time part of the signal.

Selecting the 'All inputs equal' option makes it easy to configure all signals in the same way.

Most signals require the application of a windowing function before transformation to the frequency domain. The windowing function can be chosen in the Window control.

The octave analysis is based on a Butterworth 3rd order band-pass filtering. The number of band pass filters in each octave is given in the Oct resolution control.

The start and stop frequency for the octave analysis are given in their respective control.

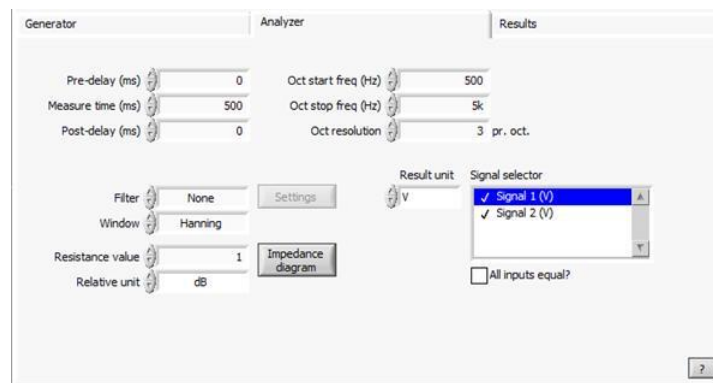


Figure 65: Waveform analyzer configuration panel

Finally, the parameters Resistance value and Relative unit are used by the two channel measurements like impedance and gain. The parameters can be specified for each signal pair, Signal 1 and 2, 2 and 3 etc.

6.11.4 Result Settings

The Waveform Analyzer returns the following analyzes as vector results.

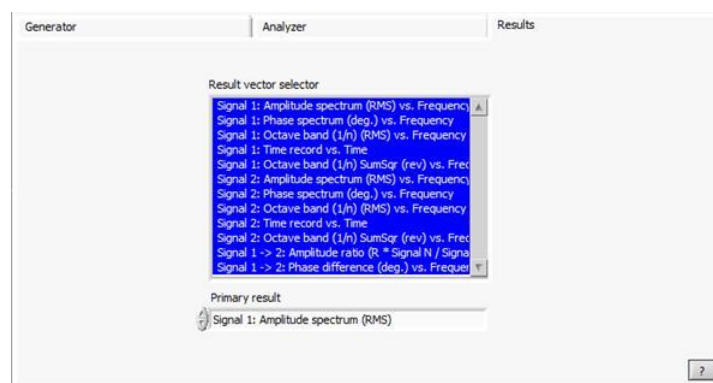


Figure 66: Waveform analyzer result selection

Index	Analysis	Unit	Description
0	Primary y values vs. primary x values	As primary y	Copy of the result vector you chose as the primary result.
1	Amplitude vs. frequency	Unit* (rms)	Amplitude spectrum with the applied Window function

Index	Analysis	Unit	Description
2	Phase vs. frequency	Deg.	Phase spectrum with the applied Window function
3	Octave vs. frequency	Unit* (rms)	Octave analysis as defined in the three octave controls (Hanning window used).
4	Time record vs. time	Unit*†	Time record (optionally calibrated) on the input channel.
5	Amplitude ratio (Signal N / Signal N+1) vs. frequency	Relative unit	Relative amplitude spectrum of two input channels.
6	Phase difference	Deg.	Phase difference between amplitude spectra of two input channels.
<p>* You can specify the unit of measurement on each channel for this control.</p> <p>† If output unit is dB SPL the result unit is changed to Pa.</p>			

6.11.5 Special Usage Hints

This section lists some useful hints and caveats to the use of this plug-in.

- Useful for measuring noise and other situations, where information about the signal structure is not required
- Output can only be calibrated in a single point (no frequency correction)
- No distortion analysis
- Phase difference requires that the two input channels are sampled simultaneously.
- The default expression represents the formula:

$$\sin\left(\frac{2\pi(f_2 - f_1)}{\Delta T}t^2 + 2\pi f_1 t\right)$$

where f_1 and f_2 is the start and end frequency in Hz, and ΔT is the sweep rate in seconds.

- The formula and expression for a pure sine wave:
 $\sin(2\pi f t)$ or $\sin(\pi(2)*f*t)$
 where f is the frequency in Hz
- The expression for a random signal with amplitude values in the range ± 1 :
 $2*\text{rand}(n)-1$

7 LimiTEST

The CATS step types includes support for LimiTEST. LimiTEST is an advanced tool for evaluating waveform results in TestStand.

LimiTEST has a full featured user interface and can easily be integrated in any TestStand application.

Using LimiTEST, it is possible to define limits for waveforms. Limits for waveforms are tunnels, windows, and points the curve must pass for the entire test to pass. The graphical limits includes tunnels (upper, lower, or combinations of these), Window limits (Valley, Knee point, Falling curve, Rising curve and many others), and point limits. All of these can be combined and can also be used as so-called floating limiters.

LimiTEST also includes a system for pre-processing the curve before the limits are evaluated. Typical pre-processing is noise filtering or FFT conversion.

LimiTEST also includes key parameter calculations. Key parameters is a technique for applying post-processing to a measured result and converting the results to simple parameters that are easier to interpret and test than a curve. Typical key parameters are the RMS value or maximum peak value and maximum peak position.

LimiTEST supports a number of different waveform representations. Some examples are National Instrument Analog Waveform type, simple numeric 1D array or vector (array of x, y coordinates).

7.1.1 Overview

The LimiTEST system takes a measured waveform as input. The waveform is filtered and evaluated using Graph Limits and/or Key Parameters.

The following figure illustrates the basic components of LimiTEST.

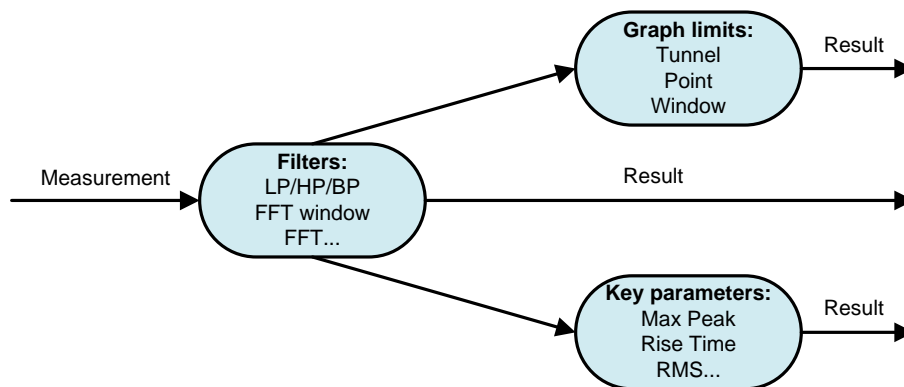


Figure 67: LimiTEST overview

The measurement can optionally be processed ("filtered") before it is passed to the curve evaluation. (The filter function can also be used as a standalone tool).

Output from the filter function is then used by the Graph limits for evaluation the curve against tunnels, Points and Windows. Each of these limits specifies criteria that the curve must fulfill for the test to pass.

Output from the filter function can also be used by the Key Parameters for calculation and evaluation of simple measurements extracted from the curve.

Graph Limits

The following figure illustrates a typical example of a curve evaluation.

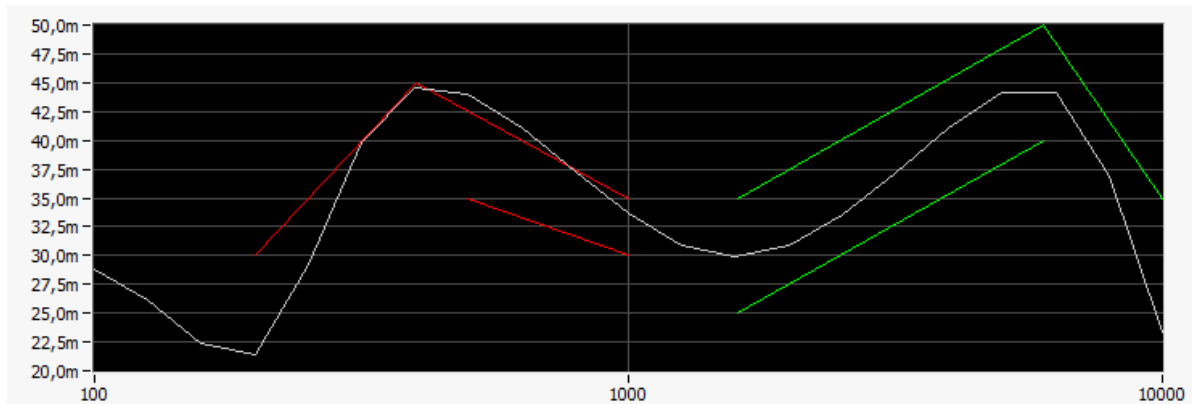


Figure 68: Example of response evaluation

The example shows a curve that must be evaluated against two tunnels. The tunnel from 200 to 1000 Hz fails because the curve is above the upper limit, and the tunnel from 1200 to 10000 Hz passes because the curve is within the limits (colours added for illustration).

Key Parameters

Key parameters are a method to extract numeric information from a curve and compare this value with simple limits. Typical examples of key parameters are volt RMS, Maximum value, and Minimum values.

More Key Parameters can be calculated for a curve and each of these can have their own set of limits. Limits can even be TestStand expressions, which make it possible to make dynamic limits on key parameters.

Waveform Processing

Before testing the limits of a curve, it might be necessary to remove noise or even to convert a time signal to FFT. This can be done by using the Waveform Processing feature of LimiTEST.

7.1.2 Using LimiTEST in TestStand

The LimiTEST step type is available both as part of Audio Test steps or as separate test steps as explained in section 4, User Interfaces.

7.2 LimiTEST Data Source

Because TestStand makes a copy of the context tree when executing a test step or sequence, the results are erased by default when the execution ends. This makes it difficult to configure limits etc. because it is not immediately possible to view the results while editing the limits.

To overcome this problem, LimiTEST implements a data transfer mode for CATS. Every time CATS executes a measurement, the result vectors are transferred to LimiTEST. The result is stored until a new result is written. It is recommended to use the internal data transfer method to configure the LimiTEST data source so that you can browse through these available result vectors by name.

The CATS simulate demo.seq sequence file shows an example of how to use the Graph limits and Key parameters.

All LimiTEST features require a data source to be specified. This is the source to the array or waveform containing the signal to be evaluated. Examples of data sources are:

- `Step.result.VectorRes`
- `StationGlobals.VectorArray`
- `RunState.Sequence.Main["ButterWorthFilter"].Result.NumericArray`

The first example shown above (`Step.result.VectorRes`), points to an array in the current step. This syntax can be used in the Audio Test or LiveView step types that are merged with the LimiTEST test type. This gives the simplest handling of results.

In the second example, the result is stored by the measurement function in a global variable.

The last example shows the data source with a common syntax for the solution where the measurement step is followed by a LimiTEST step. In this case, the LimiTEST step must refer to the result in the previous step. This can be done in many ways, and one of them is shown in this example.

7.2.1 Internal Data Transfer

Using internal data transfer, you can use the LimiTEST data source viewer to select data from a list of available measurements made in CATS. This functionality is available on the Data Source tab, as shown in the following figure.

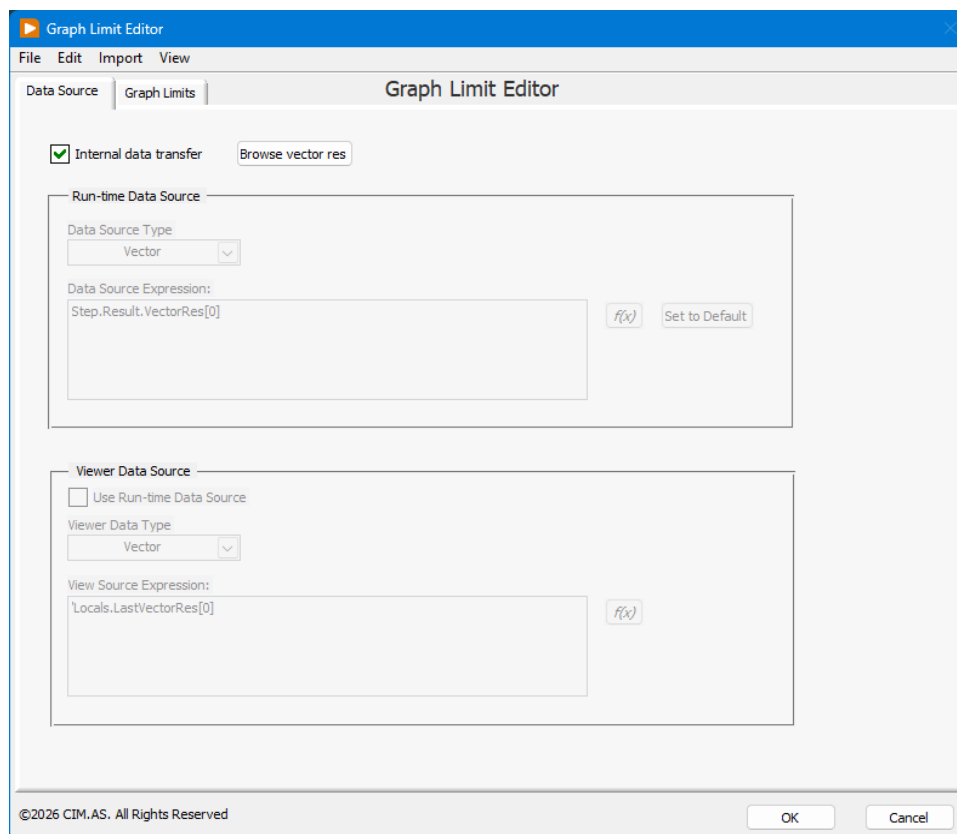


Figure 69: Configuration UI of the Graph Limit Editor after the update

If you place a checkmark in the Internal data transfer checkbox, the Browse vector res button is enabled, and all other controls are disabled. Click the Browse vector res button to show the user interface of the vector result viewer. The viewer displays the vector results of the acquisition step (such as the CATS plug-in) last executed, as shown in the following figure.

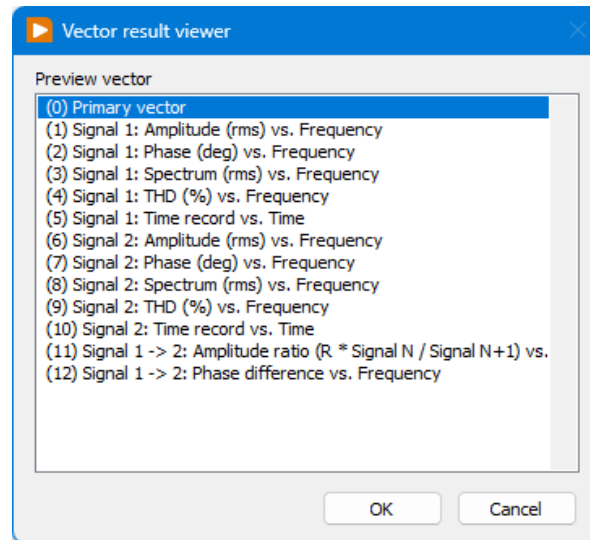


Figure 70: Configuration UI of the Vector result viewer

7.2.2 View Data Source

For the Waveform Processing and the Graph Limits functions, a viewer data source can also be specified. This data source is used for viewing the measurement while modifying the test sequence. When TestStand executes a sequence, the results only exist during the execution. As soon as the execution ends, all results are no longer available. In order to use the viewer function, the results must therefore be stored in a safe place before the execution ends. To do this, you can specify a global or local variable with the Shared flag set so that TestStand will hold the data after executing the step. This global or local variable can then be specified as viewer data source since the contents of the variable still exists after the execution ended.

The result can be transferred to the global variable in the post expression of the measurement step. An example of this is shown in the following figure.

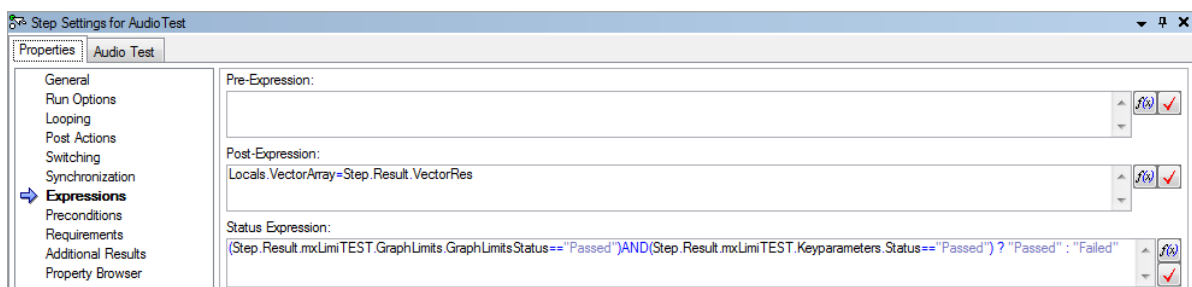


Figure 71: Post expression to store result in a global variable

A common method is to have one global variable for all waveform results. To show the result in the LimiTEST Editor, the step(s) in question must then first be executed. This will store the last result in the global variable. When one of the LimiTEST editors is open, it will show the stored result from the global (if viewer data source is set to point to this variable).

Care should be taken to remove the copying of data to the global when the test program has been finally debugged. This should be done to increase the speed of the final test program. For later use the post expression can be commented out or made dependant of a local or global flag (Boolean) for example called "debug". The following expression is an example of a conditioned post expression.

- StationGlobals.Debug ? StationGlobals.MyArray=Step.Result.NumericArray: ""

This example will transfer the result to the global array if the debug flag is set. Otherwise, no (or an empty) expression is evaluated.

7.3 Waveform Processing

The LimiTEST Waveform processing feature can be used to apply filter, etc. to the result vector before testing against graph limits and key parameters.

When used with CATS steps no waveform processing is normally needed as CATS provides common filtering internally.

The Waveform Processing feature of LimiTEST includes a number of processings that can be made on the waveform.

The result of the processing can be used as input for Graph Limits and Key Parameters or saved as a result in TestStand without making any further limit testing.

The Waveform Processing Editor contains the following components.

Available Processings

This control lists the available processings in a tree structure. The available processings can be inserted in the list of selected processings by clicking Insert. Remove removes the selected processing from the list of processings.

Selected Processings

This is the list of processings that will be used for the signal. The processings are executed in the order they are listed starting with the first processing in the Selected Processings list. In the example in the following figure, three processings are selected.

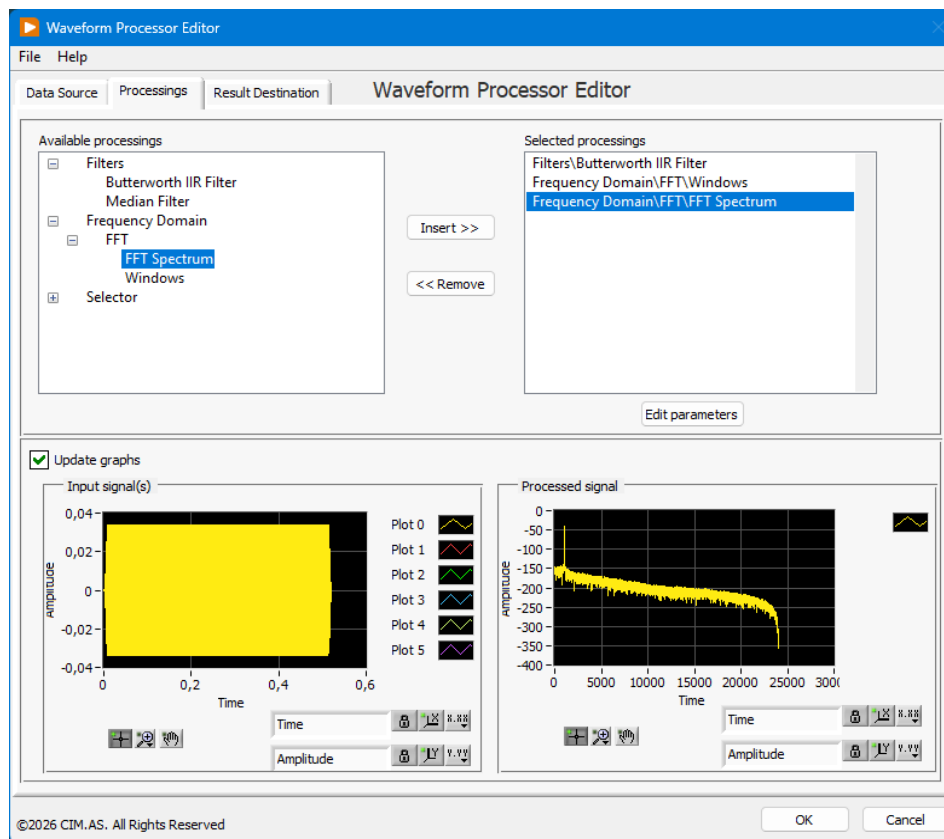


Figure 72: Waveform Processor. Processings tab shown

The first processing makes a low pass filtering of the signal. The second processing applies a Hanning window, and the third processing converts the time signal to a FFT spectrum in dB.

More of the processings have additional parameters, such as the cut off frequency of the low pass filtering. These parameters can be set by pressing the Edit Parameters button.

Graphs

The two graphs show the signal before and after the waveform processing. This can be used for verifying that the processing is correct.

Data Source

The signal used for the graphs can be defined on the Data Source tab as explained in section 7.2.

Result Destination

On this tab, the location for the result can be defined. By default, the waveform processing overwrites the measurement used as input. The Result Destination tab is shown in the following figure.

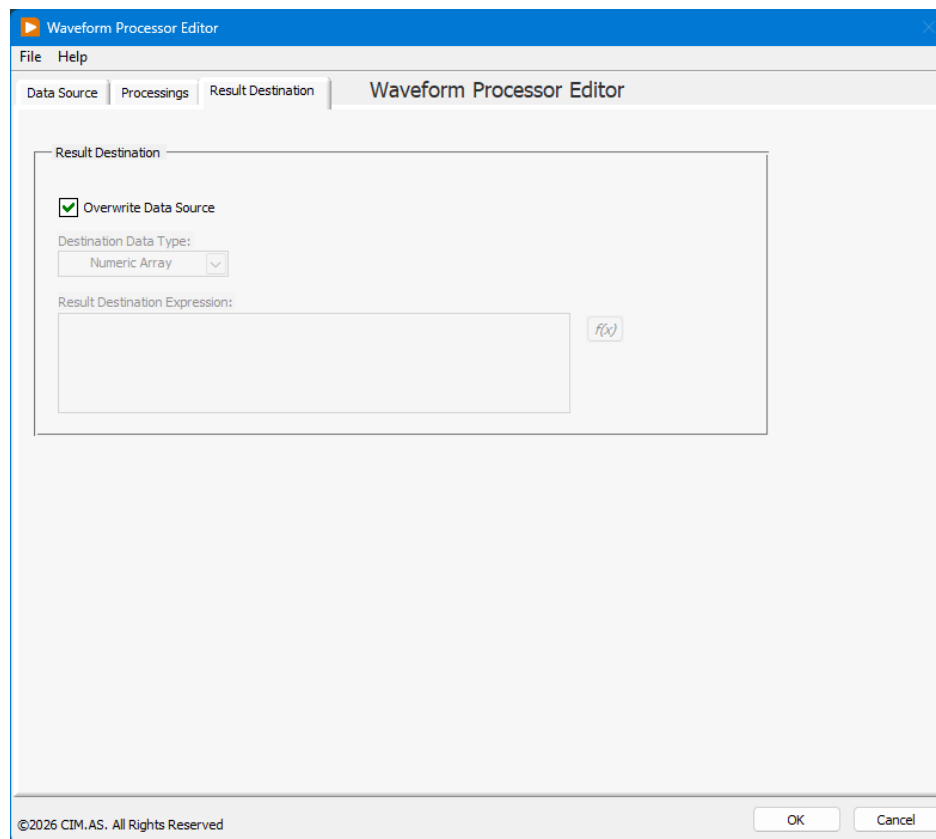


Figure 73: Waveform processing. Result destination tab

7.4 Graph Limits

Graph limits in LimiTEST offer a much more advanced functionality for testing waveforms than simple limiters offer.

The following figure illustrates types of limiters in the vector limits system.

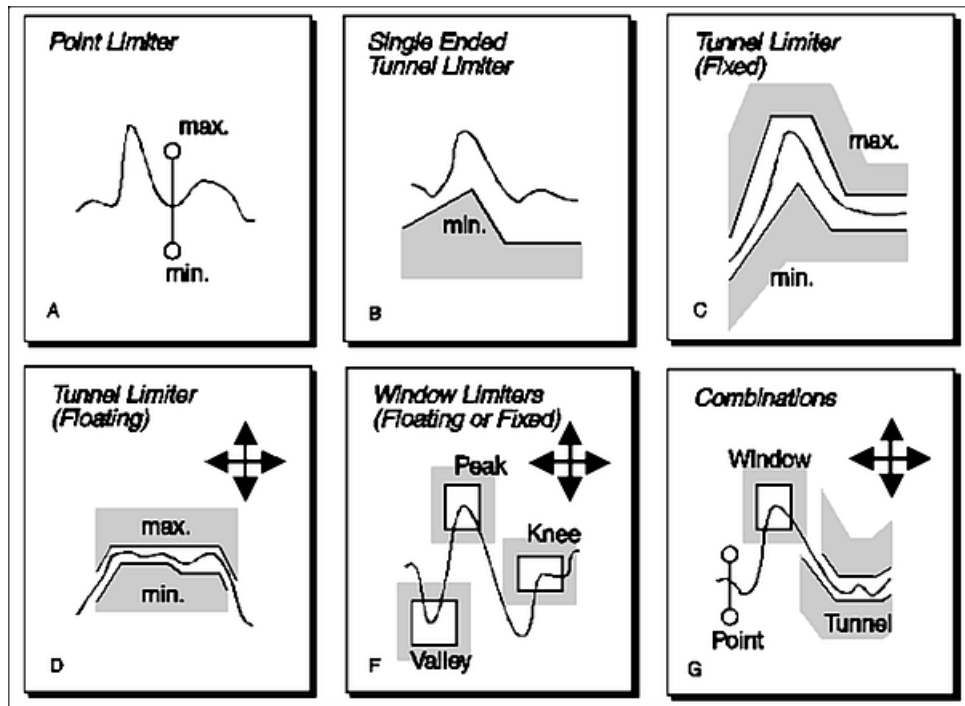


Figure 74: Limiter functions offered with the vector limits system. Vector limits can be point limiters, tunnel limiters, and window limiters. Furthermore, a limiter may be fixed or floating. Combinations of several limiters are possible.

Limiters can be programmed as single-ended elements or multi-level limiters, and have individual reference points, such as the upper and lower level values in the X and Y directions.

Point Limiters

In its simplest form, the limiter function may just be a single-ended or double-ended point limiter (A). The point limiter defines the testing limit at any given point, whether that is a single ended test point or actually a test point having a min and max value.

Tunnel Limiters

Tunnel limiters are often used in mixed-signal testing. A tunnel limiter simply defines the envelope in which a signal has to conform. If the signal exceeds the boundaries of the tunnel limiter, the test will be classified as failing to the given set of tunnel limiters. The single-ended version of the tunnel limiter (B) defines a boundary condition that the given signal must either be above, or below, depending on what test conditions are chosen by the user. Tunnel limiters may be fixed (C) or floating (D) at discretion of the user.

Window Limiters

The window limiters (F) are the more specialized type of limiters. These limiters specify one or more window(s) or box(s) within which special signal characteristics must occur. Examples of window limits are:

- Peak detection
- Valley detection
- Knee detection

A peak window means that the signal must have a local peak within the boundaries of the specified window. The size of the window is fully defined by the user. Similarly, a window limiter having defined a valley requires the signal to have a local valley inside the specified conditions. The knee version of the window limiters defines a breaking point, which means that a signal must feature a specified boundary crossing. For example the signal must pass the window from left-to-top, left-to-bottom, bottom-to-right, or top-to right.

7.4.1 Floating Limits

A given limiter - whether it is a tunnel limiter or a window limiter - can be fixed or programmed to float in the X-direction, the Y-direction, or both X and Y directions, as illustrated in the following figure. This feature is of utmost importance in many practical test situations.

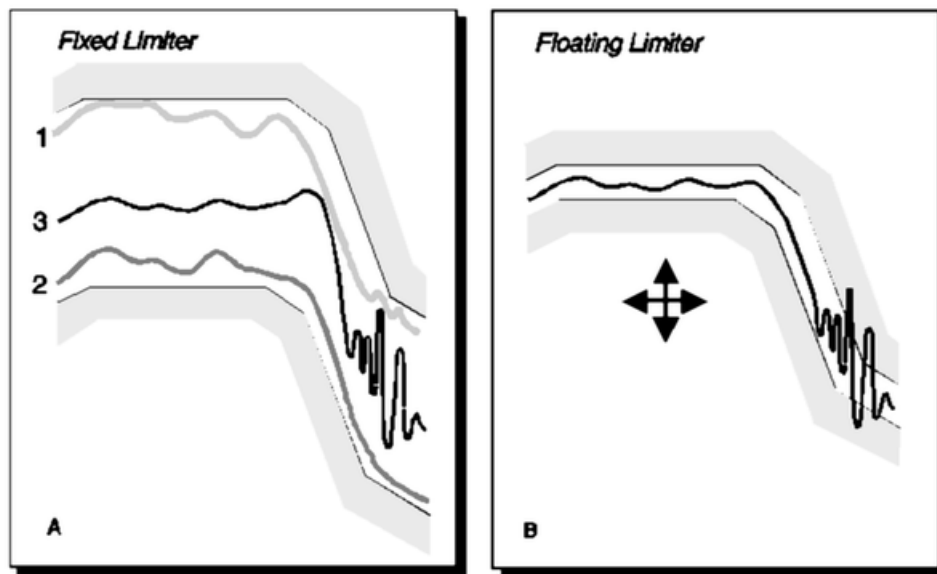


Figure 75: Floating limits. In figure A tolerances must be set wide because of production tolerances. Signal 1 and 2 must pass but 3 should fail but does not fail in this case. By using floating limiters as seen in figure B, 1 and 2 will pass and 3 will fail.

Take for example a situation where a device has to be tested for the behaviour at frequency roll-off. The frequency itself may not be of critical importance and varies as a result of production tolerances, which may be as much as $\pm 20\%$. However, the requirement for the actual shape of the frequency fall-off response of the given signal is high with respect to slope, over-shoot/under-shoot, ringing phenomena, etc. This calls for a rather tight tolerance of the limiter envelope. Using the floating limiter option this can be obtained. A tight envelope may be programmed - and at the same time the entire limiter can be set to float for example $\pm 20\%$ in the X-direction.

Another situation of interest could be one of flatness of an amplifier over a given frequency range. Here the actual gain may be allowed to vary ± 1 dB, but the flatness of the amplification over a specified frequency range may have to stay within 0,1 dB. Here the limiter can be programmed to float in the Y-direction and then perform the flatness test using tighter control limits.

7.4.2 Graph Limit Editor

The limit editor allows the limits to be defined graphically. It is possible to combine any number of limits and limit types as needed by the actual measurement.

First, specify the data source as explained in section 7.2.

Specifying Graph Limits

After selecting the desired vector, switch to the Graph Limits tab and click the Update Graph button to show the data in the graph. It may be necessary to change the numeric format and scaling to make the graph visible.

The graphical view shows the currently defined limits and, optionally, the signal from the last measurement.

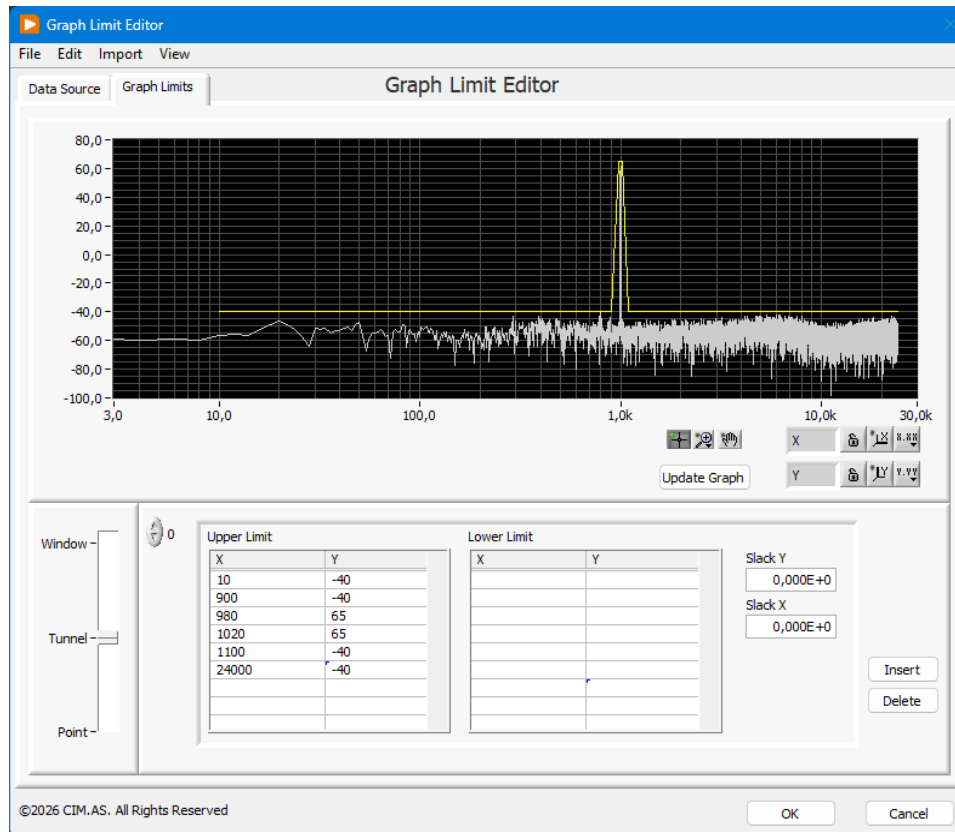


Figure 76: Specify graph limits

Specify the limit values in the table below the graph. It is possible to specify multiple upper and/or lower limits that are automatically drawn on the graph.

You can specify all limit values using engineering notation (for example, 1.020E+3) or using the International System of Units (SI) (for example, 1.267k). When you specify tunnel limit values, when you close and reopen the Graph Limit Editor, any engineering notation values are rounded to three decimal places and any SI values you specified appear in engineering notation and are rounded to three decimal places. When you specify window or point limit values, the rounding and value format change occurs when you click outside of the limit value field after editing the value.

7.4.3 Window Limits

When selecting window limits, the limit specific parameters change window limit settings as shown in Figure 77. Window limiters are used to verify peaks, values and/or knee-points in a curve. Window limiters consist of pattern search functions like peaks or valleys and functions for entry/exit values.

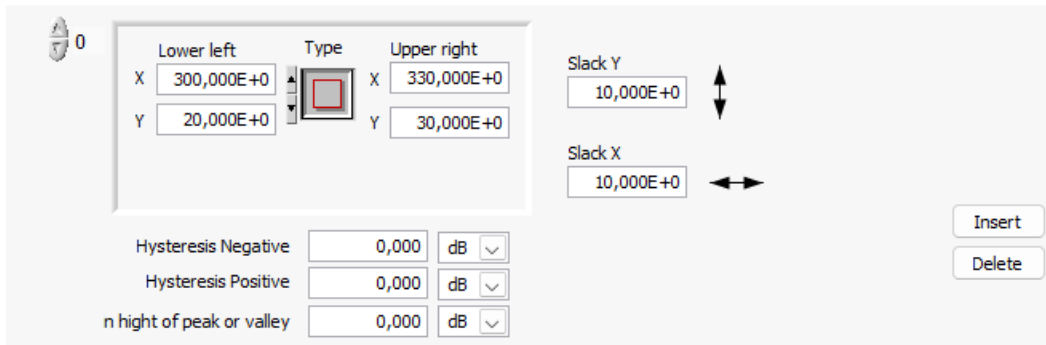


Figure 77: Window limit parameters

To set a window limiter, first select the limiter type Window on the vertical scroll bar to the lower left of the user interface.

Window Position

A limiter window is defined by the upper, right corner and the lower, left corner. Start defining the (X, Y) values, for the lower, left corner and then do the same for the upper, right corner of the window. Note that values in the upper, right corner must always be larger than the values in the lower, left corner. For example, $X_U > X_L$ and $Y_U > Y_L$. Otherwise, you will experience a failure.

Window Type

Select the type of window limiter by clicking the square field in between the upper, right corner values and the lower, left corner values. The 10 pictograms of the window selector pop-up menu show the various types of window limiters.

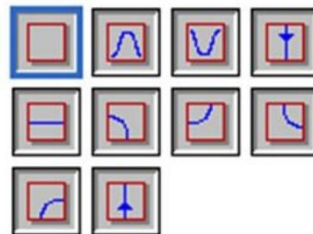


Figure 78: Window limit functions

Slack

The tolerance on floating conditions for a window limiter is controlled by the slack-X and slack-Y control windows. Use the mouse on the scrolling facilities or enter the values from the keyboard.

Insert and Delete

A new windows limiter can be inserted using the Insert function. The Insert and Delete operations take place on the window limiter at the top. For example, if the selector in the upper, left corner is set to "3", the Insert function will insert a new limiter numbered "3", and the "old 3" will become "4", etc.

It is similar for the Delete operation. Delete will remove the upper limiter and shift all subsequent limiters one step.

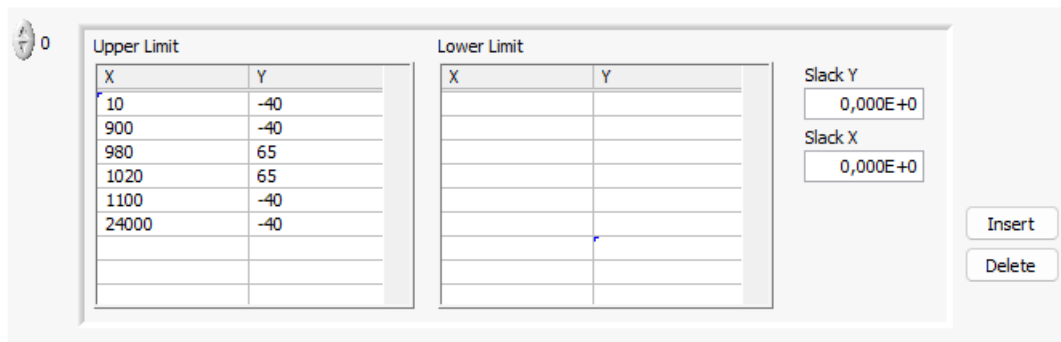
Hysteresis Settings

The parameters are used as simple filters to reduce noise sensitivity, so only real local extreme are found. The parameters are:

- Positive hysteresis — Minimum difference between two ordinate points when the differential derivative is positive.
- Negative hysteresis — Minimum difference between two ordinate points when the differential derivative is negative.
- Min height of peak or valley — The difference between the extreme and the surrounding points before it is accepted as an extreme.

7.4.4 Tunnel Limits

When selecting tunnel limits the settings will display as illustrated in the following figure.



Upper Limit		Lower Limit	
X	Y	X	Y
10	-40		
900	-40		
980	65		
1020	65		
1100	-40		
24000	-40		

Slack Y: 0,000E+0
Slack X: 0,000E+0

Insert
Delete

Figure 79: Tunnel limits

A tunnel limiter simply defines the envelope in which a signal has to conform. If the signal exceeds the boundaries of the tunnel limiter, the test will be classified as failing to the given set of tunnel limiters. The single ended version of the tunnel limiter, illustrated in Graph Limits as diagram B, defines a boundary condition that the given signal must either be above, or below, depending on what test conditions are chosen by the user. Tunnel limiters may be fixed or floating, illustrated in Graph Limits as diagrams C and D.

Upper Limits and Lower Limits

These coordinates define the curves for the upper and/or lower limits given as X and Y coordinates. Each of the coordinates will be connected by a linear curve to form a complete limit curve.

Slack

The tolerance on floating conditions for a tunnel limiter is controlled by the slack-X and slack-Y. Use the mouse on the scrolling facilities or enter the values from the keyboard.

7.4.4.1 Editing Tunnel Limits

Using the cursors, a part of a tunnel limit can be edited. Selecting View->Cursors to centre will show two cursors. These cursors can be moved with the mouse by using the cursor tool selected from the graphs controls. The tunnel limits within the cursors can be moved by selecting Edit->Shift tunnel limit.

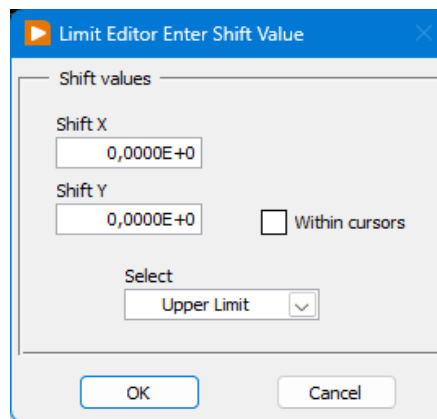


Figure 80: Shift tunnel limits. This example will move the upper tunnel limit 0.2 up within the cursors.

A tunnel limit can be deleted by selecting Edit->Delete current tunnel.

The Edit->Crop tunnel within cursors function is used to skip a tunnel limit except for what is within the two cursors. This is, for example, useful if the tunnel is created automatically from imported data.

Automatic generated limiters can sometime have excessively high numbers of points, less desirable for a limiter, because it can slow operation and is difficult to modify. To reduce the number of points, select Edit->Reduce Resolution.

Importing Tunnel Limits

You can import tunnel limits, also known as vector limits, from a CSV file, which is an ASCII comma-separated file, using the following format:

- <ULIMIT>
- <x>,<y>
- ...
- <LLIMIT>
- <x>,<y>
- ...

ULIMIT and LLIMIT define the start of the limits. Each of the lines following ULIMIT and LLIMIT are the coordinates of the tunnel. ULIMIT and LLIMIT must be written in upper case. The sequence of ULIMIT and LLIMIT are optional, and they can both be omitted if not used.

When defining tunnel limits in a CSV file, you must separate the coordinates using a comma (,) and define decimal values using a decimal point (.).

Example:

- ULIMIT
- 100,10.0
- 200,20.1
- 300,30.2
- LLIMIT
- 100,1
- 200,2
- 300,3

7.4.5 Point Limits

When selecting point limits, the settings will be shown as illustrated in the following figure.



Figure 81 shows a dialog box for setting point limits. It contains three input fields: 'X' with the value '1,2520E+3', 'Minimum Y' with the value '-35,0000E-3', and 'Maximum Y' with the value '35,0000E-3'. To the right of these fields are two buttons: 'Insert' and 'Delete'.

Figure 81: Point limits

Point limiters are a special case of a tunnel limit, where there is only one abscissa point in the limiter.

7.5 Key Parameters

Key parameters are actually a technique for applying a post-processing function to a measured vector result. In many situations, simple parameters are easier to interpret and test than a curve.

Specifying the Data Source

As for graph limits, the data source for the key parameters must be specified. The source may be different from the source selected for graph limits, but again only one source is available to the key parameter functions. Specify the data source in the Data Source tab illustrated in the following figure.

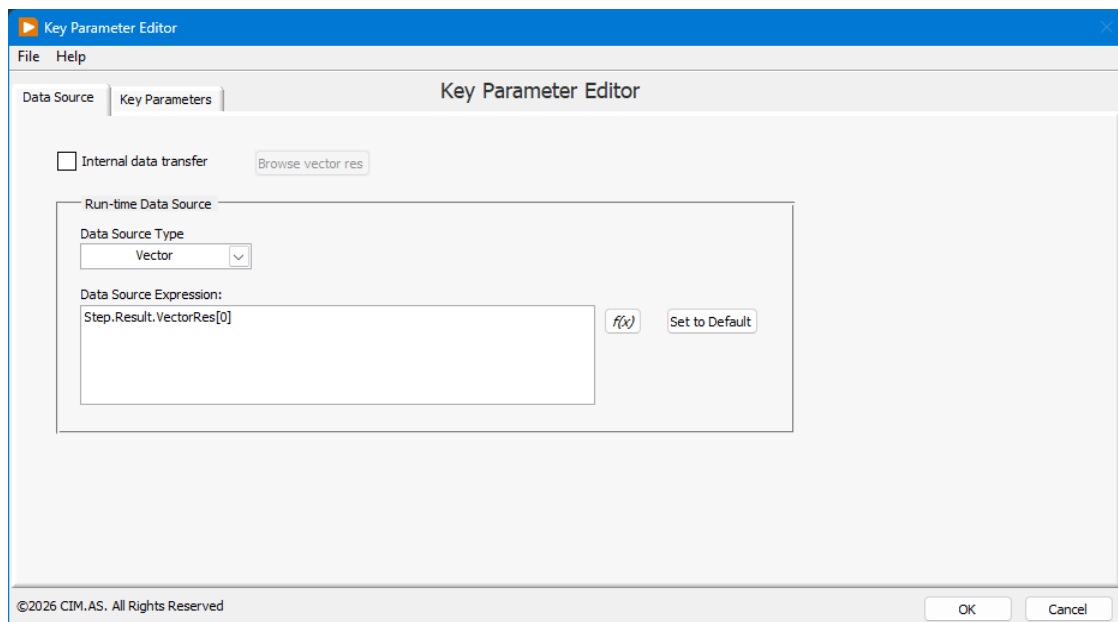


Figure 82 shows the 'Key Parameter Editor' dialog box, specifically the 'Data Source' tab. The dialog has a menu bar with 'File' and 'Help'. Below the menu bar are two tabs: 'Data Source' (selected) and 'Key Parameters'. The main area contains a checkbox for 'Internal data transfer' with a 'Browse vector res' button next to it. Below this is a section for 'Run-time Data Source' which includes a 'Data Source Type' dropdown menu set to 'Vector'. Underneath is a 'Data Source Expression' text box containing the text 'Step.Result.VectorRes[0]', with a 'f(x)' button and a 'Set to Default' button to its right. At the bottom left is the copyright notice '©2026 CIM.AS. All Rights Reserved', and at the bottom right are 'OK' and 'Cancel' buttons.

Figure 82: Key Parameter Editor. Data source tab.

On this tab, the data sources for the run-time execution can be specified. Besides that, data can be read from the internal data transfer. Refer to LimiTEST Data Source for a detailed description of data source.

If a waveform type or waveform array type is used, the expression must return a value of type NI_AnalogWaveformType or an array of NI_AnalogWaveformType, respectively.

If a vector type or vector array type is selected, the expression must return a value of type `Mx_Vector` or `mx_VectorArray`, respectively.

Specifying Key Parameters

Once the data source is selected a number of key parameters can be specified using the Key Parameter Editor.

The Key Parameter Editor has the following entries:

Name: Can be any description of the measurement.

Function: Select the key parameter function to use

Min, Nom, Max: Specify the minimum and maximum limit for the result. This can be a TestStand expression that evaluates to a number. Min and Max can also refer to the nominal value by using the following functions:

- dB10
- dB20
- ppm
- %
- +rel
- -rel

An example of this is illustrated in the following figure.

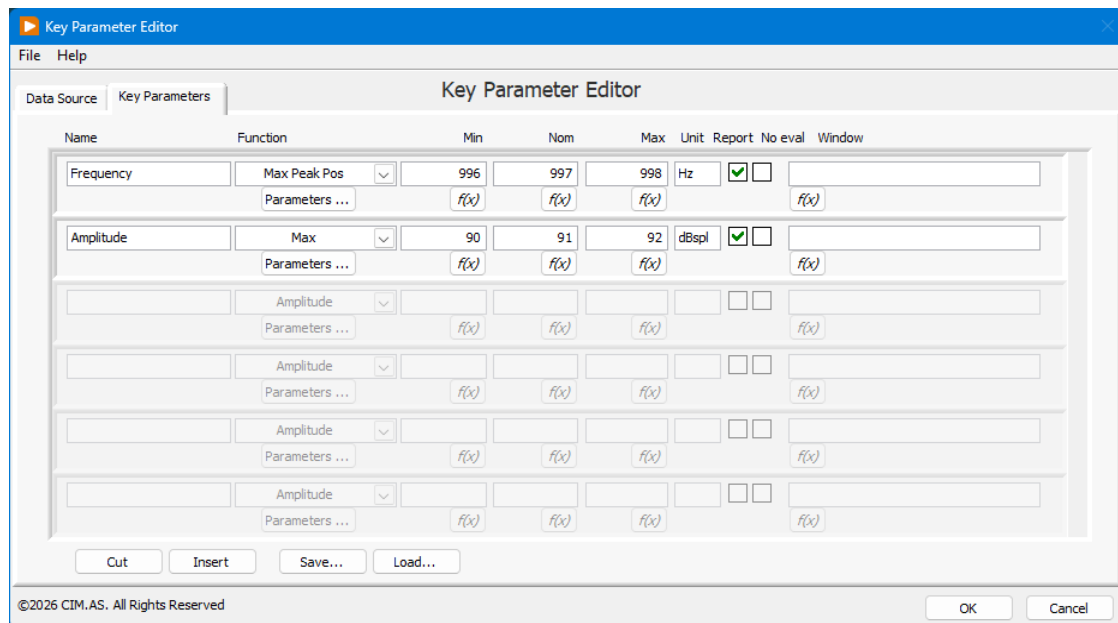


Figure 83: Key parameter editor

Unit: A text label that can be used for reporting. If you begin the unit string with an SI prefix, LimiTEST scale the values in the Min, Nom, and Max fields to the prefix during evaluation. If you do not want LimiTEST to scale the Min, Nom, and Max fields to the specified unit, include a space at the beginning of the unit string to use the unit as a label. For example, if the Min field contains 3.5 and you specify mV without a space for the unit, LimiTEST evaluates the Min field value as 3.5 mV, or 0.0035 V. If the Min field contains 3.5 and you specify mV preceded by a space, CATS evaluates the Min field value as 3500 mV, or 3.5 V.

Rep: Checkmark to use if the key parameter should be included in the report. This functionality is now also included in TestStand.

No eval: No evaluation can be set if the key parameter should be calculated but the final result should not be included in the overall Pass/Fail result.

Window: Defines which part of the measured data will be used for the calculation. The range of the window must be specified in the same unit as the x-axis of the waveform. Normally this is time or frequency. A TestStand expression that evaluates to a string of the below described format can be used for the window parameters.

You can use a colon (:) to separate the values that compose a range and a semicolon (;) to separate multiple ranges.

Examples are:

- Range: 0:200e-3 (0 to 200 msec)
- Single point: 500 (500 hertz)
- More windows or points: 10E-3;200E-3;300E-3
- TestStand Script: StationGlobals.myWindow (where myWindow could be "0:200E-3")

7.6 Status Evaluation

The Graph Limits and Keyparameter functions store the results in the context tree in the following path:

- Step.Result.mxLimiTEST

Each function has an overall result stored in Status (Pass or Fail):

- Step.Result.mxLimiTEST.GraphLimits.Result
- Step.Result.mxLimiTEST.Keyparameters.Result

In order to make a complete status for the step, the LimiTEST test type contains a default status expression displayed in the following figure.

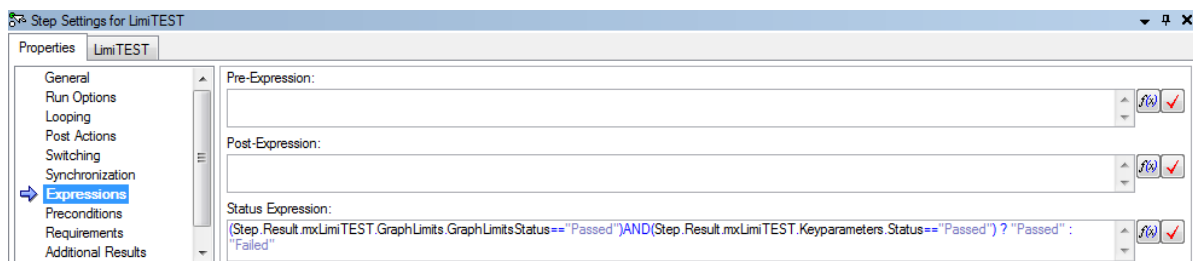


Figure 84: Default status expression

The expression will insert "Passed" in Step.Result.Status if both the Graph Limits and Keyparameters passed.

The expression in the step can be modified if other conditions are needed for the final status.

7.7 Additional LimiTEST Steps

Inserting a stand-alone LimiTEST step and using the internal data transfer can also test the value of other vectors from a previous measurement step.

This is a practical feature because it can save a number of time-consuming data acquisitions. Based on a single acquisition CATS will calculate a number of result vectors that can be evaluated in the following LimiTEST steps.

For stand-alone LimiTEST steps the default run-time data source is the result of the previous step: RunState.PreviousStep.Result.VectorRes[0].

This means the by default it will evaluate the primary vector of the previous step.

When using more than one LimiTEST step with CATS, use the relevant data source tabs to change the data source to refer to the desired vector using the internal data transfer.

An alternative solution is to let the measurement step store its results in a TestStand variable but remember that the direct data transfer has improved performance and is more user friendly in browsing for the desired result vector.

8 LiveView

This test step is used in a test sequence to show a pop-up window that continuously displays a vector measurement result on a graph.

The Live View test step is practical for positioning and adjusting the device under test, such as a hearing aid in a test chamber. If a tunnel limit is defined for the test step it will automatically appear on the graph in the pop-up window.

Most applications require a stimulation signal output to the DUT, which you can achieve by inserting a step for the stimulation signal immediately before the Live View step. You can then use any CATS measurement function that returns a vector result. Live View returns the last vector result returned by the measurement function. This vector result is verified against the limits programmed for the test, and the outcome of this test is the pass/fail result of the step.

The front panel is divided into a "Measurement Setup" section for defining the measurement function at the left, and a "Viewer Setup" to the right defining the appearance of the graph representing the measurement result.

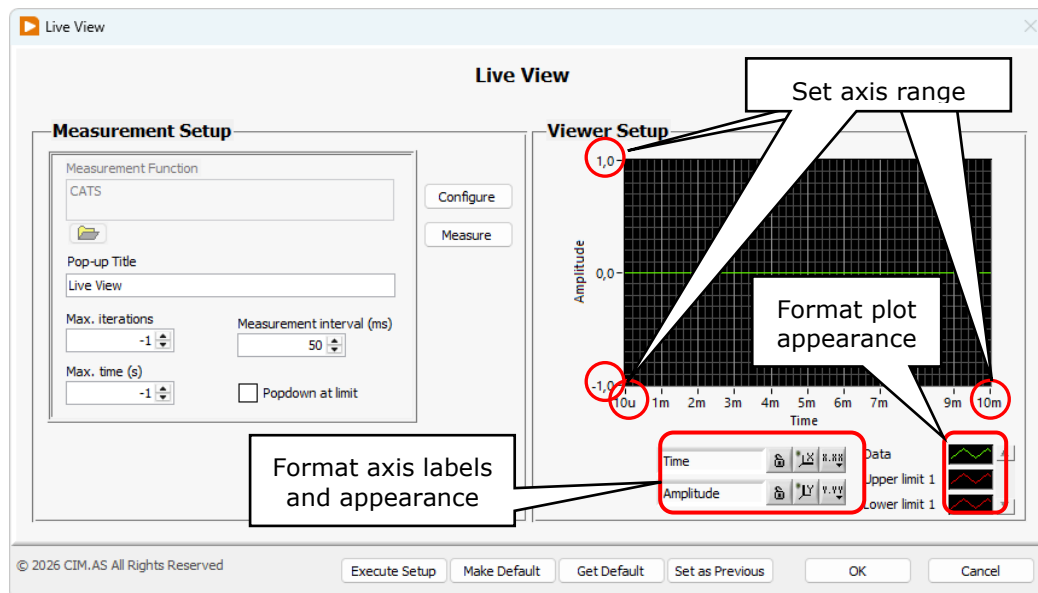


Figure 85: LiveView configuration panel

The measurement function is locked to CATS.

When selecting the Configure button, the configuration window of the measurement function is opened, and the measurement parameters can be specified.

Selecting the Measure button calls the measurement function and opens the pop-up window. Now the window can be resized and moved to the position where it will open when the test step is executed in the test program.

Other options can be set by formatting the "Viewer Setup" graph template on the right-hand side of the configuration panel of Live View.

8.1 Measurement Setup

These settings are used to select the measurement function that produces the result vector to display in the pop-up window and set the overall behaviour of the pop-up window and measurement repetition.

Measurement Function

The measurement function is locked to CATS. If you run a test sequence created with AudioMASTER 6.2 or earlier that calls measurement functions other than AudioMASTER, you will receive an error message.

Max. Iterations

Specifies the maximum number of iterations of the measurement function. -1 means no limit. That is; run until the OK button is pressed.

Max. Time

Specifies the maximum time in seconds to run the measurement function. -1 means no limit. That is; run until the OK button is pressed.

Pop-up Title

Specifies the title that will be displayed in the title bar of the Live View viewer window.

Pop-down at Limit

Determines if the Viewer window is popped down (and the test step ended) when the time/iteration limit is exceeded.

If the state of the check mark is unchecked, the Viewer window will be displayed until the user presses the OK button.

If the check mark is checked, the Viewer window will be popped down, when the time or iteration limit is exceeded, and the test sequence will continue.

Measurement Interval (ms)

Specifies the time period in milliseconds at which to run the measurement function. Zero means run without pauses.

Configure

Configure the measurement function. Pops up the measure function window and allows the user to configure the window.

Measure

Pop up the viewer window, run the measurement function and display the measured result in the window. The measurement function is run continuously, or if a maximum limit has been specified until the limit is exceeded. During this time, the viewer window may be moved and resized. When the window pops down, the final position and size of the window is remembered, and in run mode the window will appear with the remembered position and size.

8.2 Viewer Setup

The Viewer setup allows the user to specify how the measured vector result is presented to the user in the Viewer. Use the controls marked in the following figure to format the appearance of the graph to suit the type of measurement data.

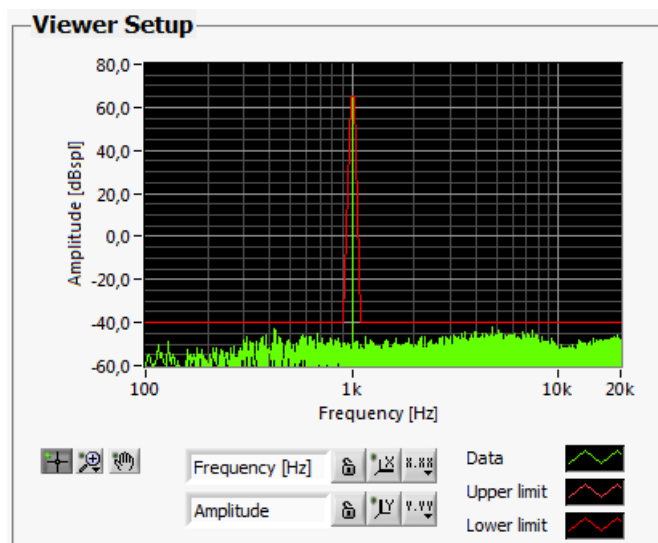


Figure 86: Viewer setup

Refer to Measurement Setup for a detailed description of the controls and menus available for controlling the appearance of the graph.

When using the Measure or Execute setup functions the size and location of the pop-up window can be adjusted.

Pop-up Window Examples

For FFT spectra the graph can be formatted as a bar-plot with no interpolation, fill baseline to $-\infty$ and no point.

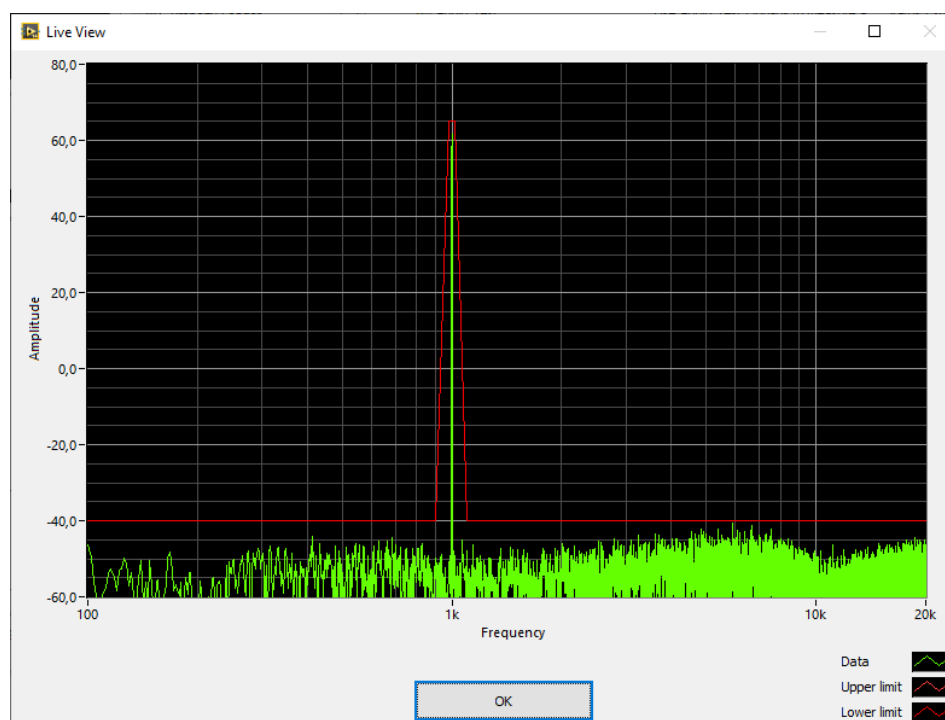


Figure 87: LiveView FFT graph window example

For frequency response type of measurements, a line plot is more suitable and here the point style has been set to emphasize the measurement values.

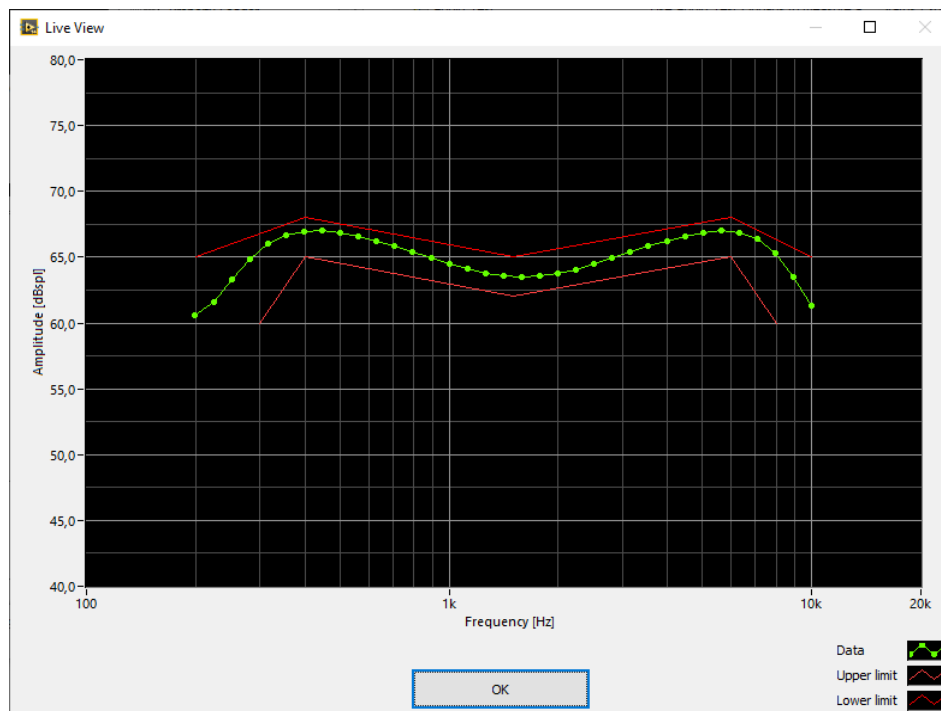


Figure 88: LiveView response graph window example

8.3 Test Function Buttons

The buttons located in the lower right corner of the Live View configuration window are keys that control the configuration of the test function. The functions are described in the following sections.

Ready/Busy

This indicator shows the state of the test function.

- Ready: Live View is ready for user-action.
- Busy: Communication is active.

Execute Setup

Run the measurement function continuously and display the result live to preview the pop-up window.

Make Default

Store the current configuration as default for use by "Get Default".

Get Default

Get the default configuration as saved by the latest executed "Make Default". If no "Make Default" operation has been executed previously, the configuration is set to a factory default.

Set as Previous

This button presently has no effect.

Cancel

Leave the setup mode and skip any changes made to the configuration.

OK

Leave the setup mode and update the current configuration with the changes made.

8.4 Results Returned to TestStand

The vector result returned by the Live View test function is the last vector result returned by the measurement function. This vector result is verified against the limits programmed for the test, and the outcome of this test is the pass/fail result of the step.

9 Calibration

In electrical measurement systems that use high quality data acquisition boards with accurate voltage reference, there is typically no need to use the internal CATS calibration system. In such systems the calibration system is simply disabled on each channel (refer to Channel Configuration for details).

For measurements that involve external transducers, e.g. loudspeakers and microphones, the calibration system enables users to do measurements that are calibrated to physical units.

The sections within this book describe the modules of the calibration system and their features:

- **Calibration tool** – stand-alone tool for creating and maintaining calibration data in a calibration file.
- **Calibration manager** – test function that checks calibration intervals and make calibration data available to test functions.
- **Copy calibration data** – test function that streamline calibration procedures by re-using calibration data.

After inserting a Calibration manager or Copy calibration step it is configured by selecting Edit configuration on the context menu. The context menu on this type of steps also includes an option to open the Calibration tool.

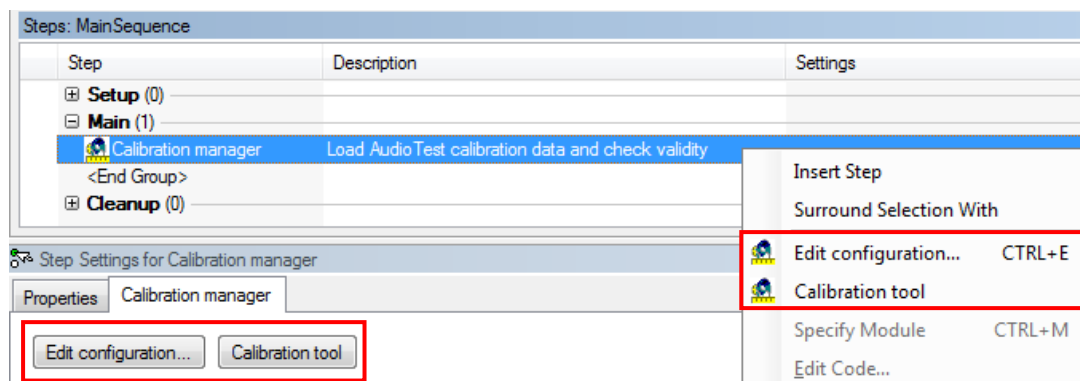


Figure 89: Context menu of Calibration manager step

The Calibration tool and Calibration manager includes a Calibration wizard that guides the user through the calibration procedure.

9.1 Calibration Tool

This is an engineering tool for creating and maintaining calibration data in a calibration file. Calibration data is used by the CATS measurement functions to convert between the internal voltage representation and external physical quantities.

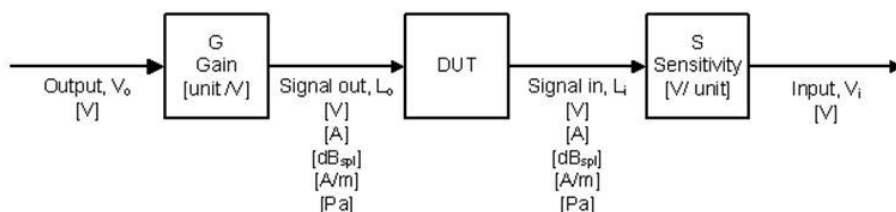


Figure 90: Calibration calculation overview

Several calibration data items can be stored in one calibration file. The file name and path of the calibration file is specified in the Windows registry and is loaded automatically when the calibration tool is started.

9.1.1 Using the Calibration Tool

One calibration data item contains conversion factors for one input and one output channel, including some channel settings and parameters as shown in the figure below.

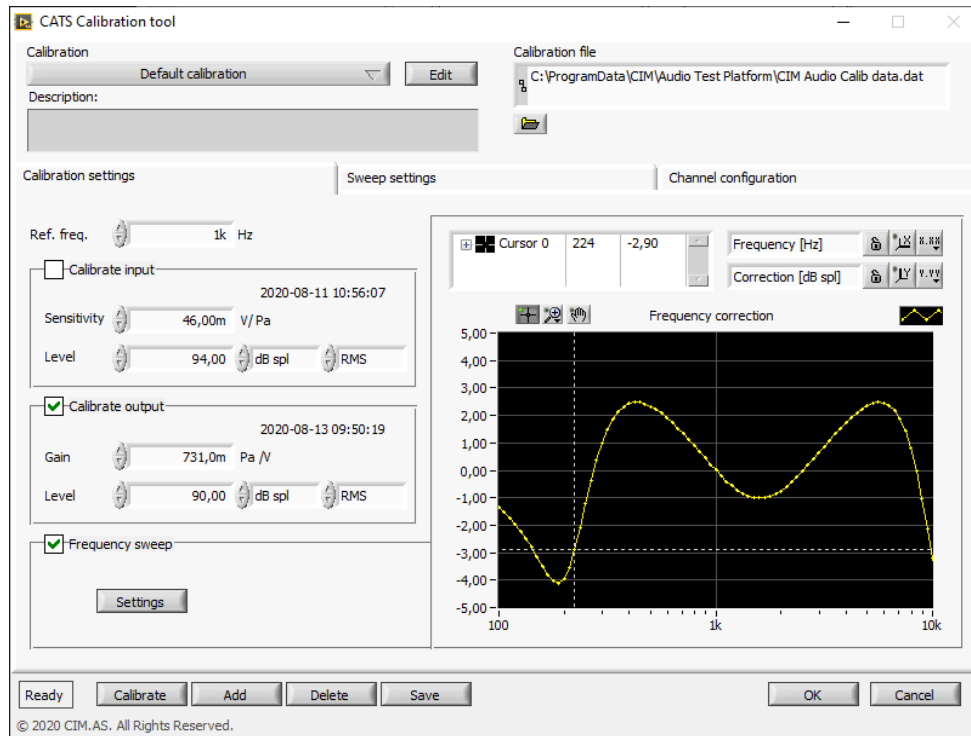


Figure 91: Calibration tool functions

NOTE: The three check boxes to enable or disable calibrations steps is not part of the calibration data but are used only to select which steps to execute when selecting Calibrate.

Use the calibration data selector to browse through the calibration data items that are stored in the file. The calibration values and settings can then be viewed in the controls and on the graph.

Calibration data items can be added to and deleted from the selector list by the Add and Delete buttons at the bottom of the window.

The calibration tool works like a sketchpad where the data can be modified, and the calibration procedure can be executed, and the results review without changing the calibration data file. The calibration file is only updated when OK or Save is selected. Selecting Cancel discards all changes and leaves the file unchanged.

When a new calibration data item is added the Edit dialog is opened to easily type a name and description. The new data is a default calibration item where all values are displayed in red colour to show that they are marked as invalid.

The valid flag on calibration values is a type of reminder for the status of the value. The valid flag is reset when data is created or modified and set as a result of a successful calibration or when explicitly accepted by the user.

9.1.2 Calibration Procedure

The CATS calibration procedure consist of three steps that are controlled by a calibration wizard:

1. **Input calibration**
Measures the input level of an external reference signal with known amplitude and frequency.
2. **Output calibration**
Adjusts the output level until the specified level and frequency is measured on the input.
3. **Output frequency correction**
Sweeps the output signal over a specified frequency range at constant output voltage and calculate corrections from the measured input.

It is possible to adapt the wizard to specific applications by creating a HTML file for each step in the calibration procedure. The files are located in the CATS data folder and the names are listed in relevant sections below.

Because the input calibration is used during output calibration to adjust the output level, the signal level unit must be the same. In hearing aid testing this is typically not a problem because the DUT is stimulated with a sound pressure and the output is also a sound pressure. In systems where the DUT itself converts an input signal in one unit to an output signal in another unit, the input must be calibrated with the new unit.

Each step of the procedure can be executed separately provided that the calibration values of preceding steps are valid and reflect the actual properties of the channels.

9.1.2.1 Verifying Invalid Calibration Data

When the calibration procedure is started the valid flags are checked versus the selected calibration steps. If an invalid value exists for a step that is not selected to execute in the procedure, one of the following dialogs are shown:

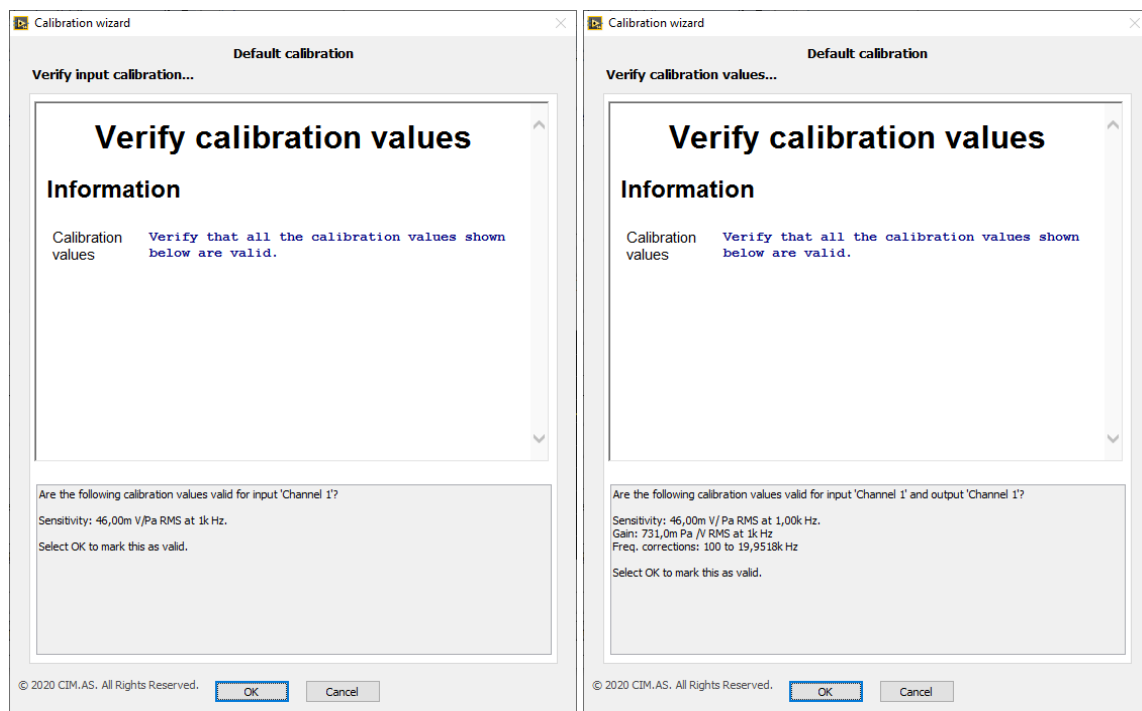


Figure 92: Calibration tool - verifying invalid calibration data

Modify 'CIM Calib – default calib verify.htm' to customize the info frame in this dialog.

The value must be accepted as valid by selecting OK in order to continue the calibration procedure. Selecting Cancel exits the procedure without changing any values and returns to the calibration tool.

9.1.2.2 Input Calibration

Before the measurement of the input sensitivity can start the user must connect the external calibrator to input transducer. The expected signal properties of the external reference signal source are shown to assist the user. The reference signal source must match these values exactly to ensure that the sensitivity is calculated correctly.

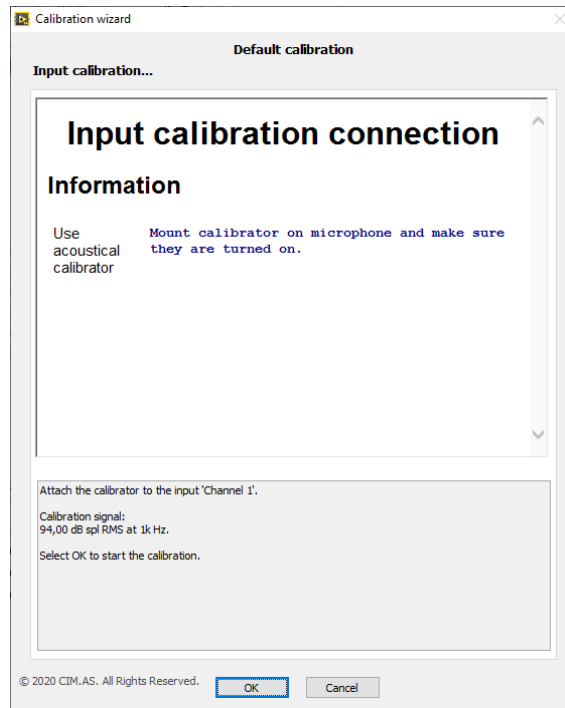


Figure 93: Calibration tool - preparations for input calibration

In acoustical systems this typically means fitting the acoustical sound level calibrator or pistonphone on the microphone, while in electrical systems it typically means connecting a reference generator to the input terminals.

Modify 'CIM Calib – default input connect.htm' to customize the info frame in this dialog.

When you select OK, the actual measurement is started, and the status is shown:

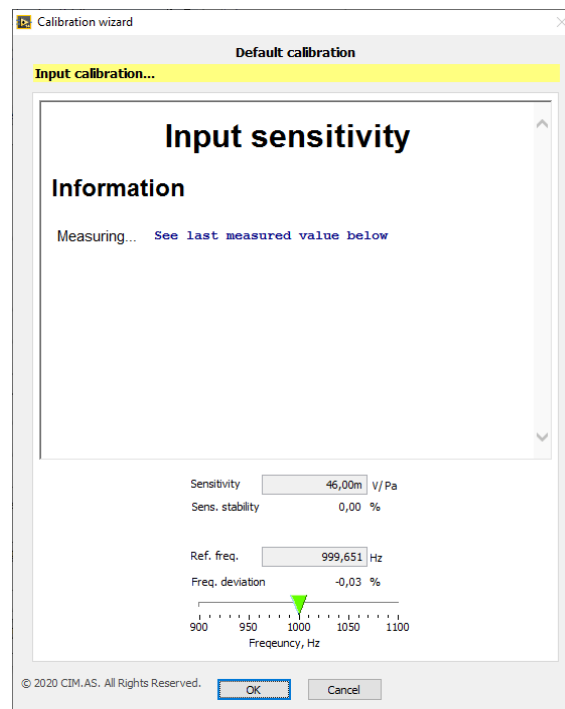


Figure 94: Calibration tool - input sensitivity measurement status

The values are updated repeatedly while the measurement runs, to show when the system is stable. Selecting OK accepts the new value, while Cancel exits the procedure.

Modify 'CIM Calib – default input meas.htm' to customize the info frame in this dialog.

9.1.2.3 Output Calibration

Before starting the output gain measurement, the user must ensure that the input transducer is placed correctly to measure the reference signal from the output.

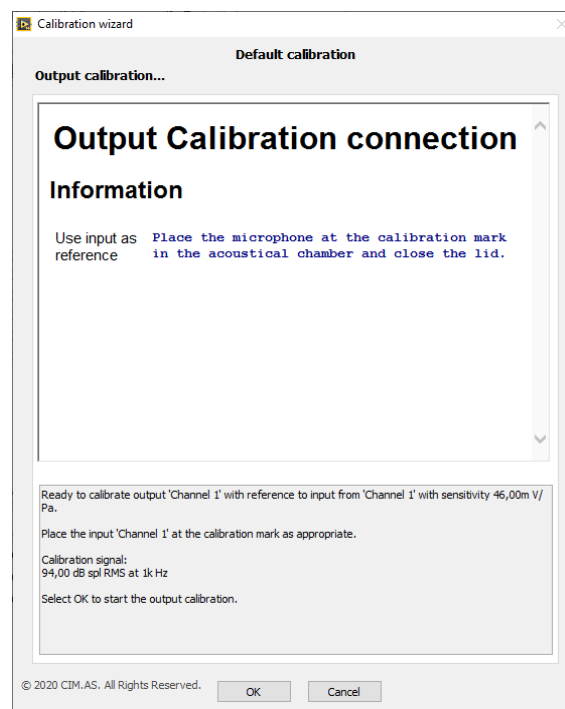


Figure 95: Calibration tool - preparations for output gain measurement

In typical acoustical systems, this means placing the microphone at the reference mark in the acoustical chamber, while in an electrical system the output terminals would typically be connected directly to the input terminals.

Modify 'CIM Calib – default output connect.htm' to customize the info frame in this dialog.

When you select OK, the actual measurement is started, and the status is shown:

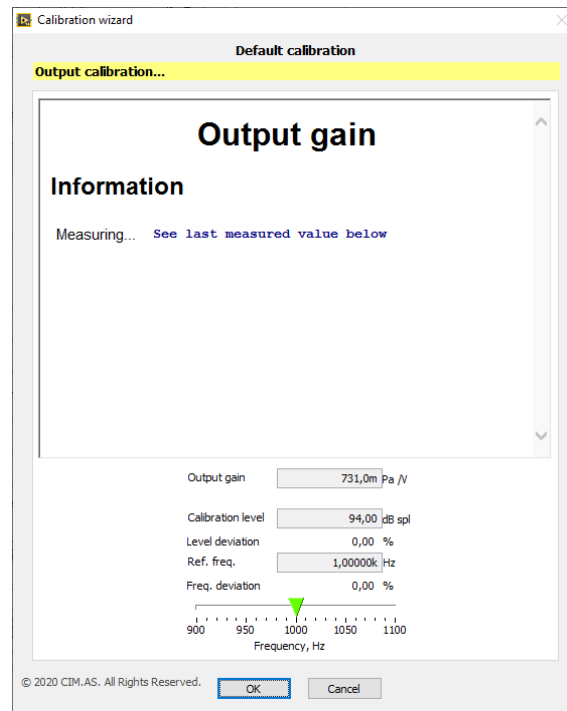


Figure 96: Calibration tool - output gain measurement status

The values are updated repeatedly while the measurement runs, to show when the system is stable. Selecting OK accepts the new value, while Cancel exits the procedure.

Modify 'CIM Calib – default output meas.htm' to customize the info frame in this dialog.

9.1.2.4 Output Correction Calibration

This step relies on running just after the output gain measurement. If the step is to run alone the user must prepare the system as described in Output Calibration.

No status is shown during frequency correction measurement and the wizard closes automatically when the sweep ends, and the results are shown in calibration tool window.

When frequency corrections are measured in a re-calibration procedure by Calibration manager, the result is displayed for accept by the user.

Modify 'CIM Calib – default accept.htm' to customize the info frame in this dialog.

9.1.3 Calibration Data Details

The calibration data contains both the conversion factors for input and output and also the parameters needed by the calibration procedure. In the calibration tool, tabs and frames group the calibration data setting.

The main calibration data is located on the Calibration settings tab, while frequency correction settings and channel configuration settings are located on other tabs.

The upper main frame contains input setting and the lower frame contains output settings. Each frame is further divided to emphasize the individual calibration steps and separate the calibration values from the parameters.

'Ref. Freq' is a common parameter that specifies the frequency of the external signal used for the input calibration. Also this value is used for the output calibration and is the reference for the output correction sweep.

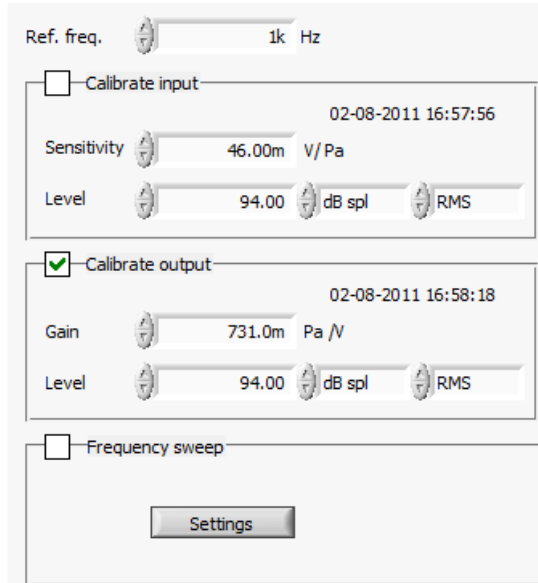


Figure 97: Calibration data

The graph to the right of the data on the above figure shows the frequency corrections. Internally the corrections are stored as a unit free factor relative to the reference gain value. On the graph, the correction factors are shown as an additional amount to the reference gain in the selected unit.

9.1.4 Calibration Parameters

Input Calibration Parameters

- **Sensitivity**
Is the conversion value of the input transducer. The selected calibration level unit determines the sensitivity unit. The sensitivity is typically measured by applying an external calibrator with the specified reference frequency and level; or the value is typed in from the transducer data sheet.
- **Level**
Specifies the reference level and physical unit of the external calibrator.

Output Calibration Parameters

- **Gain**
Is the conversion value of the output transducer. The selected calibration level unit determines the gain unit.
- **Level**
Specifies the reference level and physical unit of the output signal. Because CATS use the reference frequency and input sensitivity for calculating the gain, the output unit must be the same as input unit when performing the output calibration.

Output Correction Sweep Parameters

The parameters for correction measurement are located on the Measurement settings tab to allow addition of new methods in the future.

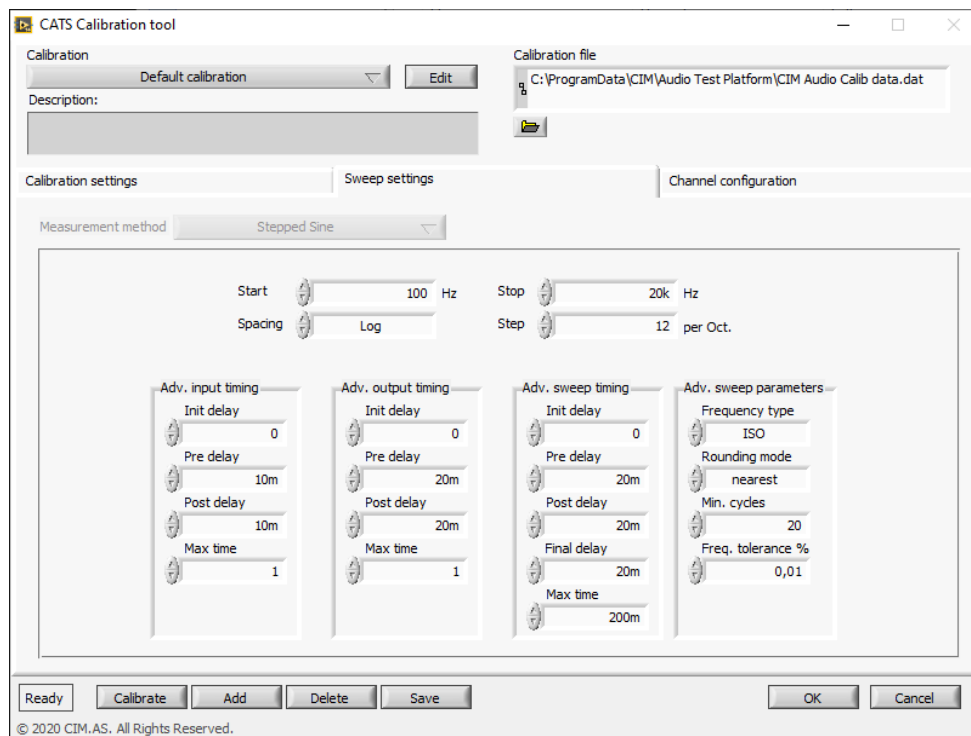


Figure 98: Calibration tool - sweep settings

The stepped sine sweep method includes the following settings:

- **Start/Stop**
Specifies the frequency range in Hz for the output correction sweep.
- **Spacing**
Selects linear or logarithmic frequency spacing for the sweep.
- **Step**
Specifies the frequency resolution for the sweep. In linear mode steps are specified in Hz, and in tones per octave ($1/N^{\text{th}}$ octaves) in logarithmic mode.

The advanced timing parameters at the bottom can be used to adjust the measurement timing for the calibration measurements. Normally these values should be left at their default values and only changed by an expert user if there is a specific reason.

9.1.5 Channel Configuration

On this tab the setting for the input and output channel is specified, including sample frequency. It is also possible to reset the devices in case of setting invalid parameters that may bring NI-DAQmx into a state of error.

The 'Auto mode accept limits' are used to specify criteria for accepting calibration measurements in the automatic mode issued by the 'Calibration manager'.

The accept limits specify acceptable variation in % on the calibration value, from iteration to iteration. Measurements are stopped if they are not accepted after the maximum number of iterations.

One set of values applies to input calibration and one set applies to output calibration.

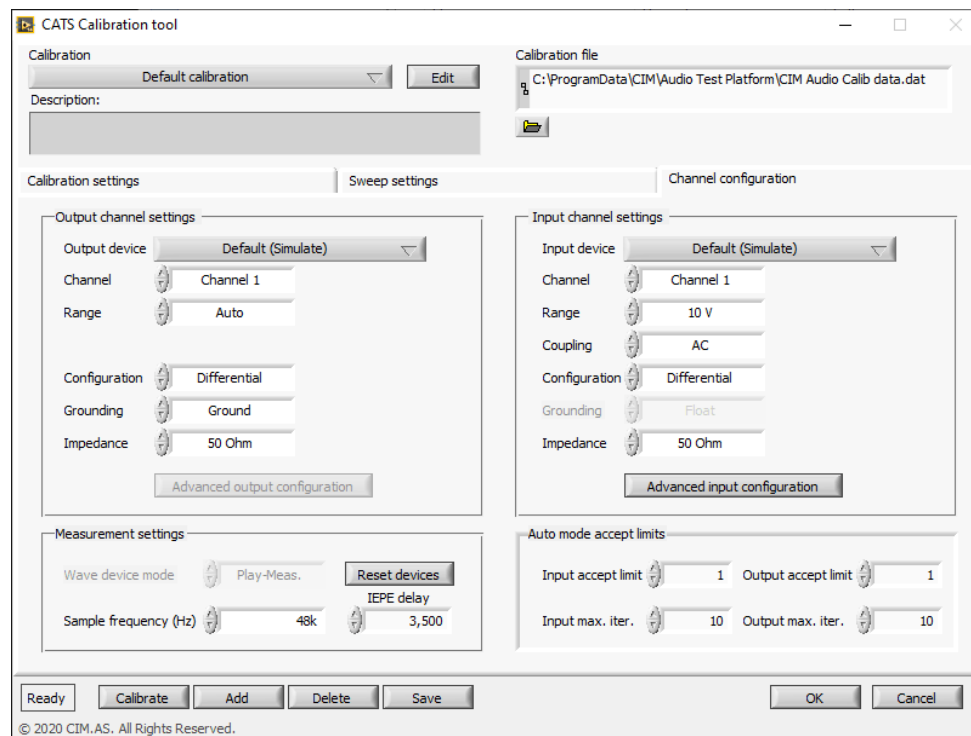


Figure 99: Calibration tool - channel configuration

- **Output device/Input device**
Select which hardware platform to use for output and input respectively. Normally the same platform is used for both.
- **Channel**
Selects which channel to use when measuring the input from the transducer.
- **Range**
Selects the hardware output and input range to use for the measurement. The values are specified in Vpeak as maximum voltage that the hardware can handle. It is important that the range selected carefully to be suitable for the test signal level and type of transducer.
- **Configuration, Grounding, Impedance**
These controls must be set according to the transducers in use and capabilities and the hardware capabilities to avoid ground loops and free-floating inputs.
- **Advanced configuration**
Depending on the selected hardware platform the 'Advanced configuration' opens a panel that allows setting special features, fx. turning on build-in IEPE-power.
- **Sample frequency**
Specifies the sample frequency. This is common to both input and output.
- **IEPE delay**
Specifies an initial delay for measurements when 'Advanced configuration' has been set (presumed to be IEPE power on). The calibration measurement is delayed allowing the IEPE microphone to stabilize after power up.
- **Auto mode accept limits**
These vales specifies the acceptable variation limits in % and maximum number of iterations for input and output calibration respectively. They are used in the automatic re-calibration procedure of the calibration manager.

9.2 Calibration manager

This setup function is used to select which calibration data that should be available to the CATS measurement functions in a test program and validate the calibration interval.

When the calibration manager runs, it reads the specified calibration file and checks the calibration date of the named calibrations in the validation list.

If the calibration date is within the specified calibration interval, the calibration data in the file is accepted. Otherwise, the selected Action if expired is executed.

Any calibration data in the file that is not listed in the validation list is handled according to the selected Default action.

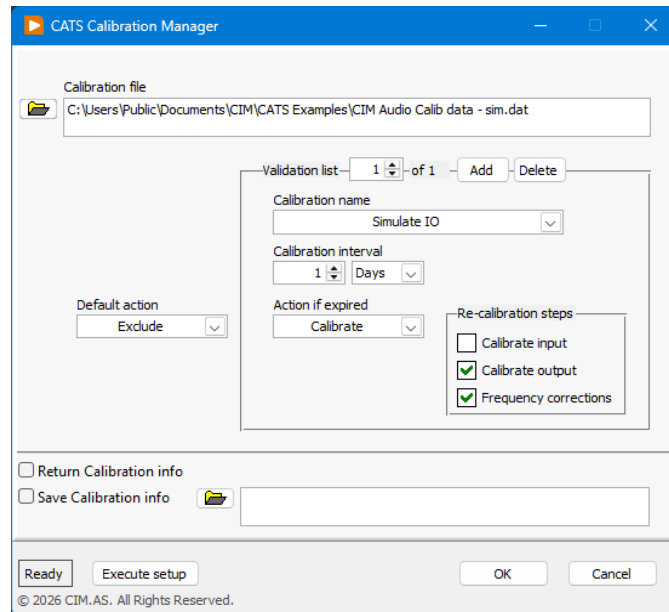


Figure 100: Calibration manager – settings

- **Calibration file**
This is the path and filename to the calibration file. Use the browse button to browse for another file. If a relative path is specified, it is relative to the user data path (C:\ProgramData\CIM\Audio Test Platform).
- **Validation list**
Contains the names and settings of calibration data to validate. Use the numeric control and *Add* and *Delete* buttons to browse and edit the list.
- **Calibration name**
Selects the name of the calibration data to validate. The list contains names of all calibration data in the calibration file.
- **Calibration interval**
Specifies the period of time that the selected calibration data is accepted as valid. The interval can be specified as *Hours*; *Days*; *Weeks*; *Months* or *Years*. Specifying a value of zero will always evaluate the calibration data as invalid.
- **Action if expired**
Selects what to do when the calibration data gets too old. The options are:
Exclude prevents expired calibration data to be used by the measurement functions.
Include makes the expired calibration data available to the measurement function without showing a warning dialog.
Warn shows a dialog to let the user select OK or Cancel to accept or reject the expired calibration data.
Calibrate starts the re-calibration procedure for the expired calibration data.
- **Re-calibration steps**
Select the specific step to execute during the re-calibration procedure.
- **Default action**
Selects what to do with calibration data that are found in the file but is not listed in

the validation list. The options are the same as in the validation list, except that calibrate is not available as default action.

- **Return Calibration info**
Select this option to return all validated calibration information in XML format to TestStand. The XML string is stored in 'step.Result.CalibInfoXML'.
- **Save Calibration info**
Select this option to write all validated calibration information to the specified file in XML format.

9.2.1 The Recalibration Procedure

The recalibration procedure uses the same basic measurement functions as the calibration tool, but with a simplified user interface it is faster to use and prevents the user from changing any parameters of the calibration.

This means that the recalibration procedure provided by the calibration manger is suitable for operators in a production environment.

Input Calibration

If the input calibration step is selected, the user is prompted to connect the external calibrator to the input channel. The calibration parameters are shown in the window to guide the user through the procedure.

Selecting OK continues the procedure, while Cancel exits without changing the calibration data.

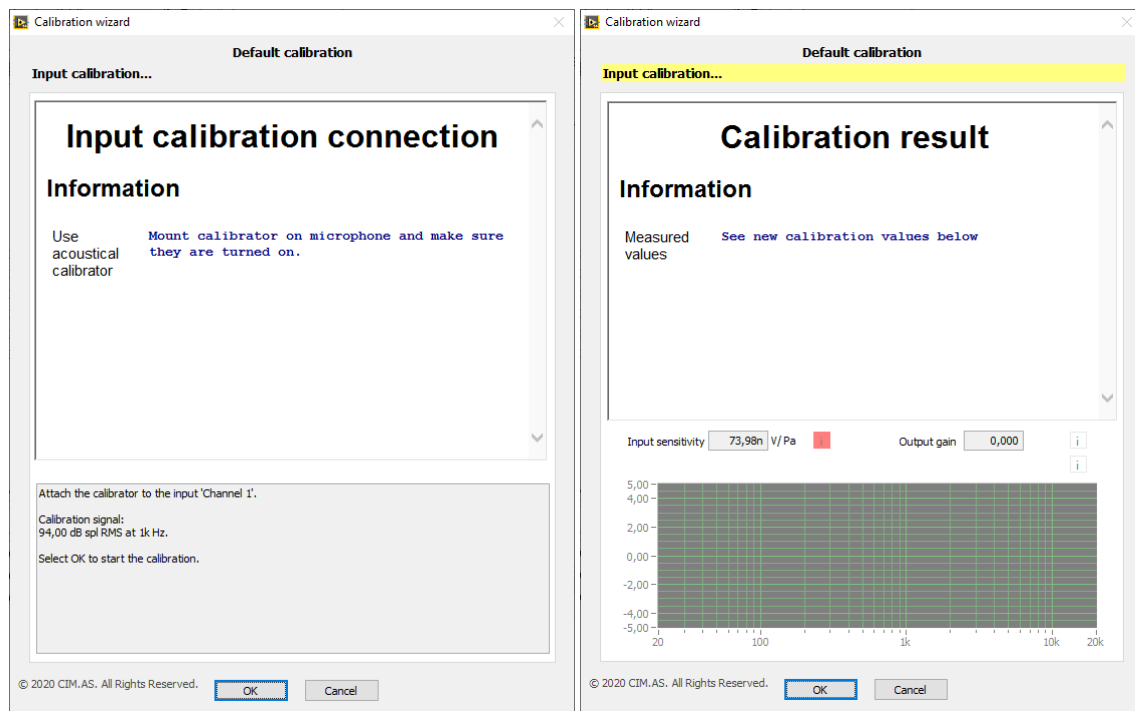


Figure 101: Re-calibration - input calibration step

While the input calibration runs, the status is shown directly in the result view window. Note the **[i]** indicator that is highlighted in red if the value is not in the expected range. Hover the mouse cursor over the indicator to get detailed information in the tool tip.

In re-calibration mode the actual measurement of the output calibration value is done automatically, and measurement is stopped when the level is stable, typically within $\pm 1\%$. Also, the measurement will stop when the maximum number of iterations is reached.

If the input calibration step is not selected, this step is skipped, and the procedure starts from the output calibration step.

Output Calibration

Because the output calibration relies on the input calibration, its validity is checked before the output calibration step is started. If the input calibration value is marked as not valid the user is prompted to accept the value. Selecting OK marks the value as valid and continues the procedure, while Cancel exits without changing the calibration data.

The procedure prompts the user to select OK before the output calibration is started. This allows positioning the DUT and closing the acoustical chamber before the measurement starts.

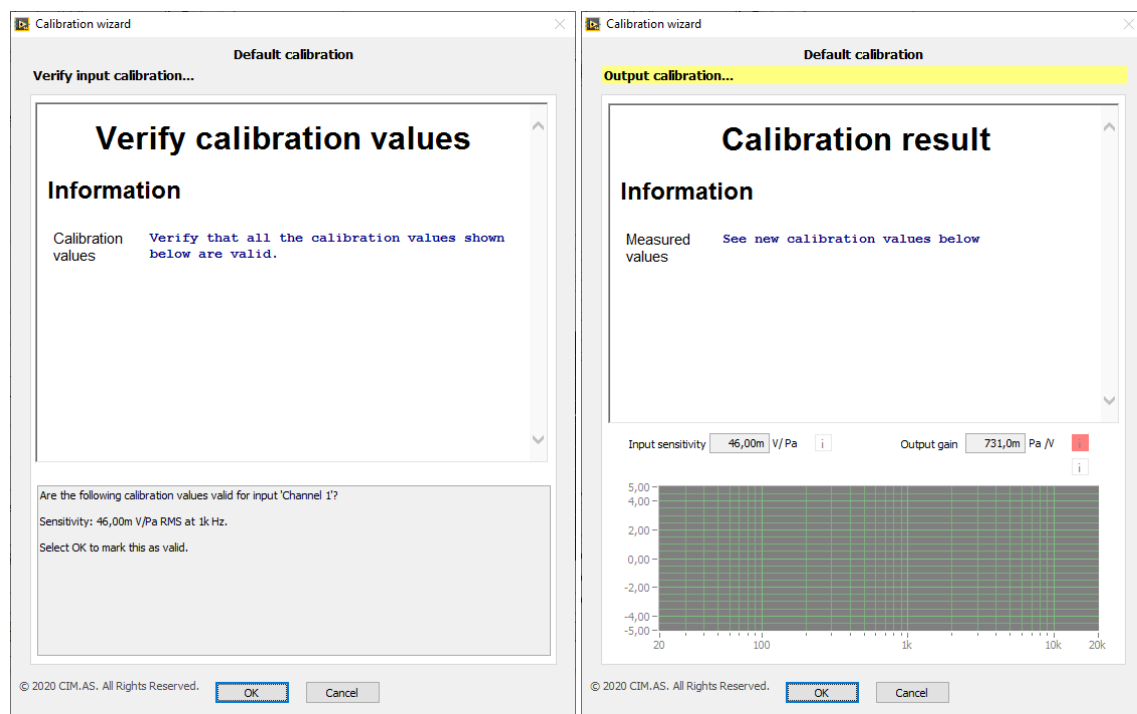


Figure 102: Re-calibration - preparations for output calibration

While the output calibration runs, status is shown directly in the result window. Note the **[i]** indicator that is highlighted in red if the value is not in the expected range. Hover the mouse cursor over the indicator to detailed information in the tool tip.

In re-calibration mode the actual measurement of the output calibration value is done automatically, and measurement is stopped when the level is stable, typically within $\pm 1\%$. Also, the measurement will stop when the maximum number of iterations is reached.

Output Frequency Corrections

If the frequency correction step is selected, the output signal is swept over the specified range with a constant level corresponding to the parameters and calibration values for input and output.

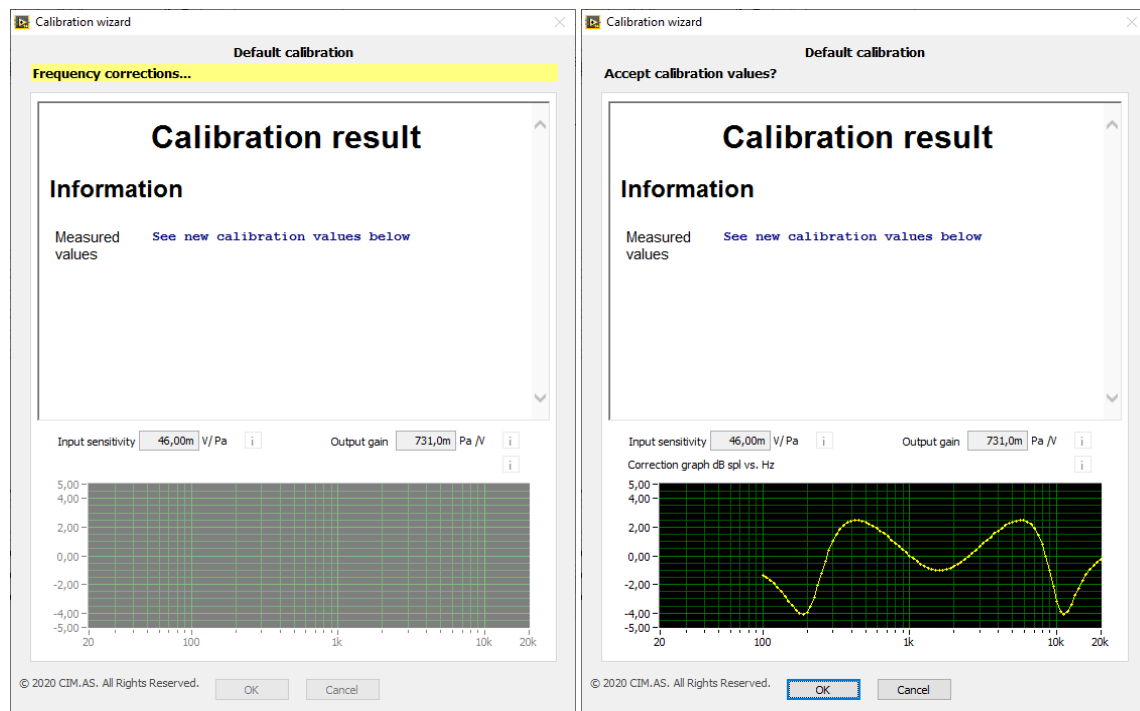


Figure 103: Calibration manger - frequency corrections measurement

There is no status display during the sweep.

Accept Calibration Values

Finally, the user is prompted to accept the calibration values. The OK button saves the calibration data, while Cancel leaves the data unchanged.

NOTE: Only the calibration dates of changed values or values that have been marked as valid are updated.

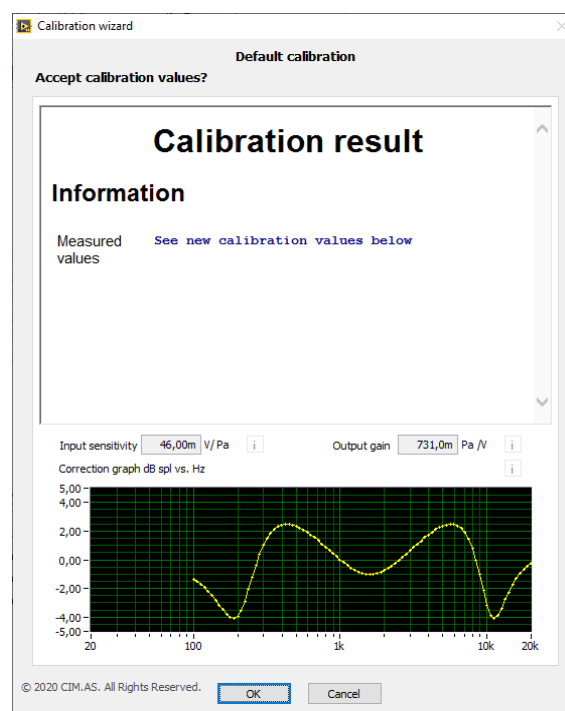


Figure 104: Re-calibration - accept calibration values

9.3 Copy calibration data

This test function is used to copy parts of one calibration data item to one or more items in the same CATS calibration file.

This feature is practical in a system, where the same transducer is used for different types of measurements using different units.

It is possible to create a list of copy actions that are executed in sequence.

Each copy action selects the name of a source to copy from in the drop-down box and selects which parts to copy.

Normally the 'Update other date values' must be selected, to ensure that the CATS calibration manager does not mark the calibration item as expired.

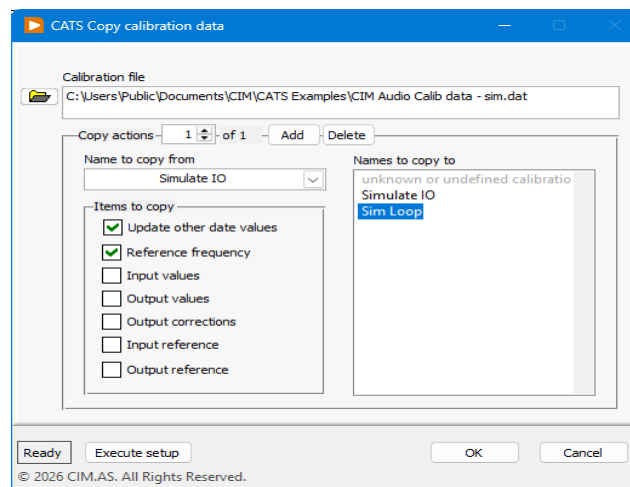


Figure 105: Copy calibration data

The new options 'Input reference' and 'Output reference' enables copying the reference level and units of the input and output calibrator respectively. This is practical to keep the actual calibrator level aligned in a system where the same calibrator is used for multiple signal paths.

In the 'Names to copy to' list box one or more destination can be selected.

10 Advanced CATS Features

This chapter describes various advanced features in CATS.

10.1 Modifying CATS parameters using LabVIEW VI's within TestStand

Using the LabVIEW VIs for CATS within a test step in TestStand, you can change the configuration of CATS measurement steps at runtime.

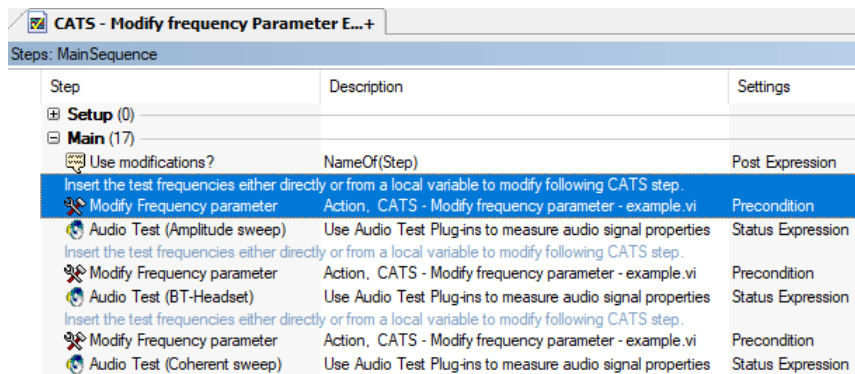
The VIs allows you to access and operate the configuration data that is stored as a numeric array on a given TestStand step. The numeric array contains a flat binary string that represents various types of clusters. All configuration data is stored in the numeric array on the test step because of the complexity of the CATS configuration. The configuration for an CATS test step in TestStand includes settings for hardware platforms and one of the measurement plug-ins, resulting in a configuration data structure that changes depending on the configuration.

NOTE: Because the complex structure of and dependencies between some configuration parameters might not be immediately apparent, ensure you change values in a meaningful way.

Example Sequence

To view an example of how to modify the configuration of the CATS test step in TestStand using the VIs, open the 'CATS - Modify frequency Parameter Example.seq' file, installed in the following location:

C:\Users\Public\Documents\National Instruments\TestStand 2014 (32-bit)\Components\StepTypes\CIM\CATS Parameter API\Examples



Step	Description	Settings
Setup (0)		
Main (17)		
Use modifications?	NameOf(Step)	Post Expression
Insert the test frequencies either directly or from a local variable to modify following CATS step.		
Modify Frequency parameter	Action, CATS - Modify frequency parameter - example.vi	Precondition
Audio Test (Amplitude sweep)	Use Audio Test Plug-ins to measure audio signal properties	Status Expression
Insert the test frequencies either directly or from a local variable to modify following CATS step.		
Modify Frequency parameter	Action, CATS - Modify frequency parameter - example.vi	Precondition
Audio Test (BT-Headset)	Use Audio Test Plug-ins to measure audio signal properties	Status Expression
Insert the test frequencies either directly or from a local variable to modify following CATS step.		
Modify Frequency parameter	Action, CATS - Modify frequency parameter - example.vi	Precondition
Audio Test (Coherent sweep)	Use Audio Test Plug-ins to measure audio signal properties	Status Expression

Figure 106: Modify parameters example sequence

The typical process for using CATS VIs within TestStand involves the following steps:

1. Launch TestStand at Start » All Programs » NI TestStand.
2. Add a CATS Setup test step to the test sequence.
3. Configure the CATS Setup test step.
NOTE: It is recommended to specify the basic configuration for the CATS test step, such as channels and result selections, using the CATS Configuration panel.
4. Add a LabVIEW Action step to the test sequence.
5. Develop a VI to modify specific parameters of the measurement plug-in that you determine you want to modify.
6. Adapt the connector interface to pass in the new parameters.
7. To pass data to the VI, point to a local variable or expression in TestStand that returns the desired values.

In this example the LabVIEW VI that modifies the CATS parameters looks like this:

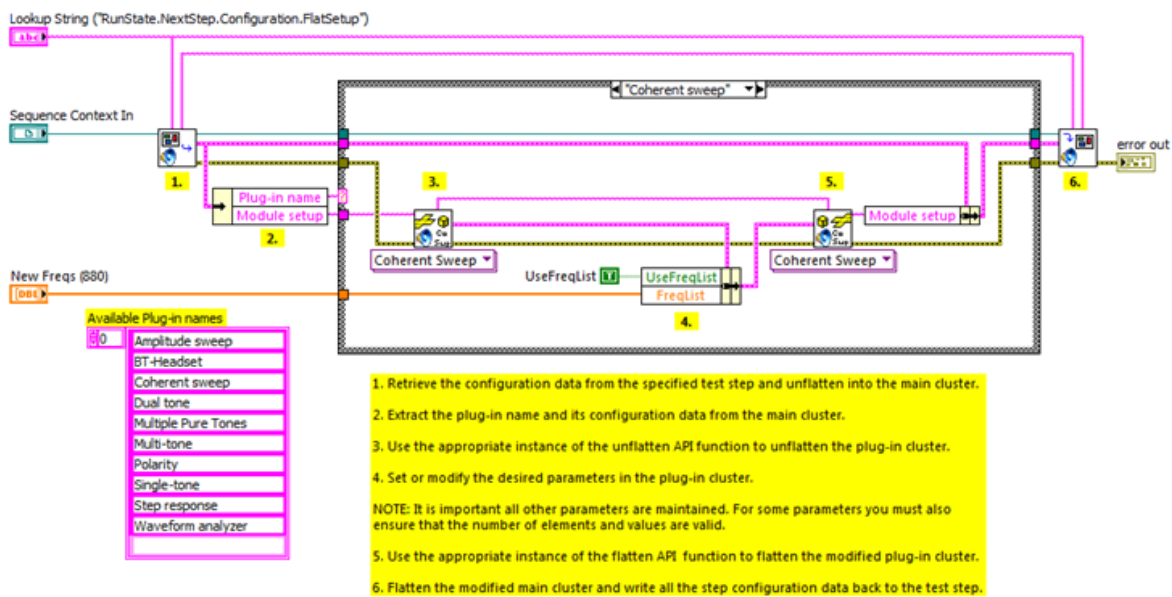


Figure 107: Modify parameters example LabVIEW code

Because the example VI can only be used within a TestStand step, this example VI is installed in the same folder as the example test sequence in the following location:

C:\Users\Public\Documents\National Instruments\TestStand 2014 (32-bit)\Components\StepTypes\CIM\CATS
Parameter API\Examples

You can copy and paste plug-in names from the list of available plug-in names into the case selector. If the plug-in name in the main cluster does not match the expected name in the case selector, an error is returned, and the configuration does not change.

10.2 Harmonic analysis in CATS

Many plug-ins provide harmonic analysis in form of various THD values, SINAD, individual harmonics and rub'n buzz.

This section outlines the details of calculations and result handling.

10.2.1 THD calculations

CATS calculates the following THD value types based on the test frequency and number of harmonics for THD.

$$THD = \sqrt{\frac{\sum Harmonics^2}{Detected^2}}$$

$$THD_{IEC} = \sqrt{\frac{\sum Harmonics^2}{Detected^2 + \sum Harmonics^2}}$$

$$THD + N = \frac{1}{SINAD} = \frac{RMS(Residual)}{RMS(Detected)}$$

The values are returned in the selected THD unit: None (unit free ratio), % or dB.

In a sweep...

10.2.2 Harmonics A and B

Each of the harmonic amplitudes are returned in the specified amplitude unit.

In a sweep the selected harmonic may be above the Nyquist frequency and cannot be detected.

In this case the CATS result vector will only include values for the test frequency where the harmonic is below the Nyquist frequency. Consequently, the Harmonic A and Harmonic B vectors may have different number of elements and be shorter than the Amplitude vector.

11 Hardware platforms

To handle the differences between different types of hardware, CIM Audio Test implements a number of different hardware platforms. This helps provide a simpler and more uniform user interface to set up various hardware properties.

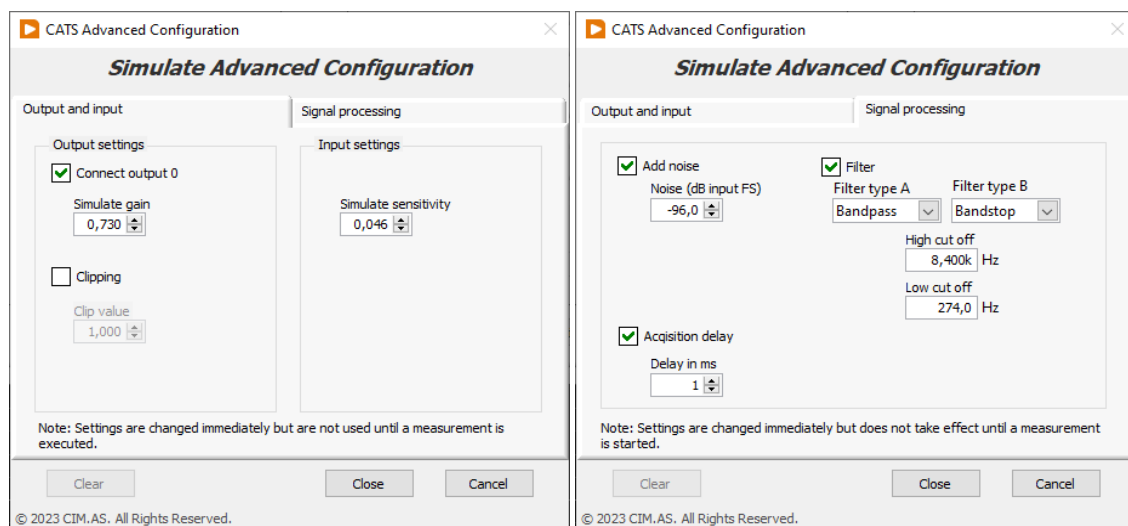
The following sections describe special configuration features that apply to each platform. These configuration panels are reached from the 'Channel configuration' tab in Audio Test configuration panel, as shown in section 5.

The configuration of which devices are connected to each platform is done in the Hardware Configuration Tool as outlined in section 3.2.

11.1 Simulate platform

The Simulate platform provides a convenient way test CATS without having any hardware available. The default settings provide a simulated loop back signal including gain, noise and response function etc.

On the Simulate platform configuration panel, you can change how the simulated output and input signals are processed:



Notes:

- When using the simulate platform at sample rates lower than 17 kS/s with the default filter enabled, you will get the message:
Error -1801 occurred at Start value too large in Get Waveform Subset.VI.
Ensure that the specified cut off frequency is below the Nyquist frequency or disable the filter.

11.2 DAQmx 446x Platform

This platform supports selected devices in the NI DSA 446x family:

- PCI-4461, PCI-4461
- PXI-4461, PXI-4462
- PXIe-4468

On the additional settings panel you can configure various properties of the hardware.

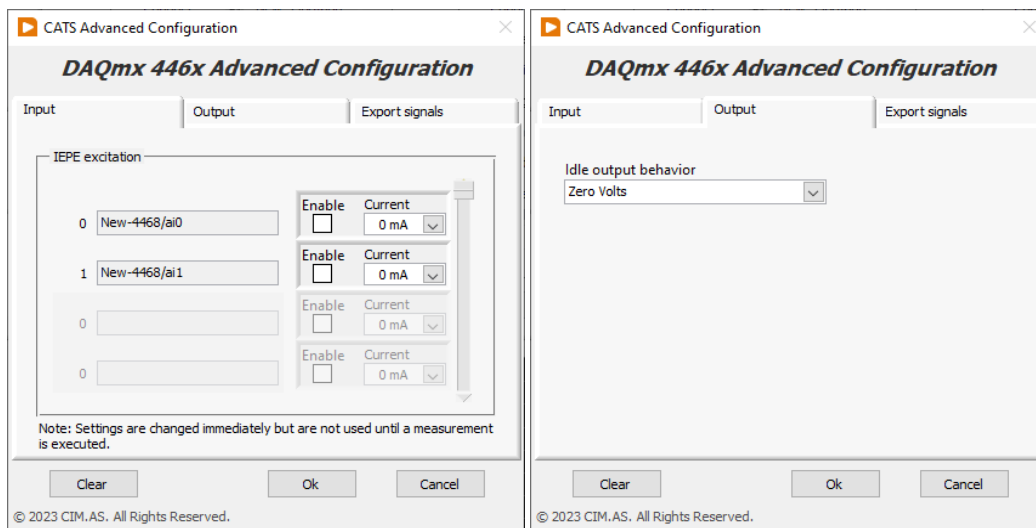


Figure 108: DAQmx 446x Advanced Configuration - input and output

Input – IEPE excitation

Each input channel has the option to enable IEPE excitation to supply current for microphone or accelerometer. Enable the IEPE and select the current level according to the requirements of the attached front-end.

Output – Idle behaviour

Select how output channel shall act when no signal generation is active:

- Zero Voltage
- High Impedance
- Maintain Existing Value

Export signals

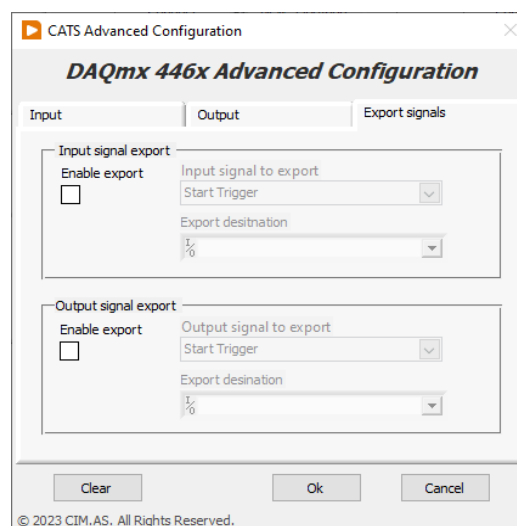


Figure 109: DAQmx 446x Advanced Configuration - export signals

You can enable export of various DAQmx signals to selected output pins. Not all available options are valid, depending on specific hardware type and NI-MAX configuration.

Refer to NI-446x specifications and NI-MAX help for further details.

11.3 DAQmx 4431 Platform

This platform is similar to the DAQmx 446x platform in most features. See description above.

11.4 DAQmx 636x Platform

This platform supports NI-6363 and NI-6361. It has no advanced configuration panel. If you select the button, you will get the following error message:

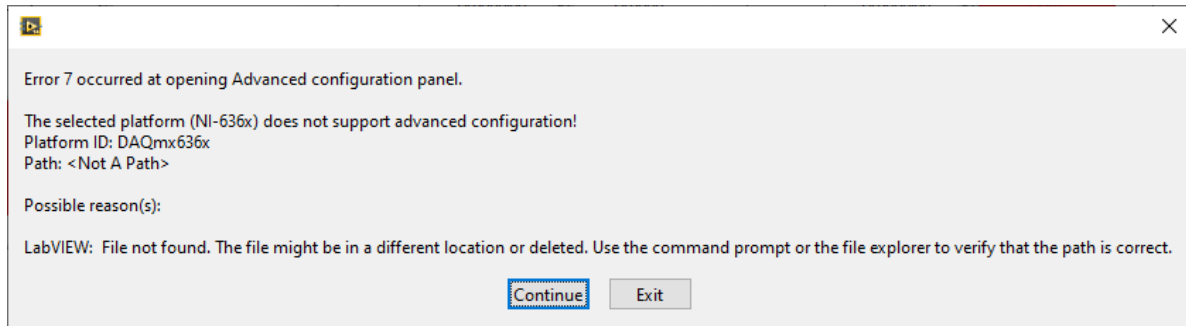


Figure 110: Advanced configuration panel error dialog

Select 'Continue' to go back to the CATS configuration panel.

11.5 DAQmx 92xx Platform

This platform supports selected compact DAQ modules, like NI-9250, NI-9233 or NI-9234. These modules support inputs only.

11.6 Sound Card Platform

With the SoundCard platform you can use the Windows sound input and output devices in CATS.

In contrast to NI-DAQmx devices, CATS have limited control over the devices, and you may need to configure some properties of the sound input and output devices using the Windows control panel.

Also, the timing is subject to larger variations and the devices may be connected or disconnected during use. When devices like USB or Bluetooth headset is connected, the sound devices are re-enumerated in Windows.

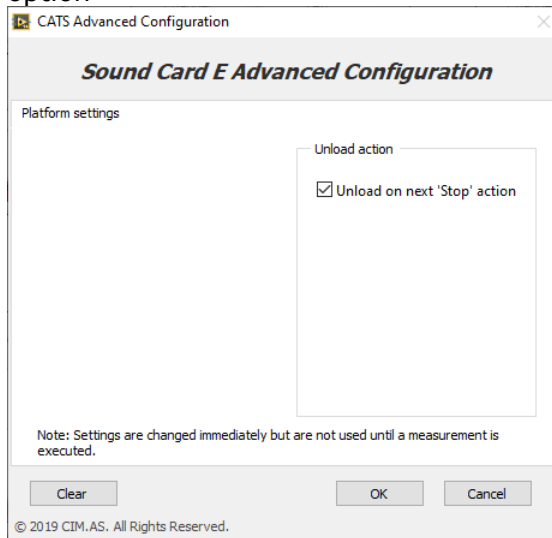
You must account for these properties in your test sequence, fx. setting longer delays or use continuous output function.

11.6.1 Un-loading SoundCard platform

Because CATS use the LabVIEW sound interface, it cannot detect that sound devices are re-enumerated. To detect new devices, you must un-load the sound card platform using the advanced configuration panel.

- Insert a CATS step and select sound card platform for input or output.
- Set the wave device mode to 'Stop'.

- Open the Advanced configuration panel and check the 'Un-load on next Stop action' option



- Select OK to store the setting

When the step runs, the sound card platform is un-loaded. The platform will be loaded automatically again when it is needed in a following test step.

Note: The platform is also re-loaded immediately by the CATS configuration panel that queries the platform properties.

11.6.2 Changed channel behaviour

With the new sound card platform that supports unloading, the output channel behaviour has changed.

On the previous sound card platform, the sound card was always configured to acquire two channels and when selecting only one signal, you could choose to use 'Left' or 'Right' in the signal configuration tab.

On the new platform, the sound card is configured in mono mode (one channel) when selecting only one signal. The 'Left' signal will contain the mono signal. Selecting 'Right' will return no signal.

12 Error Handling (Trouble shooting)

12.1 Common errors

Error code	Error Description	Corrective Action