Volume

1

AnCAD INCORPORATED

Simply Faster

Visual Signal DAQ Express User Guide

ANCAD INC.

Visual Signal DAQ Express User Guide

© AnCAD Inc.
No. 1 Baosheng Rd. • 16 Floor
Yonghe District, New Taipei City 234-44 Taiwan
Phone +886 2 8660 8000 • Fax +886 2 8660 1100

Table of Contents

Chapter

1

Introduction to Interface

Function List

Visual Signal is split into three versions, Professional, Standard, and DAQ Express. The functions made available to you depend on which version you have. The function list is located at http://www.ancad.com.tw/VS_Online_Help_1.4/index1.html?page=ar01.html

Notational Conventions

This manual uses the following notational conventions:

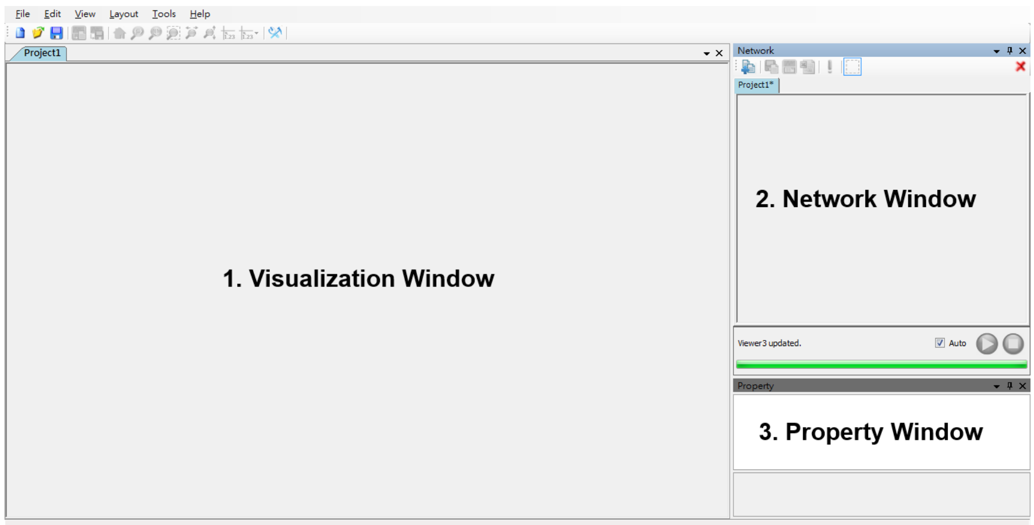
Convention	Explanation	Example
`THIS TYPE STYLE`	Parameters in property settings of functions that can be customized to the users specific needs.	*`SamplingFreq`*, *`Upsamplingmethod`*, *`StartPosition`*
`THIS TYPE STYLE`	Denotes a specific function	**`Channel Viewer`**, **`Fourier Transform`**
This type style	Used for denoting specific windows and command actions in toolbars.	**Network Window, File, View**
This type style	Used to represent mathematical functions and variables.	$y_i = \int_0^{t_i} x_i dt$

1.1 Application User Interface

This section will get you familiar with the layout of the program and the commands you will have at your disposal.

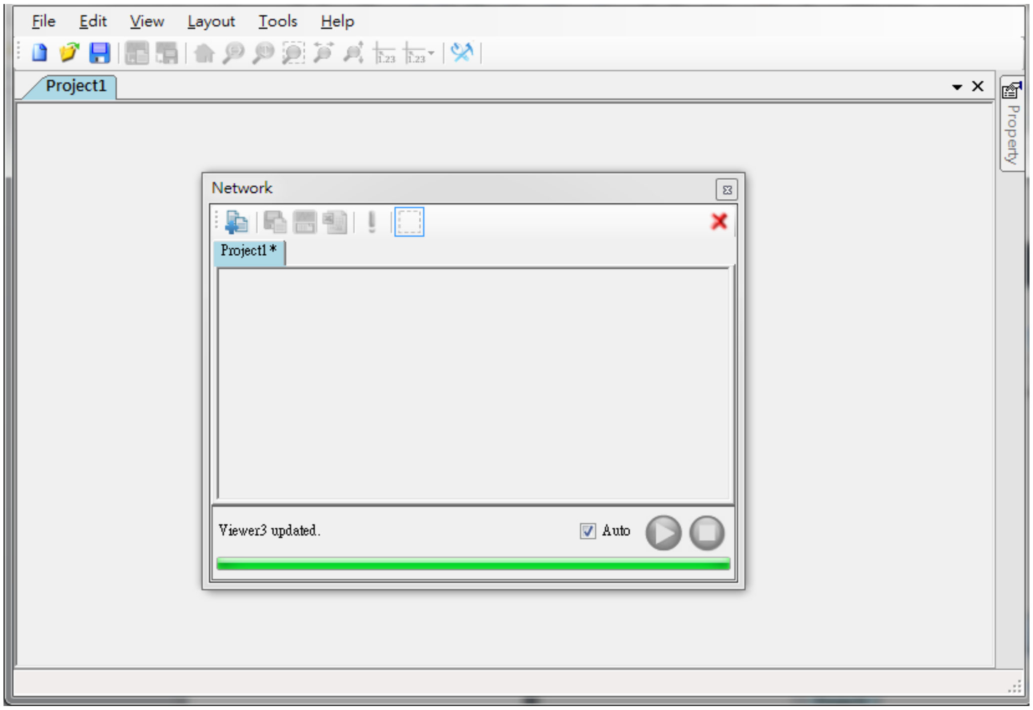
1.1.1 Graphical User Interface

First off, we need to be familiar with the interface in Visual Signal. The interface is divided into three major parts that are independent of the Visual Signal desktop. Each window can be closed and opened again.

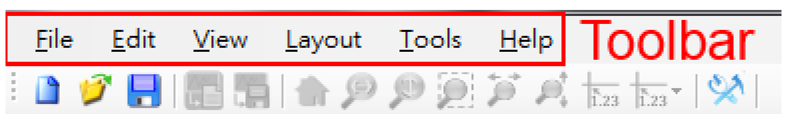


1. **Visualization Window** – This window is where the drawing occurs. Whenever a graph or chart is drawn it will show up in this window.
2. **Network Window** – This window is where the components are edited. Choosing what data to input, how to visualize it and how they connect is all done in this window.
3. **Property Window** – This window shows the specific parameters and settings of the components. Module settings are also looked at in this window.

Note: Double-clicking the title of a window will pop the window out; double-clicking it again will put it back on the original desktop. Clicking the pin icon  in the windows will set the window to auto-hide which places the windows in tabs on the right side of the screen. The figure below shows the **Network Window** being popped out and the **Property Window** being hidden.



1.1.2 Introduction To The Toolbar



The **File** menu will give you the option to create a **New Project**, **Open** an existing project or **Save** your current project. It also gives you the option to **Close** your current project or all projects you have opened.

Note: The file extension of Visual Signal projects (vsn) saves all modules in the network component link, parameter links, graphic settings, and DAQ settings of a project. When saving a project you can decide whether or not to save its intermediate data. If you save the intermediate data then all components of calculation results will be stored and the next time you open the project the calculations and drawings will be saved. If you select 'No', then all components will have to be recalculated when you open the project.

The **Layout** menu will allow you to set the viewing options of the **Network Window** or **Property Window**, or to set the window order of Visual Signal to default.

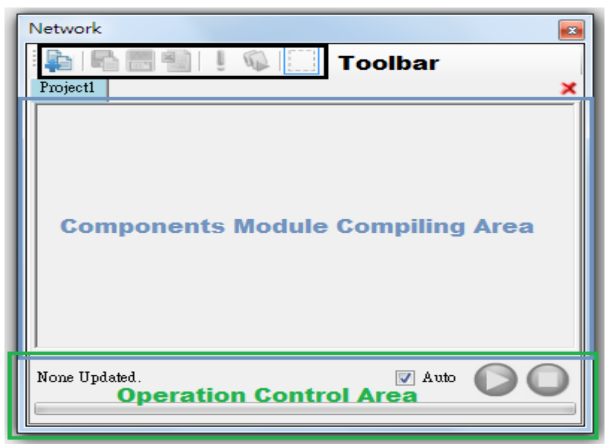
The **Tool** menu brings up the **Preferences** where you can change Visual Signal's default settings.

The **Help** menu brings up **Reference Guides** that help you understand component algorithms applied to the signals and guides to help you utilize the program. The **License Manager** is located here and allows you to renew, add, or remove licenses to Visual Signal. **Update…** will check the internet for new updates to install. **About…** will show you what version your software is on and what license you are currently using.

The **Edit** and **View** menus are used to control the function of the drawing area. For detail on these menus refer to specific drawing area section.

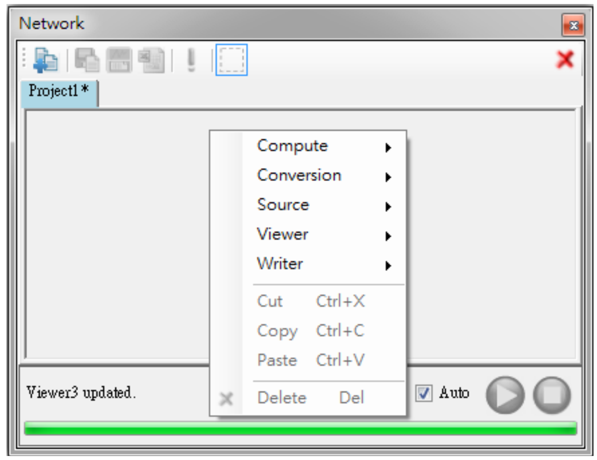
1.1.3 Network Window (Component Editor Window)

The **Network Window** is the area where you connect various components by linking them together. Dragging the mouse from one component to the other will link the components together and dragging an arrow back to its original component will remove the connection. This intuitive process allows fast combination of signal processing required for calculation and analysis.
The picture below breaks down the **Network Window** into three main parts: the components modules compiling area, the toolbar, and the operation control area.

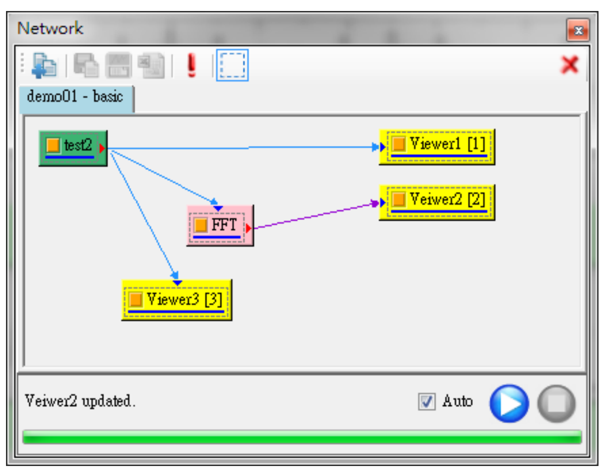


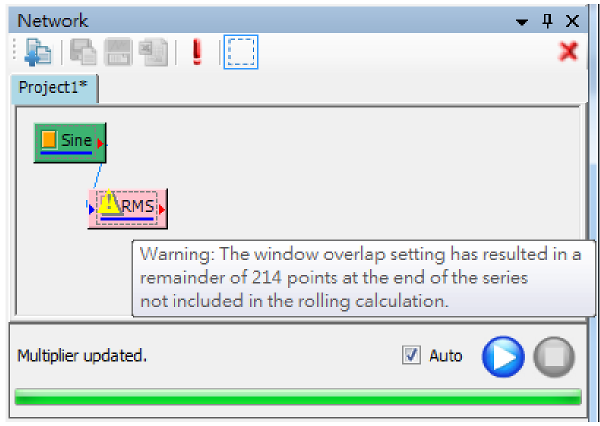
1.1.3.1 Components / Module Compiling Area

The compiling area is the core of Visual Signal. This area is the operating area where the editing of the signal processes and visualizations takes place an intuitive way. Below are brief descriptions of signal inputs; how to perform calculations, and output components.

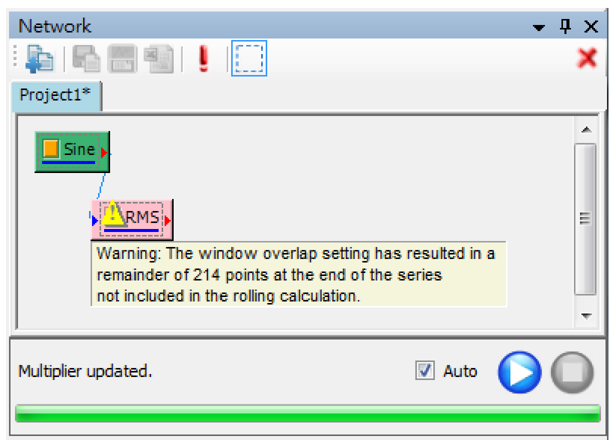


The picture above shows the symbol-editing menu that comes up after right-clicking the module compiling area. The menu is divided into five major groups, **Compute**, **Conversion**, **Source**, **Viewer**, and **Writer**. The component of each module describes its method of operation and Chapters 2, 3, and 4 go into each method in more specific detail. The components editing area works like a flow chart controlling operations of the signal processes you want analyzed. By connecting the signals and modules in the way you choose, complex signal analysis is possible in a few simple steps. With these options, Visual Signal will give you the ability to analyze signal processes, signal front-end processes, signal analysis algorithms and render them into visual graphics. Below is an example of what a typical signal analysis project could look like.





If there is a problem with the connected components, the **Network Window** will display a flashing warning sign () or an error sign () over the component. Placing the mouse over the sign will display a tooltip describing the detail of the problem. Double-clicking the error sign will place the warning inside the project which is helpful if you have multiple errors to keep track of at once.



1.1.3.2 Component Editor Toolbar

The component editor toolbar is responsible for data operation commands such as inputting and outputting data and are listed below:

 Import data from file

 Save data to file

 Open Data Viewer

 Export data to Excel

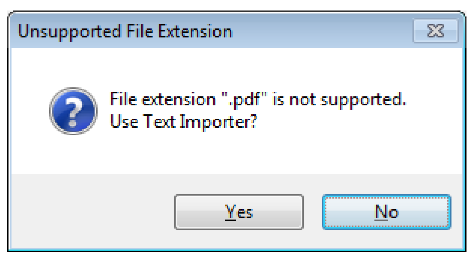
Descriptions of each command are listed below:

1.  Import data from file:

   This command makes Visual Signal read import data from an external file. Acceptable file formats are Time Frequency Analysis file, plain text (ASCII file), and a variety of other different formats. If the file is in plain-text format or comma separated values format, the **Text Importer** window will appear for you to set up information for the data.

File Extension	File Type
tfa	Time Frequency Analysis File
txt	Plain text file
uff	Universal file format
vsb	Binary file for Visual Signal
eeg	SleepScan and Ceegraph EEG data file
csv	Comma-separated values
wav , mp3 , aac , ac3 , mp4 , m4a , amr , ape , wma	Audio files
dat	ADLINK DAT file

If you want to import a data file that is not in a supported format a warning message will appear asking you if you want to read the file in plain text format. Selecting yes will bring up the **Text Importer** to attempt to read the file. You must use the **Text Importer** to setup the signal timeline, such as units of time, the sampling range, and the data range. A more detailed explanation of this feature is in Section 2.



2.  Save data to file and  Export data to Excel

   These two commands save the data in the file format tfa, txt, and csv etc. Sound signals can be saved as a wav file or other audio formats. These two features are mainly used in the **`Writer`** function and are explained in more detail in Section 4.5 **`Writer`** (Signal Output Modules).

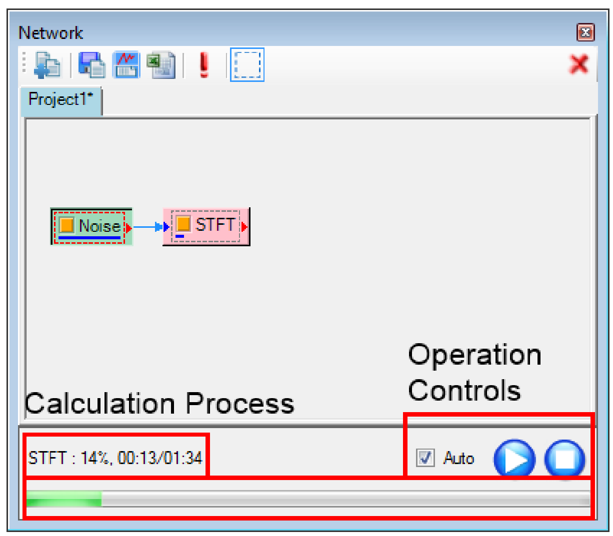
3.  Open Data Viewer

   This command opens the data viewer window which allows you to select the components of calculation results. Also allows you to view the data browser which will detect the output data type, automatically adjust the way the data is presented, and display graphics with data of the output device. This command is described further in Section 1.3.4.

4.  Force update
Click to execute the project from start to finish including all module components.

1.1.3.3 Operation Control Area

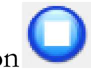
The Operation Control Area is located below the Components Module Compiling Area and controls and displays the calculation process. The text on the left side is the calculation process in real-time, displaying the percentage of progress made in the implementation of the current program while the text below the progress bar immediately displays the progress of the current component being calculated. There are three controls on the right to provide you with control over the computing process, and are described below:



1. Auto function 

This function determines whether or not the program will be updating the calculations in real time as you modify them. If the Auto box is checked, whenever the user changes device parameters the program will immediately recalculate the components and all of the following components in the component output port. You may encounter a situation where you need to modify multiple components before wanting the program to update after each change, if that is the case uncheck the Auto box.

2. Perform operation 
If the Auto button is not checked, then pressing the perform operation button will have the program run the modified components. Compared to the "Auto" option this can be looked at as a "Manual" option.

3. Abort operation 

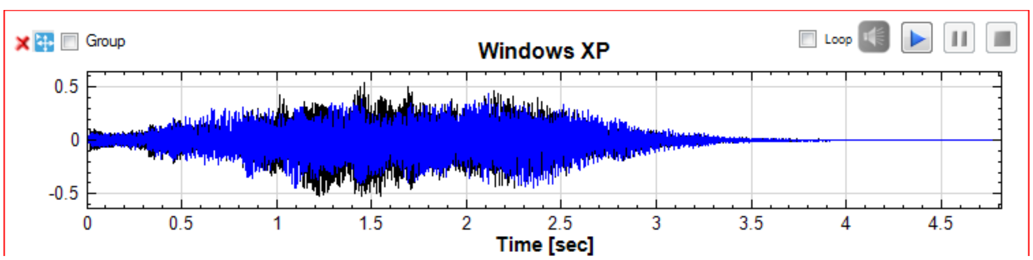
If program is in the course of running operations and you want to terminate the processes you can press the abort operation button. Something to note though is that this feature only terminates the components of a single process and thus only one element and its following processes will be terminated, the program will still continue the implementation of the other components.

1.1.3.4 Data Viewer

There is lots of information that needs to be shown such as component outputs, data types, query signals, etc., so we provide a fast tool to view that data with the data viewer. The interface of the data viewer window shows the data value, waveform, and signal information as long as their components are in the component editor. Click a specific component and press the data viewer button and the browser will accommodate the different signal types (such as signal spectrum analysis results, numeric data or time-frequency analysis) and display relevant information. The browser window interface will correspond to the different signal types being used.

1.1.4 Visualization Window

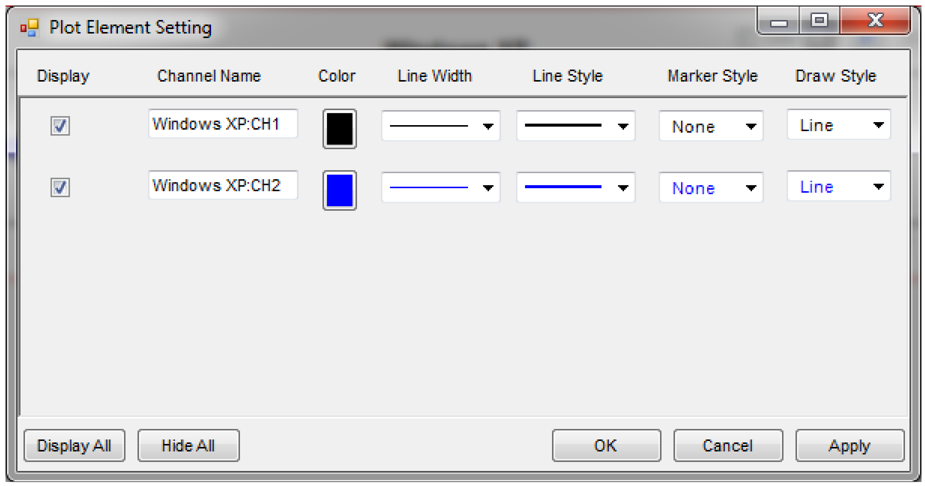
The **Visualization Window** is where the diagrams appear after creating a viewer component in the **Network Window**. In this window you will have a few options available to customize your diagrams and we will go through them in this section. More detailed customization is done through the **Property Window** and is covered in Section 1.5.



Note: The group/move options located in the top left will only show up when your curser is hovering in that area.

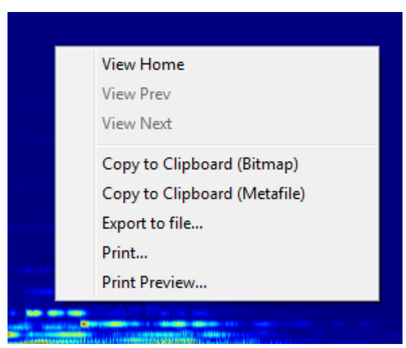
Above is an example of audio signal displayed by a basic **`Channel Viewer`** diagram. The red outline around the diagram signifies that the diagram is currently being selected in the **Network Window**. Because this is an audio file, there are five options located in the top right that allow you to continuously **Loop**  (if box is checked), **Mute** , **Play** , **Pause** , or **Stop**  the audio file and will only show up if the data is in audio format.

When a diagram is selected (as indicated by the red outline around it) scrolling the mouse wheel will zoom in and zoom out the x-axis for more specific viewing. To select a specific diagram, you can either click the diagram directly or click the viewer component it is associated with in the **Network Window**. Double-clicking the viewer component will bring up **Plot Element Setting** window which allows you to choose whether or not a certain channel is displayed, the channel name, and what type of color and line will be associated with the channel. This is very useful when working with a diagram that has multiple channels.



Note: Clicking **Display All** or **Hide All** will either check all the boxes under Display or uncheck them all.

Hovering your cursor in the top left of a diagram gives you the options to **Delete**, **Move**, or **Group** Group your diagram. Checking the **Group** box gives you the option of placing the specific diagram in a group numbered 1-5 and choosing whether to sync the X axis, the Y axis, or both with other diagrams in the same group.

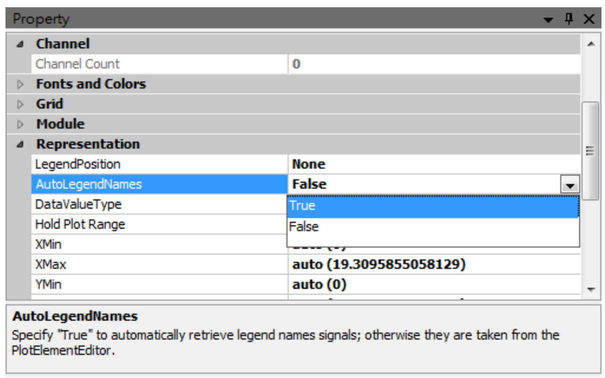


Right-clicking a diagram will bring up a list of options dealing with going back to specific views of the diagram and how to export the specific diagram into other areas. The first three options **View Home**, **View Prev**, and **View Next** deal with viewing the diagram after making edits such as zooming in and out. **View Home** brings the diagram back to its default view, **View Prev** can be compared to an undo button while **View Next** is a re-do button.

The next set of options deal with exporting the specific diagram to other programs or documents. The first two options, **Copy to Clipboard (Bitmap)** and **Copy to Clipboard (Metafile)** place the diagram on your clipboard to easily paste it into a program or document. **Export to file…** allows you to save the diagram as a file in multiple picture formats. **Print…** will bring up a print settings window for you to directly print the diagram.

1.1.5 Property Window

The **Property Window** shows the properties of whichever diagram you are selecting in the **Visualization Window** or whichever component in the **Network Window**. When you select something the **Property Window** will display a list of its properties that can be edited either through manual typing or through a drop down menu. You can also see what module the program is using for its computations.

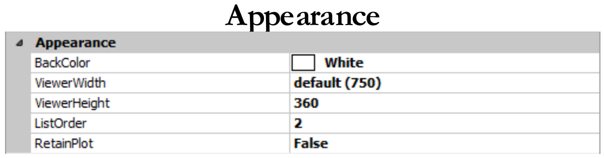


Note: When the arrow to the left of a section is black the options are being shown, when it is clear the options are being hidden. Clicking the arrow will either expand the list or contract it. The bottom of the Property Window gives a detailed explanation of what editing each property will affect and is very helpful if you don't know what a certain property does.

1.1.5.1 Properties of a Diagram

Selecting a diagram from the **Visualization Window** will bring up its properties in the **Property Window**. Details of the diagram can then be altered to suit your needs. Almost every aspect of your diagram can be changed from the **Property Window**. It is important to note that if you do not know what changing a certain property will do, the bottom box in the **Property Window** gives a detailed explanation of what the property does.

Appearance



The first section of properties dealing with your diagram has to do with its appearance. Here you can specify the background color and the height or width of your diagram.

The `ListOrder` specifies where the diagram is listed in your **Visualization Window** and `RetainPlot.`

Channel

Channel	
Channel Count	1

This section allows you to choose how many input channels are in your diagram but also depends on how many input channels your specific type of diagram can support.

Fonts and Colors

Channel	
Fonts and Colors	
Title Font	Arial, 12pt, style=Bold
Title Color	0, 0, 0
X-Axis Title Font	Arial, 11pt, style=Bold
X-Axis Title Color	0, 0, 0
Y-Axis Title Font	Arial, 11pt, style=Bold
Y-Axis Title Color	0, 0, 0
X-Axis label Font	Arial, 10pt
X-Axis Label Color	0, 0, 0
Y-Axis label Font	Arial, 10pt
Y-Axis Label Color	0, 0, 0

Here you will be able to change the fonts and colors of all the text in your diagram from the title to the axes.

Grid

Grid	
Horizontal Grid Type	Coarse
Vertical Grid Type	Coarse
Major Grid Style	Solid
Major Grid Color	LightGray
Minor Grid Style	Solid
Minor Grid Color	LightGray
X Major Grid Spacing	Auto
X Major Grid Anchor	Auto
X Minor Divisions	Auto
Y Major Grid Spacing	Auto
Y Major Grid Anchor	Auto
Y Minor Divisions	Auto

This section allows you to make changes to the grid overlay behind the data. Many of the spacing/anchor is already set to `Auto` by default which makes the program calculate the best fit for the diagram. If you change the value and want to restore it to its default, type `Auto` again.

Module

Module	
Class	ChannelViewer
Name	Viewer
Input Port Side	Left
Execute Time	0.0410023 sec
Acceptable Data Types	Real/Complex Single/Multiple-Channel Signal

This property menu shows the details of the module being used. You will be able to see what class of viewer is being used, the time it took to execute, and the acceptable data types it can use. You can also change the name of the module to make organizing your **Network Window** easier.

Representation

Representation	
TimeUint	sec
LegendPosition	None
AutoLegendNames	True
XAxisType	LinearAxis
Plot Elem Editor	PlotEditor
DataValueType	Magnitude
Hold Plot Range	False
XMin	auto (0)
XMax	auto (19.318821)
YMin	auto (-0.953729)
YMax	auto (0.975787)
DateTime Format	Auto

This section allows you to edit how your data is represented in the diagram. You will have options such as adding a legend, specifying what type of axes to use and what maximum and minimum values to use for your axes.

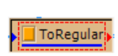
Title

Title	
Title	{default}
XTitle	{default}
YTitle	{default}

Here is where you will be able to change the labels on your title, x-axis, and y-axis. Typing in `{default}` will revert back to the original label and typing `{all}` will show the whole title. You may also use the index to show which component name you want to see.

1.1.5.2 Properties of a Component

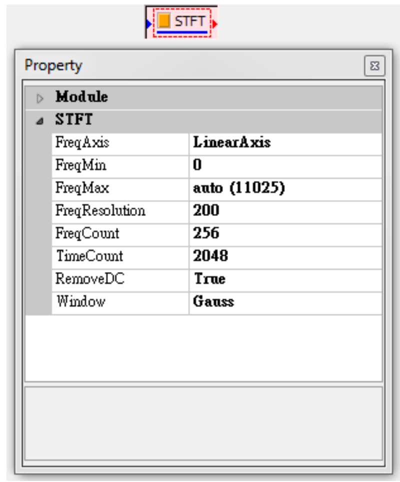
Selecting a component from the **Network Window** will bring up its properties in the **Property Window**. Every component in the **Network Window** has options that can be modified from the properties. The **Property Window** also displays detailed information about the component. For example if you select a data component, the **Property Window** will list where the file location is, how many channels are in the data, the sampling frequency, etc. Depending on which component you have selected, different options will be available to modify. There is never one set way to analyze data and is why these options are made available to you. In this section we will use examples from a compute component and a convert component.



Property

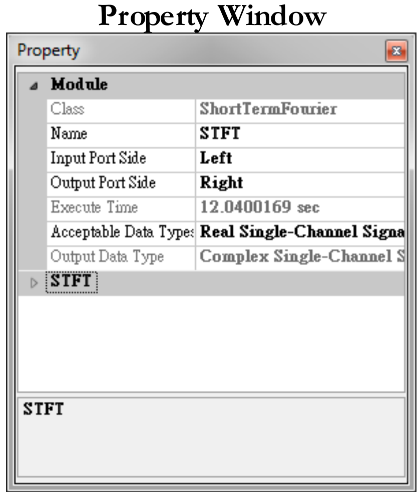
Convert To Regular	
ConvertMethod	FillGap
FillMethod	LinearInterpolation
Sampling Period	8.99999999965928E-05
Unit	sec
AutoDetect	True
Module	

The first example shows the properties of a **`Conversion→Convert to Regular`** component. Here you will have options relating to how you want to convert your indexed into regular data. For example, convert method allow you to choose whether you want the sampled data to be filled in by missing points or for the gaps to be removed. If you don't know what a certain property does remember that when it is selected a brief description will appear in the bottom box of the **Property Window**.



The second example shows the properties of selecting a **`Compute`→`TFA`→ `ShortTerm Fourier Transform`** component. You will see a different set of properties than the **`Convert to Regular`** component as different components have different sets of modifiable properties. In this set of properties you can customize how you want the specific **`ShortTerm Fourier Transform`** component to operate. Maybe for your first transformation you want a linear axis as the frequency type and a log axis for your next transformation, all these details are modified from the

Property Window



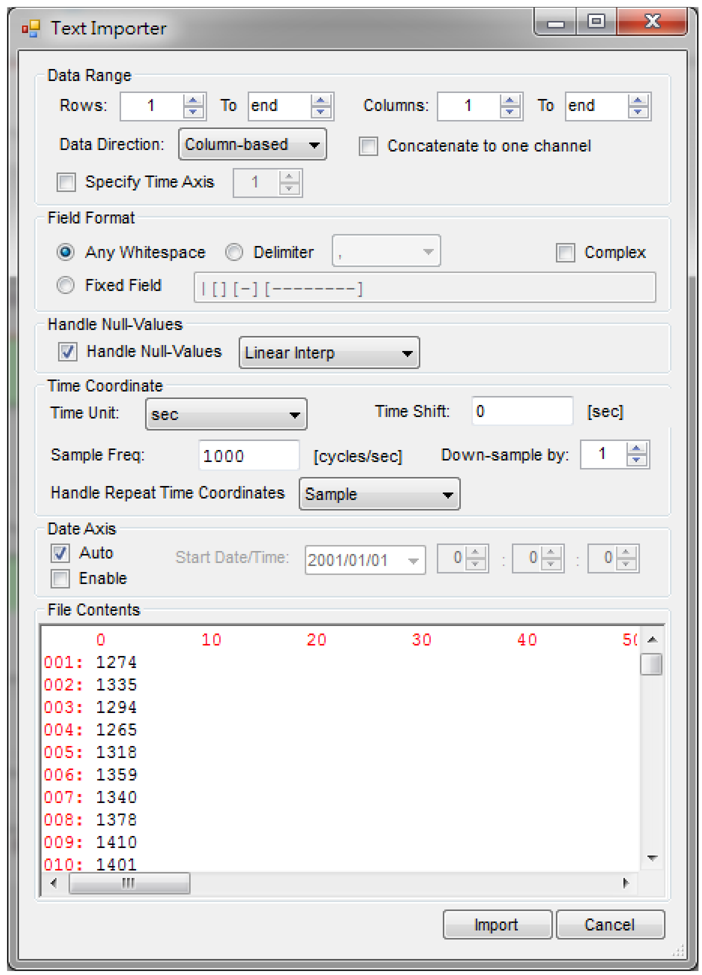
The **Module** section of these components is similar to the diagram module section. Specific information will be shown about the module being used for the component such as the class module, the execution time, the acceptable data types, and its output data types. You will also have the option to change the name of the module and its input and output port side.

1.2 Importing Your Data

1. Importing data is done by going to the **Network Window** and clicking  **Import data from file**.

2. Locate your data file and see if it is a supported format, if it is not the program will ask you if you want to use **Text Importer** to format the data.

3. Once the **Text Importer** window is opened, a preview of how your data will be formatted will be under **File Contents** near the bottom of the window. This preview will adjust in real-time as you make changes to the options.



4. Start by specifying the data range of your data. Here is where you specify which rows and columns you want your data to start and end at and whether you want the direction of the data to be column-based or row-based. By default the data will be separated into multiple channels depending on how your data is formatted, but by clicking the `Concatenate to one channel` checkbox your data will be set as a single channel regardless of how many columns or rows you have. You can

also specify which axis is your time axis by checking the `Specify Time Axis` box and choosing the corresponding axis.

5. Next area is the `Field Format` which will determine how the data will be separated. The default option is `Any whitespace` which will separate the data each time there is white space between them. The `Delimiter` option allows you to choose specific symbols that separate the data. If you want to specifically format the data yourself, you can use the `Fixed Field` option and see how it will be formatted in the file contents section. The import data can be merged to complex by checking `Complex` check box.

6. In the event that there are null-values in your data set, there is a `Handle Null-Values` option that is checked by default. Here you can choose how Visual Signal will handle your null-values when it imports your data set as there are a variety of options such as having fixed values or having Visual Signal calculate a linear interpolation.

7. The next set of options has to deal with the `Time Coordinate`. Specify the unit of time your data set uses by selecting the options under `Time Unit`. If you need to shift will also be able to choose the `Sampling Frequency` and whether or not you need to down sample your data set in the case your amount of data is too large and you want to reduce the number of sampling points to make computation easier.

8. The final sets of options are related to adding dates to your data set. The default option checked is `Auto` which will check if you have dates associated with your data and will set up accordingly. If you want to add dates manually check the `Enable` box and specify the starting date and time options to its right. After you are finished adjusting all the settings press import and your data should show up as a component in the network window.

Chapter

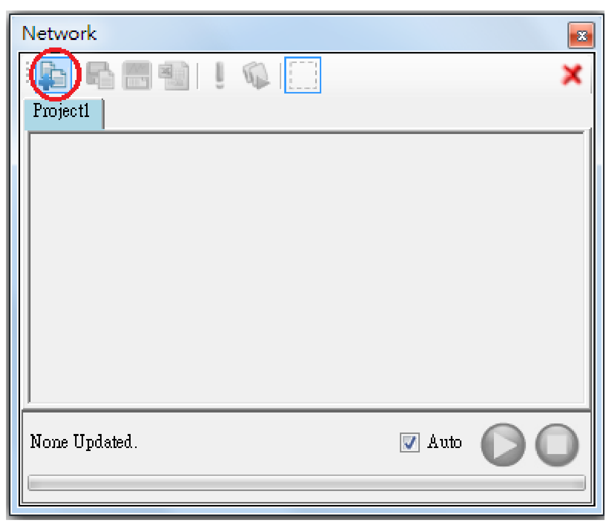
2

Example Projects

2.1 Your First Project

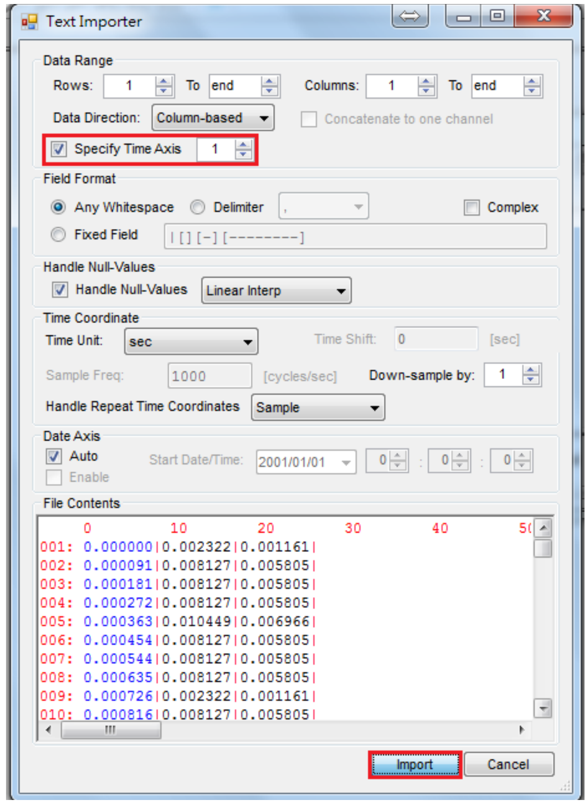
This section will go through an example of how you would start a project, use the modules, and export your new diagrams. In this example we will take raw frequency data from the audio file "WindowsXP.wav", which is located in C:\Program Files\AnCAD\Visual Signal\demo\Basic, compute a ShortTerm Fourier Transformations and display two separate time-frequency spectrograms usable in a document or presentation.

1. Select **File→New Project** from the toolbar.
2. Go to the **Network Window** and click  **Import data from file**



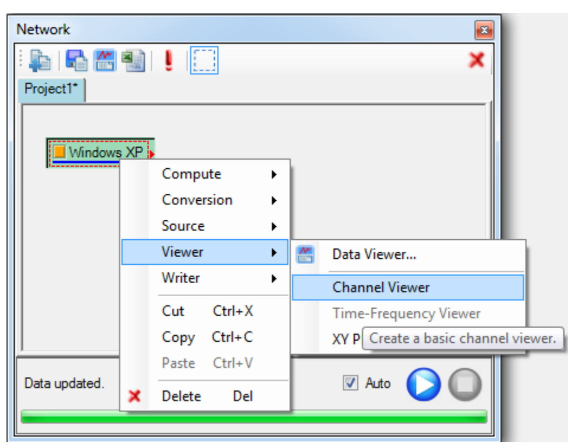
3. Navigate to C:\Program Files\AnCAD\Visual Signal\demo\basic and open the file "WindowsXP.wav".
   Note: File locations will be different depending on platform (x86 or x64) or the installation path you selected.

4. Check the `Specify Time Axis` box and click `Import`, the settings should like the picture below.



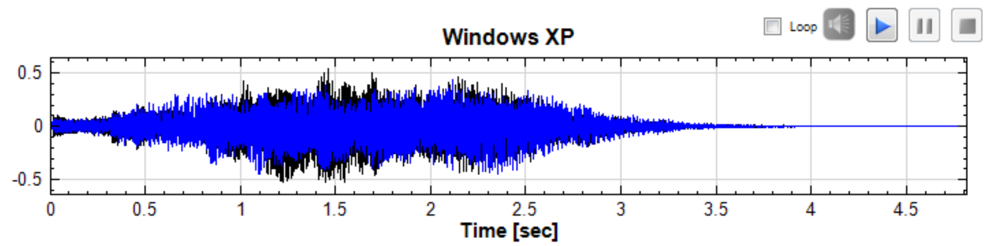
5. A warning will pop up saying the data is "Indexed" and should be converted to "Regular". Click **Okay**.

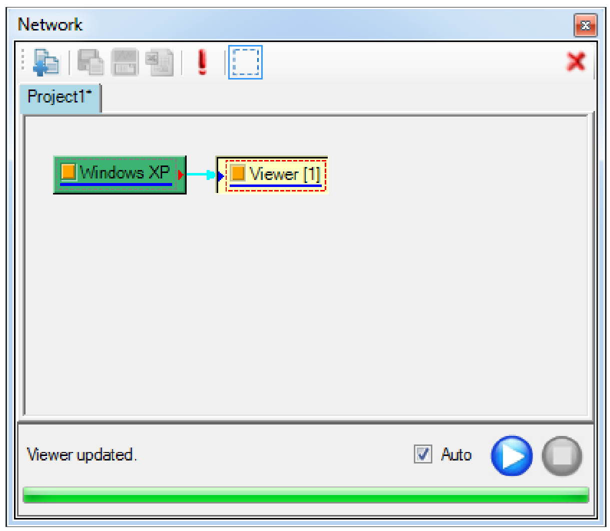
6. A green component named Windows XP will now be in the **Network Window**, right-click the box and select **`Viewer→Channel Viewer`**.



Note: Hovering your curser over an option or component will show details on what the option does.

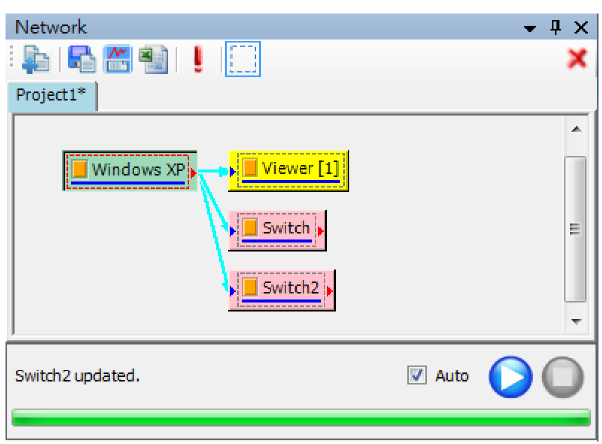
7. A **`Channel Viewer`** component will now be linked to the Windows XP component by an arrow and a graph will be displayed in the **Visualization Window**. The graph shows that the data set has two separate channels and is shown by the two different colors.



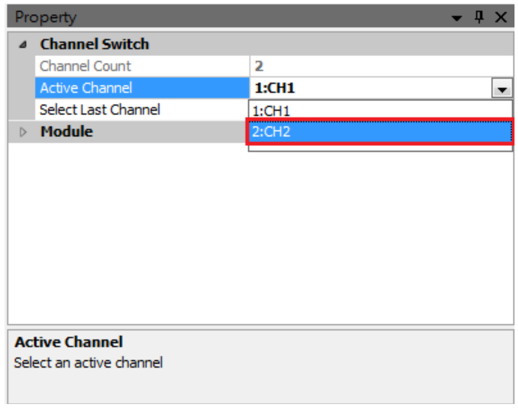


Note: Clicking on the square inside a component disables/enables the component, while clicking the component directly as shown in the picture above outlines in red the specific graph the component is associated to.

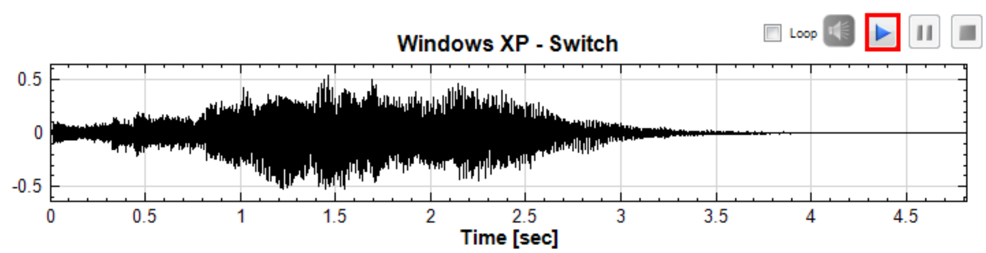
8. Right-click the Windows XP component and select **`Compute→Channel→ Channel Switch`**. Repeat this step so you have two switch components linked to the Windows XP component.



9. Click on the **Switch2** component. Locate the *Active Channel* section under the **Property Window**. Click the arrow on the right of *1:CH1* and click *2:CH2* from the drop down menu.



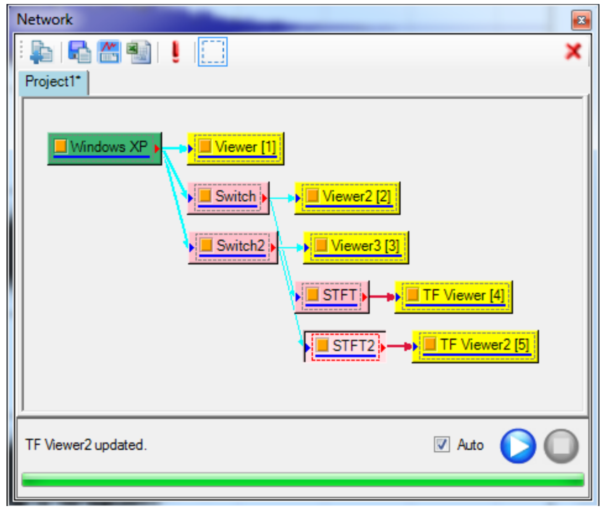
10. Right-click the newly made **Switch** component and select **Viewer → Channel Viewer**. A new diagram will appear in the **Visualization Window** with the option to listen to the audio. Do the same thing to the **Switch2** component.



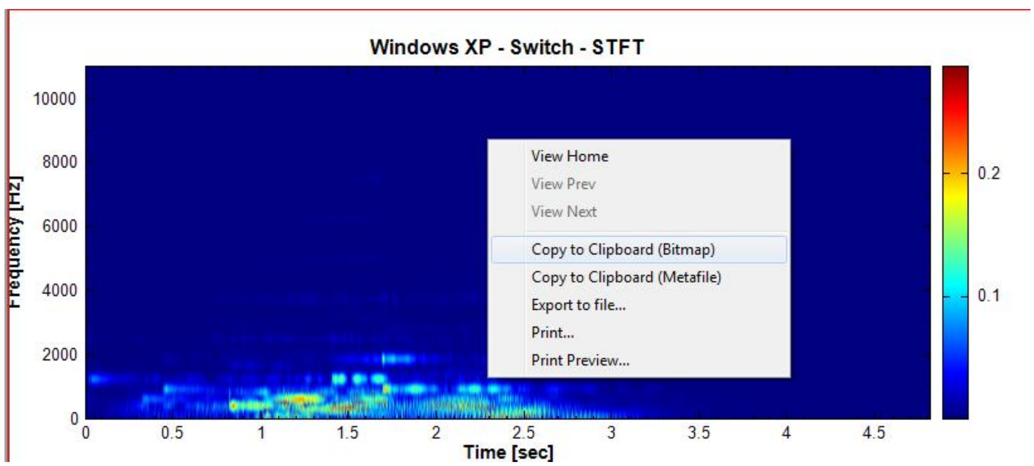
11. Right-click the **Switch** component and select **Compute → TFA → ShortTerm Fourier Transform**.

12. Right-click the **STFT** component and select **Viewer→Time-Frequency Viewer**. A time-frequency spectrogram will now show up under the **Visualization Window**.

13. Repeat steps 11 and 12 for the **Switch2** component. You will now have two time-frequency spectrograms in your **Visualization Window**. Following these steps should yield a **Network Window** similar to the figure below.



14. Select the TF Viewer [4] component. The **Visualization Window** should now bring up the first time-frequency spectrogram. Right-click the diagram and select **Copy to Clipboard (Bitmap)**. The diagram will now be copied onto your clipboard and can be pasted in documents or presentations.

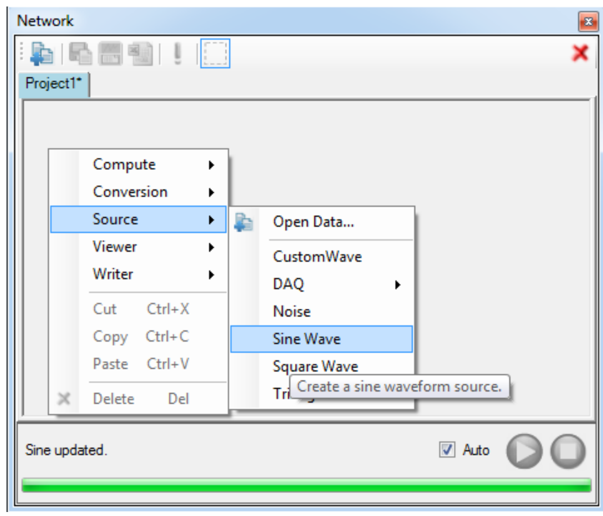


2.2 Spectrum Analysis

2.2.1 Setting Up and Analyzing a Fourier Transformation

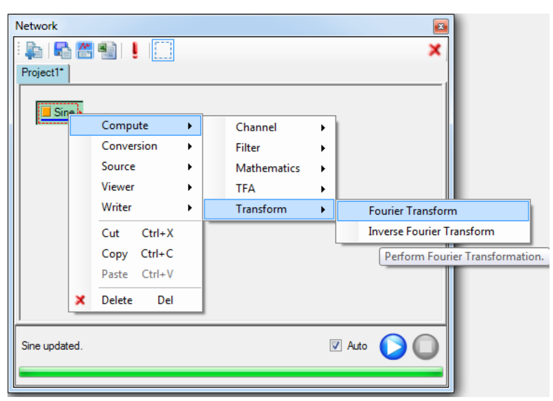
In this section we will go through an example of how to set up a Fourier analysis of a sine function.

1. Right-click the component editor window inside the **Network Window** and select **`Source→Sine Wave`**.



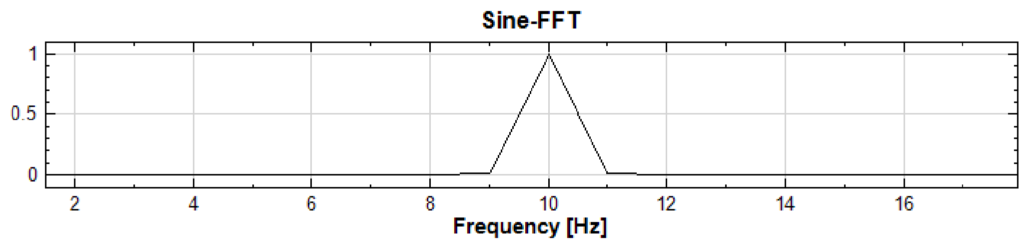
2. Right-click the new component labeled **`Sine`** inside the **Network Window** and select **`Viewer→Channel Viewer`**. This will display a graph of a sine wave that you can analyze and modify in the **Property Window**. In this example we will keep the sine wave as default.

3. Right-click the component labeled **`Sine`** again and select **`Compute→Transform→Fourier Transform`**.

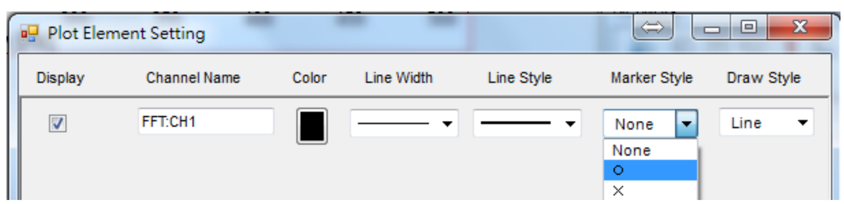


4. Right-click the new **`FFT`** component and select **`Viewer→Channel Viewer`**. A graph of the Fourier transformed sine wave will now be displayed in the **Visualization Window**.

5. On the graph that you just created labeled Sine-FFT click inside the graph and drag your mouse cursor from the second notch in the x-axis and to zero as if you were highlighting the triangle in the graph. This will zoom in the graph to the section you highlighted.

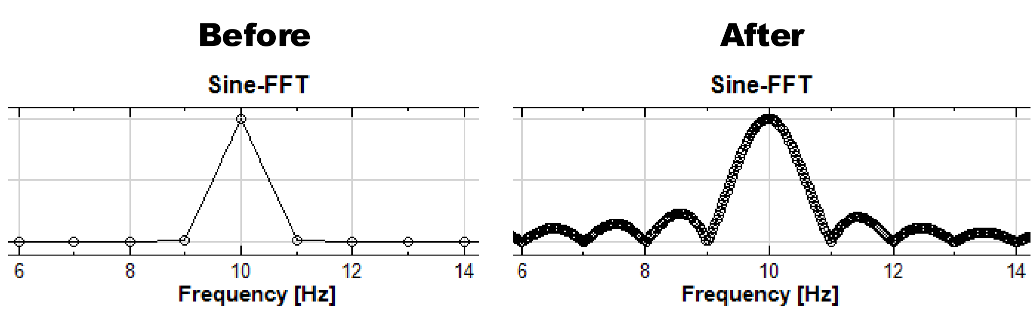


6. Double-click the Viewer [1] component and select the circle from the *`Marker Style`* drop down menu in the *`Plot Element Setting`* window, then hit *`Ok`*.

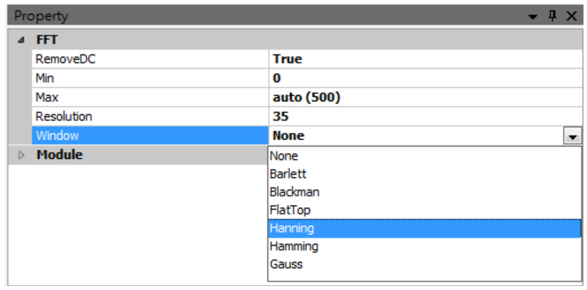


7. The graph will now be marked with circles at every frequency point. You should notice that the main curve is only made up of three points, creating a jagged curve. To achieve a more accurate curve we need to add more frequency points that make up the curve.

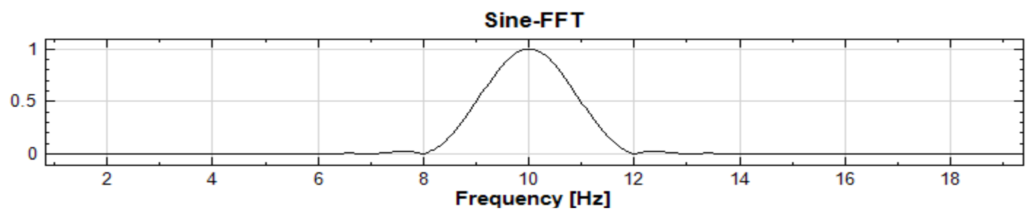
8. Select the **`FFT`** component and locate the *`Resolution`* setting in the **Property Window**. This is the multiplication factor of frequency resolution of the Fourier transform. Increasing the number will increase the number of frequency points resulting in a smoother curve. Replace 1 with 35 and highlight the curve on the graph like you did in step 5.



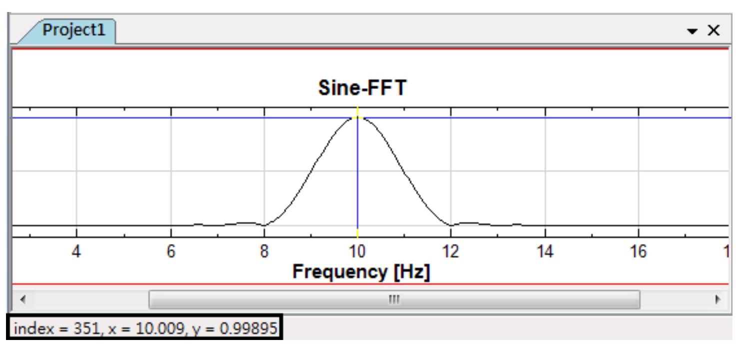
9. The curve now has side lobes as a result of the multiplication factor but we can fix this by selecting a window function to apply before the Fourier transformation. Select the **FFT** component and locate the **Window** setting in the **Property Window**. Then select the `Hanning` option from the drop down menu.



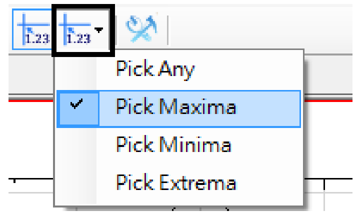
10. Double-click the 'Viewer [1]' component and select `None` under the `Marker Style` to remove the circle indicators.



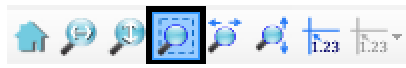
11. Click the  **Show Value** button in the toolbar then place your mouse cursor over any section of the curve. This will allow you to analyze the values of the curve with your mouse cursor. The values are displayed on the bottom-left corner of the **Visualization Window**.



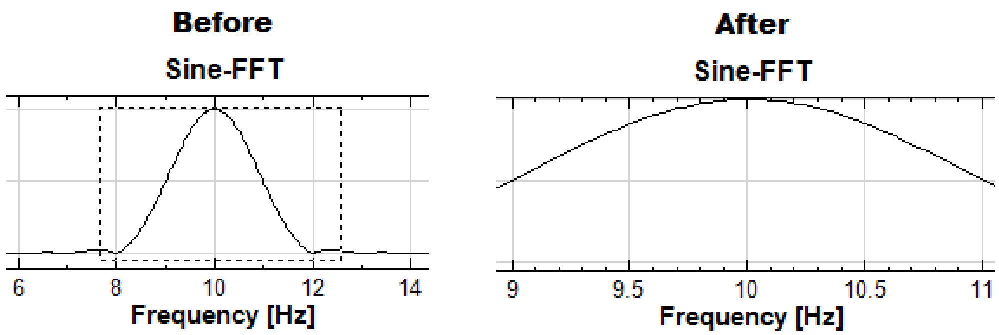
12. Click the button to the right of the `Show Value` button with the drop down menu to select `Pick Maxima`. The value cursor will now lock to the nearest maxima on the curve making it easier to find specific values.



13. To leave the `Show Value` mode, select the `Rect Zoom` buttons in the toolbar.



14. Click and drag a rectangle over the curve, the graph will then zoom in to the area inside the rectangle.



15. Using the other zoom functions in the toolbar will make it easier for you to display specific areas of the curve.

Chapter

3

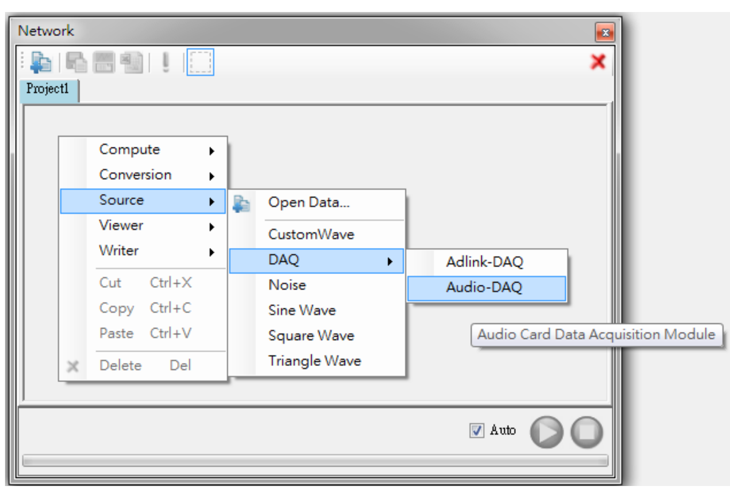
Data Acquisition Quick Start

Data acquisition is the process of sampling signals that measure real world physical conditions and converting the resulting samples into digital numeric values that can be analyzed on the computer with programs such as Visual Signal. Visual Signal works in conjunction with a variety of data acquisition systems (DAQ) such as ADLINK that allows real-time data acquisition and the ability to convert the recorded analog waveforms into digital values for processing. The data can be directly sampled from within Visual Signal for simple accessibility and manipulation. This user guide will go through examples of how to integrate the ADLINK data acquisition system with Visual Signal and how to record audio with your computer.

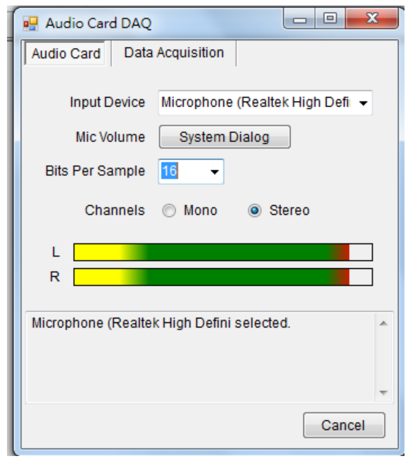
3.1 Recording Audio with Computer

This section will go through an example of how to record and analyze an audio signal recorded from your computer microphone. If your computer has a microphone, you can directly record an audio signal into Visual Signal to analyze.

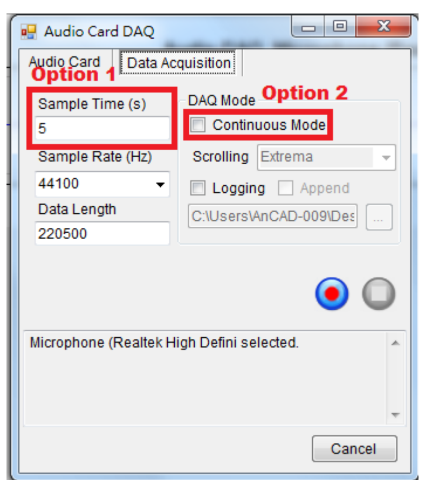
1. Right-click the **Network Window** and select **`Source→DAQ→Audio-DAQ`**.



2. Double-click the 'Audio-DAQ' module to open the Audio Card DAQ window. Under the **Audio Card** tab select the input device for your microphone from the drop down menu under the **Input Device** option. You can adjust the volume of your microphone by clicking on the **System Dialog** button next to **Mic Volume**. This will open up your microphone properties and allow you to make proper adjustments. Next select the bit rate of your recording under `Bits Per Sample` and then choose whether you want a `Stereo` or `Mono` channel. You can then test your microphone by looking at the microphone level bars for the left channel (capital letter "L") and right channel (capital letter "R").



3. Once your microphone is set up and the settings of your recording are correct click the **Data Acquisition** tab. Here you will have two options for recording your audio. The first option is recording for a set amount of time that you set under the `Sample Time (s)` in seconds. The second option is checking the `Continuous Mode` option, this option constantly records audio and displays the data in real-time until stopped by clicking the square button (Stop button) in the network window. The `Sample Rate (Hz)` applies to both options of recording.



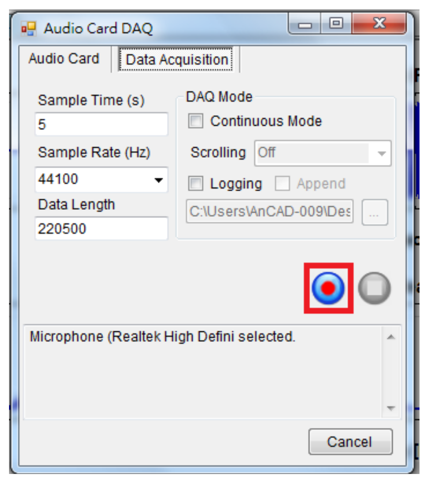
3.1.1 Recording audio data in a set amount of time (Option 1)

1. Set the amount of time you wish to record for in the `Sample Time (s)` section in seconds.

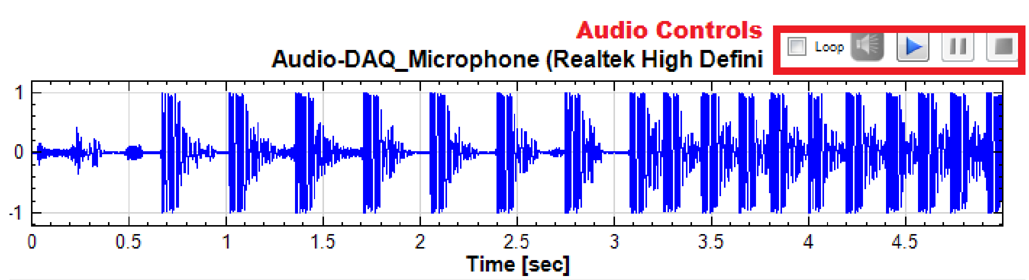
2. Select the sound quality of your recording by adjusting the `Sample Rate (Hz)`. The higher the sampling rate, the higher the sound quality. The standard sampling rate of most recordings is 44100 Hz.

3. The `Data Length` is the number of points recorded during the recording. Typically, this does not need to be changed as it will automatically update on its own based on your sample time and sample rate.

4. Click the red circle (Record button) when you are ready to begin your recording.

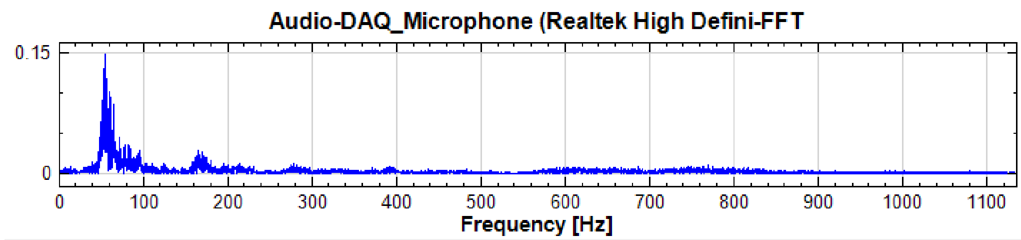


5. After the recording is finished, the data will be stored inside the **Audio-DAQ** component. From here you can attach a **Channel Viewer** by right-clicking the component and selecting **Viewer→Channel Viewer** to view the recording and play the recording back.



6. Right-click the **Audio-DAQ** component and select **Compute→ Transform→Fourier Transform**. This will convert the audio

recording into the frequency domain. Right-click the **FFT** component and select **Viewer→Channel Viewer** to view the resulting graph.

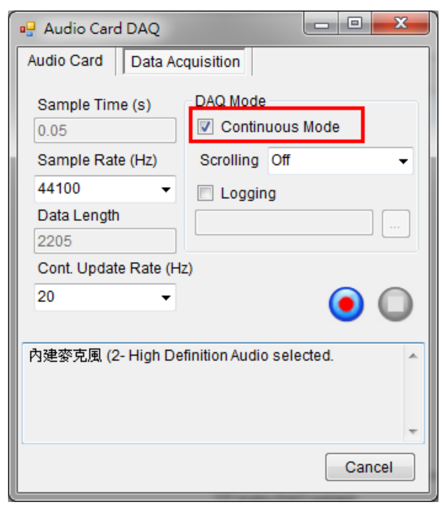


3.1.2 Recording audio data in real-time (Option 2)

1. Right-click the **Network Window** and select **Source→DAQ→Audio-DAQ**. Attach a **Channel Viewer** to the **Audio-DAQ** component if you want to view the recording in a time vs. amplitude graph or attach a time-frequency viewer if you want to view the recording in a time vs. frequency graph.

2. Double-click the **Audio-DAQ** component and select the **Data Acquisition** tab.

3. Check the `Continuous Mode` box under DAQ.



4. Select the sound quality of your recording by adjusting the `Sample Rate (Hz)`. The higher the sampling rate, the higher the sound quality. The standard sampling rate of most recordings is 44100 Hz. When DAQ mode is checked the sample time and data length option will be disabled and grayed out since it will automatically be calculated for you.

5. Change the refresh rate of the graph and the data length by adjusting the `Cont. Update Rate (Hz)`. The higher the update rate, the lower the

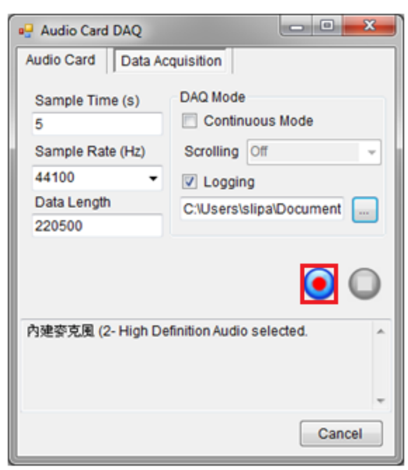
data length. If you want to get a finer resolution in the spectrum or spectra, the data length should be longer. So setting a higher update rate will result in a higher refresh rate in the graph. Set a lower update rate to get a finer resolution.

6. Select the type of scrolling mode you want when the data is represented on your attached viewer. The way scrolling works is that for every *n* data points sampled, which is decided by setting `Cont. Update Rate (Hz)`, only one data point is used for plotting, and the scrolling mode allows you to choose how you pick that one data point. The different modes include **Off**, **Sample**, **Average**, **Extrema**, and **STFT**.

   Off : This mode will have no scrolling and will represent whatever audio signal is currently being recorded on the viewer.
   Sample : This mode picks the first point of the data sampled and plots the point.
   Average : This mode takes the average of the *n* points and plots the point.
   Extrema : This mode picks the point with the largest absolute magnitude of the *n* points and plots the point.
   STFT : This mode will output time-frequency data in real time and requires you to attach a time-frequency viewer instead of the standard channel viewer.

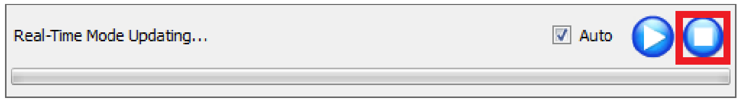
7. Once your settings are set, you can decide whether or not you want to have the recording saved to a specific file by checking the `Logging` box and specify the name and location. If the specified file already exists, the old content will be replaced by the new file.

8. Click the red circle (Record button) when you are ready to begin your recording.

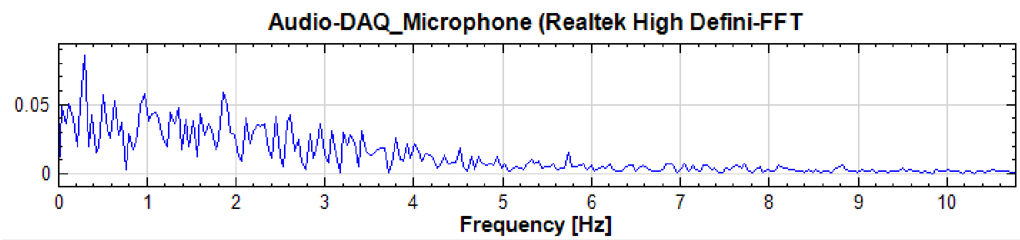


9. Visual Signal will now go into Real-Time Mode Updating which means data is constantly being recorded into your **`Audio-DAQ`** component. The viewer

you attached to the module will constantly be updated to show you the recorded data in real-time. To finish recording, press the stop button in the network window.



10. After the recording is finished, the data will be stored inside the **`Audio-DAQ`** component. Right-click the **`Audio-DAQ`** component and select **`Compute→ Transform→Fourier Transform`**. This will perform a Fourier transform and convert the audio recording into the frequency domain. Right-click the 'FFT' module and select **`Viewer→Channel Viewer`** to view the resulting graph



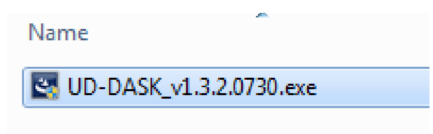
.

3.2 Using ADLINK

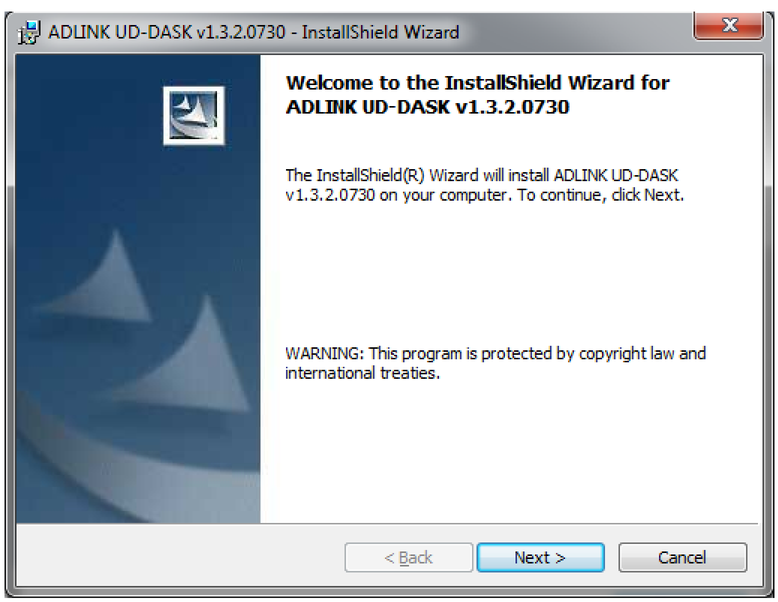
In order to use the ADLINK hardware with Visual Signal, the latest driver for the device must first be installed. UD-DASK is the kernel driver for the ADLINK hardware and has support for 32-bit/64-bit Windows 7/Vista/XP OS. Please note that only UD-DASK versions 1.3.2.0730 and later can support the USB-2405 module.

3.2.1 Installing the UD-DASK Driver

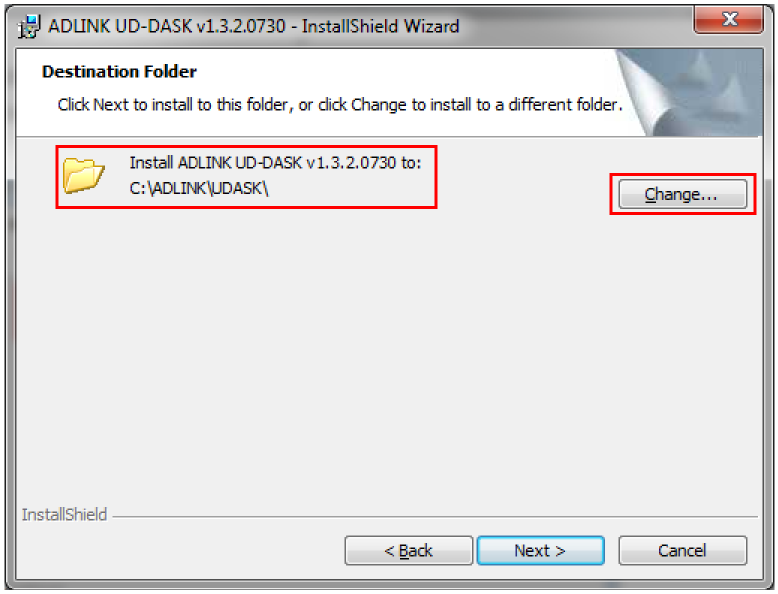
1. Locate the file named UD-DASK. This will be in the driver installation CD provided with the device or you can download the latest driver from http://www.adlinktech.com/PD/Download/. The latest version at the time this guide was created is v1.3.2.0730.
   Note:
   a. Make sure you are the Administrator of the computer or have administrative privileges.
   b. Confirm you have the latest driver installed to ensure your device works properly.



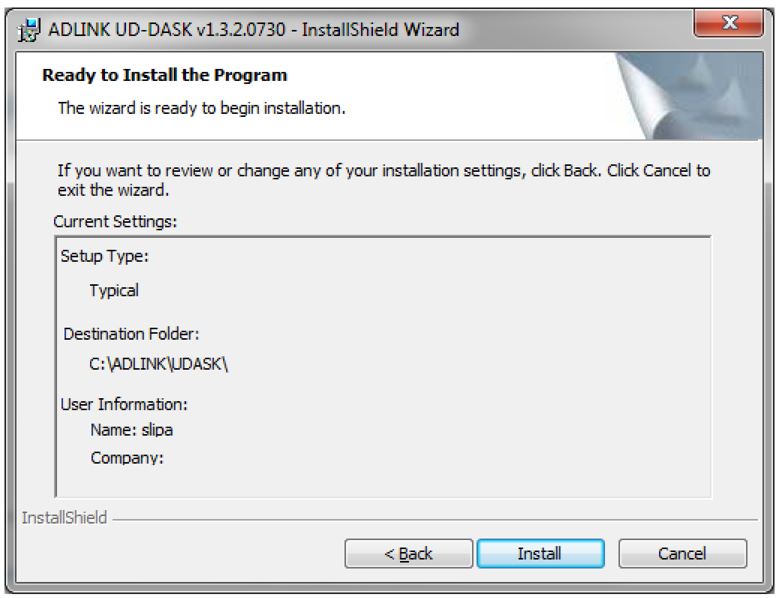
2. Once the installation wizard is opened, click **Next**.



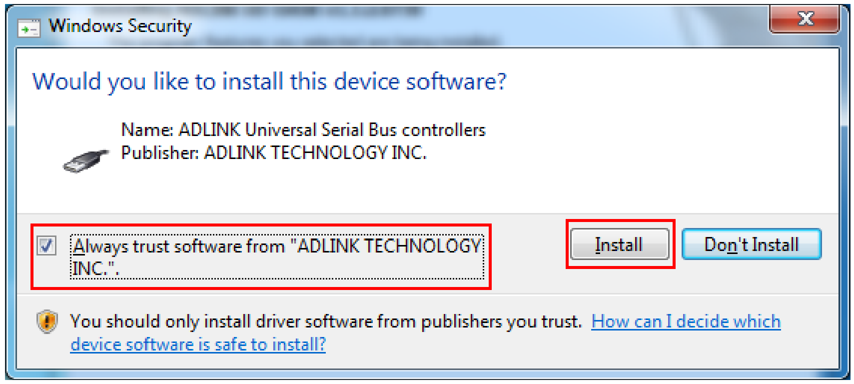
3. Click **Next** to install to the default folder, or click **Change…** to install to a different location.



4. Press the **Install** button to begin the driver installation process.

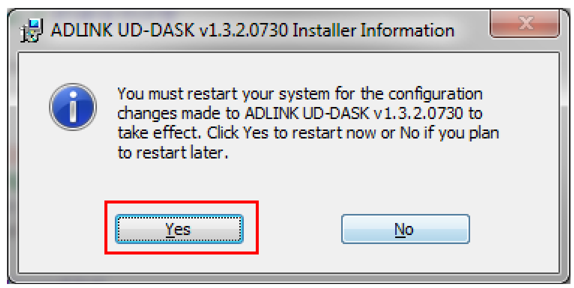


5. During the installation process, Windows will verify the installation the driver. Check the **Always trust software from "ADLINK TECHNLOGY INC."** and click **Install**.



6. Once the installation process is done, click **Finish** to close the wizard.

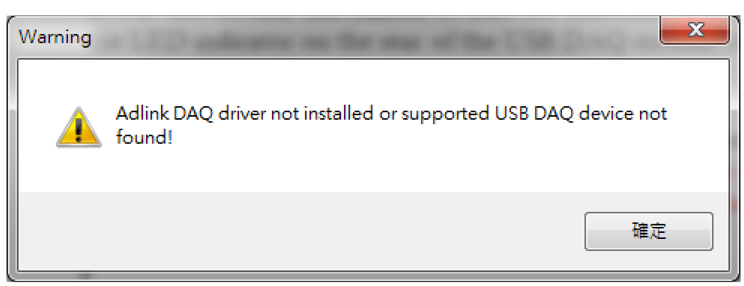
7. A prompt will ask to restart your computer to finalize the installation, click **Yes**.



3.2.2 Connecting the device

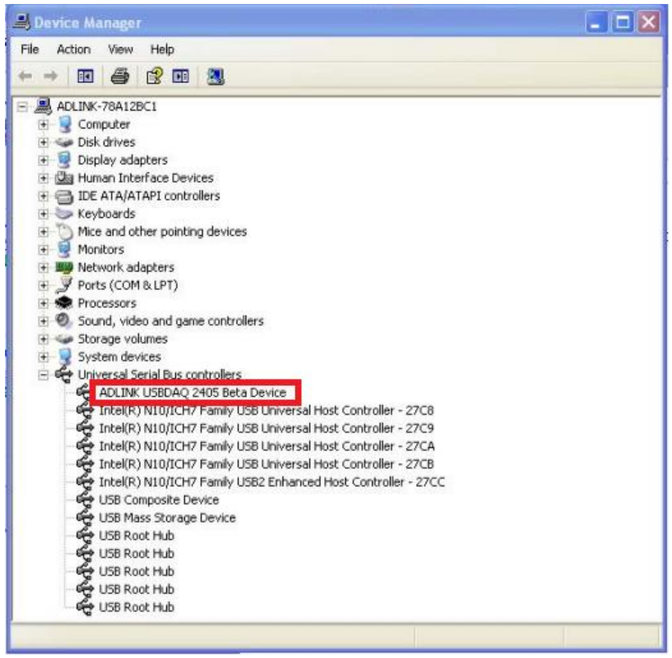
1. Connect the ADLINK USB DAQ module to a USB port on the computer using the included USB cable.

2. The first time the device is connected, a New Hardware message will appear. It will take around one minute to load the firmware. When loading is complete, the LED indicator on the rear of the USB DAQ module changes from amber to green and the New Hardware message will close.
   Note: There are two requirements to starting Visual Signal DAQ Express.
   a. The driver must be installed properly.
   b. The hardware (ADLINK USB-DAQ 2405) is connected.



Note: If any of the following requirements are not met, the following error message will appear.

3. The USB DAQ module can now be located in the hardware Device Manager.

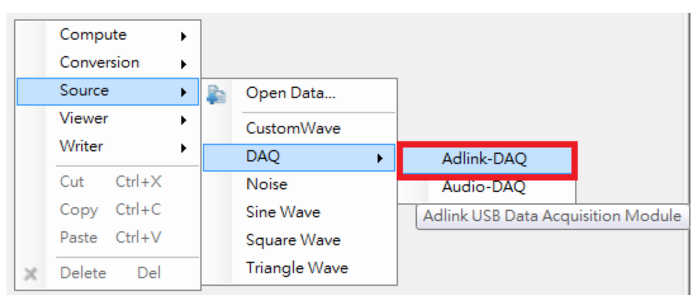


Note: If the device cannot be detected, the power provided by the USB port may be insufficient. The device is exclusively powered by the USB port and requires 400 mA @ 5 V.

3.2.3 Recording data with device

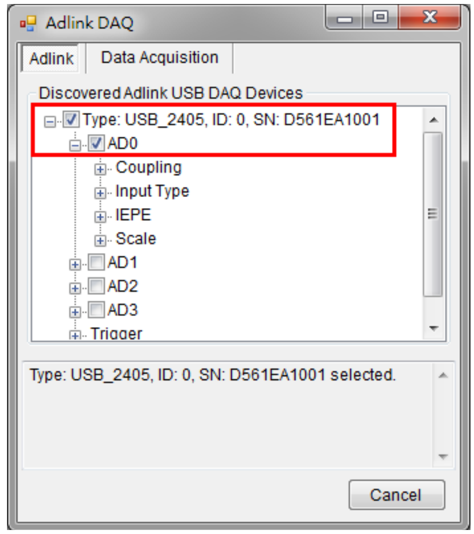
1. Open Visual Signal DAQ Express.

2. Right-click the **Network Window** and select **`Source→DAQ→Adlink-DAQ`**. Attach a **`Channel Viewer`** to the **`Adlink-DAQ`** component if you want to view the recording in a time vs. amplitude graph or attach a **`TF Viewer`** if you want to view the recording in a time vs. frequency graph.



3. Double-click the **`Adlink-DAQ`** component to open the **Adlink DAQ Window**.

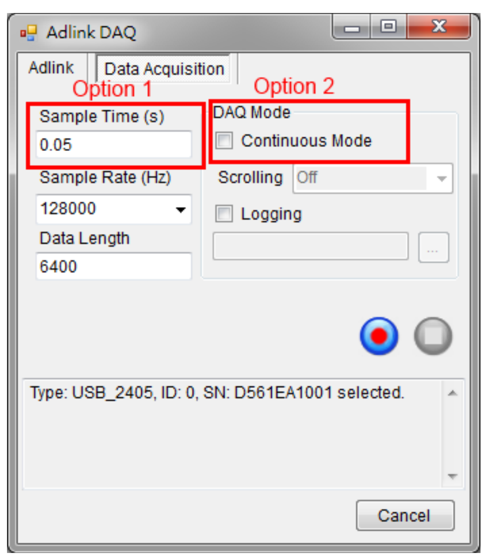
4. In the **Adlink DAQ Window**, select which channels to record from.



Note: If you have your sensor connected to channel 0 on the device, it will be channel 1 in Visual Signal. ADLINK devices start at channel 0, while Visual Signal starts at channel 1.

5. Adjust any settings such as the `Coupling`, `Input Type`, `IEPE`, and `Scale` to work with the signal you are analyzing.
   Note: Refer to the ADLINK User Guide for detailed explanations of these functions.

4. Once you are finished with selecting the channels to use and the settings click the **Data Acquisition** tab. Here you will have two options for recording your signal. The first option is recording for a set amount of time that you set under the `Sample Time (s)` in seconds. The second option is checking the `Continuous Mode` option, this option constantly records signals and displays the data in real-time until stopped by clicking the square button (Stop button) in the **Network Window**. The `Sample Rate (Hz)` applies to both options of recording.



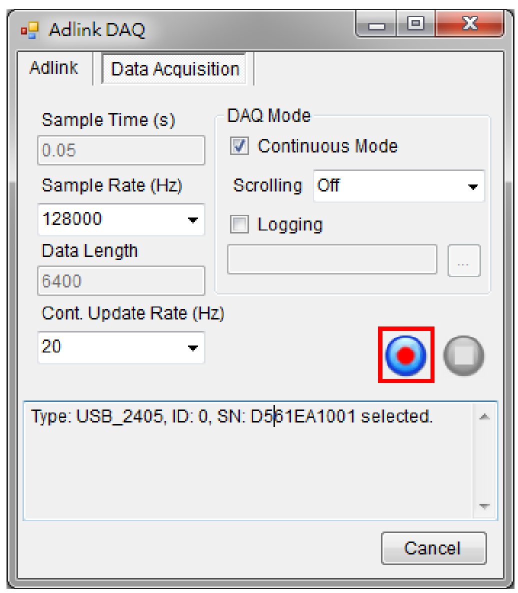
3.2.3.1 Recording signal data in a set amount of time (Option 1)

1. Set the amount of time you wish to record for in the `Sample Time (s)` section in seconds.

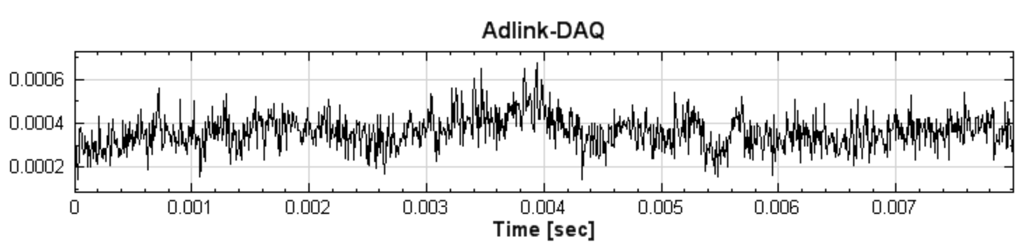
2. Select the sample rate of your recording by adjusting the `Sample Rate (Hz)`. The higher the sampling rate, the more sample points a second resulting in more accurate signals. The highest sampling rate of the ADLINK device is 128000 Hz.

3. The `Data Length` is the number of points recorded during the recording. Typically, this does not need to be changed as it will automatically update on its own based on your sample time and sample rate.

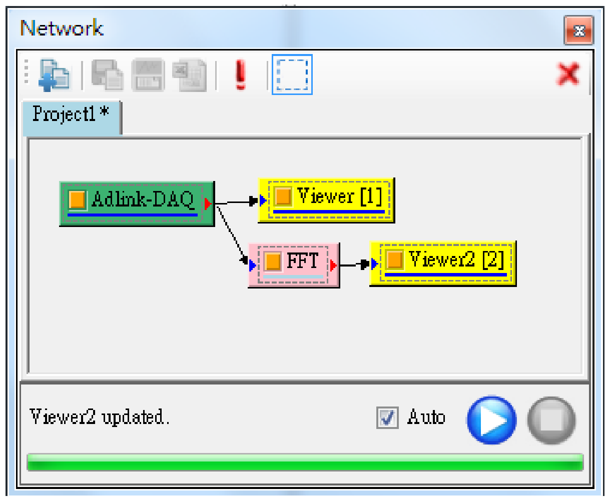
4. Click the red circle (Record button) when you are ready to begin your recording.

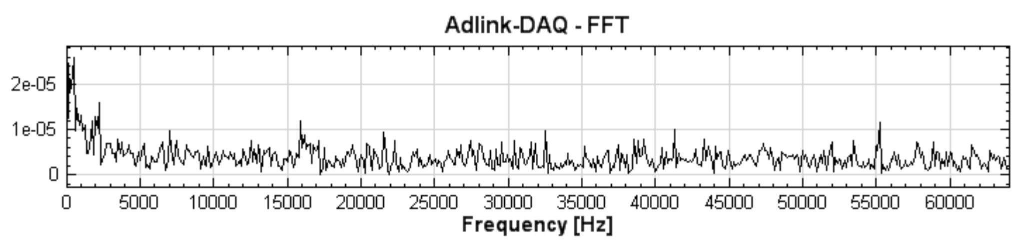


5. After the recording is finished, the data will be stored inside the **`Adlink-DAQ`** component. From here you can attach a **`Channel Viewer`** by right-clicking the component and selecting **`Viewer→Channel Viewer`** to view the signal.



6. Right-click the **Adlink-DAQ** component and select **Compute→ Transform→Fourier Transform**. This will convert the signal into the frequency domain. Right-click the 'FFT' module and select **Viewer→ Channel Viewer** to view the resulting graph.



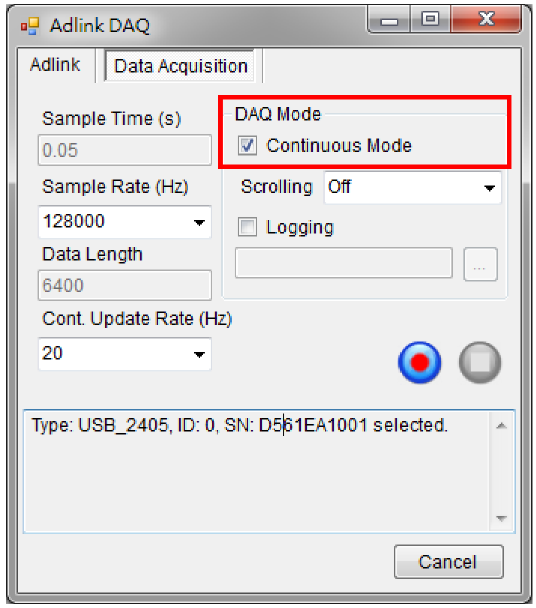


3.2.3.2 Recording signal data in real-time (Option 2)

1. Right-click the **Network Window** and select **Source→DAQ→Adlink-DAQ**. Attach a **Channel Viewer** to the Adlink-DAQ module if you want to view the signal in a time vs. amplitude graph or attach a **TF Viewer** if you want to view the recording in a time vs. frequency graph.

2. Double-click the **Adlink-DAQ** component and select the **Data Acquisition** tab.

3. Check the `Continuous Mode` box under DAQ.



4. Select the recording quality by adjusting the `Sample Rate (Hz)`. The higher the sampling rate, the more accurate the resulting signal will be. The highest sampling rate of the device is 128000 Hz. When DAQ mode is checked the sample time and data length option will be disabled since it will automatically be calculated for you.

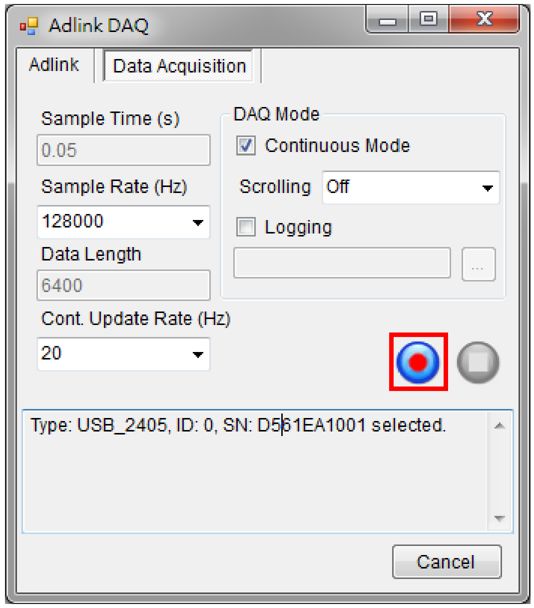
5. Select the desired `Cont. Update Rate (Hz)`

6. Select the type of scrolling mode you want when the data is represented on your attached viewer. The way scrolling works is that for every *n* data points sampled, only 1 data point is used for plotting, and the scrolling mode allows you to choose how you pick that one data point. The different modes include **Off**, **Sample**, **Average**, **Extrema**, and **STFT**.

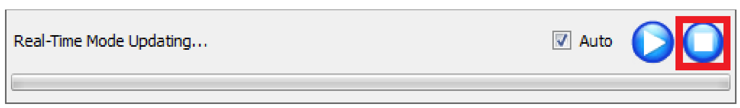
   Off : This mode will have no scrolling and will represent whatever audio signal is currently being recorded on the viewer.
   Sample : This mode picks the first point of the data sampled and plots the point.
   Average : This mode takes the average of the 1024 points and plots the point.
   Extrema : This mode picks the point with the largest absolute magnitude of the 1024 points and plots the point.
   STFT: This mode will output time-frequency data in real time and requires you to attach a time-frequency viewer instead of the standard channel viewer.

7. Once your settings are set, you can decide whether or not you want to have the recording saved to a specific file by checking the **Logging** box and specify the name and location. If the specified file already exists, the old content will be replaced with the new file.

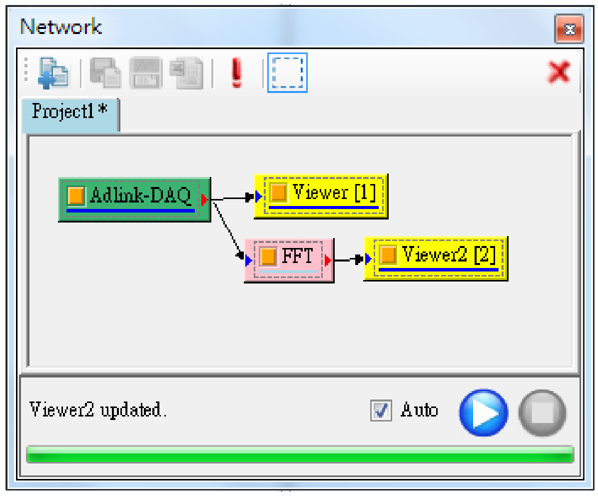
8. Click the red circle (Record button) when you are ready to begin your recording.

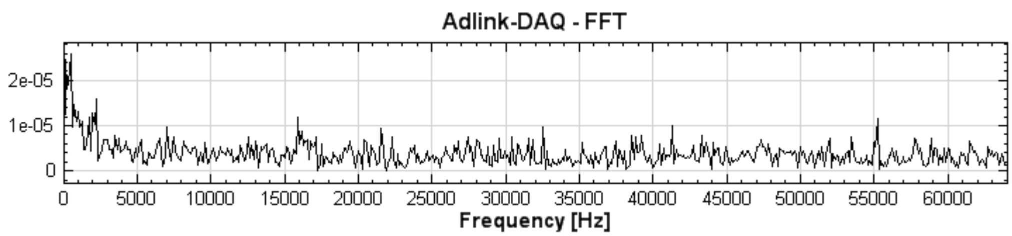


9. Visual Signal will now go into Real-Time Mode Updating which means data is constantly being recorded into your **Adlink-DAQ** component. The viewer you attached to the component will constantly be updated to show you the recorded data in real-time. To finish recording, press the stop button in the **Network Window**.



10. After the recording is finished, the data will be stored inside the Adlink-DAQ module. Right-click the **Adlink-DAQ** component and select **Compute→ Transform→Fourier Transform**. This will perform a Fourier transform and convert the recording into the frequency domain. Right-click the **FFT** component and select **Viewer→Channel Viewer** to view the resulting graph.


Network
Project1 *
Adlink-DAQ
Viewer [1]
FFT
Viewer2 [2]
Viewer2 updated.
Auto


Adlink-DAQ - FFT
2e-05
1e-05
0
0
5000
10000
15000
20000
25000
30000
35000
40000
45000
50000
55000
60000
Frequency [Hz]

Chapter

4

Function List

4.1 Computing With Signal Flow Object

4.1.1 Channel

1. **`Channel Switch`**: Select a single-channel from a multi-channel source.

2. **`Data Selection`**: Select a time frame from a source data to be analyzed.

3. **`Fill NULL Value`**: Use mathematical methods to fill any data that is missing (known as NULL value).

4. **`Input Switch`**: Accept all sorts of input signals, and one signal is chosen.

5. **`Remove Channel`**: Remove a single-channel from a multi-channel source.

6. **`Replace Value`**: Replace a particular value in the signal data.

7. **`Resample`**: Set a new sampling frequency value to a signal data.

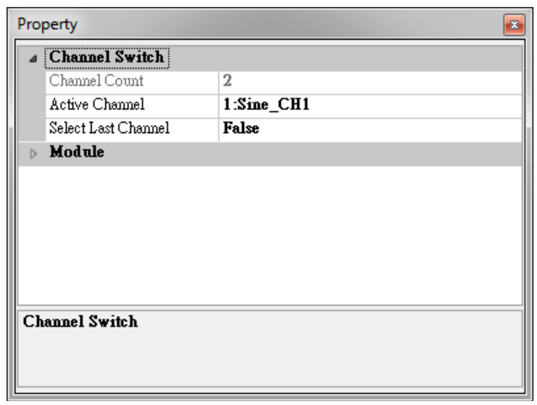
8. **`Time Shift`**: Shift the graph along the x-axis (time).

4.1.1.1 Channel Switch

Select a single-channel from a multi-channel source.

Properties

This module accepts input of Signal (which could be a real number or complex number, multiple-channel, Regular or Indexed), numeric (which could be a real number or complex number, multiple-channel, Regular or Indexed), and Audio (which could be real number or complex number, multiple-channel, Regular). The output formats are real numbers or complex numbers, single channel, and Regular signals. The properties and settings of the **`Channel Switch`** are introduced below.

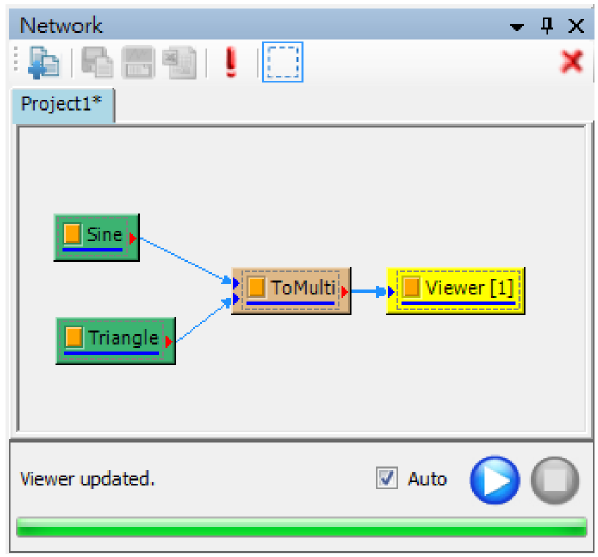


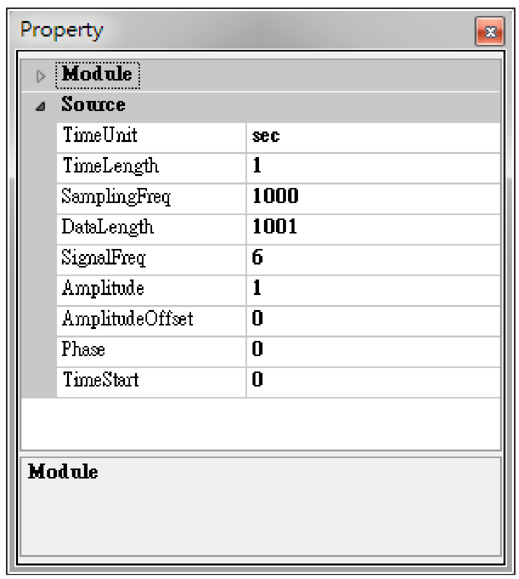
{Channel Switch} Property Name	Property Definition	Default Value
`Channel Count`	Shows the number of channels currently connected to components	Positive integer
`Active Channel`	Select the active channel	Channel 1 (the 1st channel)
`Select Last Channel`	If `Select Last Channel` is set as True, then the channel to be removed will always be the last channel	False

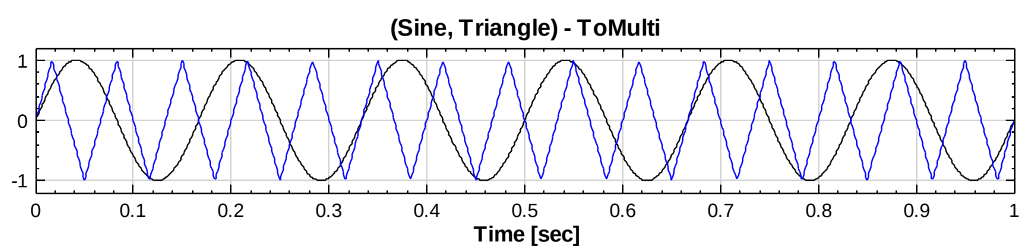
Example

Combine a sine wave data with a triangle wave data into multi-channel and use **`Channel Switch`** to select one of the signal waves to display.

1. Create a **`Source→Sine Wave`** and a **`Source→Triangle Wave`** and connect the two components into a **`Conversion→Merge to Multi-Channel`** to create a multi-channel signal data.



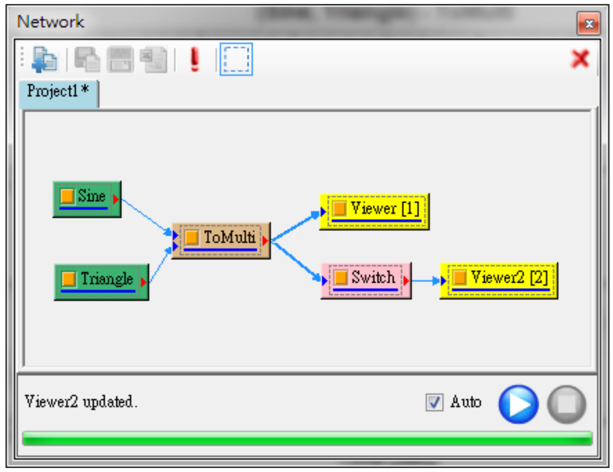


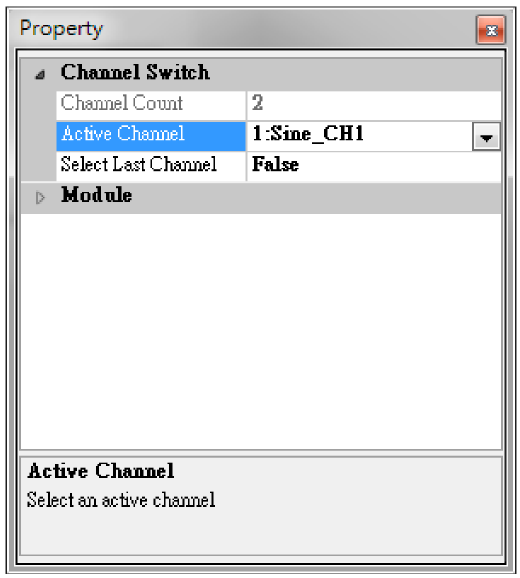


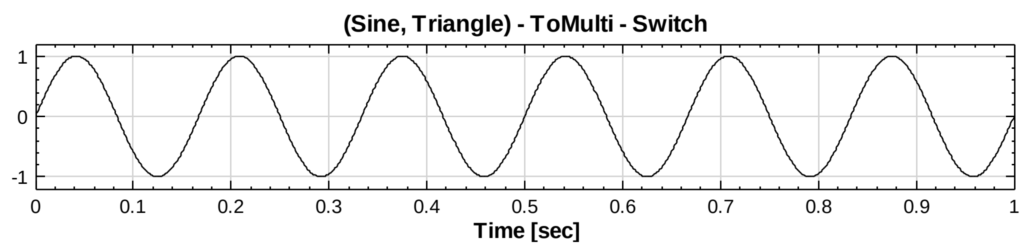
Change the `SamplingFreq` of both Sine and Triangle component to 1000, `SignalFreq` to 6 and 15 respectively.

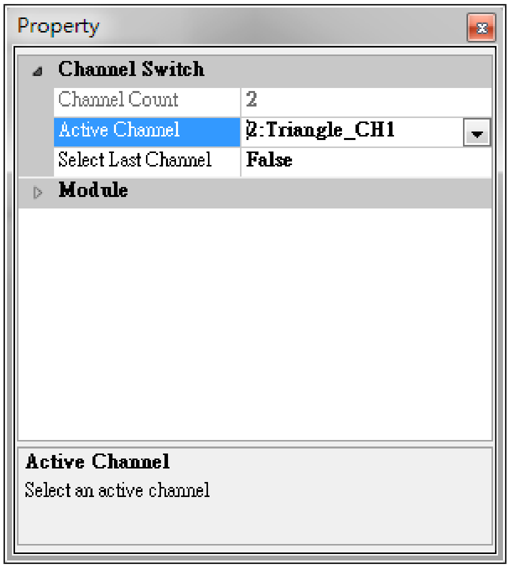
2. Output **`ToMulti`** component to a **`Channel Viewer`** component. In the **`Channel Switch`** component, you can change the `Active Channel` to

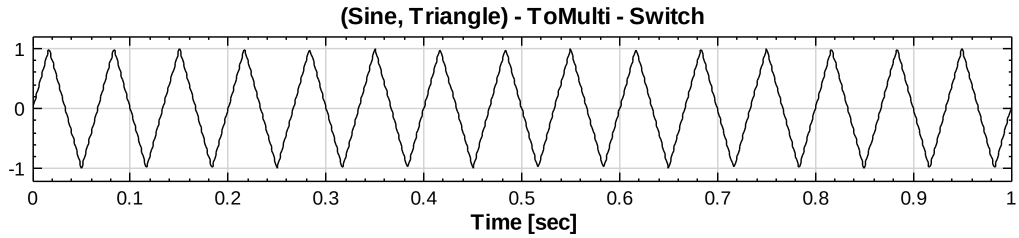
1:Sine_CH1 or 2:Triangle_CH1 to read either the sine wave signal or the triangle wave signal.











Related Functions

Merge to Multi-Channel, Channel Viewer, Source

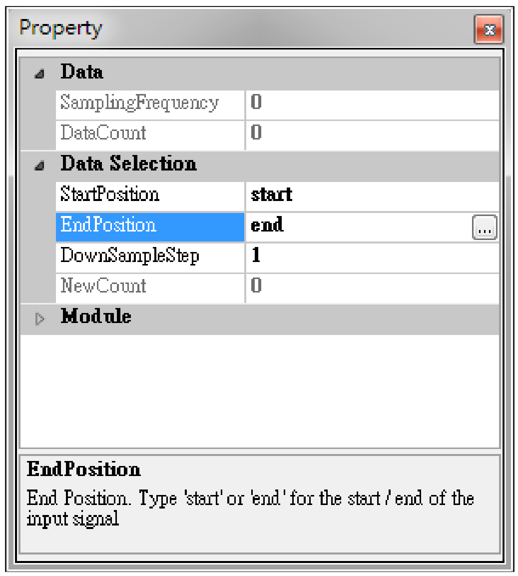
4.1.1.2 Data Selection

Select a time range from a source data to be analyzed.

Properties

This module accepts input of Signal (which could be a real number or complex number, single channel or multi-channel, Regular) and Audio (which could be a real number or complex number, single channel or multi-channel, Regular).

Enter the selected range of the signal by defining the `StartPosition` and `EndPosition` (time unit). You can also move your cursor near the entering field and the "**…**" button will show up. Click the "**…**" button at the right side of the field to enter the **Data Viewer** and select the range by mouse.



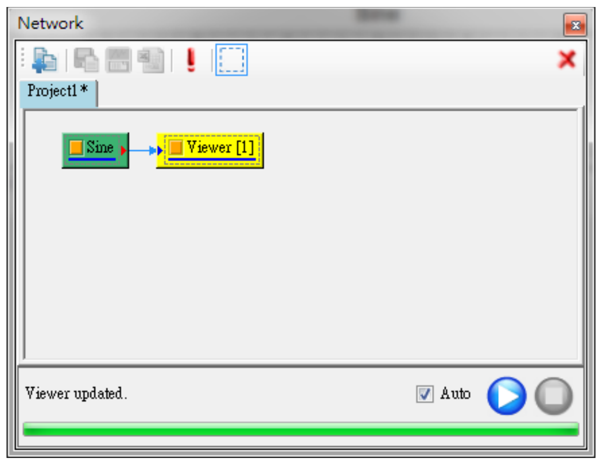
{Data} Property Name	Property Definition	Default Value
SamplingFrequency	Displays the sampling frequency of the input data	0
DataCount	Displays the sampling count of the input data	0

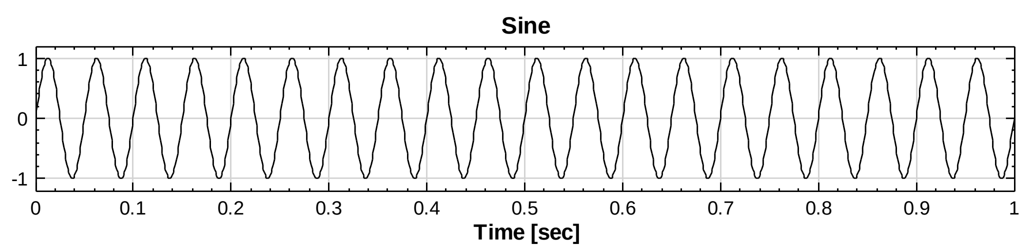
{Data Selection} Property Name	Property Definition	Default Value
StartPosition	Enters the value of the start position of the input data	The original start time for the input data
EndPosition	Enter the value of the end of the input data	The original end time for the input data

DownSampleStep	Reduce the sampling frequency of a signal by dividing an integer	1
NewCount	Display the new data count re-calculated by selected interval and *DownSampleStep*	0

Example

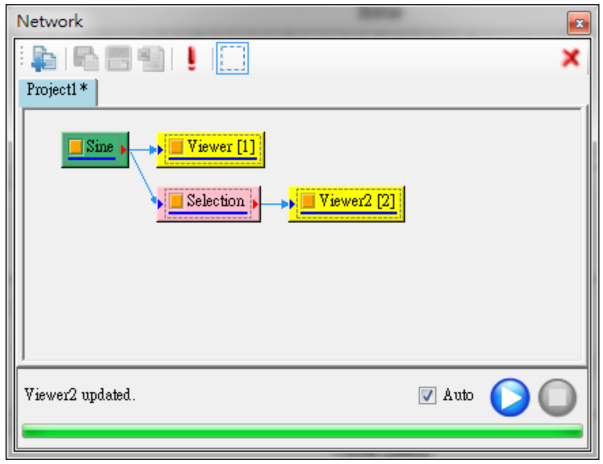
1. Create **Source→Sine Wave** and connect it to **Viewer→Channel Viewer**.

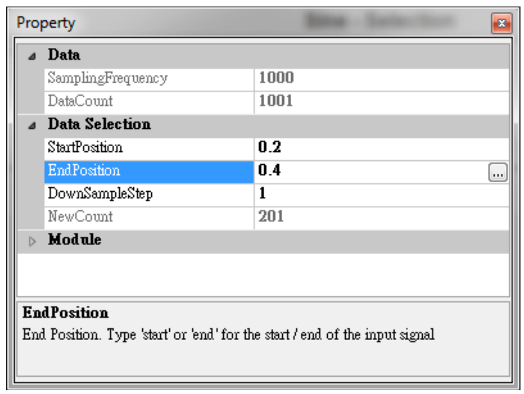


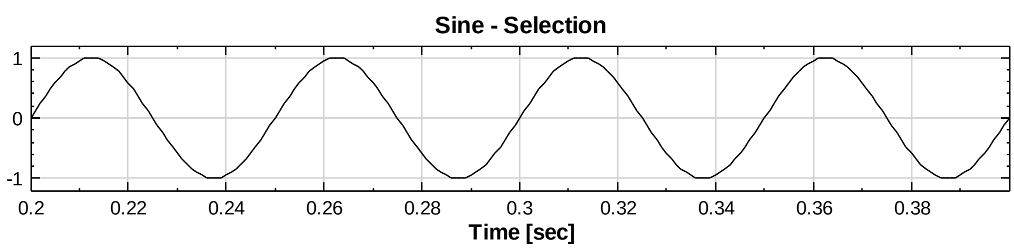


In this Sine component the *SamplingFreq* is 1000 and the *SignalFreq* is 20.

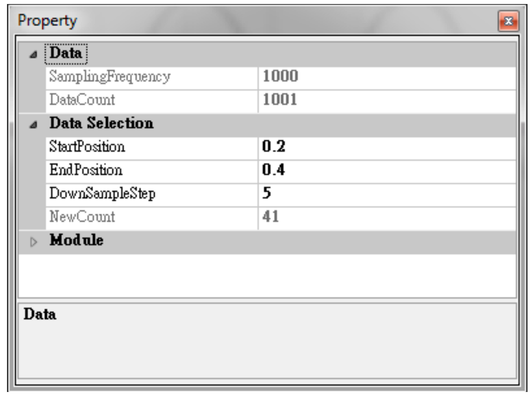
2. Connect a **Compute → Channel → Data Selection** to the Sine component then connect it to **Viewer→Channel Viewer**. In Selection component set the *StartPosition* to $t = 0.2$ and *EndPosition* to $t = 0.4$ and the new graph will be show the range between $t = 0.2$ to $t = 0.4$ (The original range is between $[0, 1]$, so the new range has to be within the original range.)

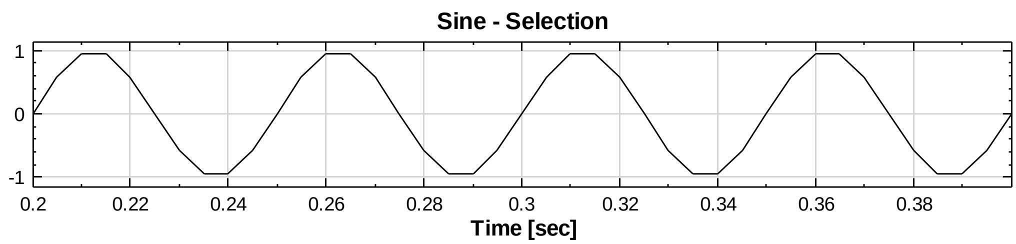




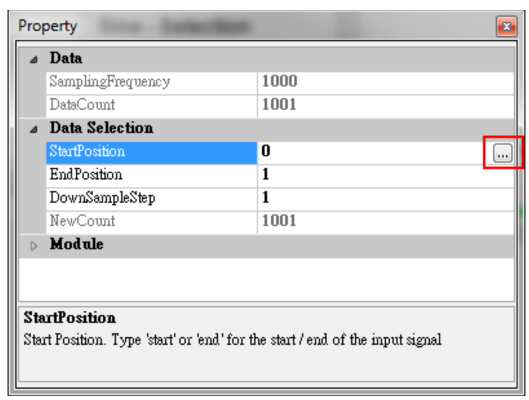


3. The range will now be between $[0.2, 0.4]$ and the `NewDataCount` is 201. Try using `DownSampleStep` to reduce sampling frequency. By setting `DownSampleStep` to 5, the sampling frequency of the sine wave will change to 200 from 1000 and the `NewDataCount` will be 41. This will cause the curve of the sine wave to become rough.

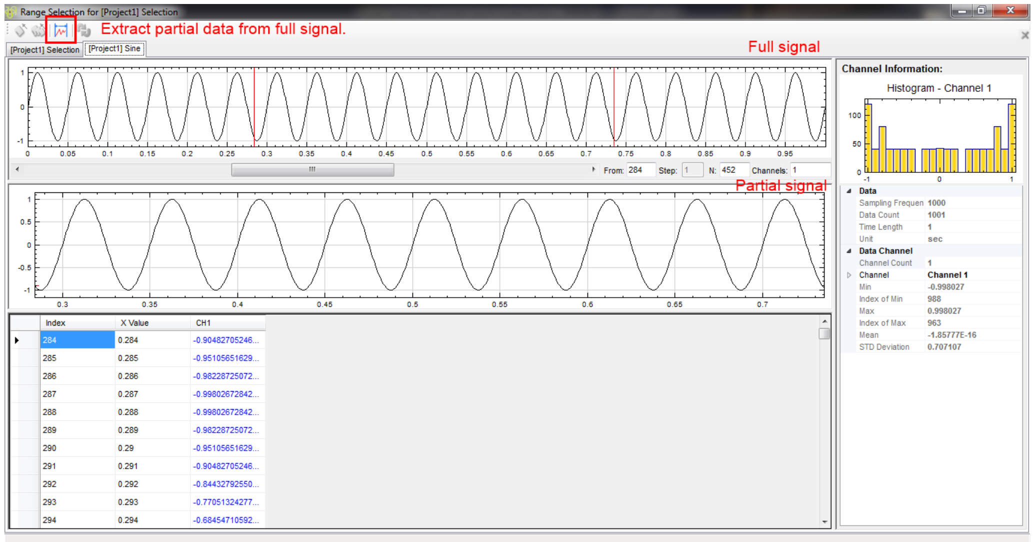




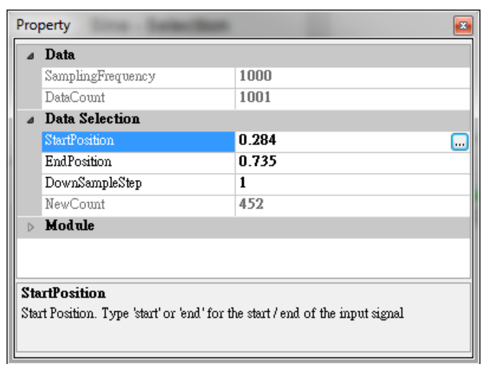
How to select a data range with Data Viewer



1. Click "**…**" to enter the **Data Viewer**. The top graph is the full signal where the range can be selected by dragging your mouse over the desired area. After selecting your range, the selected data will be shown within two red vertical lines as indicated in figure below. The second graph shows the partial data which you selected.



2. After deciding the range, click  to extract the data. The **Selection** component will automatically import the selected value of *StartPosition* and *EndPosition*.



Related Functions

Data Viewer, Channel Switch, Channel Viewer, Source

4.1.1.3 Fill Null Value

Use a mathematical method to fill any data that is missing with the NULL value.

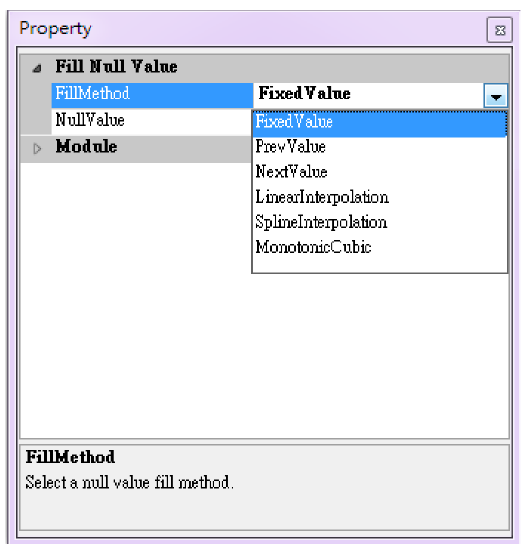
Introduction

To fill in the data signal $x = \{x_0, x_1, \cdots, x_{N-1}\}$, which contains NaN (Not a Number) or NULL values.

Properties

This module accepts input of Signal (which could be a real number, single channel or multi-channel, Regular or Indexed) and Audio (which could be a real number, single channel or multi-channel, Regular).

In the *FillMethod*, there are six methods to fill in the missing values.



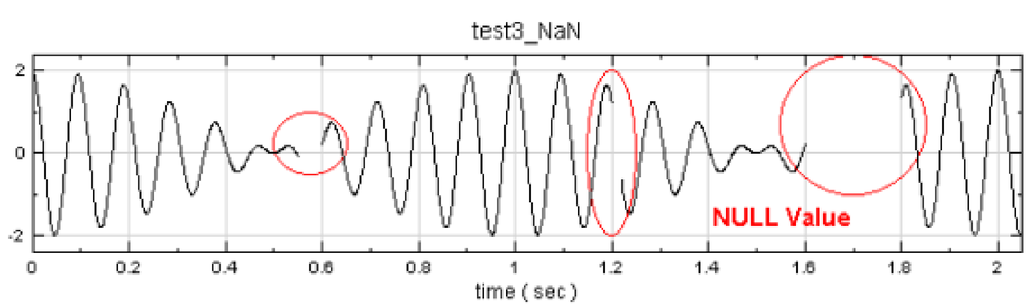
{Fill Null Value} Property Name	Property Definition	Default Value
FillMethod	There are the *FixedValue*, *PrevValue*, *NextValue*, *LinearInterpolation*, *SplineInterpolation*, and *MonotonicCubic* methods to fill the NULL value	*LinearInterpolation*
NullValue	Enter a value to replace all the NULL values.	0

Variable Option	Property Definition
`FixedValue`	A new `NullValue` variable option will appear in the **Properties Window**. Enter a value to replace all the NULL values to the values entered
`PrevValue`	The NULL value will be replaced with the previous value in the signal
`NextValue`	The NULL value will be replaced with the next available value in the signal
`LinearInterpolation`	Using linear interpolation to calculate the value of the NULL
`SplineInterpolation`	Using spline interpolation to calculate the value of the NULL
`MonotonicCubic`	Monotone cubic interpolation is a type of cubic interpolation that preserves monotonicity of the data set being interpolated. `MonoticCubic` method is better than `SplineInterpolation` method when the slope of the signal is large e.g. **`Square Wave`**

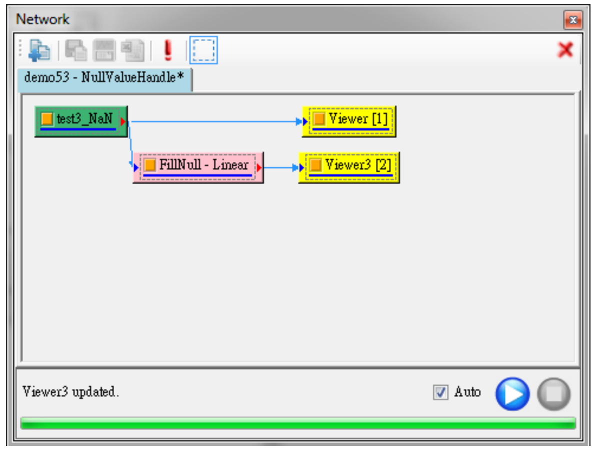
Example

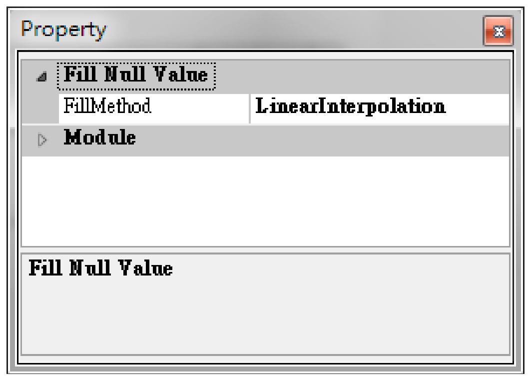
To fill in the missing values using **`Fill NULL Value`** component.

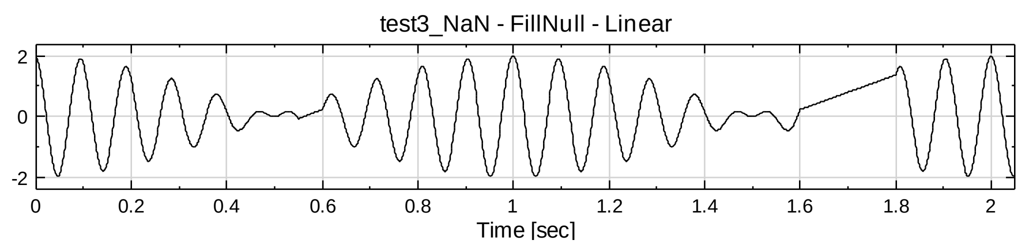
1. Open demo53 in the directory C:\Program Files\AnCAD\Visual Signal\demo. From the graph in the **Visualization Window**, you can clearly see the missing values on the graph.
   Note: File locations will be different depending on platform (x86 or x64) or the installation path you selected.



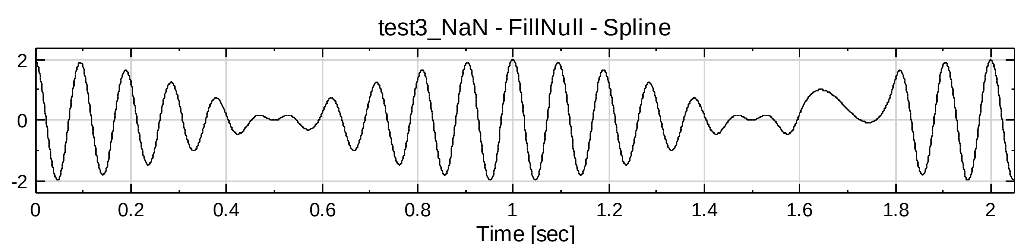
2. Connect the source signal data to **`Compute`**→**`Channel`**→**`Fill NULL Value`** and select `LinearInterpolation` in the `FillMethod` of the **`Fill NULL Value`** component.



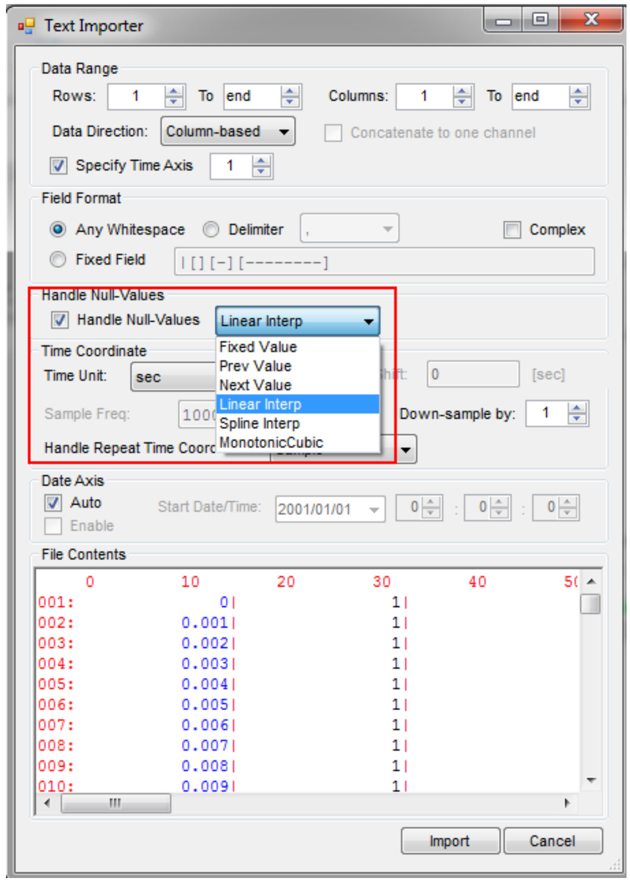




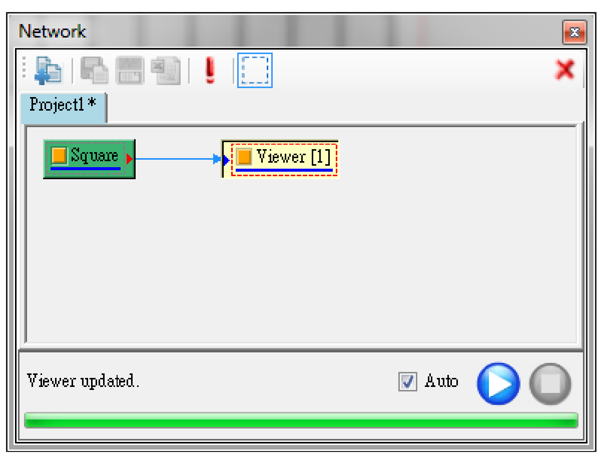
3. Select `SplineInterpolation` method instead and the way the values are filled in will be considerably different.

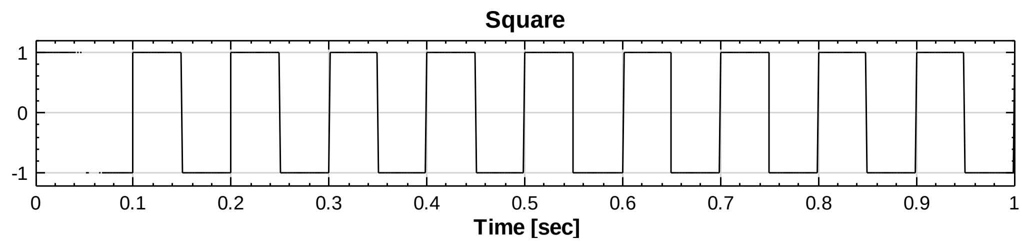


4. In the **Text Importer** there is also an option to fill in the missing value but this feature is different from the **`Fill NULL Value`** component.

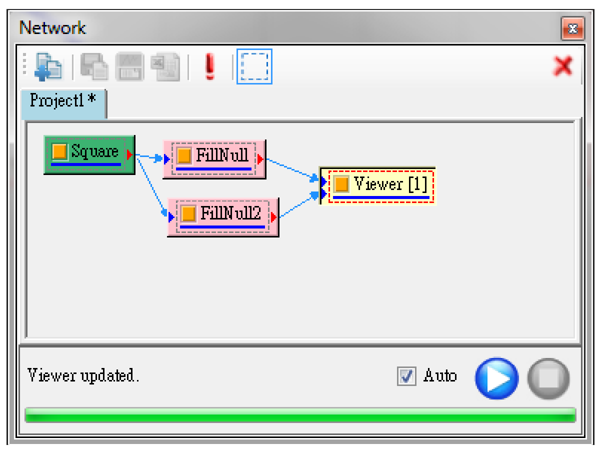


5. Import a data which has missing values and intentionally uncheck the box `Handle Null-Values` in **Text Importer**. The imported value and graph is shown in the image below.

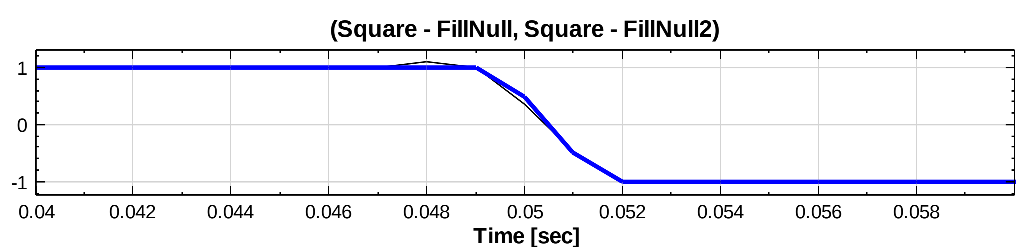




6. Now create two **Fill NULL Value** components to connect to the imported source signal data, where **Square Wave** has some removed points. The first component will use the *SplineInterpolation* fill in method and the second component will use the *MonotonicCubic* fill in method.



7. From the results shown in the **Channel Viewer**, there is an obvious difference between the *SplineInterpolation* method (thin dark line) and the *MonotonicCubic* method (thick blue line).



Related Functions

Text Importer, Resample, Channel Viewer, Source

4.1.1.4 Input Switch

Select one channel from a multi-channel input signal.

Properties

This module accepts input of Signal (which could be a real number or complex number, single channel or multi-channel, Regular or Indexed), Audio, numeric, and spectra. The definition and default value of the parameters are shown below.

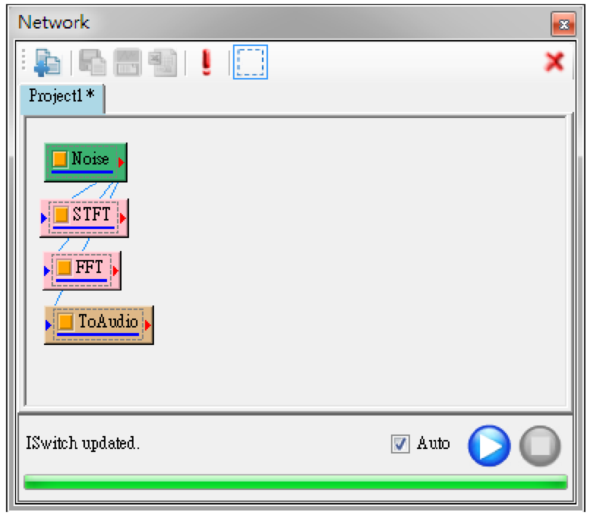
{Input Switch} Property Name	Property Definition	Default Value
Input Count	Total number of channels in this component	0
Active Input	The selected channel number	1

Example

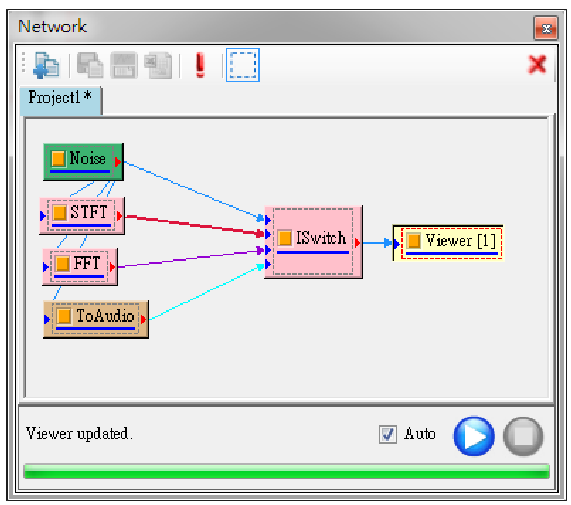
Use **Source→Noise** as the source signal, carry out different calculations on it and output the results in a different format. Then use **Input Switch** to select one of the channels.

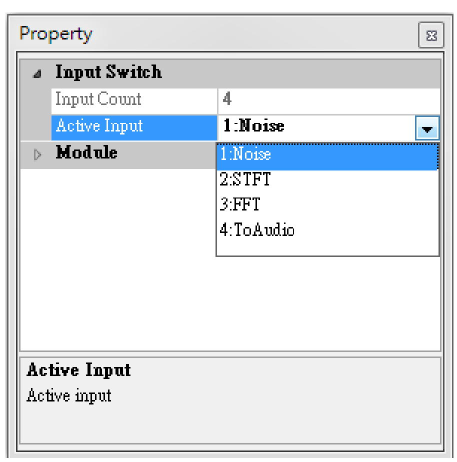
1. Create **Source→Noise** in **Network Window**. In the **Property Window** set *TimeLength* to 3, set *SamplingFreq* to 1000, and set *Amplitude* to 1.

2. Connect the **Noise** component to **Compute → TFA → ShortTerm Fourier Transform**, **Compute → Transform → Fourier Transform**, and **Conversion→Convert to Audio**, respectively.

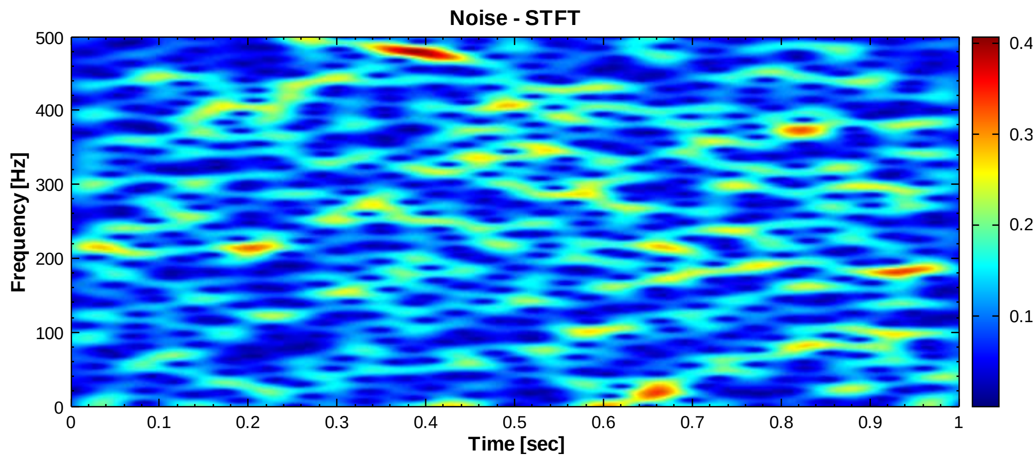


3. Connect all outputs of the calculations to **Compute→Channel→Input Switch**, and view the result with a **Channel Viewer**. Change the *Active Input* setting in **Input Switch** to view different results.





4. Connect **`Input Switch`** to **`Viewer`**→ **`TFA Viewer`** and change the *`Active Input`* setting to observe the result of the STFT.



Related Functions

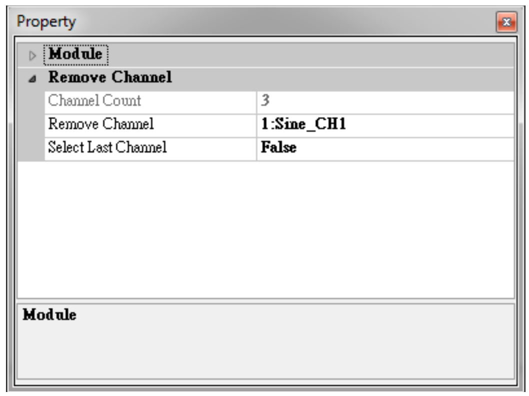
`Fourier Transform`, **`Convert to Audio`**, **`ShortTerm Fourier Transform`**, **`Time-Frequency Viewer, Source`**

4.1.1.5 Remove Channel

Remove a single-channel from a multi-channel source.

Properties

This module accepts input of Signal (which could be a real number or complex number, multi-channel, Regular or Indexed) and Audio (which could be a real number or complex number, multi-channel, or Regular).

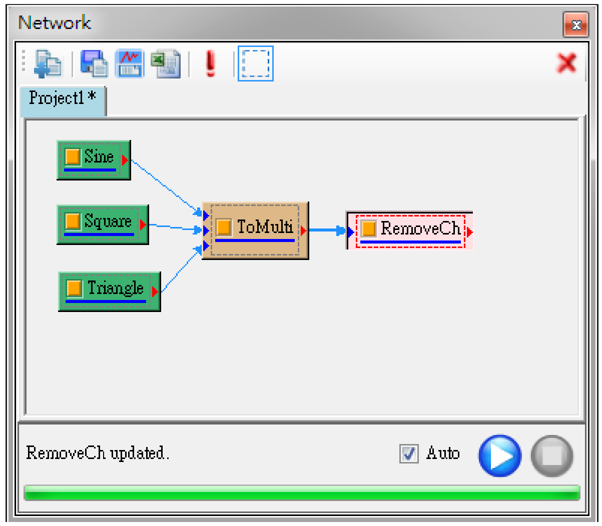


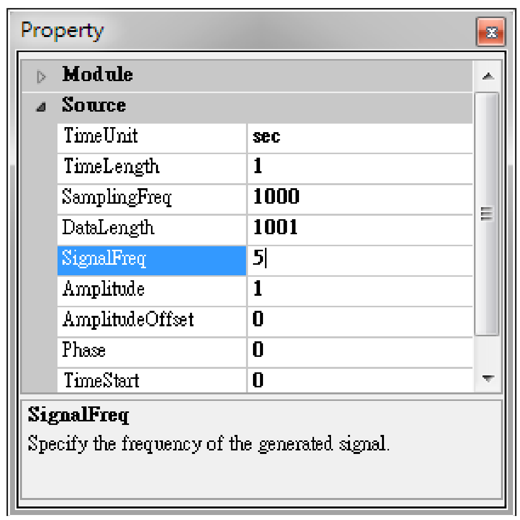
{Remove Channel} Property Name	Property Definition	Default Value
`Channel Count`	Displays the number of channels	0
`Remove Channel`	Select the channel to be removed	Channel 1
`Select Last Channel`	If `Select Last Channel` is set as True, then the channel to be removed will always be the last channel	False

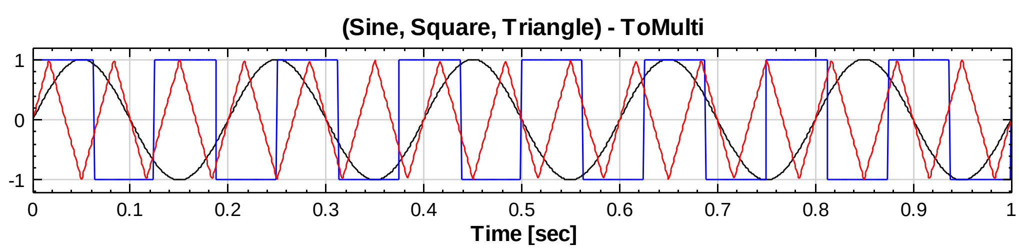
Example

Combine a sine wave, a triangle wave and a square wave together, connect it to a **`Remove Channel`** component and remove the sine wave.

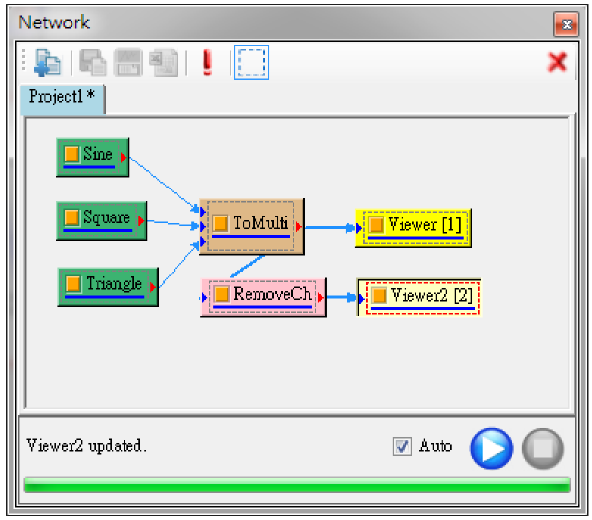
1. Create **`Source`→`Sine Wave`**, **`Square Wave`** and **`Triangle Wave`** and connect them all to a **`Conversion`→`Merge to Multi-Channel`** to make the three waves into a multi-channel signal data. Set the *`SamplingFreq`* as 1000 and set the **`Sine`** component's *`SignalFreq`* as 5, **`Square`** component's *`SignalFreq`* as 8 and **`Triangle`** component's *`SignalFreq`* as 15 and observe the different waves on the graph.

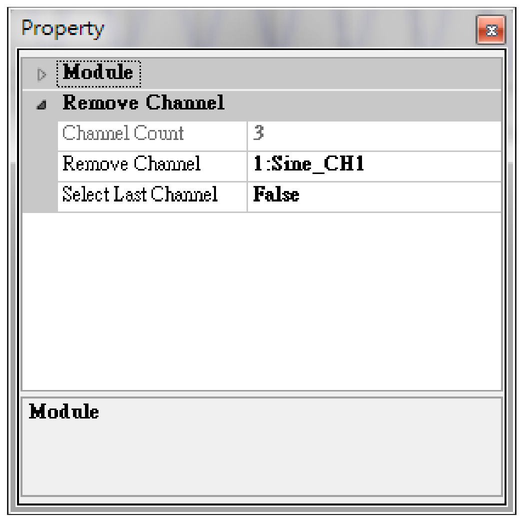


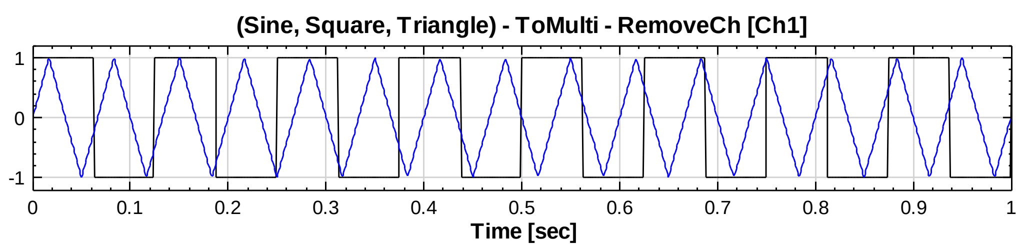




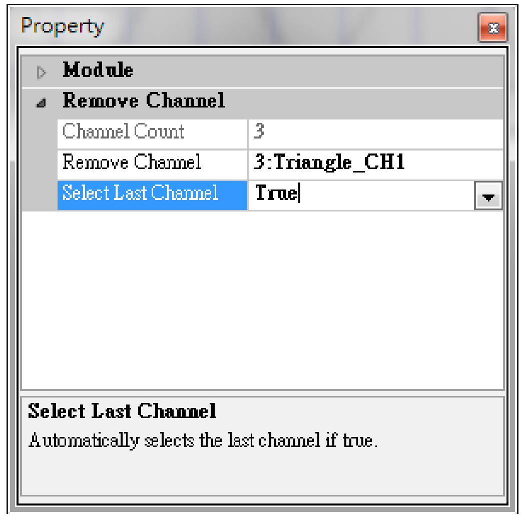
2. Connect the **`Merge to Multi-Channel`** component to **`Compute→ Channel→Remove Channel`** and set *`Remove Channel`* as 1:Sine_CH1 (sine wave).

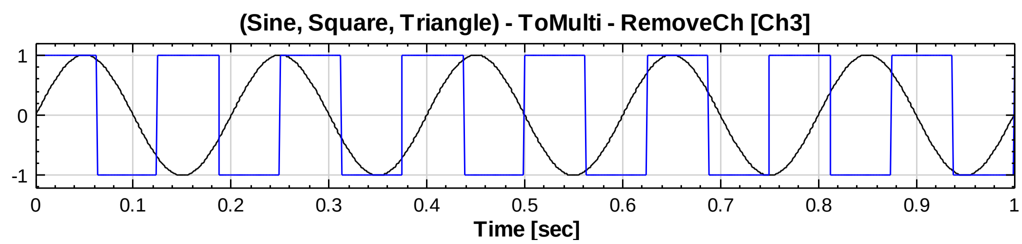






3. Now set the `Select Last Channel` as True and you will see that `Remove Channel` will automatically change to 3:Triangle_CH3 and the triangle wave signal (Channel 3) will be removed.





Important Note

`Channel Switch` and **`Remove Channel`** are completely opposite functions. **`Channel Switch`** preserves a single selected channel from a multi-channel signal data and **`Remove Channel`** removes the single selected channel from a multi-channel signal data.

Related Functions

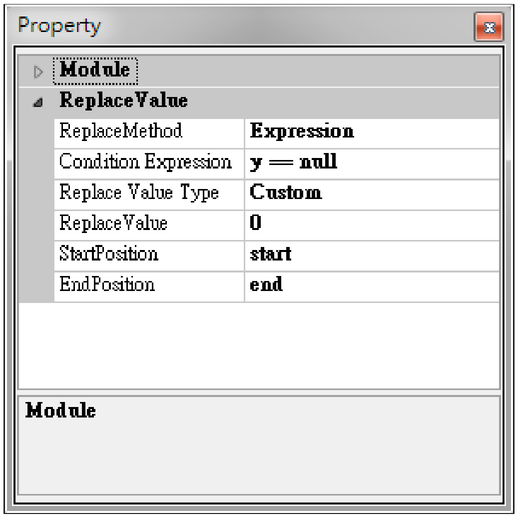
`Merge to Multi-Channel, Channel Switch, Source`

4.1.1.6 Replace Value

Replace a particular value in the signal data.

Properties

This module accepts input of a signal (which could be a real number, single channel or multi-channel, Regular) and Audio (which could be a real number, single channel or multi-channel, Regular) and replaces specific values to another value.



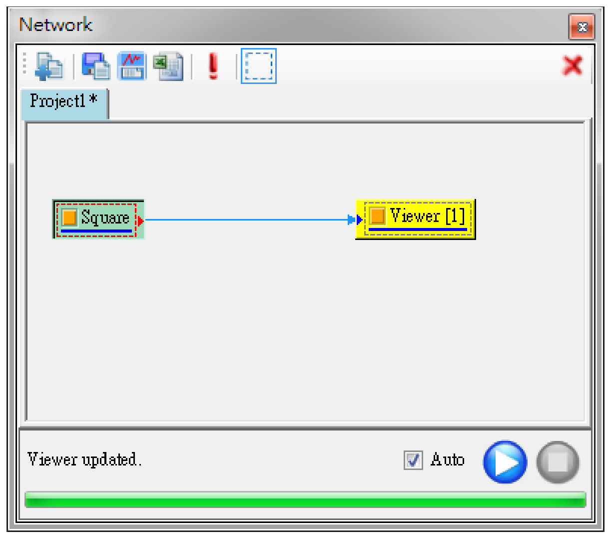
{Replace Value} Property Name	Property Definition	Default Value
`ReplaceMethod`	There are `All`, `Expression`, `Outlier`, and `ByPass` methods to replace value.	Expression
`Condition Expression`	Use the conditional expression to replace value. The arithmetic operators are available to use. Only for `Expression` mode.	y = null
`Replace Value Type`	There are three kinds of value types, `Custom`, `NULL`, `mean`, and `median`.	Custom
`ReplaceValue`	Set the value which signal data will be replaced with.	0
`StartPosition`	The start position of x-axis to begin replacing.	The start point of the input signal.
`EndPositon`	The end position of x-axis to end replacing.	The end point of the input signal.
`Outlier Boundary`	The multiples of the standard deviation. Only for `Outlier`	3

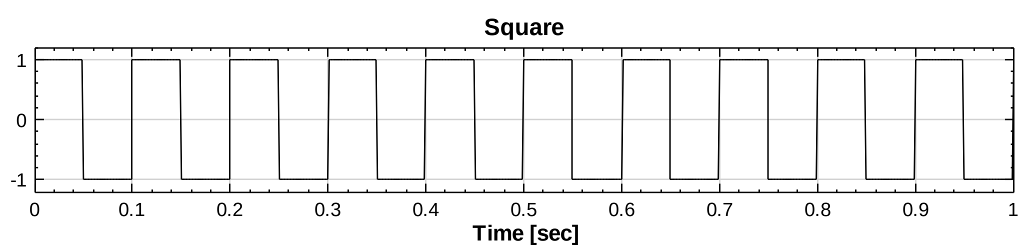
	mode.	
Sliding Window Mode	True sets the sliding window which is used for the `Outlier` mode. Only for `Outlier` mode.	False

Example

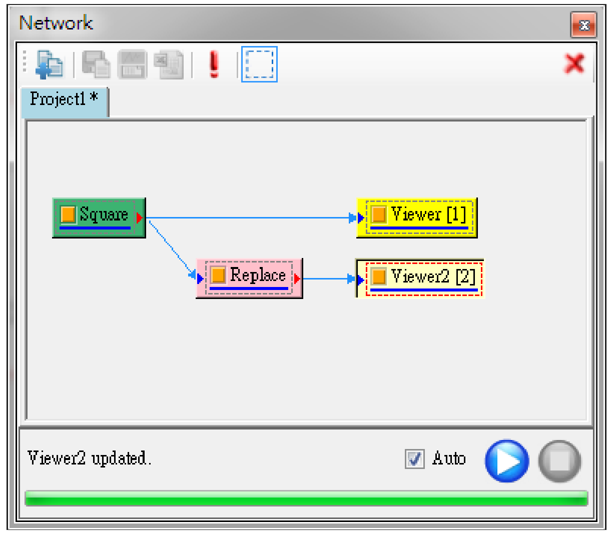
Change the maximum value of the square wave to another number

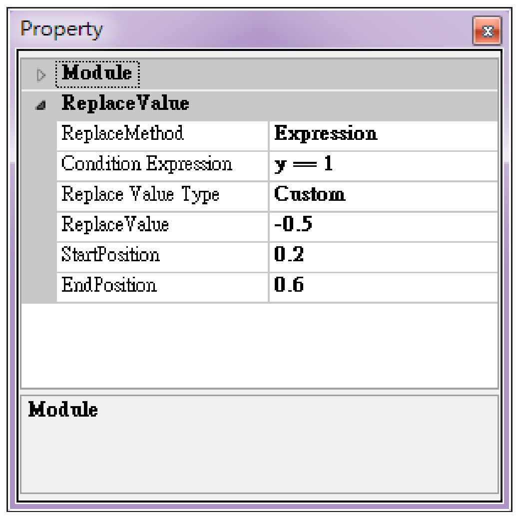
1. Create a **Source→Square Wave** and connect it to a **Viewer→Channel Viewer**.

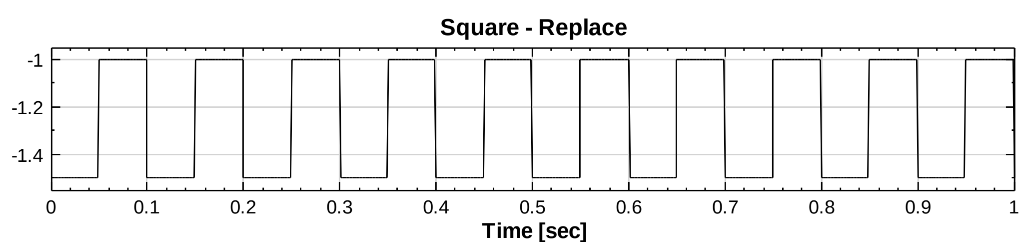




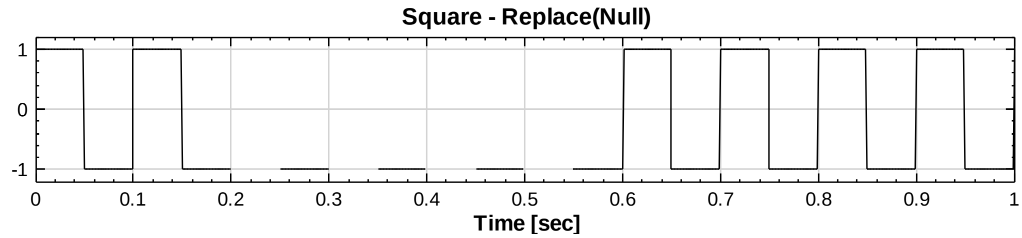
2. Now connect the **Square** component to **Compute→Channel→Replace Value** and modify the *Condition Expression* to y==1. Then set the *ReplaceValue* to -0.5, *StartPosition* to 0.2, and *EndPosition* to 0.6. Now all the values in $[0.2, 0.6]$ of the square wave which were originally 1 will become -0.5.

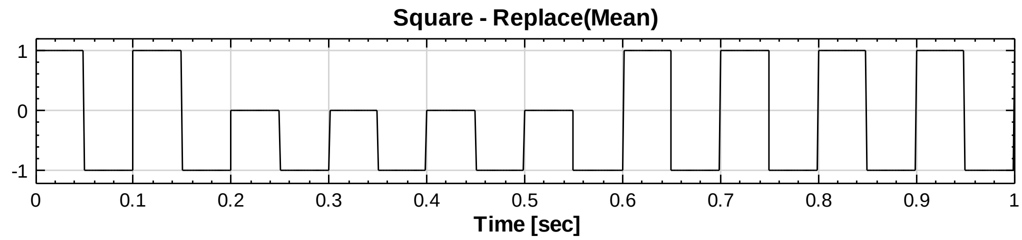


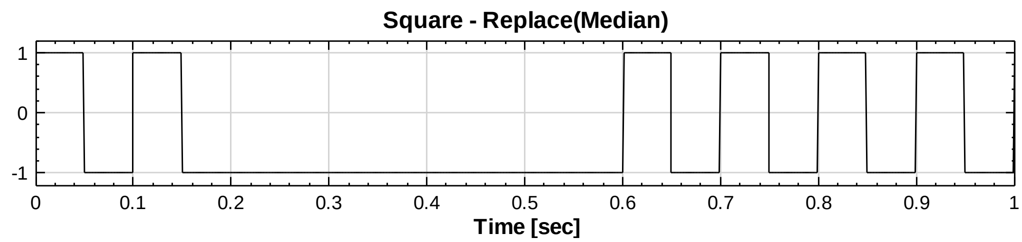




3. Change `Replace Value Type` to `Null`, `Mean`, or `Median` and observe the difference between these types. `Null`, `Mean`, and `Median` will replace values with a null value or the mean or median of a specified range, refer to the bottom figures to observe the differences.







Important Note

You can only replace one value at a time. If you want to replace multiple values then several **`Replace Value`** components will have to be created.

Related Functions

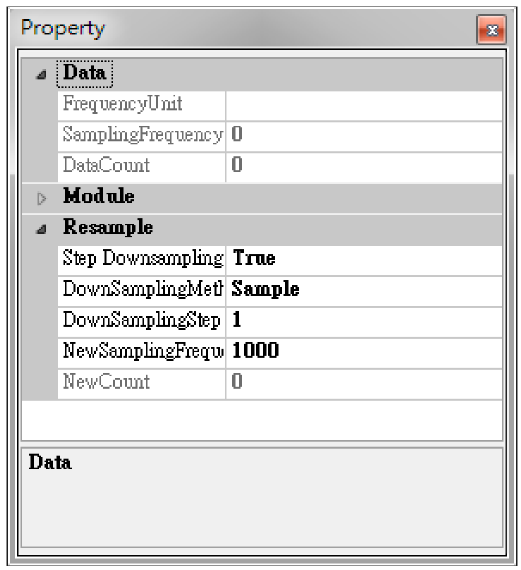
`Source, Channel Viewer`

4.1.1.7 Resample

Resample allows you to set a new sampling frequency value to a signal data.

Properties

This module accepts input of Signal (which could be a real number or complex number, single channel or multi-channel, Regular) and Audio (which could be a real number or complex number, single channel or multi-channel, Regular)



{Data} Property Name	Property Definition	Default Value
`FrequencyUnit`	Sampling frequency unit of input data	None
`SamplingFrequency`	Sampling frequency of input data	0
`DataCount`	Sampling count of input data	0

{Resample} Property Name	Property Definition	Default Value
`Step Downsampling`	If set to True, the input data will be down-sampled with `DownSamplingStep`	True
`ReSamplingMethod`	Select a resampling interpolation method: `Nearest`, `Linear`, `Spline`, and `MontonicCubic`	

DownSamplingMethod	Select a down-sampling method: *Sample*, *Average*, *MaxDetect*, *MinDetect*, or *PeakDetect*	*Sample*
DownSamplingStep	Set the integer ratio for down-sampling if *Step Downsamping* sets True	1
NewSamplingFrequency	Set the new sampling frequency	1000
NewCount	Display the new sampling count of the data output	0

{DownSamplingStep} Variable Option	Property Definition
Sample	Set the range of samples of *DownSamplingStep*. Picks the first point of samples.
Average	Set the range of samples of *DownSamplingStep*. Picks the average value of samples
MaxDetect	Set the range of samples of *DownSamplingStep*. Picks maximum value of samples.
MinDetect	Set the range of samples of *DownSamplingStep*. Picks minimum value of samples.
PeakDetect	Set the range of samples of *DownSamplingStep*. Picks maximum and minimum values of samples.

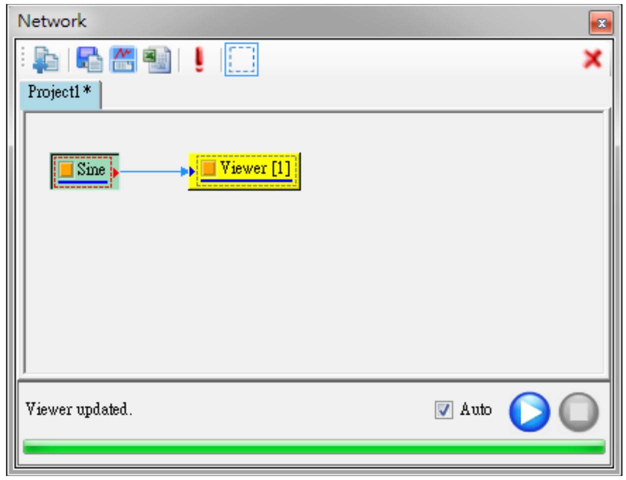
{ReSamplingMethod} Variable Option	Property Definition
Nearest	Use the nearest value to fill the new sample
Linear	Uses Linear Interpolation to calculate the value of the re-sample
Spline	Uses Spline Interpolation to calculate the value of the re-sample
MonotonicCubic	Monotone cubic interpolation is a type of cubic interpolation that preserves monotonicity of the data set being interpolated. MonotonicCubic method is better than SplineInterpolation method when the slope of the signal is large e.g. Square wave

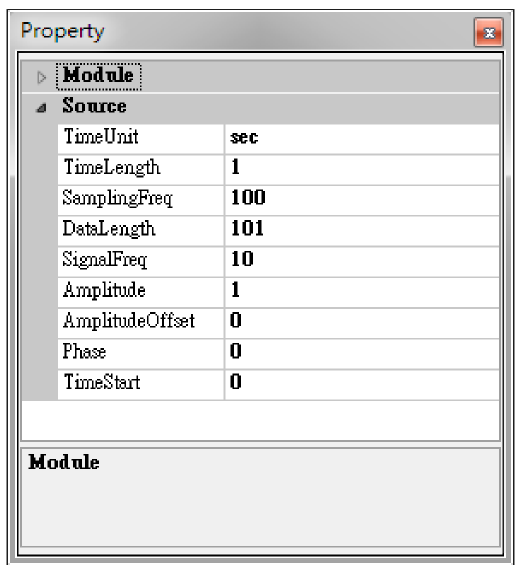
Example

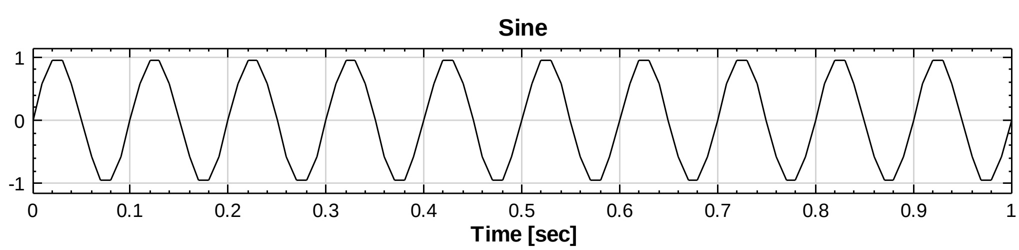
Create a **Sine Wave** component and apply **Resample** component to it.

1. Create **Source→Sine Wave** and edit the *SamplingFreq* to 100 and *DataLength* to 101. Connect the **Sine Wave** component to **Viewer→**

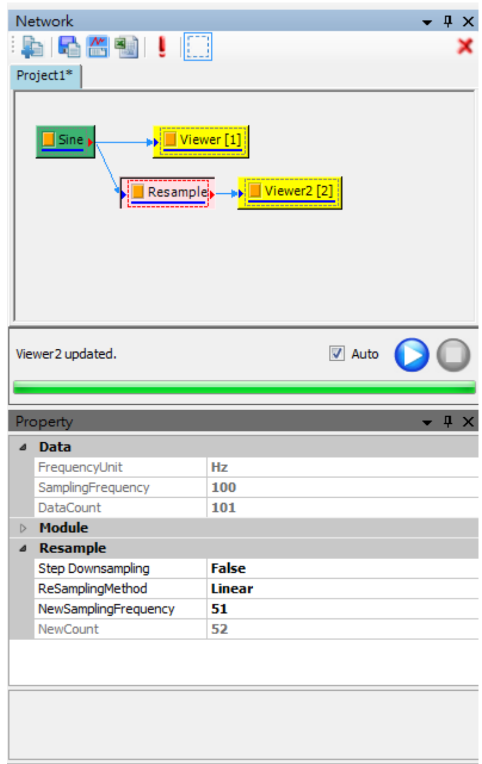
`Channel Viewer` to see the graph. You can clearly see from the graph that the wave signal is not as smooth.

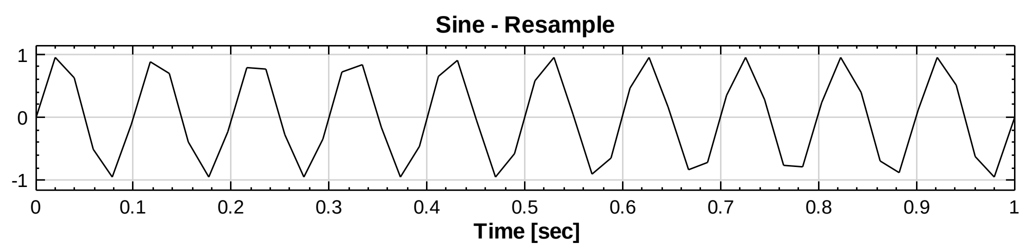




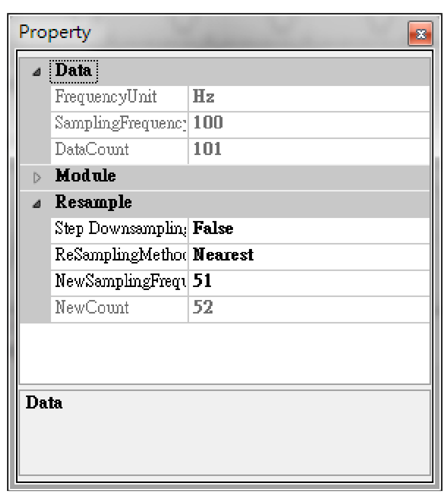


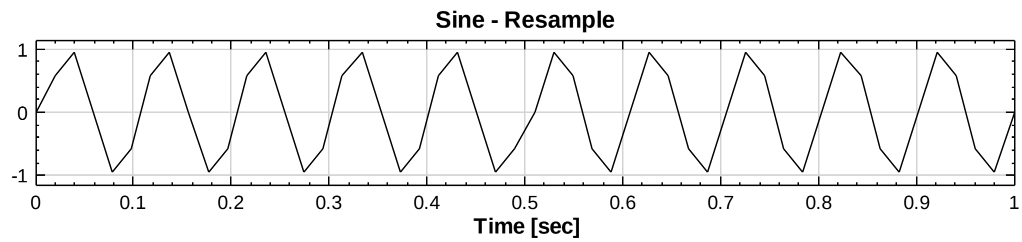
2. Connect the **`Sine Wave`** component to **`Compute→Channel→Resample`** and edit the value of *`NewSamplingFrequency`* to 51 and *`UpsamplingMethod`* to *`Linear`* to compare the difference between the two.



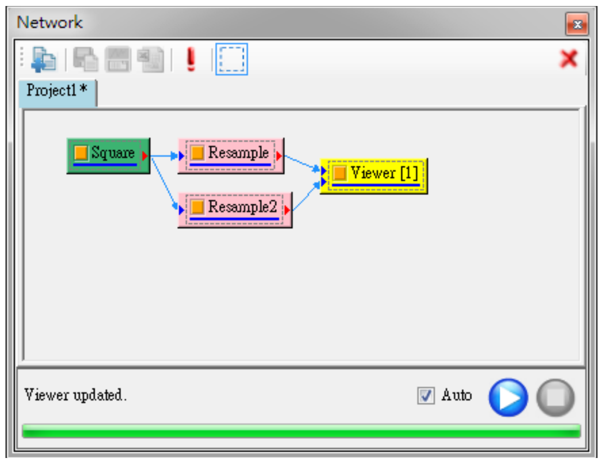


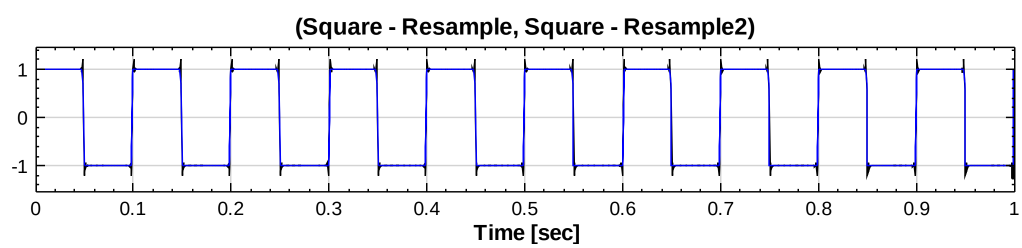
3. Now try changing the `UpsamplingMethod` to `Nearest`.



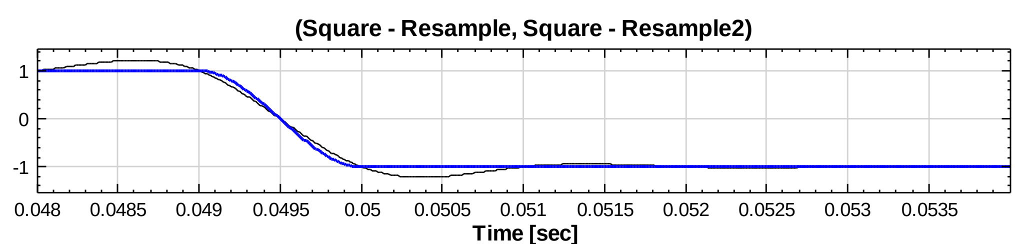


4. Create a **`Square`** component and connect it to two **`Resample`** components and set one of the **`Resample`** component's *`NewSamplingFrequency`* to 100000 and *`UpSamplingMethod`* to *`Spline`* and the other **`Resample`** component's *`NewSamplingFrequency`* to 100000 and *`UpSamplingMethod`* to *`MonotonicCubic`* and connect both **`Resample`** components to the same **`Channel Viewer`** component.





Notice the slight difference around the corners of both wave signals. Zoom into the graph for a closer look.



The thin black line is created through the *`Spline`* method and the thick blue line is created through the *`MonotonicCubic`* method. From the graph you can observe that *`Spline`* method has a tendency of overshooting while the *`MonotonicCubic`* method does not have that problem.

Related Functions

`Source, Channel Viewer, Fill NULL Value`

Reference

Numerical Recipes 3rd Edition: The Art of Scientific Computing by William H. Press, Saul A. Teukolsky, William T. Vetterlingm, Brian P. Flannery

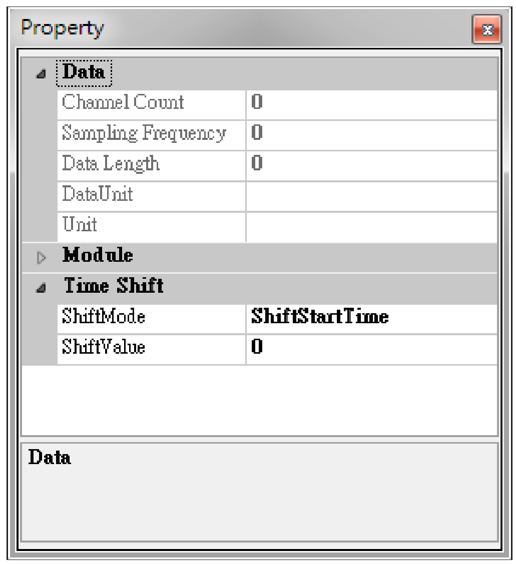
http://en.wikipedia.org/wiki/Monotone_cubic_interpolation

4.1.1.8 Time Shift

Shift the graph along the x-axis (time).

Properties

This module accepts input of Signal (which could be a real number or complex, single channel or multi-channel, Regular) and Audio (which could be a real number or complex, single channel, or multi-channel, Regular).



{Data} Property Name	Property Definition	Default Value
Channel Count	Displays the number of channels connected to the component	0
Sampling Frequency	Displays the sampling frequency of the component	0
Data Length	Displays the data length of the component	0
DataUnit	Displays the data unit of the component	None
Unit	Displays the unit of the component	None

{Time Shift} Property Name	Property Definition	Default Value
ShiftMode	Select the type of shift method to apply to the graph. There are *ShiftStartTime*, *SetStartTime*, and *SetStartDateTime* three	*ShiftStartTime*

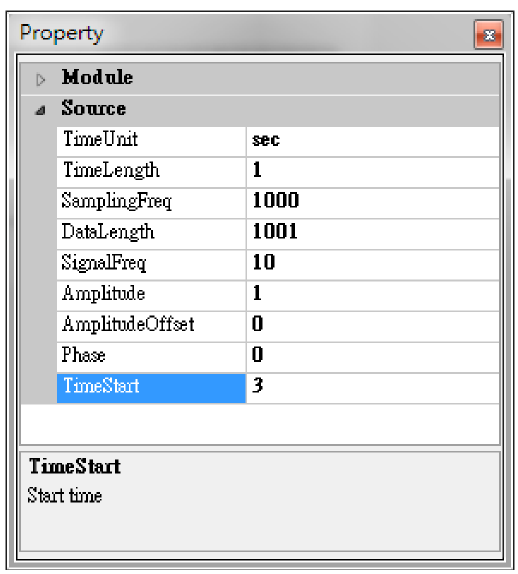
	options.	
ShiftValue	Set the shift value	0
StartValue	Set the start time value	0
StartDate	Set the start date	2000/1/1
StartTime	Set the start time	00:00:00

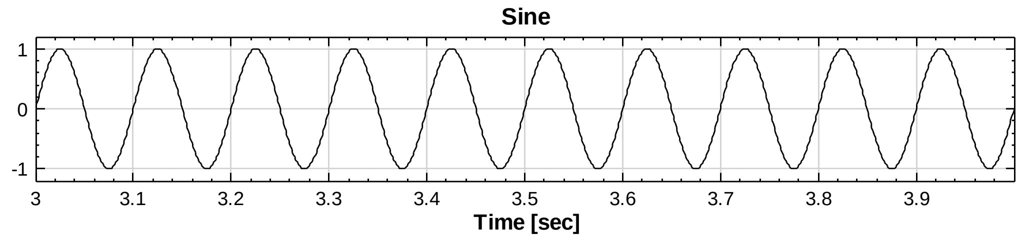
{ShiftMode} Variable Option	Property Definition	Default Value
ShiftStartTime	Shift Value. Shift the start time of the graph to the entered value (the time shift will either add to or minus from the original start time)	*ShiftValue* = 0
SetStartTime	Start Value. Set the start time of the graph to the entered value	*StartValue* = 0
SetStartDateTime	Start Date, Start Time. Set the start date and the start time of the graph to the entered value	*StartDate* = 2000/1/1 *StartTime* = 00:00:00

Example

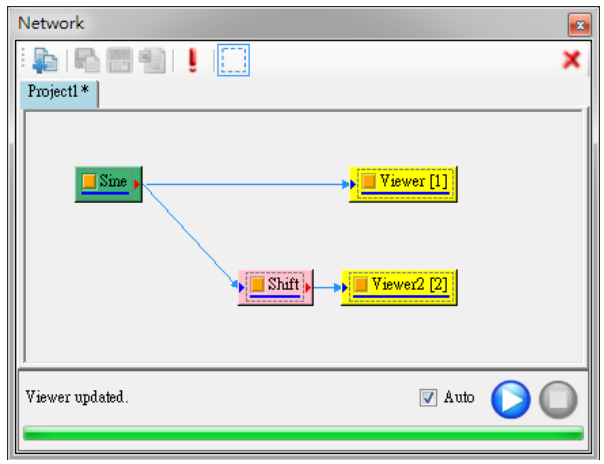
Create a **Sine Wave** component and shift its time value.

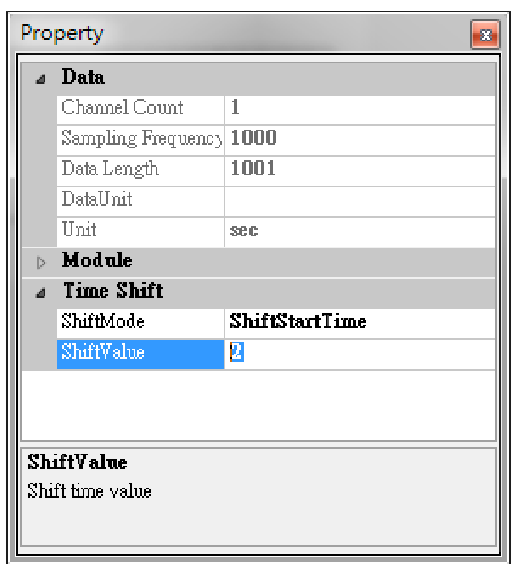
1. Create **Source→Sine Wave** and set the *TimeStart* to 3. You can see that the first point of the sine wave will begin at the 3 second mark.

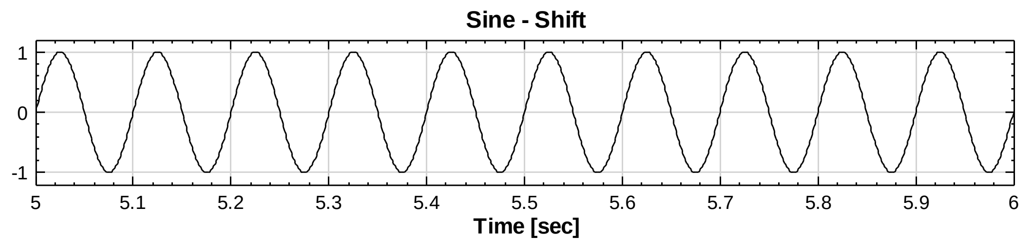




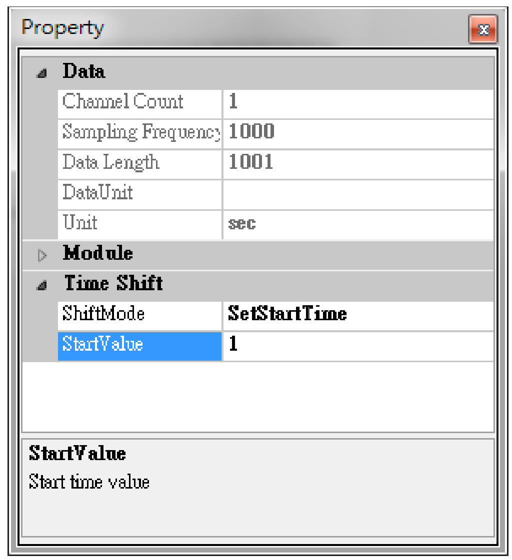
2. Connect the **`Sine Wave`** component to **`Compute`** → **`Channel`** → **`Time Shift`** and set the *`ShiftMode`* and select *`ShiftStartTime`* and set *`ShiftValue`* to 2. You will see that the start time on the graph has shifted to the 5th second.

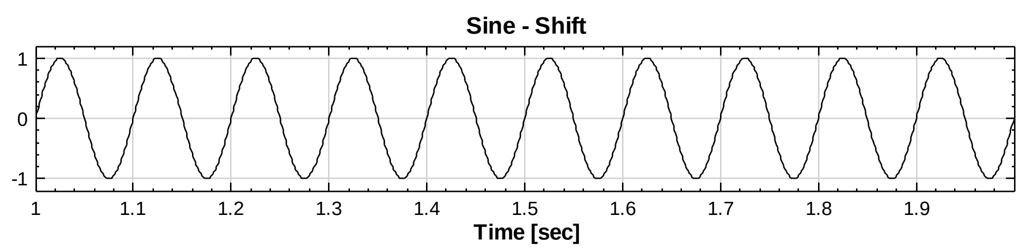






3. If you select *SetStartTime* and set the *StartValue* to 1, then the first point of the graph will start at 1 second mark.





Important

`Time Shift` allows the user to shift the graph along the x-axis and the **`RemoveDC`** function allows the user to shift the graph along the y-axis.

Related Functions

`RemoveDC`

4.1.2 Filter

This module provides several regular filters which are used to remove some components from input signals, based on different signal characteristics.

1. **`FIR Filter`**: Fundamental Finite Impulse Response Filter

2. **`Median Filter`**: Significantly reduce impulsive noises.

3. **`Moving Average Filter`**: Used to remove random noise

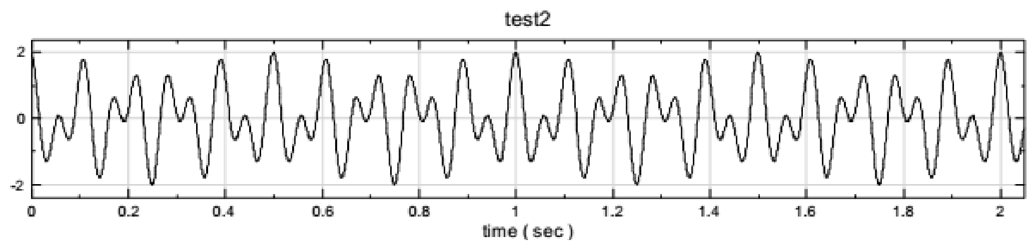
4. **`Notch Filter`**: A Notch filter is a stop-band filter with a narrow band.

4.1.2.1 FIR Filter

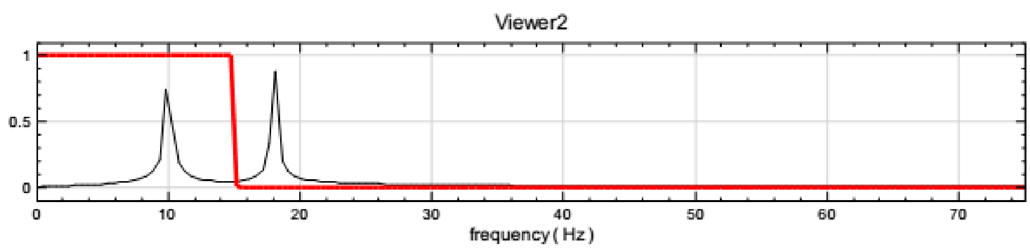
Finite Impulse Response Filter is the fundamental filter prototype in digital signal processing. It can remove a high-frequency, low-frequency or a given band frequency components. The term finite means that the filter impulse response is finite.

Introduction

Assume an input signal is given as shown below.

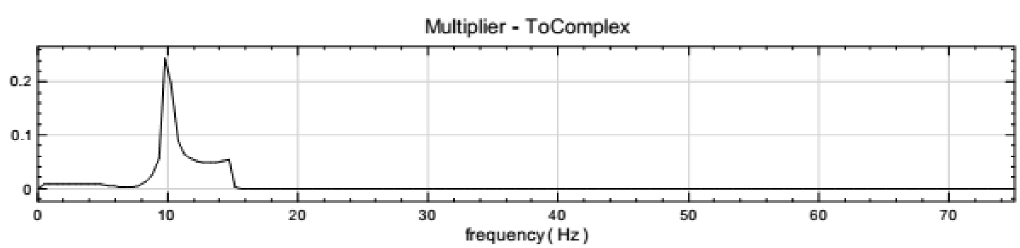


The Fourier Transform is shown below. It is desired to remove the high frequency components and preserve the low frequency components.

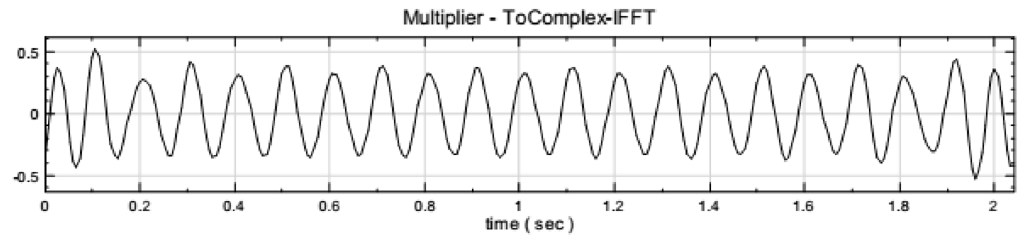


(The thin black curve represents the Fourier Transform of the original signal and the bold red curve represents the desired filter)

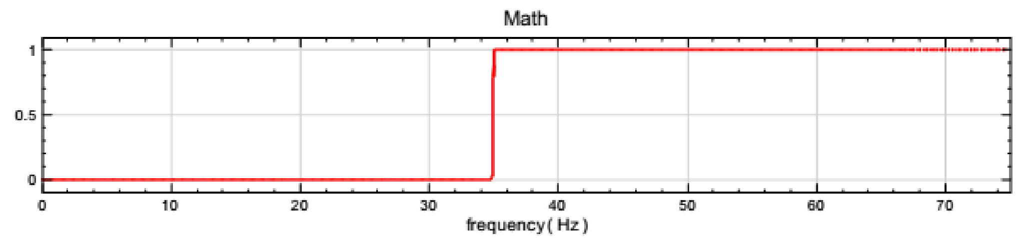
Therefore, define a function representing the filter above in Fourier Space and multiply it with the Fourier Transform of the original signal.



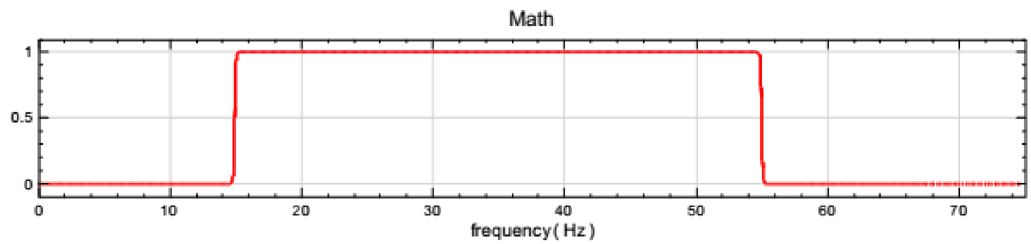
Next, conduct an Inverse Fourier Transform to remove the high frequency component. The result is shown below.



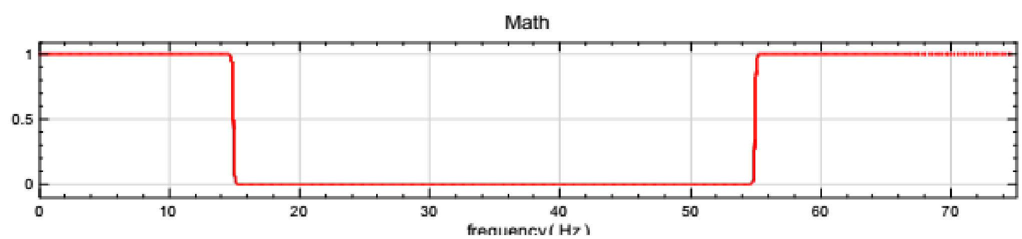
Besides the low-pass filter above, the high-pass filter is shown below.



BandPass: The BandPass filter is shown below.



BandStop: The BandStop filter is shown below.

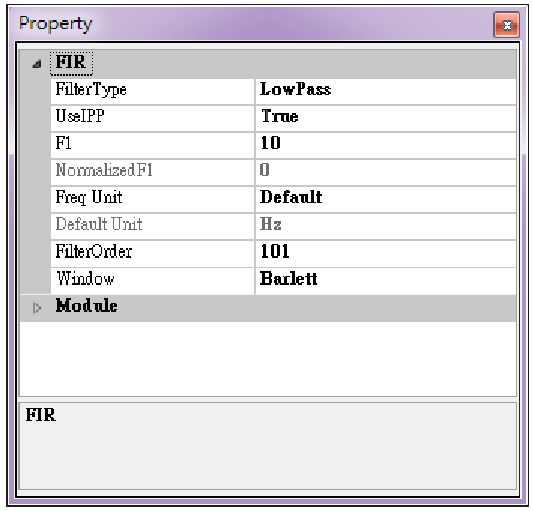


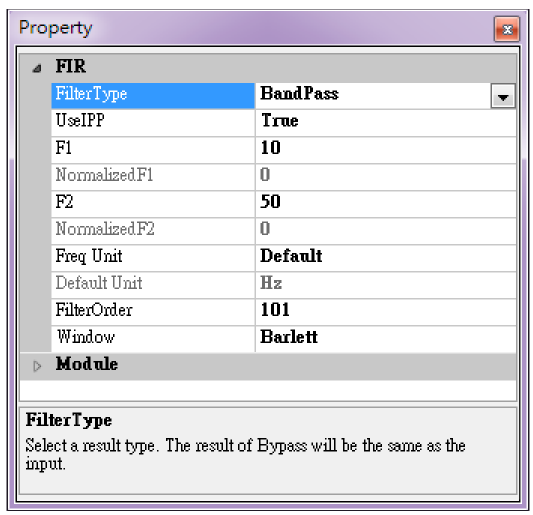
Bypass: All frequency components can pass through the filter.

Properties

This module accepts input of Signal (which could be a real number, single channel or multi-channel, Regular) and Audio (which could be a real number, single channel or 120multi-channel, Regular).

The main property of **`FIR Filter`** is *`FilterType`*, which has 5 options: *`LowPass`*, *`HighPass`*, *`BandPass`*, *`BandStop`* and *`ByPass`*. *`LowPass`* is used to remove frequency components which are higher than `F1`, while `HighPass` is used to remove components which are lower than frequency `F1`. `BandPass` is used to retain components which are between frequency `F1` and `F2` while `BandStop` is used to remove them. `ByPass` allows all components to pass through, i.e., the output signal is the input signal. Definition of properties and default values are shown below.





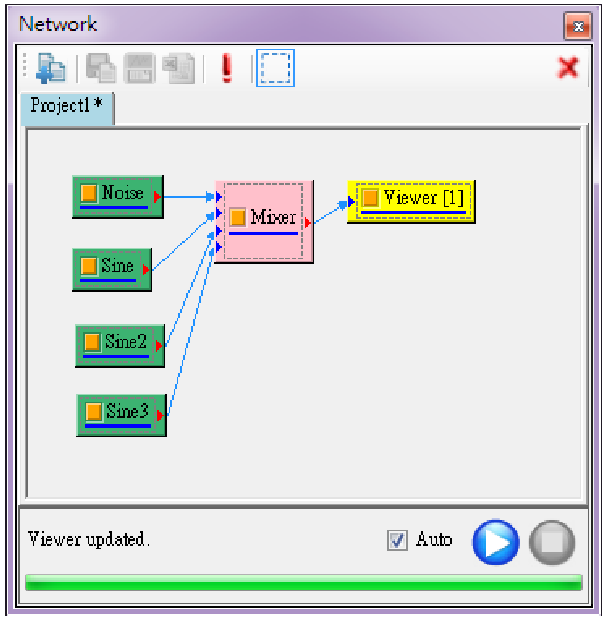
{FIR} Property Name	Property Definition	Default Value
`FilterType`	5 types are provided which are `LowPass`, `HighPass`, `BandPass`, `BandStop`, and `ByPass`	`LowPass`
`UseIPP`	Select True to perform IPP algorithms for FIR computation	True
`F1`	For `LowPass` and `HighPass`, `F1` represents the cutoff frequency.	10

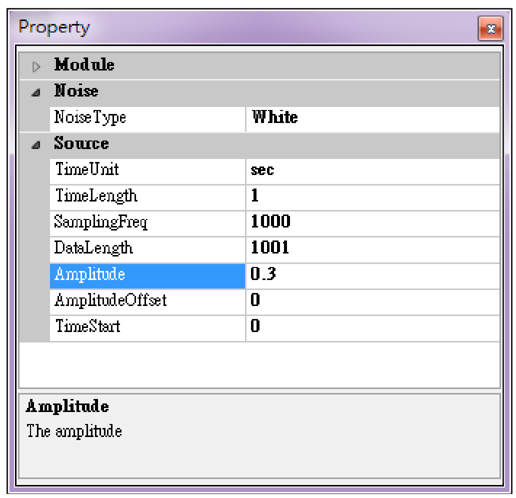
	For `BandPass` and `BandStop`, `F1` represents the frequency starting point. Unit is Hz	
`NormalizedF1`	Demonstrate the normalized `F1` based on the Sampling frequency of the input signal	Varies based on the input signal
`F2`	The frequency ending point, `F2`, for `BandPass` and `BandStop` filters. Unit is Hz	50
`NormalizedF2`	Demonstrate the normalized `F2` based on the Sampling frequency of the input signal	Varies based on the input signal
`Freq Unit`	Specify the frequency unit associated with cutoff frequency	default
`DefaultUnit`	Display the frequency unit associated with trend frequency	Hz
`FilterOrder`	The number of points in the discrete impulse response function of the filter. N means N-order Filter	101
`Window`	Use window function to reduce the leakage effect on the transform. The window functions include 5 types: Barlett, Blackman, Hanning, Hamming, and Rectangle, whose definitions are given below.	Barlett

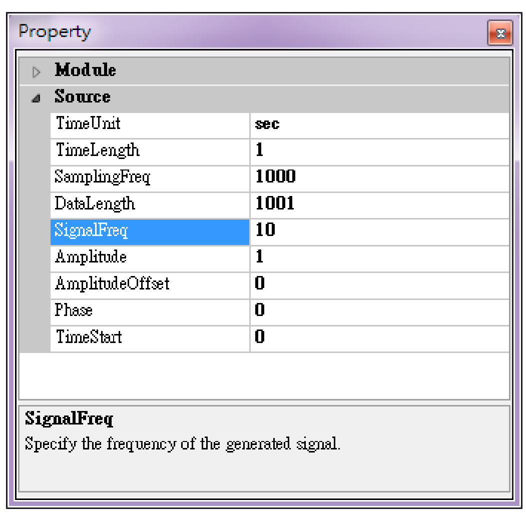
Example

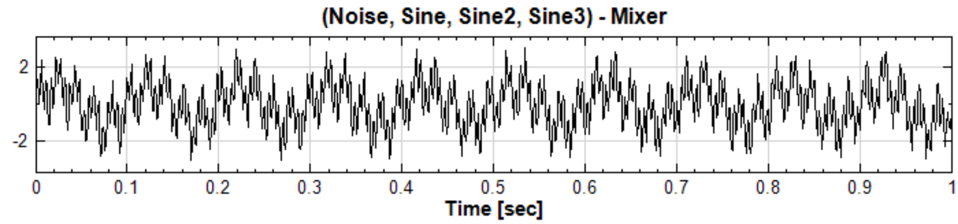
This example shows the process of using **`FIR Filter`** to remove different frequency components based on an input signal which contains 10, 51, 193 Hz sine waves plus white noise.

1. In the **Network Window**, select **`Source→Noise`** to create a white noise signal and set the `Amplitude` as 0.3. Then use the **`Source→ Sine Wave`** to generate 3 sine waves and change their `SignalFreq` to 10, 51, 193 Hz. After that, use the **`Compute→Mathematics→Mixer`** to mix the above signals and plot them using the **`Viewer→Channel Viewer`** (by dragging the Output of every signal to the Input of Mixer).

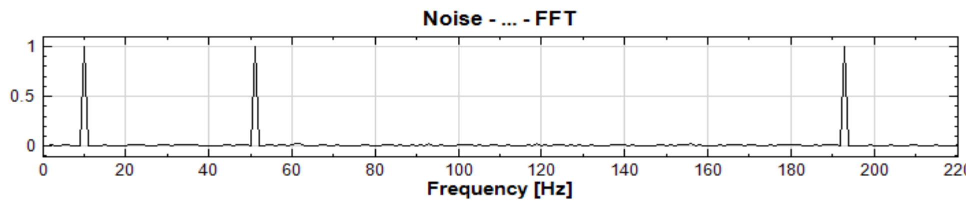

Network
Project1 *
Noise
Sine
Sine2
Sine3
Mixer
Viewer [1]
Viewer updated.
Auto


Property
Module
Noise
NoiseType White
Source
TimeUnit sec
TimeLength 1
SamplingFreq 1000
DataLength 1001
Amplitude 0.3
AmplitudeOffset 0
TimeStart 0
Amplitude
The amplitude

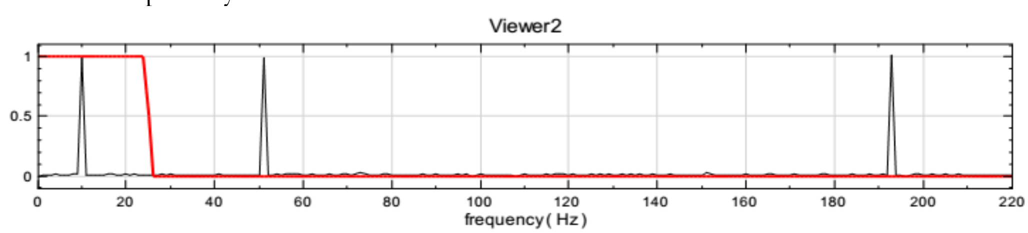

Property
Module
Source
TimeUnit sec
TimeLength 1
SamplingFreq 1000
DataLength 1001
SignalFreq 10
Amplitude 1
AmplitudeOffset 0
Phase 0
TimeStart 0
SignalFreq
Specify the frequency of the generated signal.

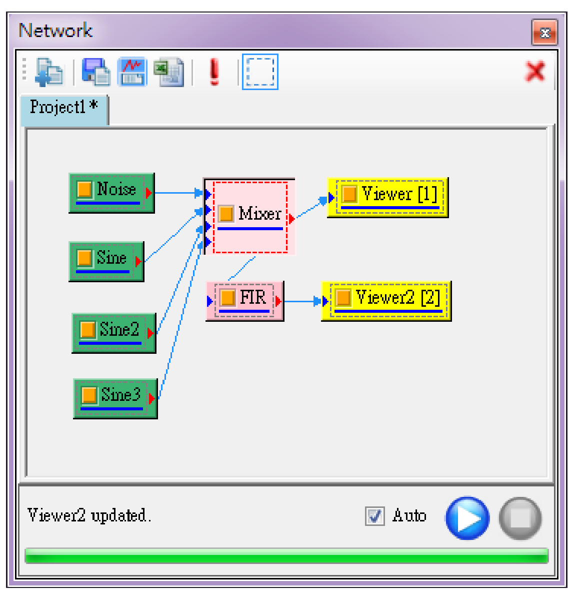


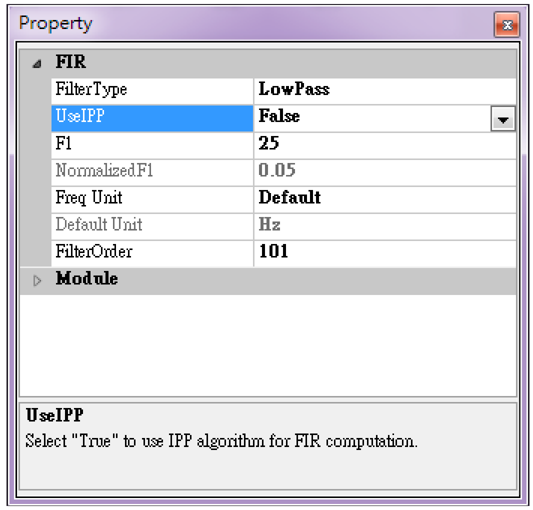
(To facilitate the following FIR Filter design, **Mixer** could be connected to FFT for frequency spectrum observation)

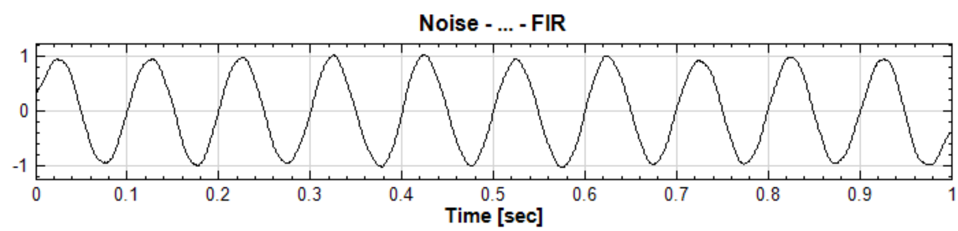


2. On the **`Mixer`** icon, select **`Compute→Filter→FIR`** Filter and change the *`F1`* to 25Hz and *`UseIPP`* to False. The default *`FilterType`* is *`LowPass`*. Then, use **`Channel Viewer`** to show the processing result. It can be seen that the frequency components higher than 25Hz are all removed and the output signal is similar to sine of 10Hz. However, because the *`FilterOrder`* is only 101, the wave is partially affected.

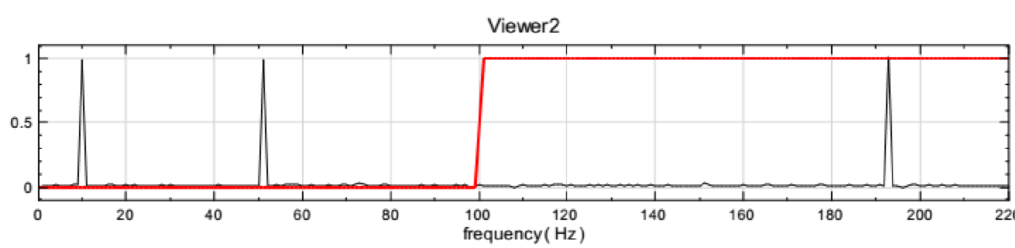


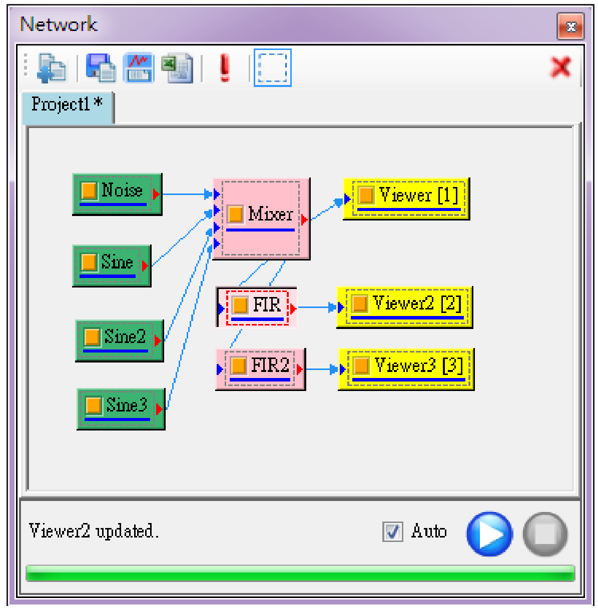


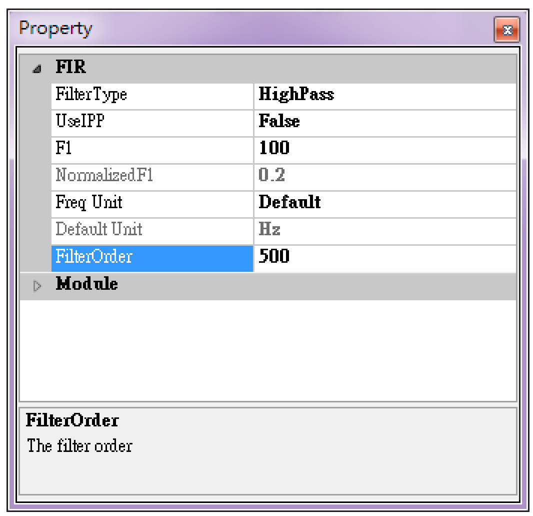


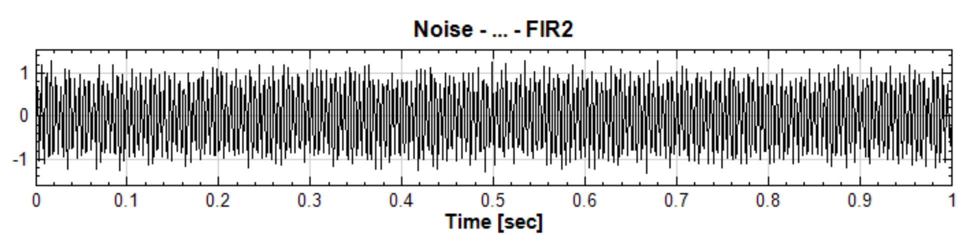


3. Change properties of `FilterType` to `HighPass`, `F1` to 100Hz, `UseIPP` to False, and `FilterOrder` to 500. As shown below, the filter removes the frequency component lower than 100Hz and it leaves a sine wave of 193Hz with the White Noise.

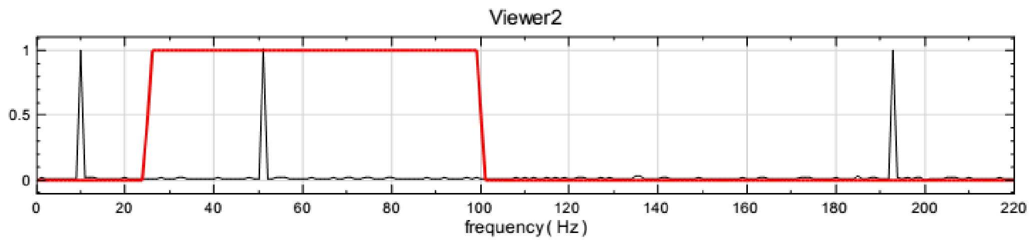


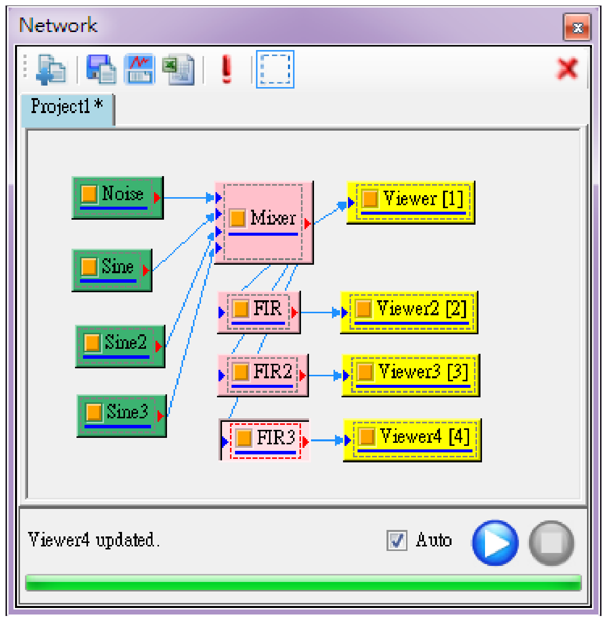


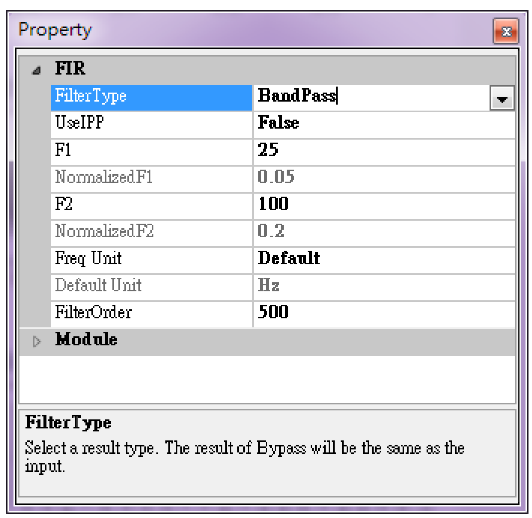


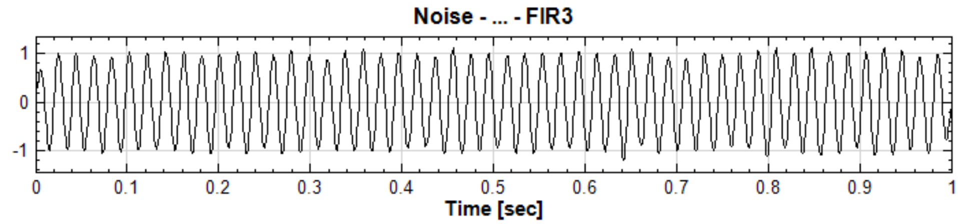


4. Repeat Step 2 and add one **`FIR Filter`**. Change the *`FilterType`* to *`BandPass`*, *`UseIPP`* to False, *`F1`* to 25Hz, *`F2`* to 100Hz, and *`FilterOrder`* to 500. The frequency components between 25~100Hz would pass through while other components would be cut off. Therefore, the output is a Sine wave of 51Hz.


Viewer2
1
0.5
0
0
20
40
60
80
100
120
140
160
180
200
220
frequency(Hz)


Network
Project1 *
Noise
Sine
Sine2
Sine3
Mixer
FIR
FIR2
FIR3
Viewer [1]
Viewer2 [2]
Viewer3 [3]
Viewer4 [4]
Viewer4 updated.
Auto


Property
FIR
FilterType
BandPass
UseIPP
False
F1
25
NormalizedF1
0.05
F2
100
NormalizedF2
0.2
Freq Unit
Default
Default Unit
Hz
FilterOrder
500
Module
FilterType
Select a result type. The result of Bypass will be the same as the input.



Related Functions

`Source, Mixer, Channel Viewer`

References:

http://en.wikipedia.org/wiki/Finite_impulse_response
http://cnx.org/content/m11918/latest

4.1.2.2 Median Filter

Median Filter is a one-dimensional non-linear filter, used to calculate the median in the range of filtering (Filter Order). Because it can reduce speckle noise significantly while retain good edge detection, it is usually used in digital image processing.

Introduction

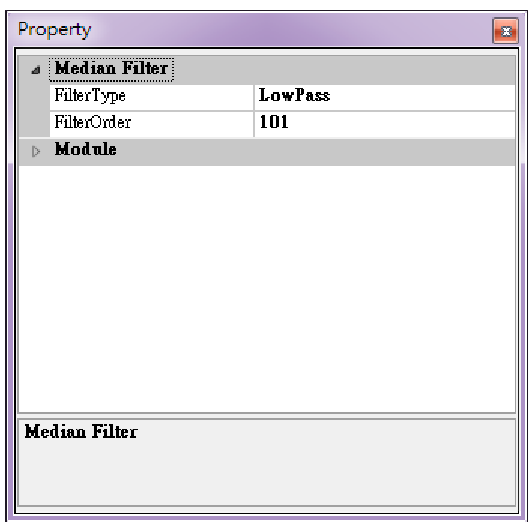
Let $X = \{x_0, x_1, \cdots, x_{N-1}\}$ be a N-length input signal, $Y = \{y_0, y_1, \cdots, y_{N-1}\}$ be the output signal, and M be the signal length for calculation. The **`Median Filter`** is defined as

$$y_i = \text{mdeian}(x_k), \qquad i - \frac{M-1}{2} \leq k \leq i + \frac{M+1}{2}$$

Centered on the i^{th} data, take $\frac{M-1}{2}$ points on both sides to construct a set of array. Then, find the median in the array to replace the i^{th} data. In the case when the number of data is insufficient, e.g., $M > N$ or $N - i < \frac{M-1}{2}$, repeat the edge data to fill the whole array. M is supposed to be an odd number. In the case of an even number, it would be made to be odd by adding 1 automatically.

Properties

This module accepts input of Signal (which could be a real number, single channel or multi-channel, Regular) and Audio (which could be a real number, single channel or multi-channel, Regular). The formats of input signal and output signal are identical.

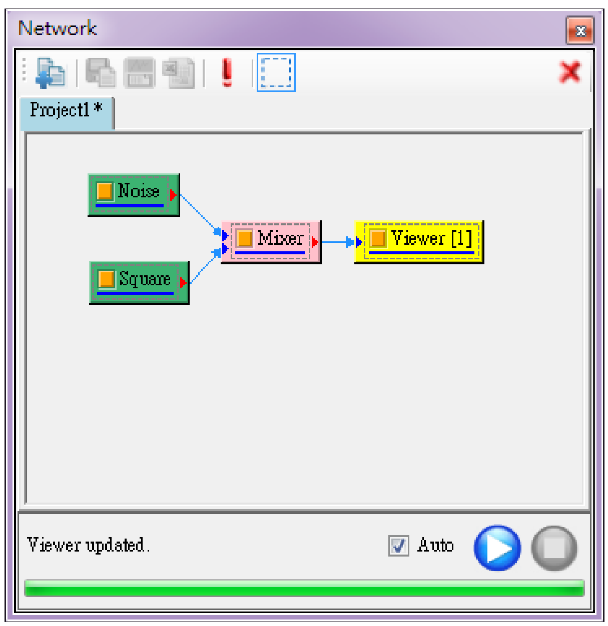


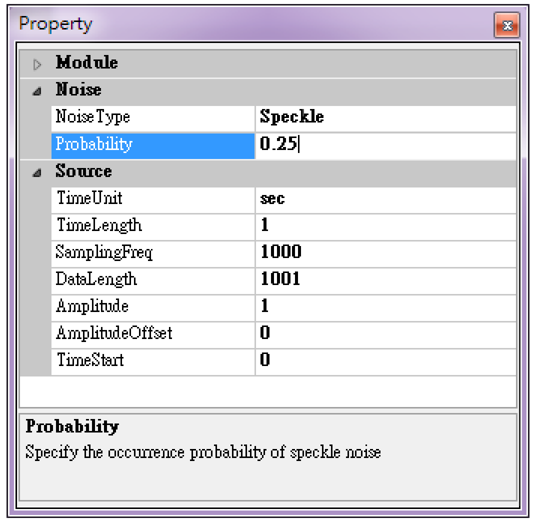
{Median Filter} Property Name	Property Definition	Default Value
`FilterType`	To set the filter to remove high-frequency or low-frequency components. The available options are `LowPass`, `HighPass`, and `ByPass`	`LowPass`
`FilterOrder`	The data length of the median filter, i.e., M, is supposed to be an odd number. In the case of an even number, it would be changed to an odd number by adding 1 automatically	101

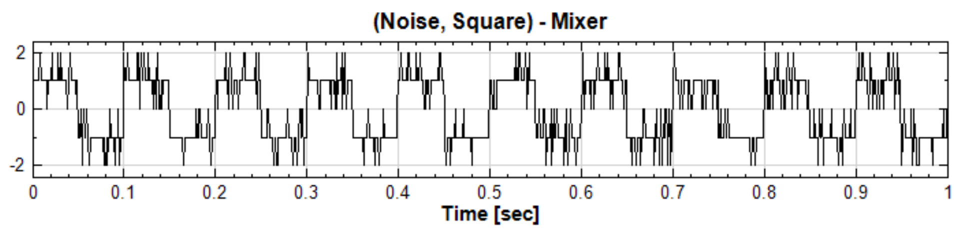
Example

The example shows the procedure of using a median filter to process a signal of a square wave and remove speckle noise.

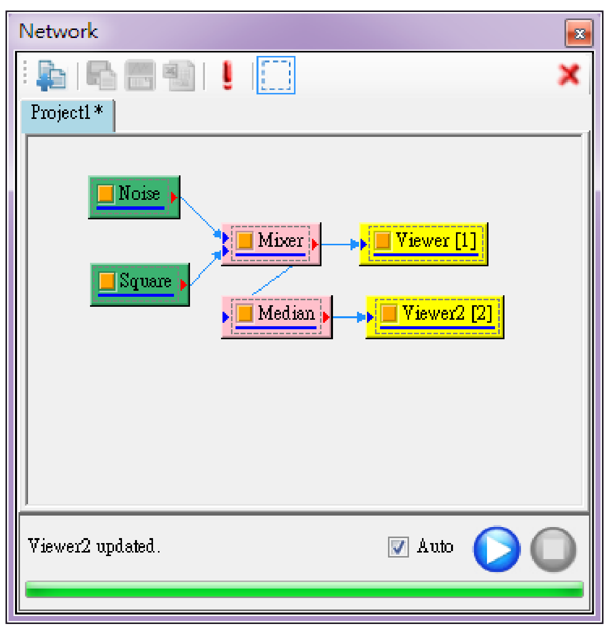
1. Right click in the **Network Window**, select **Source→Noise** to generate a noise signal. Set the `NoiseType` as `Speckle`, `Probability` as 0.25. In addition, select **Source→Square Wave** to generate a square wave. Use **Compute→Mathematics→Mixer** to mix these two signals and then use **Viewer→Channel Viewer** to show the result in the window.

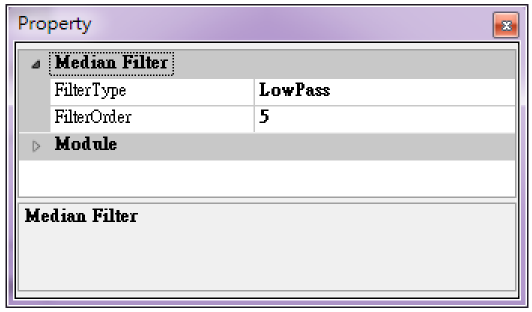


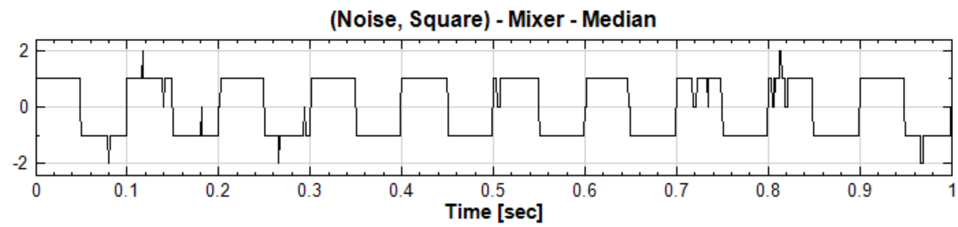




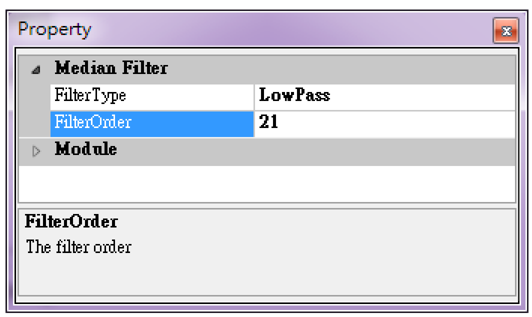
2. Click on the **`Mixer`** component to select **`Compute→Filter→Median Filter`**, then change the *`FilterOrder`* to 5. Use **`Channel Viewer`** to show the result.

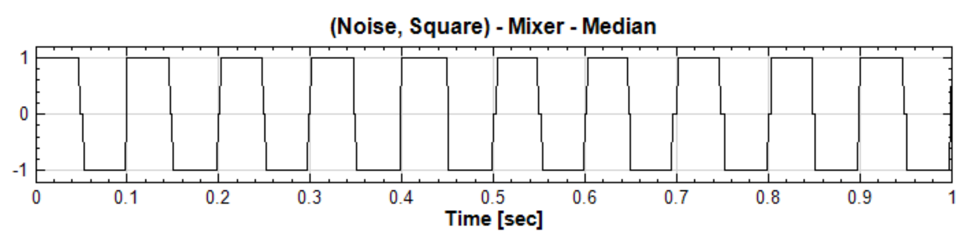




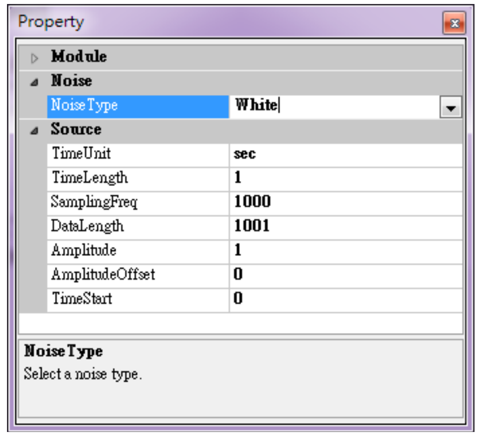


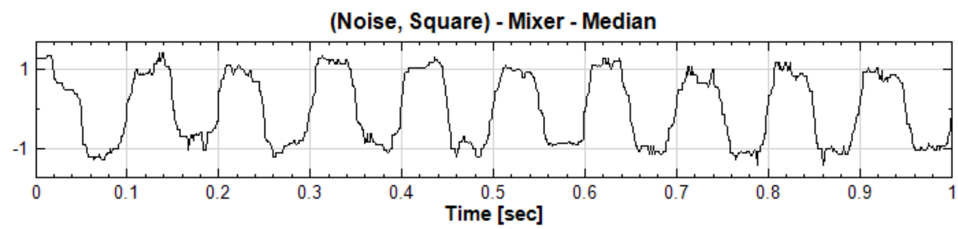
3. In step 2, a good result is achieved. As shown below, `FilterOrder` can be increased 4 times if the `Median Filter` is adjusted to 21, i.e., `FilterOrder` = 21. Not only will the speckle noise be removed completely, but the edge sharpness will still be preserved.





4. Finally, to test characteristics of `Median Filter`, come back to the **`Noise`** component and change `Speckle noise` to `White noise` in `NoiseType`. As shown below, it can be seen that the `Median Filter` cannot eliminate the effect caused by white noise completely. However, the wave edges are mostly retained. This is the main characteristic of Median filter.





Related Functions

Source, Mixer, Moving Average Filter, Channel Viewer

Reference

http://en.wikipedia.org/wiki/Median_filter

4.1.2.3 Moving Average Filter

By calculating the average of signals in the range of filtering (average length), Moving Average Filter decreases the noise in discrete time signals and increases the recognizable ability of peak. The advantages of moving average filter are: simple theory and fast calculation. However, compared with other types of filters, it has a low filtering ability to separate one band of frequencies from another. In spectrum analysis, its performance is poor.

Introduction

Let $X = \{x_0, x_1, \cdots, x_{N-1}\}$ be an N-length input signal, $Y = \{y_0, y_1, \cdots, y_{N-1}\}$ be the output. If the average length of the signal is M elements, for every signal in X, the output is

$$y_i = \frac{1}{M} \sum_{j=\frac{-(M-1)}{2}}^{\frac{M-1}{2}} x_{i+j}$$

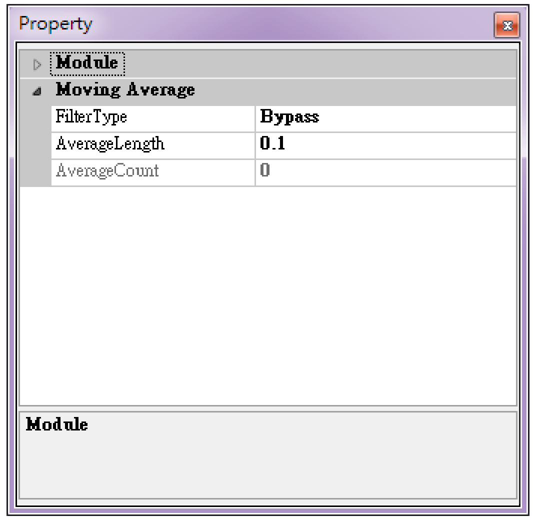
The formula above means the convolution of the input signal and a square filter which has area of 1 and length M in time axis.

Notice that this filter is similar to the Rolling Statistics on the average calculation. The difference is how to handle with the edge. On the edge, in the case when average length M_b is less than M, this filter still calculates average using M_b. Therefore, the output data length is identical to the input data length. On the other hand, the Rolling Statistics only calculates average in the range of given data and therefore, the length of output data length would be less than the length of input data.

Properties

This module accepts input of Signal (which could be a real number, single channel or multi-channel, Regular) and Audio (which could be a real number, single channel or multi-channel, Regular). The input signal format and the output signal format are identical.

`Moving Average` Filter has three properties, which are *`FilterType`*, *`AverageLength`* and *`AverageCount`*. *`AverageLength`* represents the number of data points, M, for average calculation. The unit is time. *`FilterType`* sets the type of filters which include *`LowPass`*, *`HighPass`* and *`ByPass`*. *`LowPass`* is the calculation result using the theory introduced above. *`HighPass`* is achieved by subtracting the *`LowPass`* result from the input signal. Because the original signal is equal to *`HighPass`* + *`LowPass`*, the output of *`ByPass`* is equal to the input signal. The default values of properties are shown in the table below.

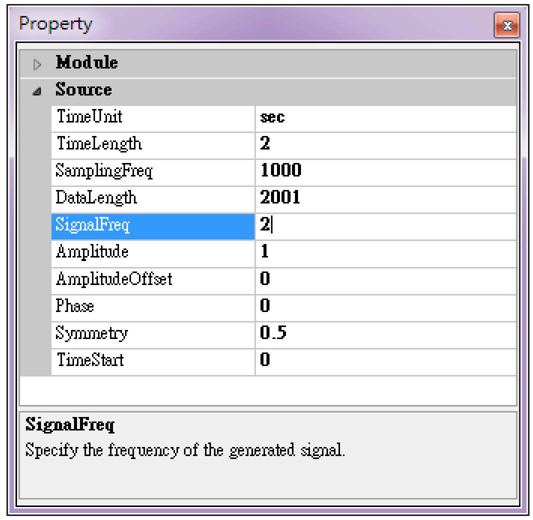


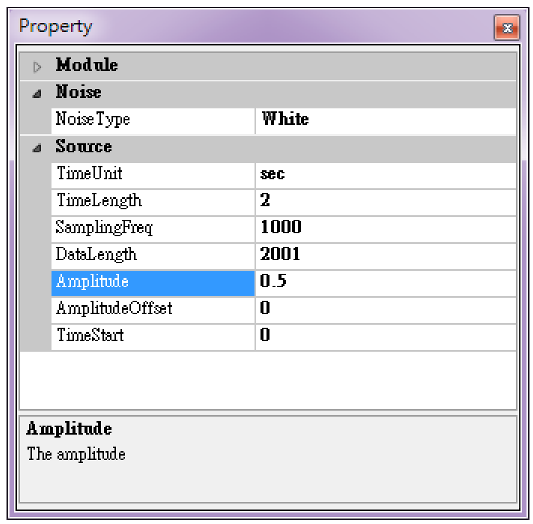
{Moving Average} Property Name	Property Definition	Default Value
`FilterType`	To set the filter to remove high-frequency or low-frequency components. The available options are `LowPass`, `HighPass`, and `ByPass`	`LowPass`
`Average Length`	The signal length for calculation of average. The unit is time	0.05 of total length
`AverageCount`	To show the number of signals corresponding to `AverageLength`	Automatically adjusts based on `AverageLength`

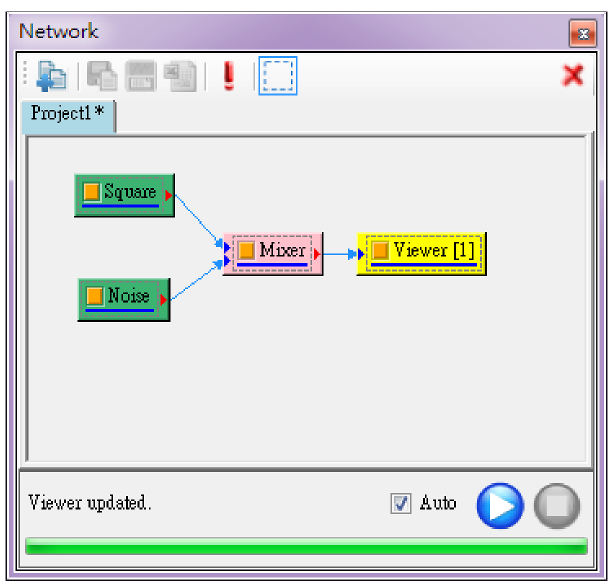
Example

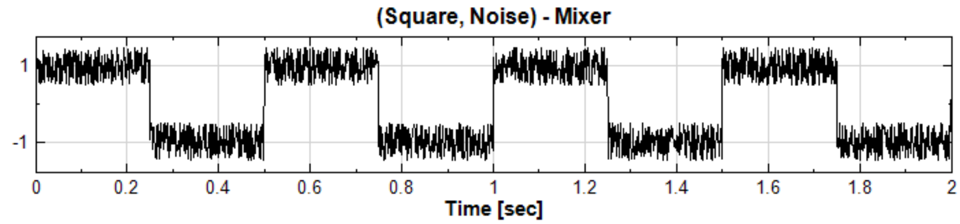
In this example, a square wave (*frequency* = 2Hz, *Amplitude* = 1, *TimeLength* = 2) is mixed with a **White Noise** (*Amplitude* = 0.5, *TimeLength* = 2) and then processed by the **Moving Average Filter**. Different *AverageLengths* are set to observe corresponding effects.

1. Right click and select **Source→Square Wave** to create a square wave. Change its *TimeLength* to 2 and *SignalFreq* to 2. Right click again and select **Source→Noise→White Noise** to generate white noise. Change the **White Noise** component's properties of *TimeLength* to 2 and *Amplitude* to 0.5. Finally, right click and use **Compute→Mathematics→ Mixer** to mix these two signals and use Viewer to plot the result.

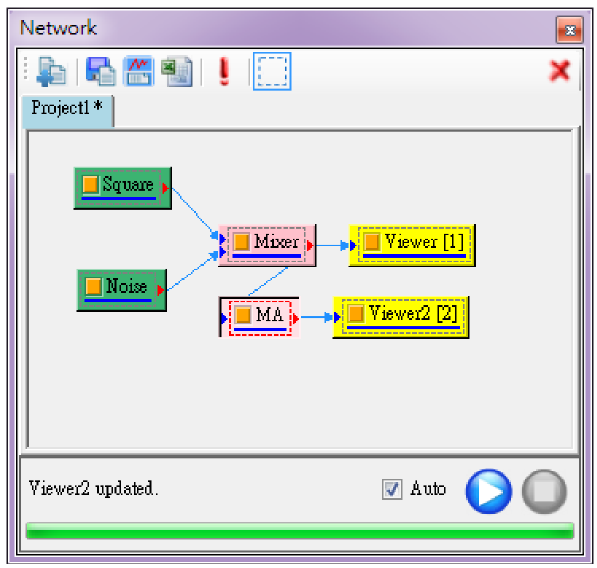

Property
Module
Source
TimeUnit sec
TimeLength 2
SamplingFreq 1000
DataLength 2001
SignalFreq 2
Amplitude 1
AmplitudeOffset 0
Phase 0
Symmetry 0.5
TimeStart 0
SignalFreq
Specify the frequency of the generated signal.

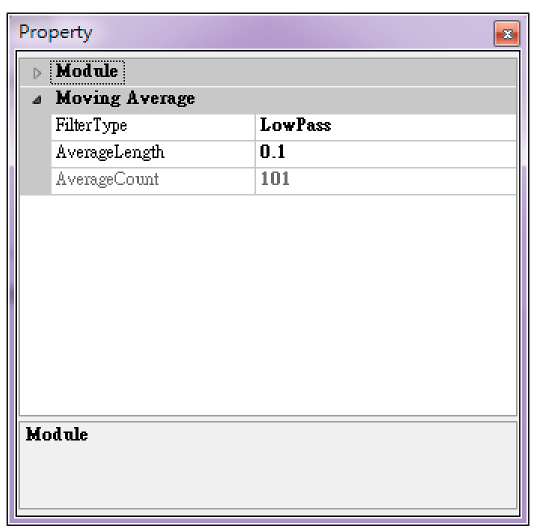

Property
Module
Noise
NoiseType White
Source
TimeUnit sec
TimeLength 2
SamplingFreq 1000
DataLength 2001
Amplitude 0.5
AmplitudeOffset 0
TimeStart 0
Amplitude
The amplitude

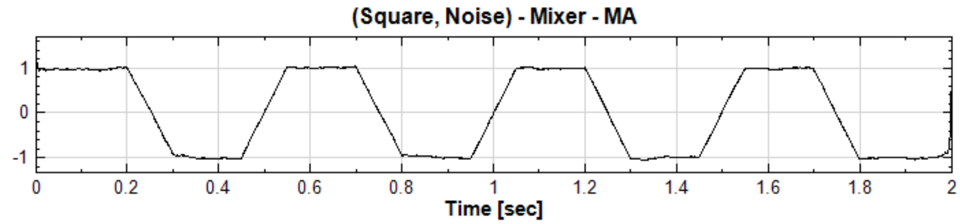

Network
Project1 *
Square
Noise
Mixer
Viewer [1]
Viewer updated.
Auto



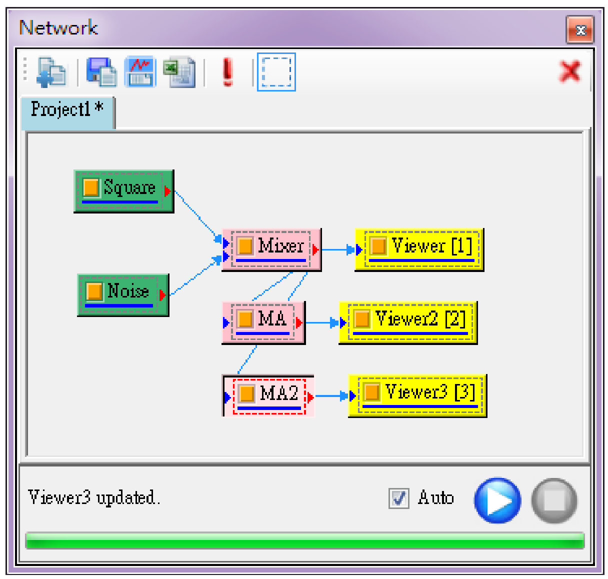
2. To conduct moving average on the input signal, right click on the **Mixer** component and select **Compute→Filter→Moving Average**. In the field of *AverageLength*, it can be seen that the default value is 0.1s. Similarly, it can be seen that the value of *AverageCount* is 101 which means that every output point of MA is the average of 101 points centering at the corresponding point in the input signal. Finally, use **Channel Viewer** to plot the result.

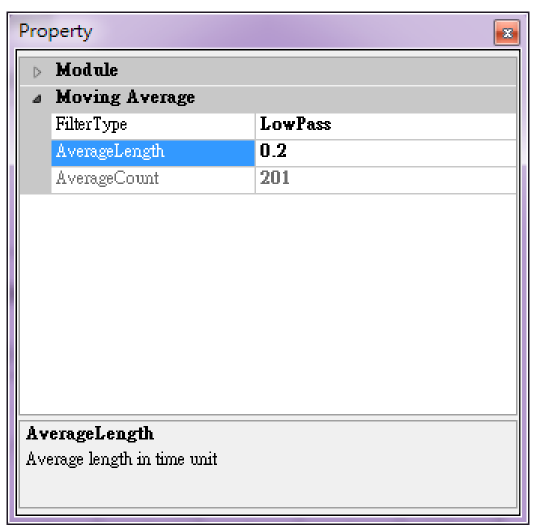


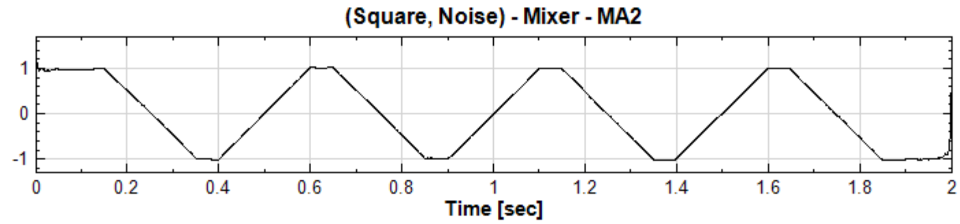




3. Following step 2, change `AverageLength` to 0.2 to perform a moving average again to generate a new figure, named MA2. Then use **`Channel Viewer`** to plot the result.

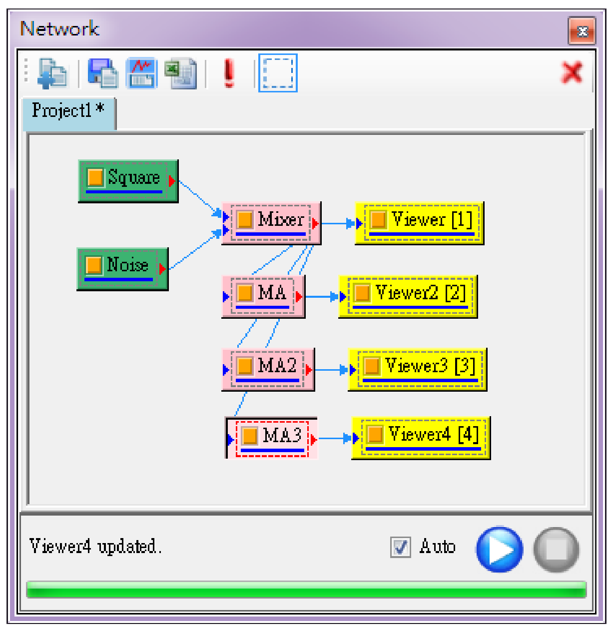


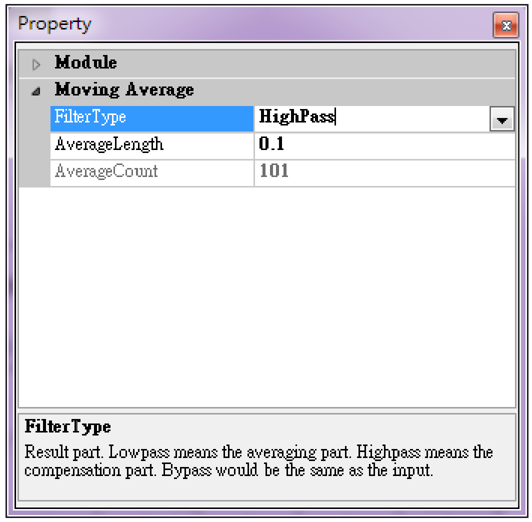


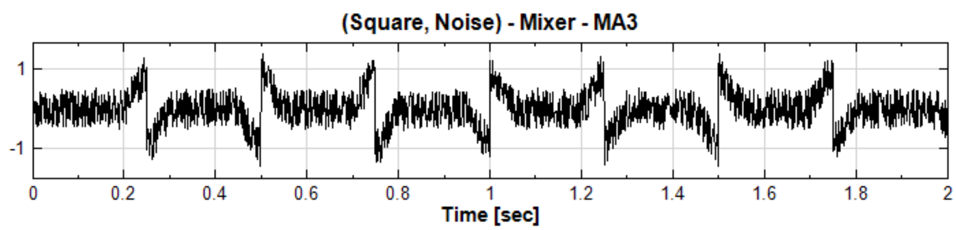


4. Comparing the results obtained in step 1, 2 and 3, it can be seen that increasing `Average Length` can reduce the noise in the input signal significantly. However, the drawback of this filter is that the sharp edge of the original square wave becomes more and more flat as the `Average Length` increases.

5. Next, after changing the `FilterType` to `HighPass`, perform a moving average again. It can be seen that the result is the input signal subtracted by the output of MA2.







Related Instructions

`Source, Mixer, Channel Viewer`

Reference

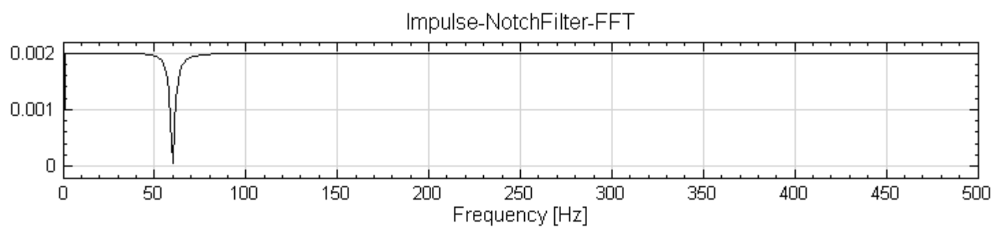
http://www.dspguide.com/ch15.htm

4.1.2.4 Notch Filter

A Notch filter is a band-stop filter or band-rejection filter. It can filter a specific frequency.

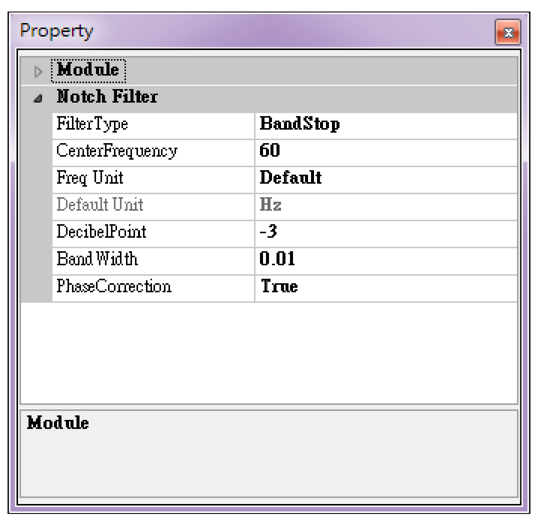
Introduction

The purpose of Notch filter is filtering a specific frequency. Suppose there is a 60Hz frequency to be filtered, its frequency response function (FRF) is shown below



Properties

This module accepts input of Signal (which could be a real number, single channel or multi-channel, Regular) and Audio (which could be a real number, single channel or multi-channel, Regular).

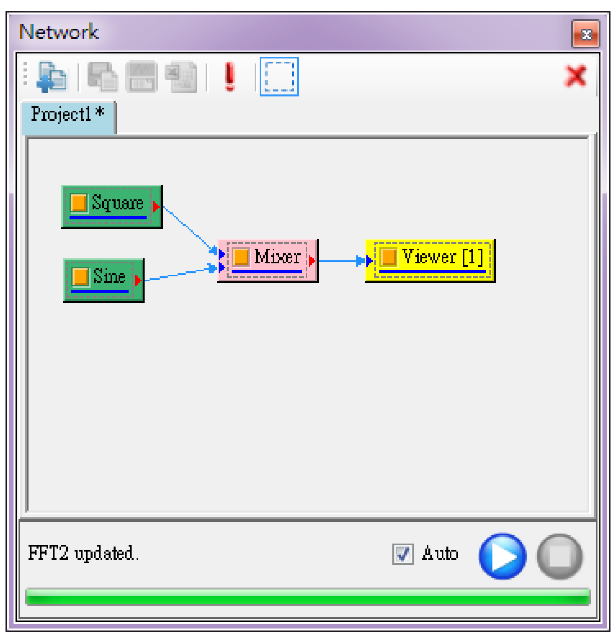


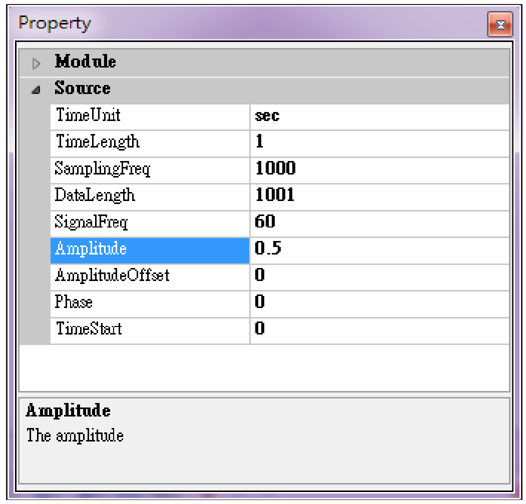
{Notch Filter} Property Name	Property Definition	Default Value
`FilterType`	To set the filter to remove specific frequency or pass specific frequency. The available options are `BandStop`, `BandPass`, and `ByPass`	`BandStop`

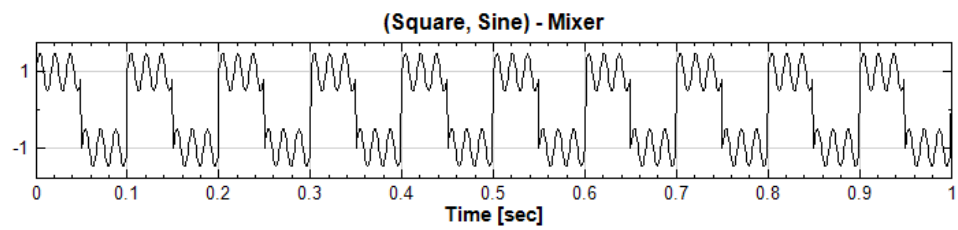
CenterFrequency	Set the central frequency be filtered	60
Freq Unit	Specify the frequency unit associated with cutoff frequency	default
Default Unit	Display the frequency unit associated with trend frequency	Hz
DecibelPoint	Set the magnitude response bandwidth at a level. The less value, the rejection-band more sharp	-3dB
Band Width	Set the band width. The definition is bandwidth of attenuation point. The unit of Band Width is(π * radian/sampling frequency)	0.01
PhaseCorrection	Set True to correction phase	True

Example

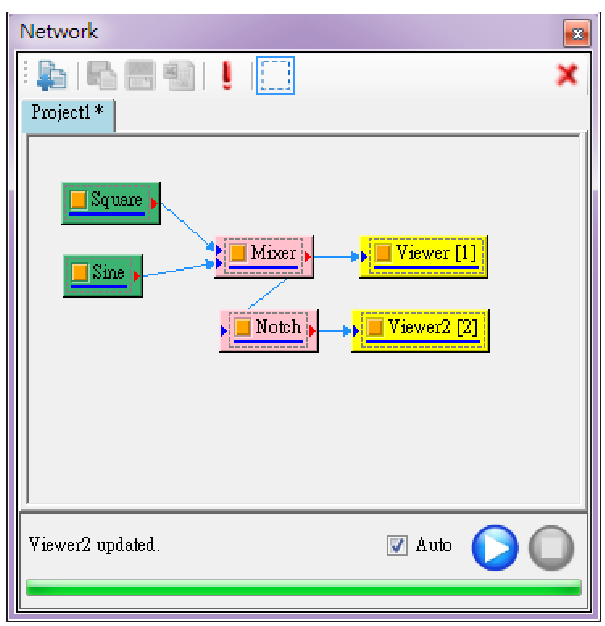
1. Right-click and select **Source→Square Wave** and **Sine Wave**. Change *SignalFreq* of Sine to 60Hz and *Amplitude* to 0.5. Connect Square and Sine to **Compute→Mathematics→Mixer** and use **Channel Viewer** to observe the graph.

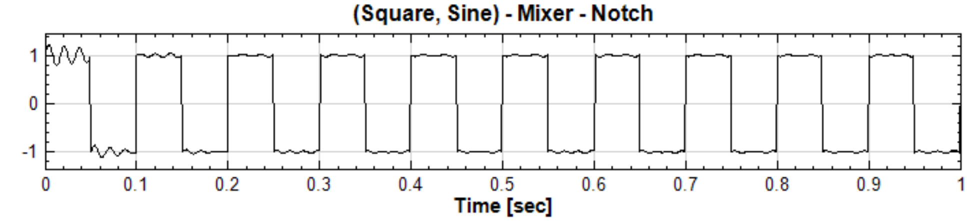




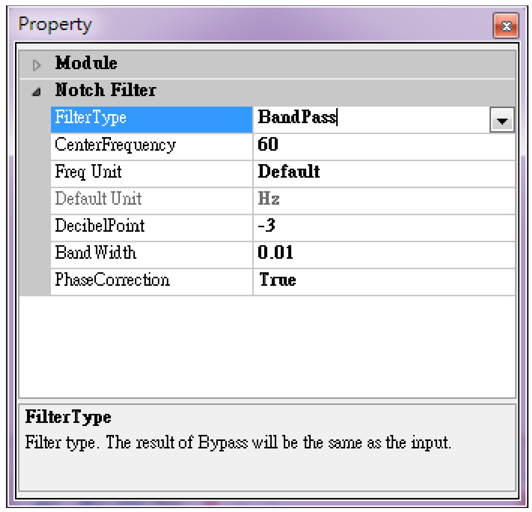


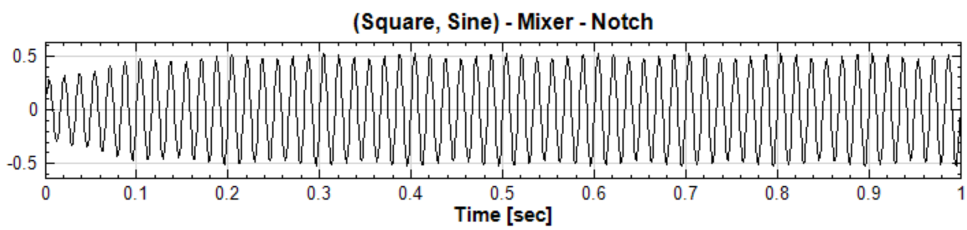
2. Use a **`Notch Filter`** to filter sine wave with 60Hz. Connect **`Compute→ Filter→Notch Filter`** to **`Mixer`** component. All settings of **`Notch Filter`** are default, and connect to a **`Channel Viewer`**. The sine wave with 60Hz will then be filtered.





3. Change *FilterType* to *BandPass* from *BandStop*. Now, the square wave with 10Hz is filtered.





Related Functions

Mixer, Channel Viewer, Source

Reference

http://en.wikipedia.org/wiki/Band-stop_filter

4.1.3 Mathematics

This group of modules process signals or relationships between signals mathematically, whose components are listed below.

1. **`Differential`**: Select a single-channel from a multi-channel source and do a numeric differential operation of the input signal.

2. **`Integrate`**: To calculate approximate integration of input signal.

3. **`Math`**: To input mathematical formula for signal calculation.

4. **`Mixer`**: To add (or subtract) several signals using an identical time scale.

5. **`Multiplier`**: To multiply several signals using an identical time scale.

6. **`Normalize`**: Normalizes the data for different normalization values.

7. **`Remove DC`**: To remove the direct current component of signal.

8. **RMS**: Compute a root-mean-square.

4.1.3.1 Differential

This component performs a subtraction or a differential operation on two signals.

Introduction

Let $X = \{x_0, x_1, \cdots, x_{N-1}\}$ be a length-N signal, the various difference/differential is defined as below.

Forward difference: $\Delta x_i = x_{i+1} - x_i$

Divided by the sampling period h, the approximate differential value is given by

$$\frac{\Delta x_i}{h} = \frac{x_{i+1} - x_i}{h} \cong \frac{dx_i}{dt}$$

Central difference: $\Delta x_i = \frac{x_{i+1} - x_{i-1}}{2}$

Dividing the central difference by sampling period h, the approximation of differential can be obtained as follows.

$$\frac{\Delta x_i}{h} = \frac{x_{i+1} - x_{i-1}}{2}\frac{1}{h} \cong \frac{dx_i}{dt}$$

Properties

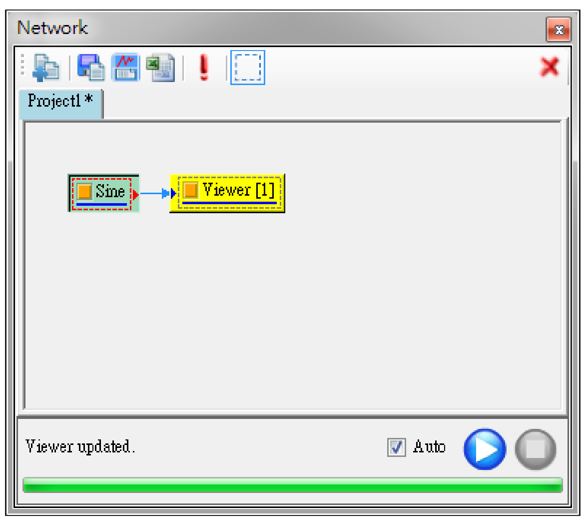
This module accepts input of Signal (which could be a real number or complex number, single channel or multi-channel, Regular) and Audio (which could be a real number or complex number, single channel or multi-channel, Regular). Settings of related properties are given below.

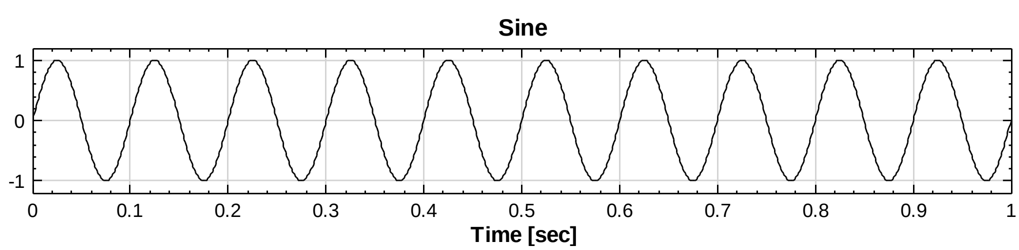
{Diff} Property Name	Property Definition	Default Value
`Zero Padding`	Specify whether to pad the first or last point with zero, or not at all. The three options are `None`, `First`, and `Last`.	`None`
`Method`	The differentiation methods include `Simple` and `Symmetrized`. `Simple` is 2 points forward difference while `Symmetrized` is central difference	`Simple`
`Differentiate`	Divide the result by the sampling period to obtain the approximation of differentiation	False

Example

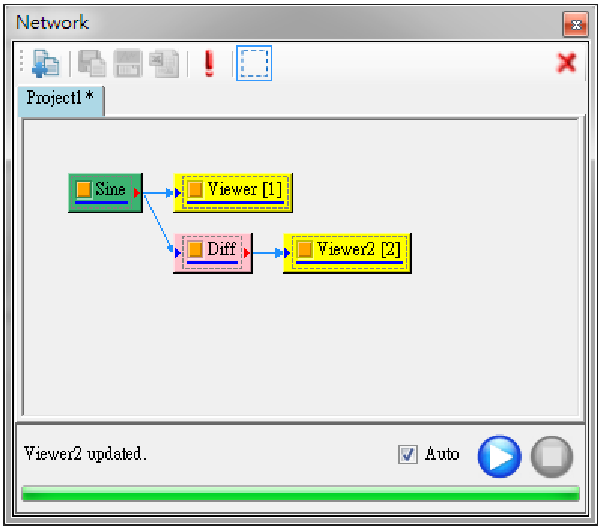
This example shows the differentiation of a sine wave.

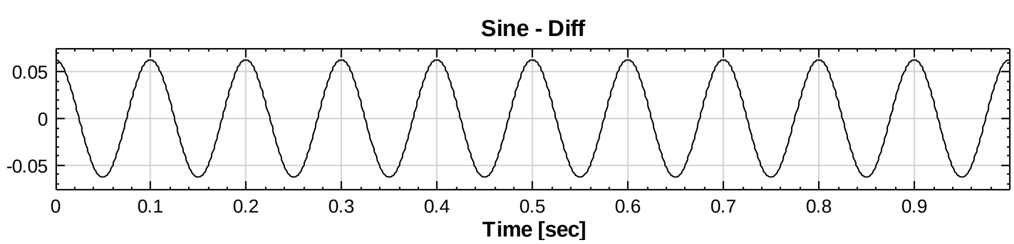
1. Right click in **Network Window** to select **`Source→Sine Wave`** to generate a sine wave, and then use **`Viewer→Channel Viewer`** to show it in the window.



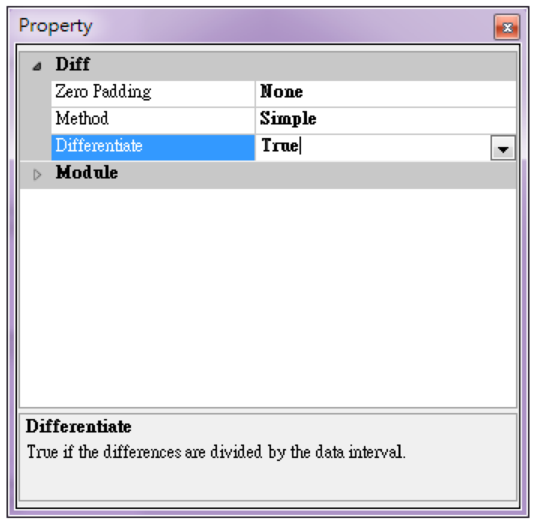


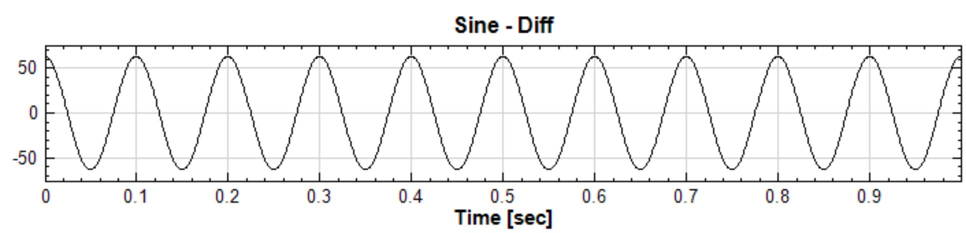
2. Right click on the **Sine** component, select **Computer→Mathematics→ Diff**, and then use **View→Channel Viewer** to plot the calculation result, as shown below. It can be seen that the sine wave is changed to cosine after **Diff** calculation. However, because the default value of *Differentiate* is False, the amplitude is very small.





3. The approximation of differentiation can be obtained by changing the *Differentiate* in **Diff** to True. Here, the **Source→Sine Wave** is $\sin(2\pi f t)$, where f is the signal frequency, t is the signal time, and the differentiation of this sine wave should be $2\pi f \cdot \cos(2\pi f t)$. In this example, f is 10 Hz, 2π is 6.28, therefore, the maximum amplitude should be $2\pi f$ = 62.83. The result can be verified by comparing with the result shown below.





Related Functions

Integrate, Channel Viewer

4.1.3.2 Integrate

This component performs integration on input signals.

Introduction

Let $X = \{x_0, x_1, \cdots, x_{N-1}\}$ be an N-length signal, $T = \{t_0, t_1, \cdots, t_{N-1}\}$ be the corresponding X-axis (time axis) coordinates, the numerical integration using n-Simple can be denoted as the formula below.

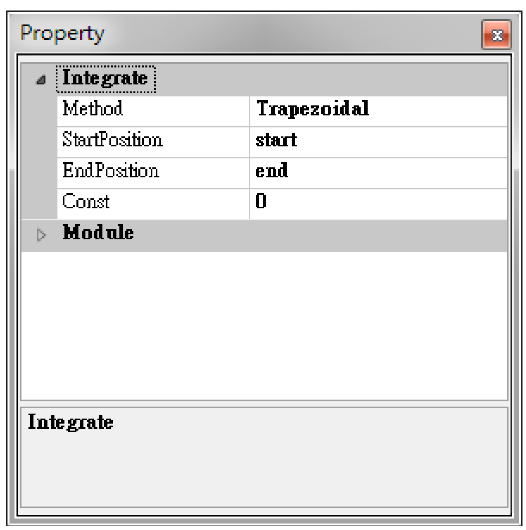
$$y_i = \int_0^{t_i} x_i dt \cong \sum_{k=0}^{i} x_k (t_{k+1} - t_k)$$

If Trapezoidal is used, the formula is denoted as below

$$y_i = \int_0^{t_i} x_i dt \cong \sum_{k=0}^{i} \frac{1}{2} (x_{k+1} + x_k)(t_{k+1} - t_k)$$

Properties

This module accepts input of Signal (which could be a real number or complex number, single channel or multi-channel, Regular) and Audio (which could be a real number or complex number, single channel or multi-channel, Regular). The related properties are introduced as in the table below.

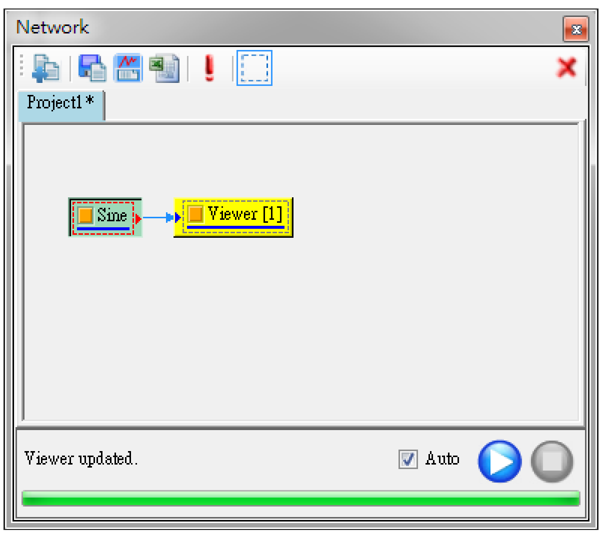


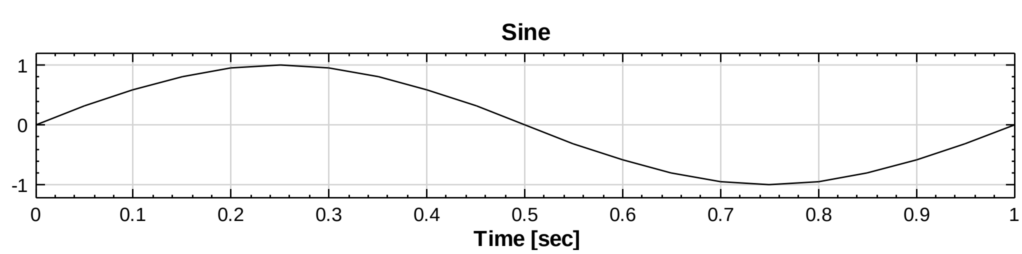
Property Name	Property Definition	Default Value
Method	The methods of numerical integration, including *Simple* and *Trapezoidal*	*Trapezoidal*
StartPosition	The start position in X-axis for integration	The starting point of the input signal
EndPosition	The end position in X-Axis for integration	The ending point of the input signal
Const	The shift along Y-axis after integration	0

Example

This example uses the integration function on a sine wave.

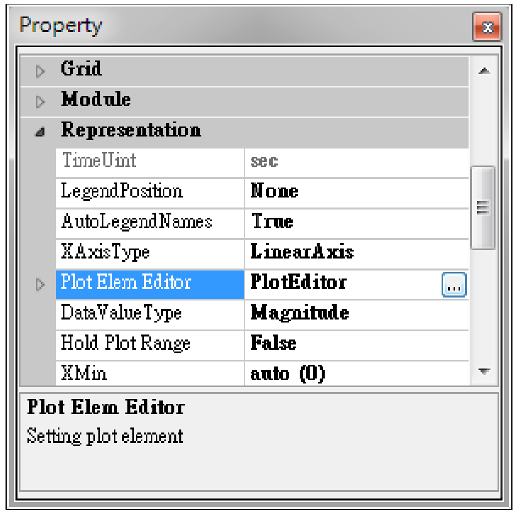
1. Right-click in the **Network Window**. Select **Source→Sine Wave** to create a sine wave, then change the *SignalFreq* to 1Hz, *SampleFreq* to 20Hz, *TimeLength* to 1 second (must be set in the final step). Then use **Viewer→ Channel Viewer** to show it in the window, as shown below.

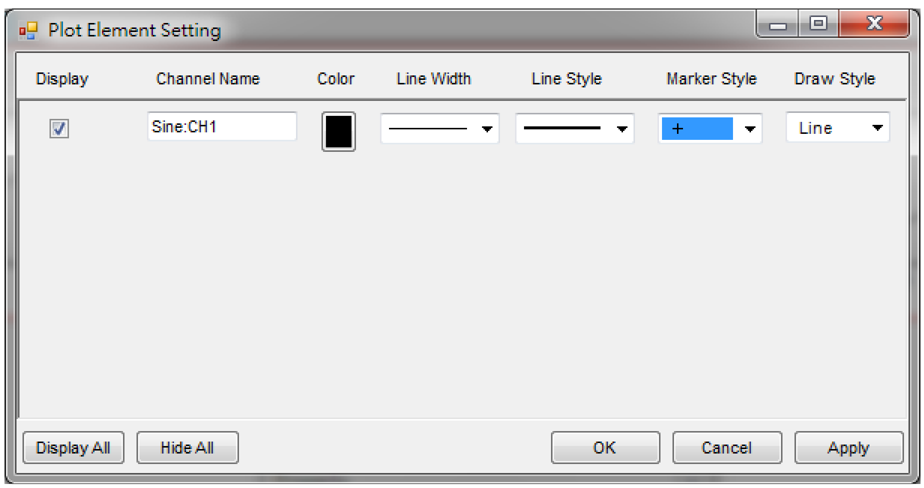


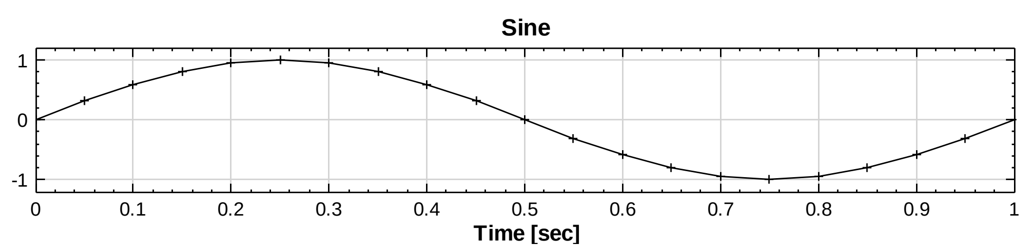


2. To show every point clearly, click *PlotEditor* in the *Plot Elem Editor* which is found in the Viewer[1] component. In the popped-up **Plot Element**

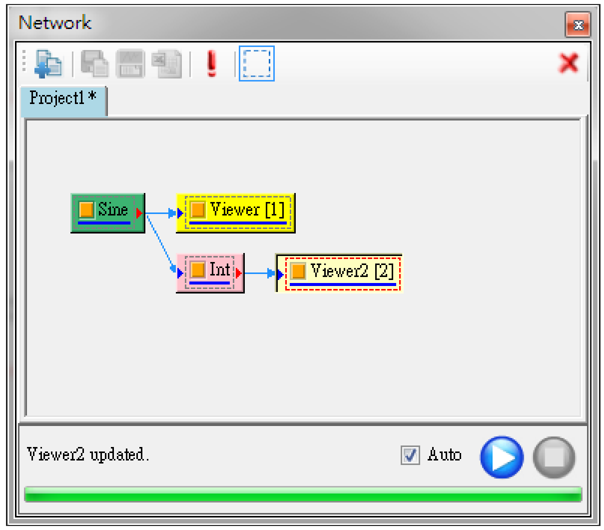
Setting Window, select 「+」 in **Marker Style** to mark every time point as a symbol of 「+」 in the curve.

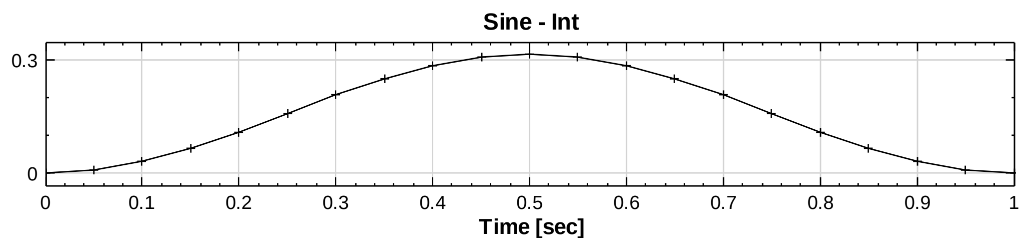




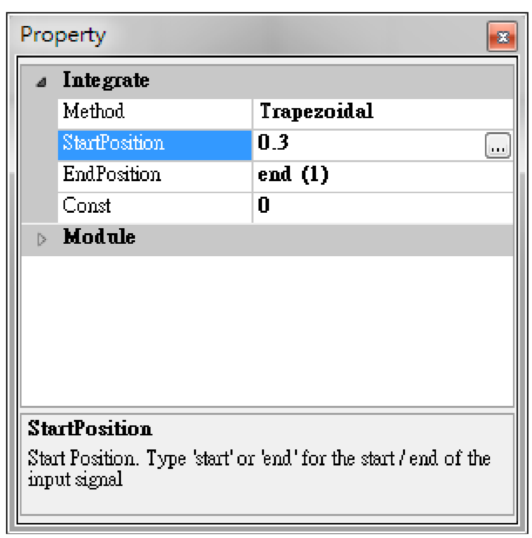


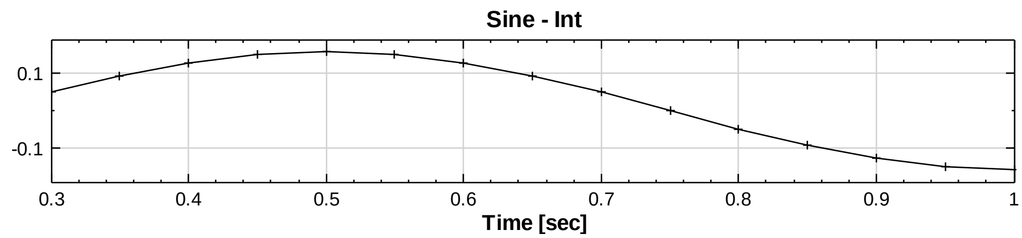
3. Perform numerical integration using **Compute → Mathematics → Integrate** on the sine wave, and change **Marker Style** to 「x」 as what has been done in step 1 and 2. The figure plotted is the integration result.

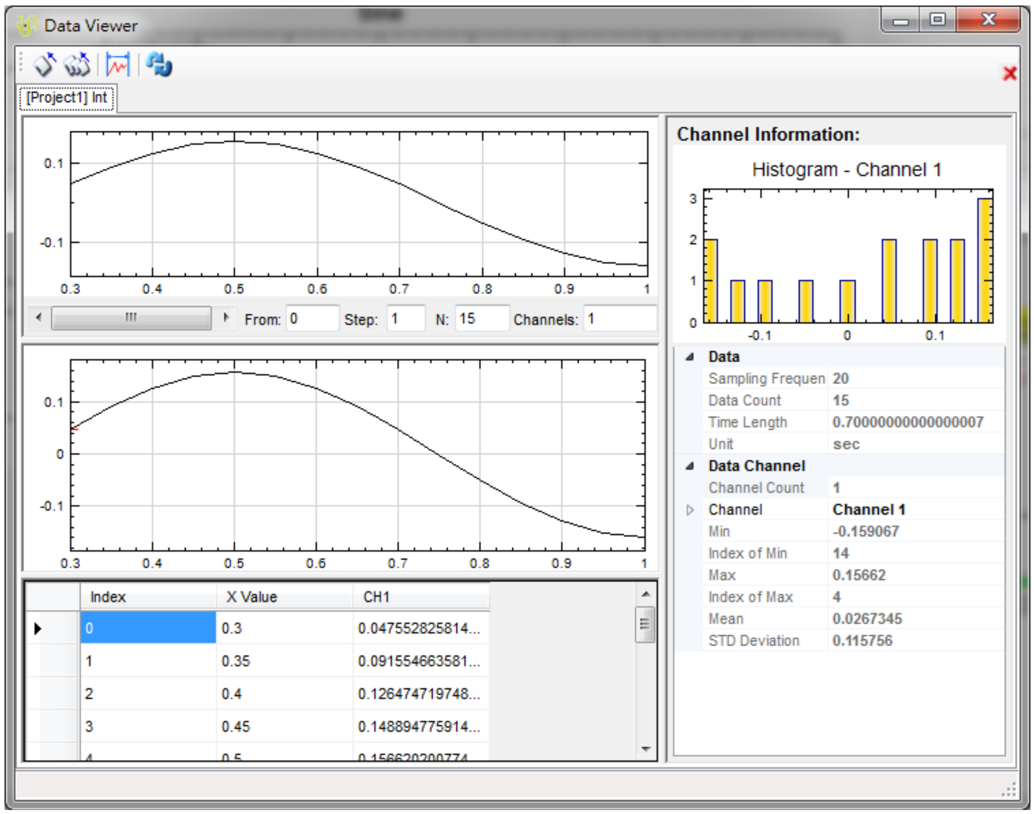




4. Change *StartPosition* of the **Int** component to 0.3, the new calculation result is shown below. Next, use **Data Viewer** to observe the signal output from **Int** component. It can be seen that the original value of 21 in *DataCount* has been changed to 15.
   Note: Changing the *StartPosition* and *EndPosition* will affect the output signal length.







Related Functions

Differential, Source, Channel Viewer

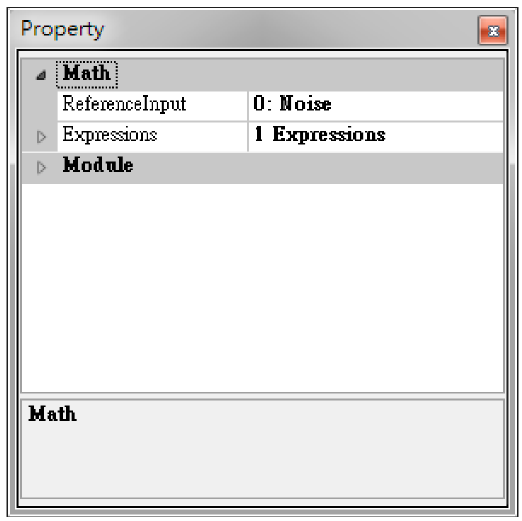
4.1.3.3 Math

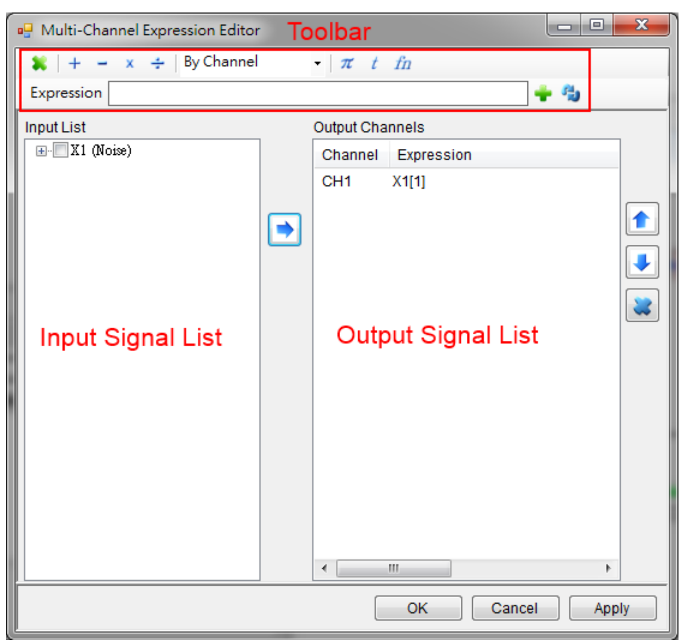
Perform point to point math calculations for an input signal.

Interface Introduction

This module accepts input of Signal which could be a real number or complex number, single channel or multi-channel, regular and audio.

To activate this module, click the "…" to the right of the field `Expressions`, or double-click the **`Math`** component. The **MultiChannel Expression Editor** will then pop up as shown below.

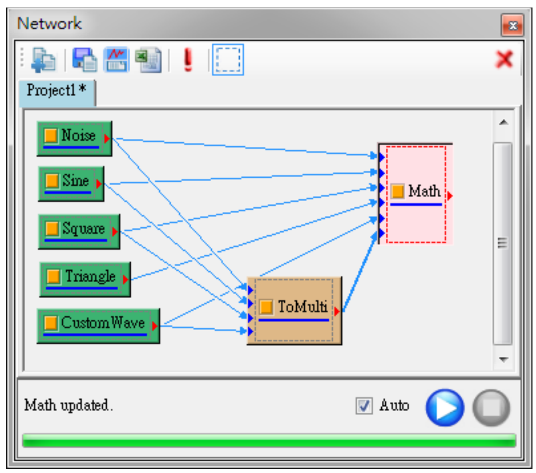


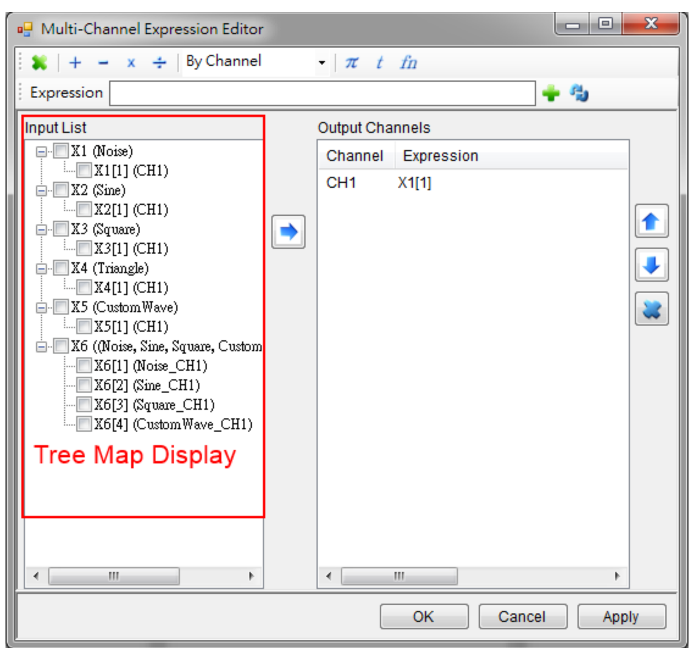


The pop-up window has three panels: Input Signal List, Toolbar, and output Signal List. The operation procedure is as following: select the signal from the Input Signal List, define math equation in the Expression field of Toolbox, the calculated result becomes one of the Output Channel in the Output Signal List. Below explains each of the functions:

Input list

Input List displays the signals connected to the input end of the **Math** component. By default, the input signal is multi-channel and each channel is displayed as a tree map shown in the graph. The 1st input signal is denoted as X1 in the Expression of Toolbox, the 2nd signal is denoted as X2, etc. The number in the square bracket represents the channel sequence of the input signal. For example, X1[1] represents the 1st channel of the 1st input signal. In addition, X1[1] can be abbreviated as X.





a. Double click a signal in the tree map. It will then be added to the Expression field.
b. If there is no calculation applied to the select input (output is the same as the input), then press  button to move the signal to the Output List.

c. When checking the checkboxes in the tree map, multi-channels can be selected for complicated calculation. In addition, the signal code (such as X, X1[2]) and math operations (such as +, -, …, sin, log) can be typed directly in the Expression field.

Toolbox

The above image shows the Toolbox of the MultiChannel Expression Editor. The Expression field is used for editing math equations.

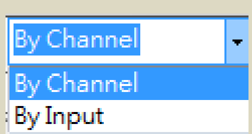
 clears all the output signals in the Output Channel panel

 add the math equation in the Expression field to the Output Channel list

 replace the math equation in one of the output signal of the Output Channel list with the equation in the Expression field

Other buttons in the Toolbox are explained below

Basic Operation	Function Definition
+	Add plus operation to Expression field, "+" can be typed in directly
-	Add minus operation to Expression Field, "-" can be typed in directly
*	Add multiply operation to Expression Field, "*" can be typed in directly
/	Add divide operation to Expression Field, "/" can be typed in directly

Special Operation	Function Definition
	Set group operation type: By Channel or By Input. By Channel: channel-by-channel calculation for selected multichannels, the output is a single channel, such as Y[1]=X1[1]+X1[2]+X2[1]+X2[2]. By Input: input-by-input calculation for selected input signals, the output is multi-channel signal, such as Y[1]=X1[1]+X2[1], Y[2]=X1[2]+X2[2].
π	Add π (pi) value to Expression field
t	Add vector of time axis t to Expression field. It is

	corresponding to the time axis of the selected input signal
	These two tools work as a group. The pull-down menu gives the internal functions. After selecting the internal function, press  button to add selected function to the Expression field. The function can also be typed in directly, such as sin(X1[1]), abs(X1[1])

Common internal functions are listed below

Function	Description	Function	Description
`abs`	Absolute value	`ceiling`	Round to the nearest integer toward infinity
`floor`	Round to the nearest integer towards minus infinity	`round`	Round to the nearest integer
`sin`	Sine	`asin`	Inverse sine
`cos`	Cosine	`acos`	Inverse cosine
`tan`	Tangent	`atan`	Inverse tangent
`sinh`	Hyperbolic sine	`cosh`	Hyperbolic cosine
`tanh`	Hyperbolic tangent	`exp`	exp(x) equals e^x
`log`	Natural logarithm	`log10`	Base 10 logarithm
`pow`	pow(x, a) equals x^a	`sqrt`	Square root
`square`	Equals x^2	`sign`	Signmum function. Returns 1 if greater than zero, 0 if equals zero and -1 if less than zero
`Truncate`	Rounds to the nearest integer towards zero. If x<0, truncate(x) equals ceiling(x). If x>=0, truncate(x) equals floor(x)	`conj`	Complex conjugate. For a complex x, $\mathrm{conj}(x) = \mathrm{Re}(x) - i \times \mathrm{Im}(x)$

In addition, there exists ">", "<", ">=", "<=", "==", and "!=" conditional signs. If the condition is satisfied, return 1. If the condition is not satisfied, return 0. There are examples below to show the usage.

Output Channels

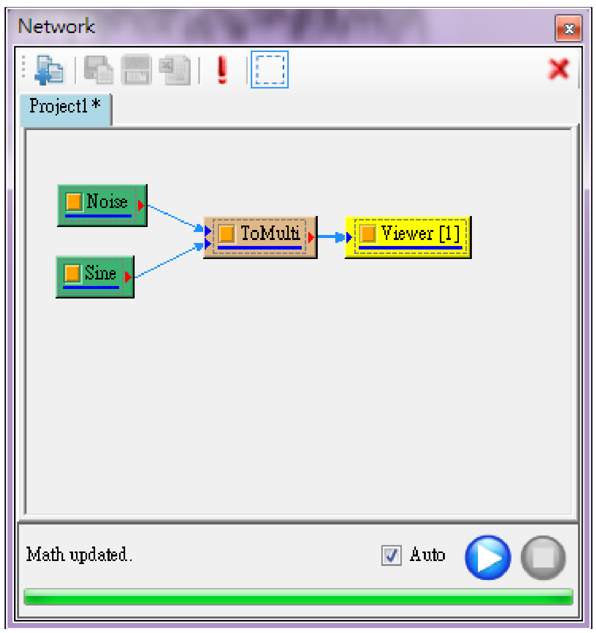
Output Channels display all output channels and defined math equations in Expression. The order of the channels in the output gives the sequence number of the signal. The sequence number/order of the channel can be changed using the button to the right of the panel,  moves up and  moves down, while  deletes channels.

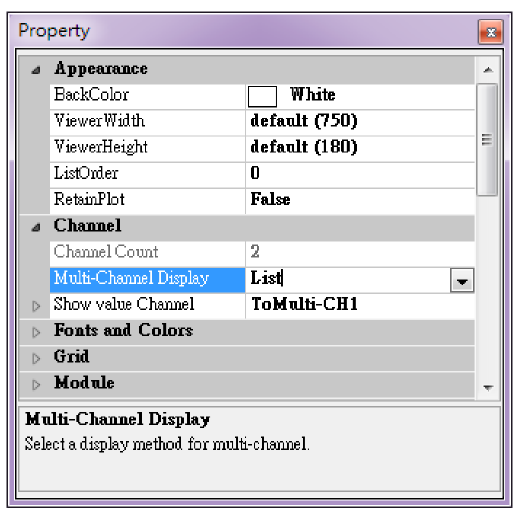
The name of the channel and the equation can be modified. Double click the channel to modify the name; if the math equation needs to be modified, select the Express of the target channel, click the mouse left-button once (similar to double click, but with a slower speed), then the Expression can be edited directly.

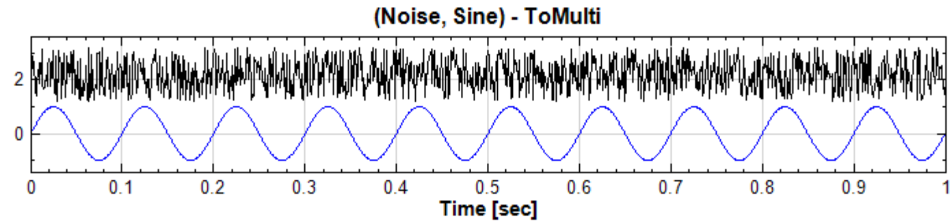
Example

Different calculation methods and functionality usages are shown below.

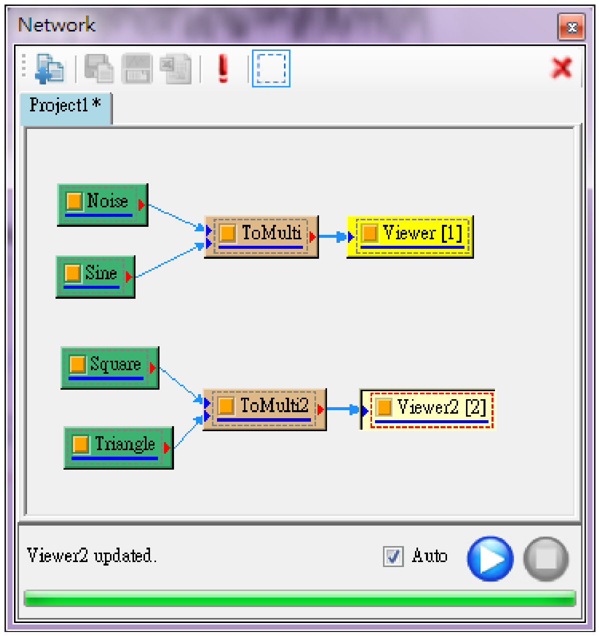
1. Create a noise and sine wave using **`Source→Noise, Sine Wave`**. Connect to **`Conversion→Merge to Multi-Channel`** to make a multi-channel signal. Then view the signals by connecting the ToMulti component to **`Viewer →Channel Viewer`**. Change *`Multi-Channel Display`* in **`Channel Viewer`** properties to *`List`*. This will cause each channel to be displayed separately.

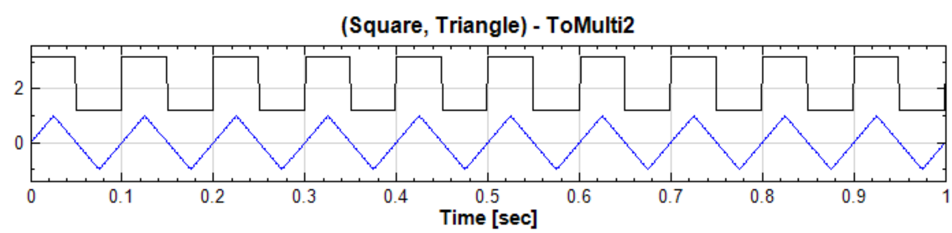




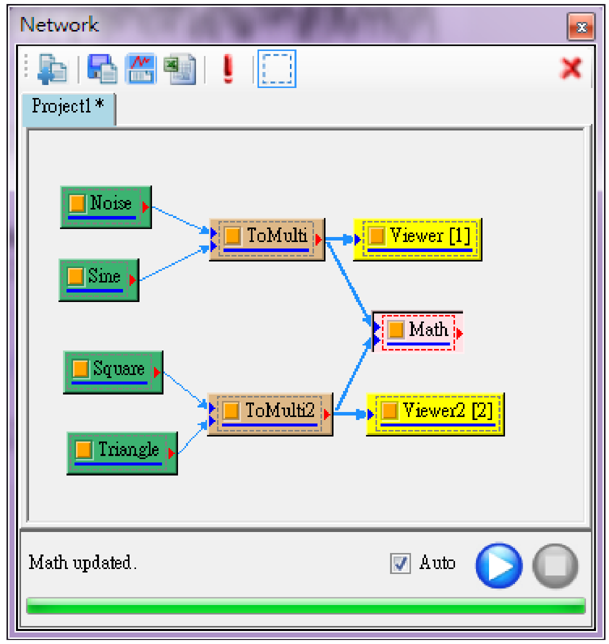


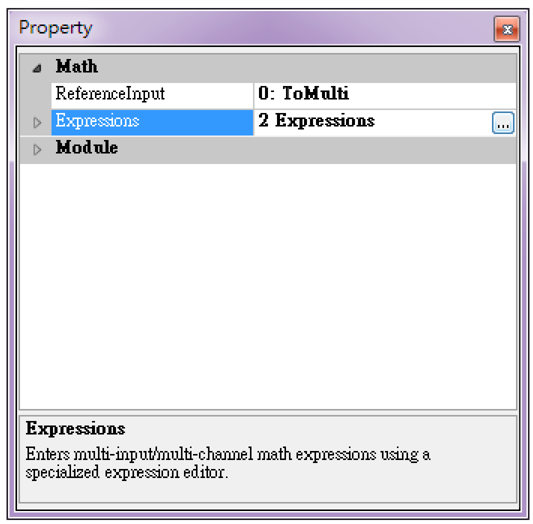
2. Do the same as step 1. However, change the **Noise** component and **Sine Wave** component to **Square Wave** and **Triangle Wave**, respectively.



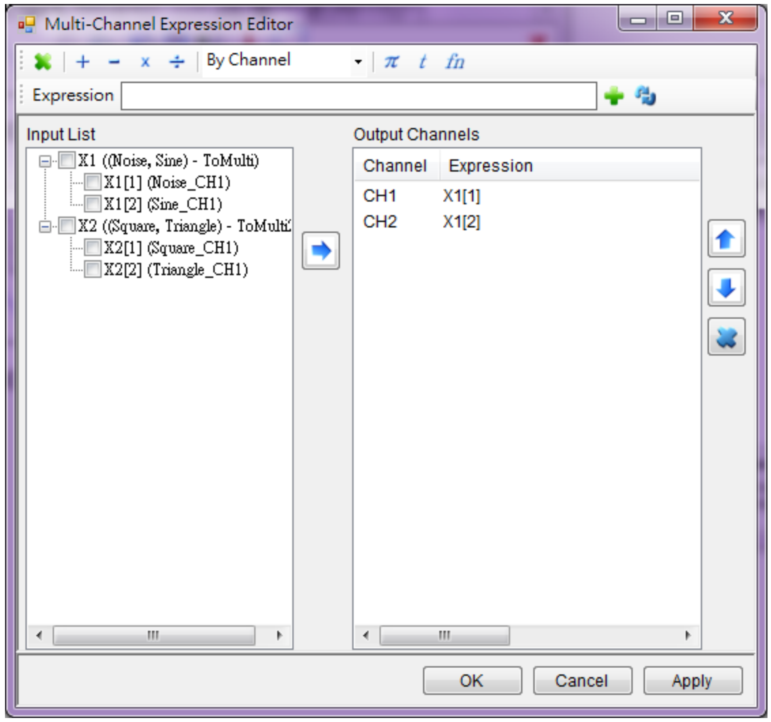


3. Connect **ToMulti** component and **ToMulti2** component to **Compute→ Mathematics→Math**. Expand the + box before Expressions in the **Math Property Window** and click *Expression Editor* at the end of the line to open the interface.

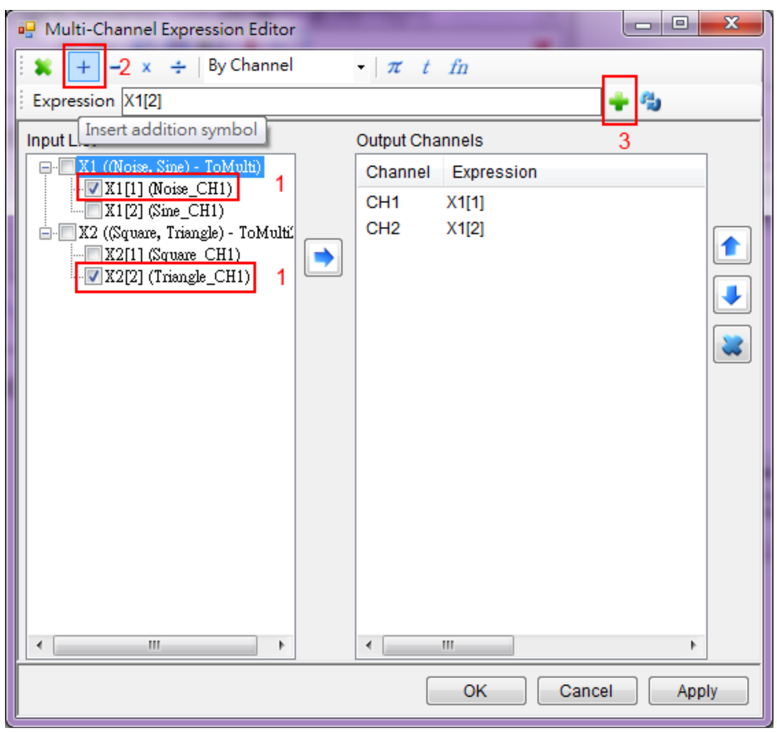




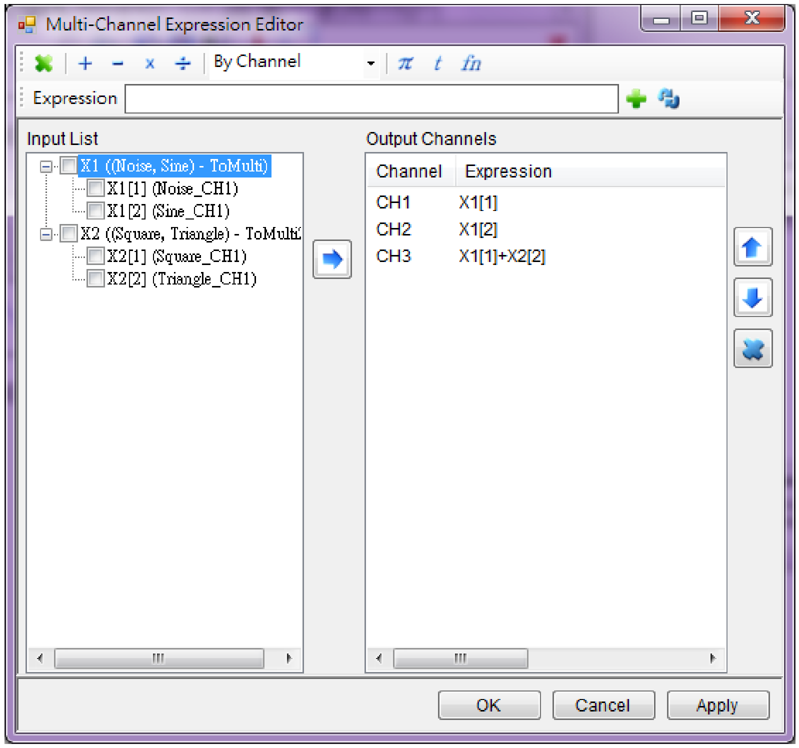
4. In **Multi-Channel Expression Editor**, open the tree map in the Input List by clicking the + sign in front of the signals. Please note that the default output signal is the 1st input signal.



5. In order to calculate the sum of channel 1 of X1 and channel 2 of X2 together, select both channels and click the basic operator "+" in the toolbar at the top of the window. The summation equation will then be added to the Expression field. This equation "X1[1]+X2[2]" can be typed in the field directly as well.

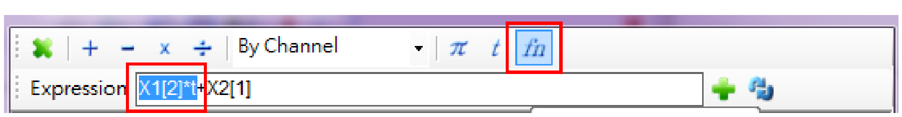


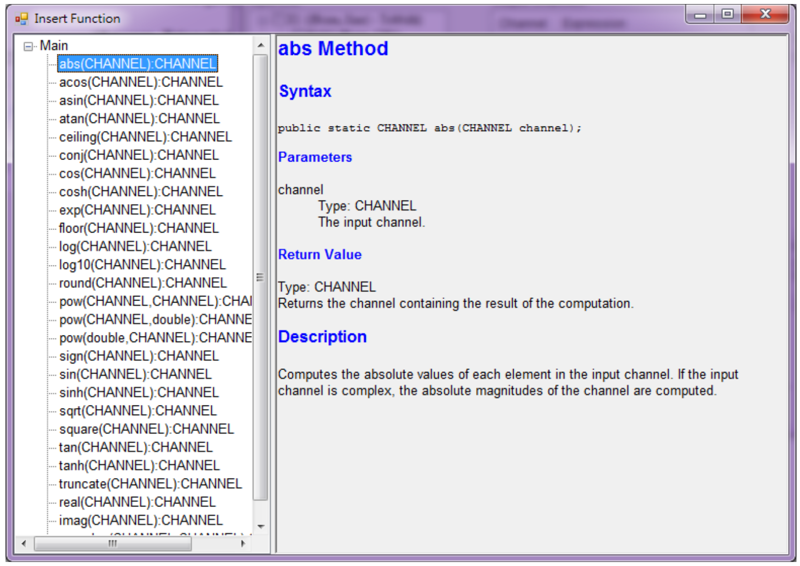
6. Press button to transfer the equation in the Expression field to the Output Channels. Here CH3 is added.



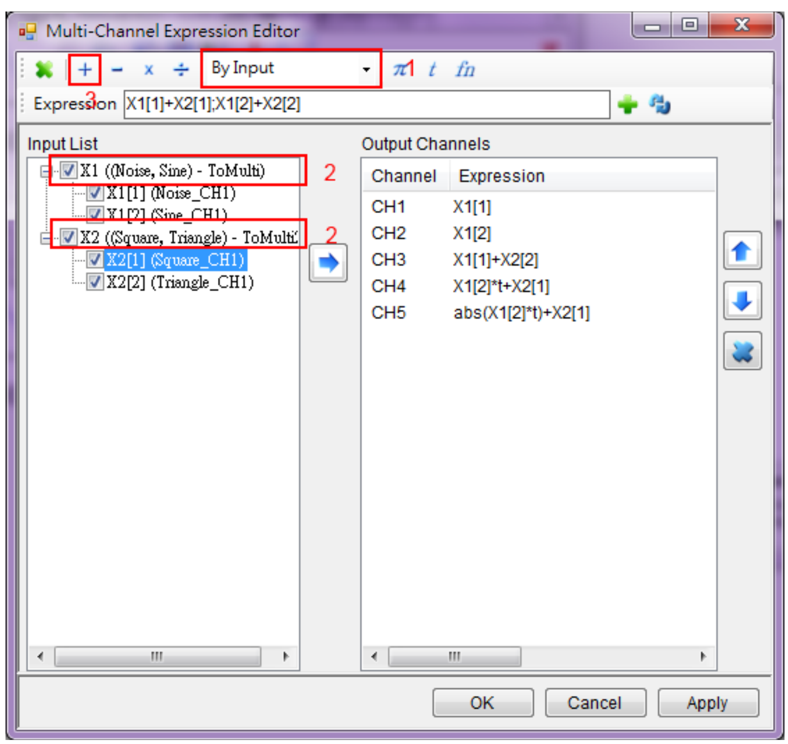
7. Next, the channel 2 of X1 multiplies the corresponding time t, then adds channel 1 of X2. Add channel 2 to the Expression field by double clicking X1[2] under X1, then click the basic operator in the Toolbox to complete the equation. The equation "X1[2]* t + X2[1]" can also be directly typed in the Expression field.

8. To get the absolute value of X1[2]* t, edit the equation directly to "abs(X1[2]* t)+ X2[1]", or highlight the "X1[2]* t" part in Expression field, then select function **abs** from the function list by pressing the **fn** button to complete the equation. All internal functions can be added this way. Finally, click the button to transfer the equation to the Output List.



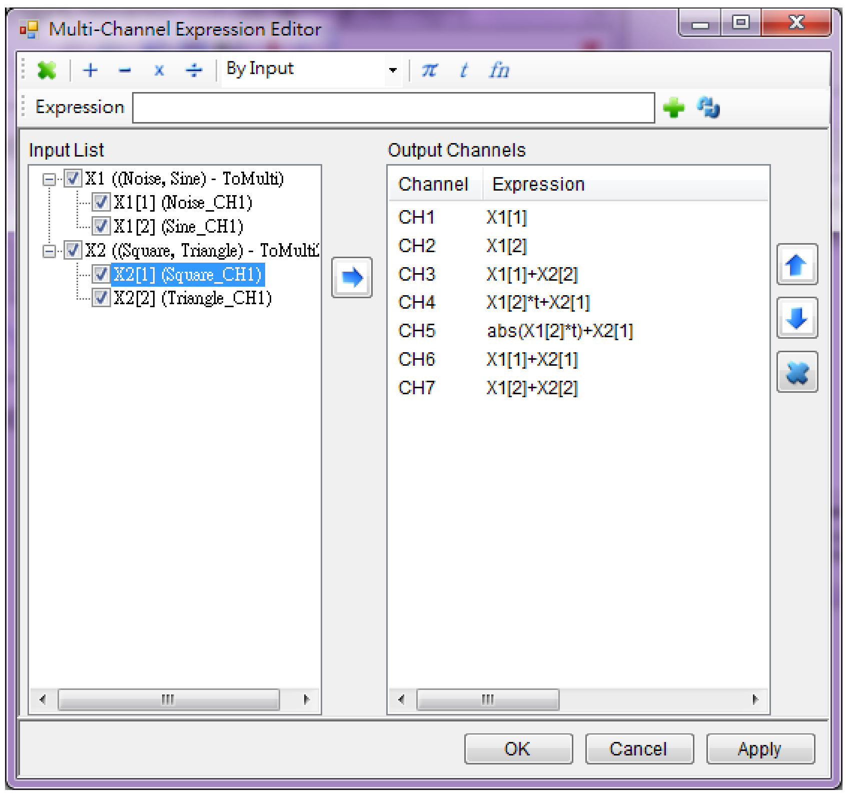


9. For the calculation of more than two input signals, such as CH1+CH1, CH2+CH2, the "By Input" option can be used. The top level selections are input signals instead of each channel under the signal. Once the signal is selected, all channels under it will be selected automatically. As shown below, select input signal X1 and X2, then select "By Input", and click basic operator "+", the full calculation equations will then be added to the Expression field.

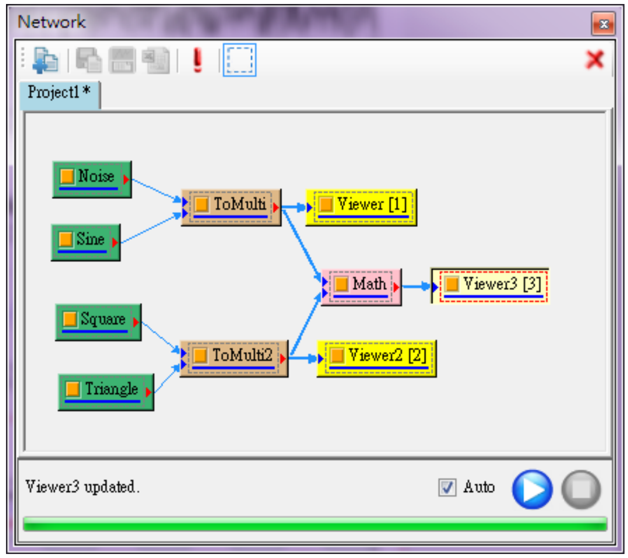


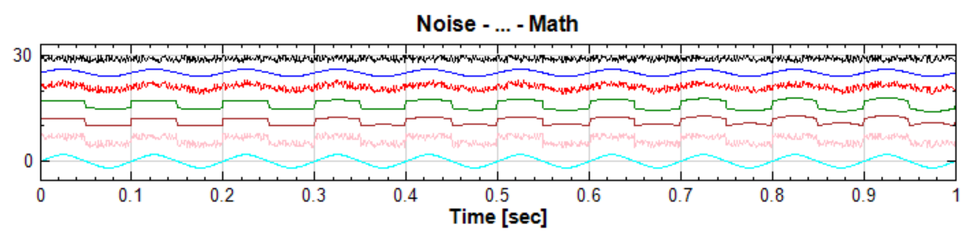
10. Finally click  button to transfer the equations to the Output Channels.

11. In Output Channels panel, there are 2 calculation channels added. There are both 2 channels in X1 and X2. If the channel counts are not same between each group, **Math** uses the lesser number for the output. There should now be 8 channels if the above steps were done correctly.

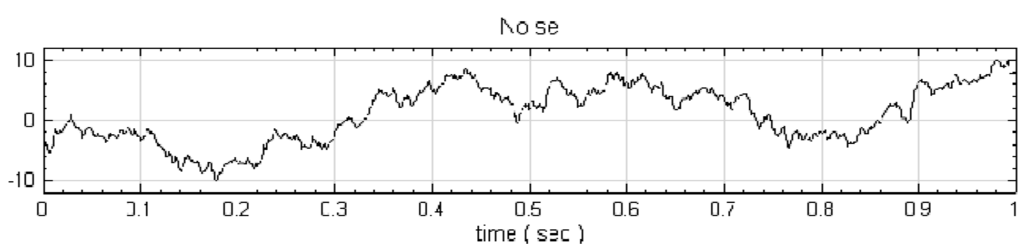


12. To the right of the Output Channels, there are 3 buttons. They can be used to move the output channel up and down, or to delete output channels. Once the output channels are ready, press "OK" or "Apply" to complete. The Output Channels can be displayed via **Channel Viewer**. Don't forget to change `Multi-Channel Display` to `List`.

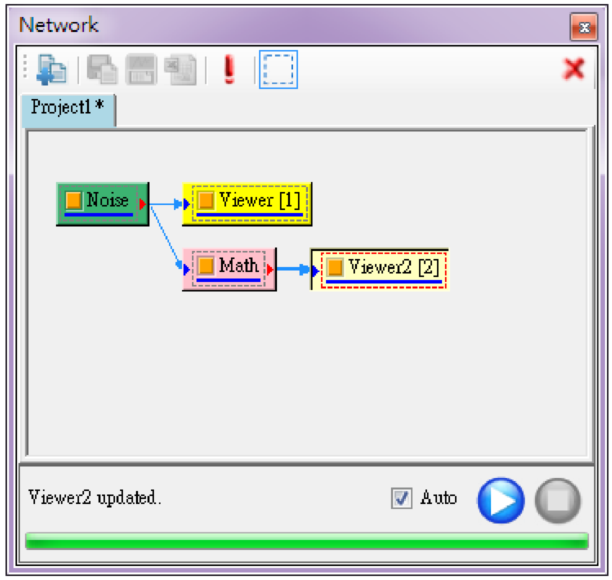




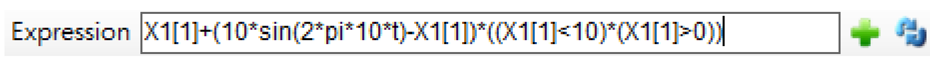
13. The Expression function can carry out calculations for ">", "<", "==", "!=" etc. Create a Noise signal using **Source→Noise** and set *NoiseType* to *Brown*, *Amplitude* to 10, then display the signal using **Viewer→Channel Viewer**.



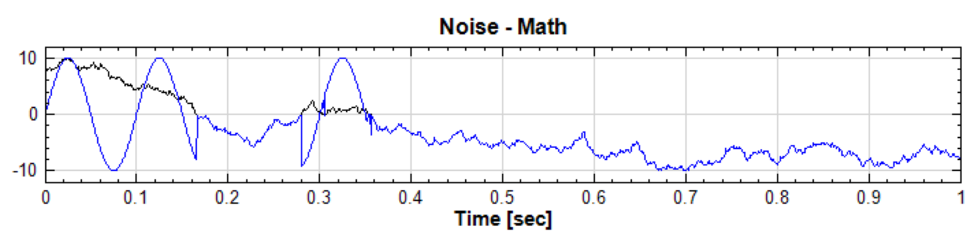
14. Connect **Noise** component to **Compute→Mathematics→Math**, and display the output signal of Math with **Channel Viewer**.



15. Replace the **Noise** signal, where the amplitude is between 0 and 10, with a **Sine** wave. The equation of the calculation is written in the Expression field.
    X1[1]+(10*sin(2*pi*10*t)-X1[1])*((X1[1]<10)*(X1[1]>0))



16. Connect the Math component to the same viewer as the input Noise signal and compare the curves.



Related Functions

Channel Viewer, Mixer, Multiplier, Source, Merge To Multi-Channel

4.1.3.4 Mixer

Mixer is used to mix several signals.

Introduction

Assume n groups of signals, $X^{(n)}(t)$, each group of which has different time axis t and sampling frequency. The mixed Signal $Z(t)$ is

$$Z(t_i) = a \cdot X^{(1)}(t_i) + b \cdot X^{(2)}(t_i) + c \cdot \left(\sum_{k=3}^{n} X^{(k)}(t_i) \right)$$

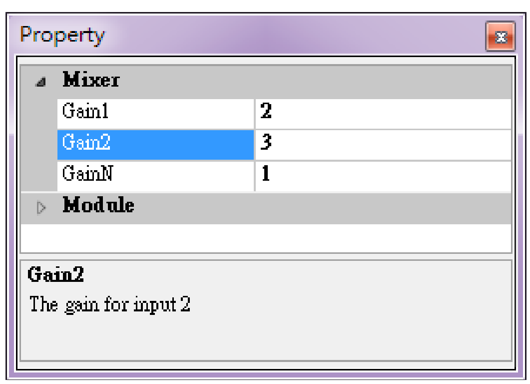
where a, b, and c are weights

In this module, because the time-axis of input signals are supposed to be different, the minimum sampling frequency $Freq_{min}$ of the input signals is extracted first, then all other signals are re-sampled by $Freq_{min}$. After the time-axis of all input signals are unified, the signals are added at every time point. Notice that the weights from the 3rd group of signals are all equal to c.

Properties

This module accepts input of Signal (which could be a real number, single channel, Regular) and Audio (which could be a real number, single channel, Regular). Multiple signal input is also allowed.

`Gain1`, `Gain2` and `GainN` are the weights of the first, the second and the third group of input signals, respectively. The difference between this module and **`Math`** is that the **`Mixer`** can perform faster addition/subtraction computation, and also perform addition/subtraction on signals with different length, while **`Math`** does not have this type of functionality.

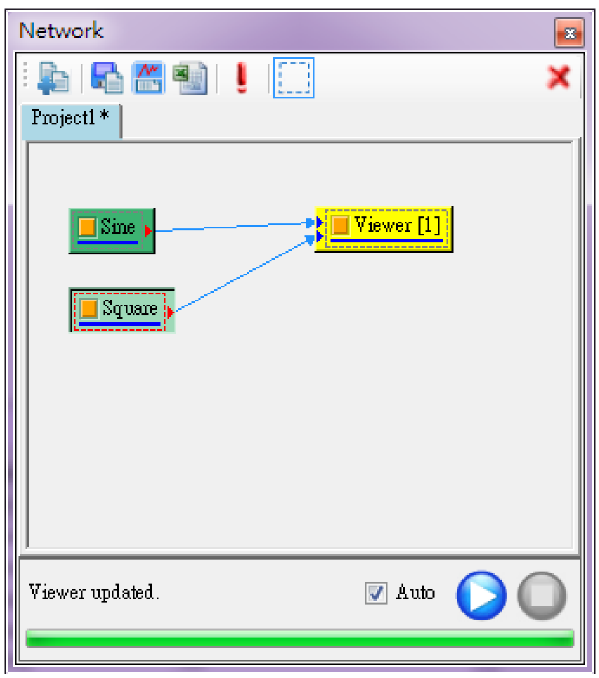


{Mixer} Property Name	Property Definition	Default Value
`Gain1`	Set the weight a for the first group of signals	$a = 1$
`Gain2`	Set the weight b for the second group of signals	$b = 1$
`GainN`	Set the weight c for the signals from the 3rd group	$c = 1$

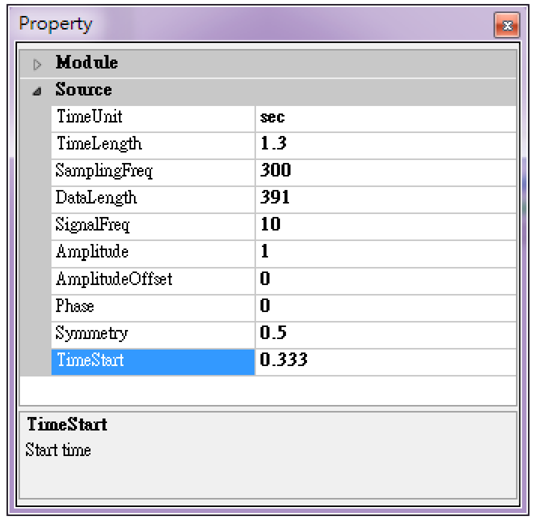
Example

This example below shows the procedure to mix a sine wave and a square wave with different time axis.

1. Use **Source→Sine Wave** to create a signal with a frequency of 5Hz, sampling frequency of 1000Hz and duration of 1.5 seconds. Then create a square wave with frequency of 10Hz, sampling frequency of 300 Hz, duration of 1.3 seconds, and time starting point of 0.333 second. Next, use **Viewer→Channel Viewer** to observe the wave.



Sine properties are shown in the table below.

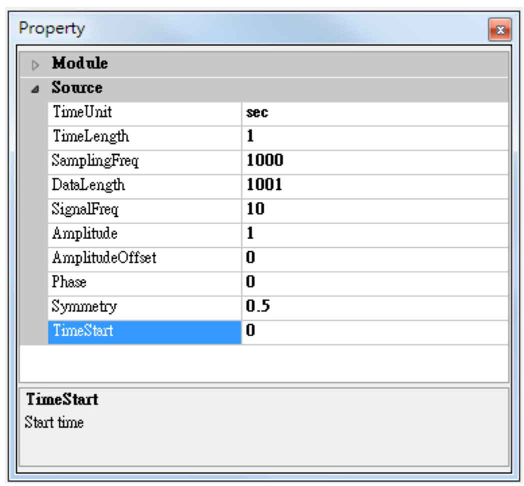


Property

Module	
Source	
TimeUnit	sec
TimeLength	1.3
SamplingFreq	300
DataLength	391
SignalFreq	10
Amplitude	1
AmplitudeOffset	0
Phase	0
Symmetry	0.5
TimeStart	0.333

TimeStart
Start time

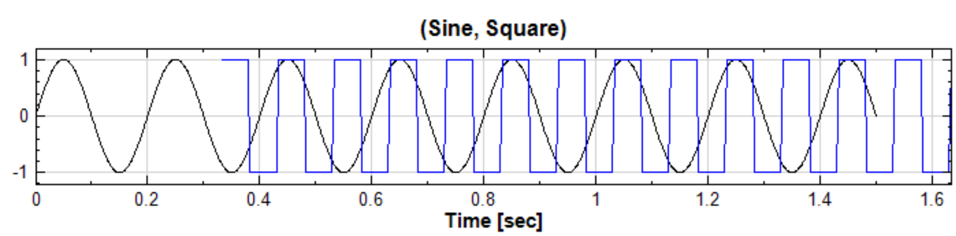
Square properties are shown in the table below



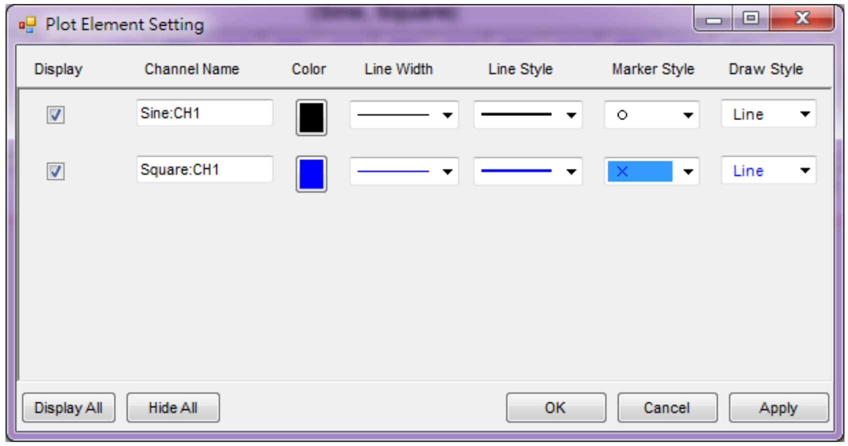
Property

Module	
Source	
TimeUnit	sec
TimeLength	1
SamplingFreq	1000
DataLength	1001
SignalFreq	10
Amplitude	1
AmplitudeOffset	0
Phase	0
Symmetry	0.5
TimeStart	0

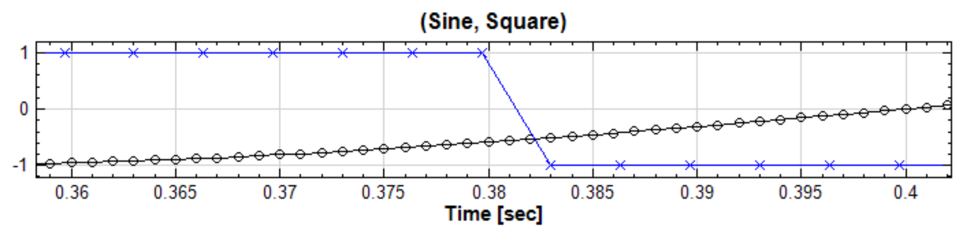
TimeStart
Start time



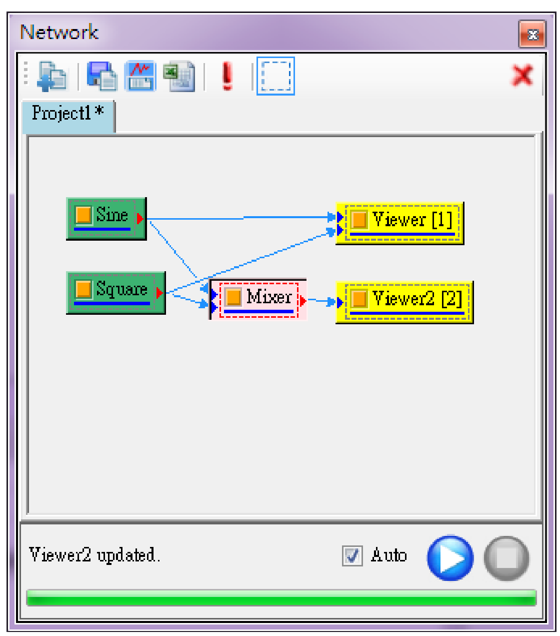
2. Select `PlotEditor` in the `Plot Elem Editor` in **Channel Viewer**. In the popped-up **Plot Element Setting Window**, add 「o」 to the Sine curve, 「x」 to the Square curve, and use the tool **Zoom X** to enlarge the overlapping point of these two signals. It can be seen that the signal data-point distributions along the X-axis (time-axis) are completely different. Settings of `PlotEditor` are given as follows.

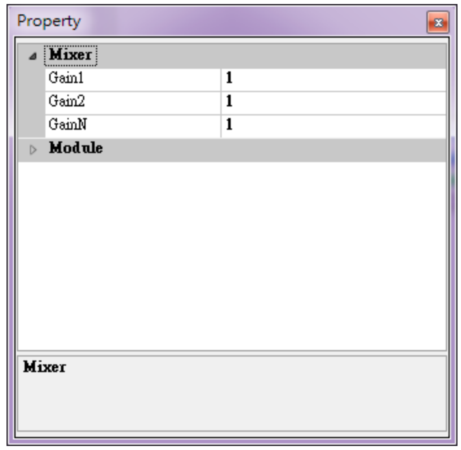


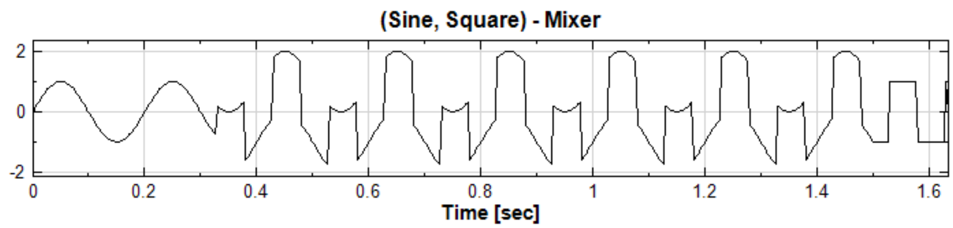
Zooming in on the result in **Channel Viewer** is shown below.



3. Add these two signals to form a new signal. As shown below in the **Network Window**, use **Compute→Mathematics→Mixer** to perform the signal mixture. The first input to **Mixer** is Sine, the second input is Square and the corresponding properties are *Gain1* and *Gain2*, respectively. Both have a default value of 1. Next, use **Viewer→Channel Viewer** to plot the mixer wave.

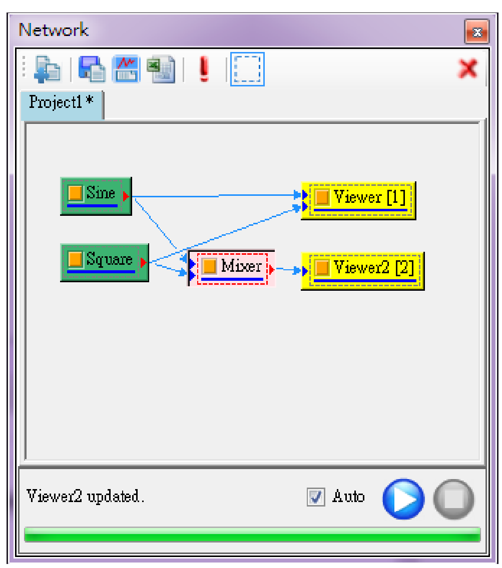


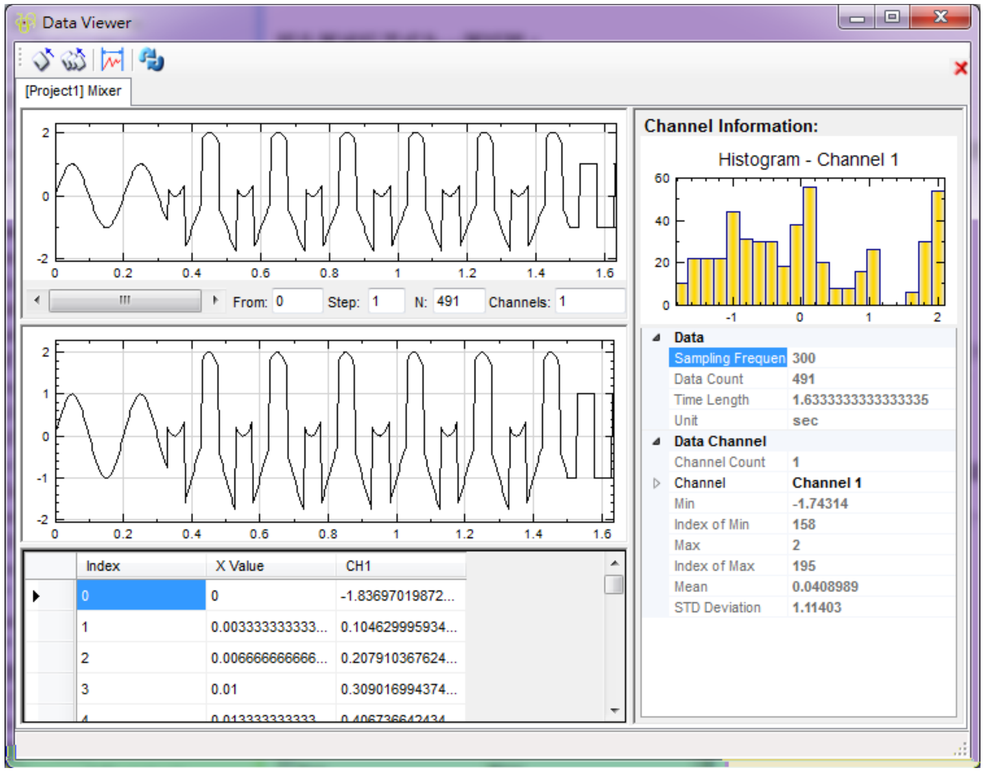




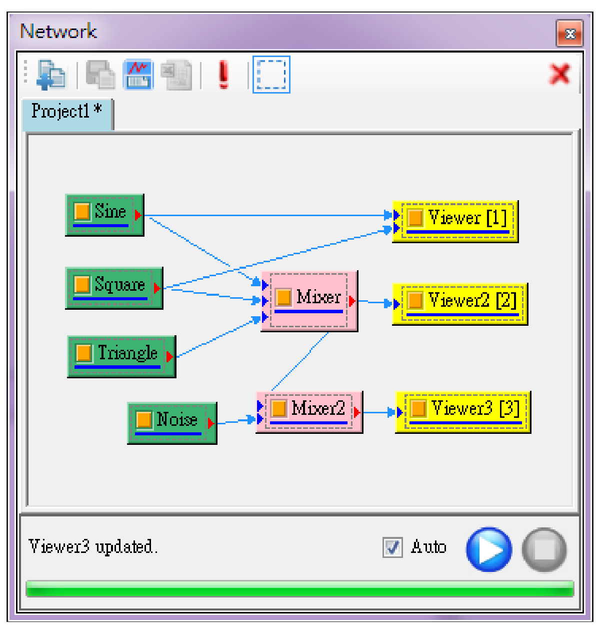
4. Now, use **Data Viewer** to check the sampling frequency and duration of the output signal from **`Mixer`**. The sampling frequency is 300 Hz. The signal starts at 0 second and ends at 1.63333 second.

   The computation method in **`Mixer`** uses the duration of mixed input signal as the total duration, selects the minimum sampling frequency from the input signals, and adds all signals after they are multiplied by corresponding weights. Therefore, to use **`Mixer`**, special attention must be paid to the sampling frequency of input and output signals.





5. Notice that more than 3 groups of data can be mixed. However, for all data groups which are higher than three, the weights will all be set as `GainN`. Therefore, it is not encouraged to use one mixer to mix more than 3 groups of data. It is recommended to use a multi-layer mixer to achieve mixture of more than 3 groups of data. The figure below shows an example using this method to mix multiple signals.



If you want to understand this function better, open Demo05 in C:\Program Files\AnCAD\Visual Signal\demo\Basic. This demo will show an example of a project utilizing the **`Mixer`** function.

Related Functions

`Channel Switch, Multiplier, Source`

5.1.3.5 Multiplier

This component multiplies multiple input signals.

Introduction

Mathematically, assume n groups of signal sources, $X^{(n)}(t)$ where time-axis t and sampling frequency of every signal are not necessarily to be identical. The mixed signal $Z(t)$ is

$$Z(t_i) = X^{(1)}(t_i)X^{(2)}(t_i)\cdots X^{(N)}(t_i)$$

In this module, because the time-axis of input signals are supposed to be different, the minimum sampling frequency, $Freq_{min}$, in all input signals is extracted first, and then all other signals are re-sampled by $Freq_{min}$. After the time-axis of all input signals are unified, the signals are multiplied at every time point.

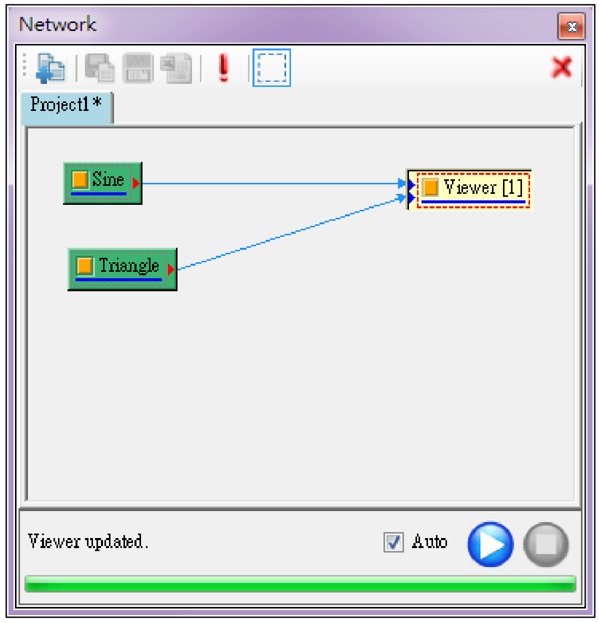
Properties

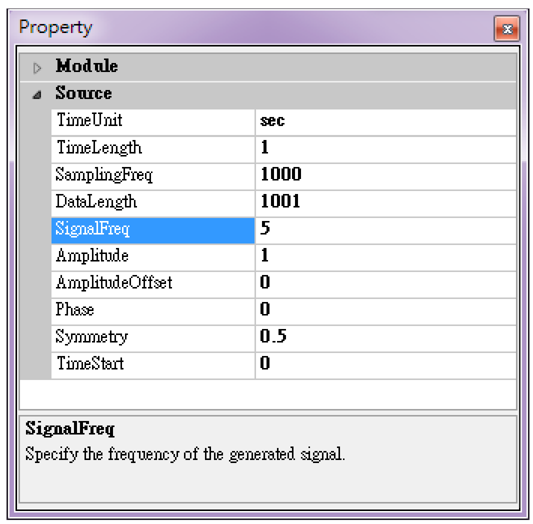
This module accepts input of Signal (which could be a real number or complex number, single channel, Regular) and Audio (which could be a real number or complex number, single channel, Regular). Multiple signal input is also allowed. This module does not require default values. It can perform multiplication on signals which have different length and sampling frequency.

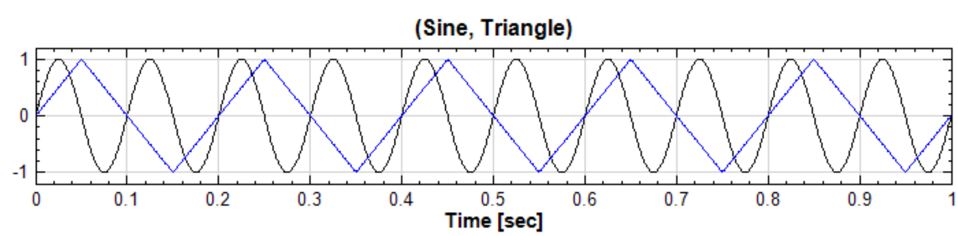
Example

This example shows the multiplication of a sine wave and a triangular wave.

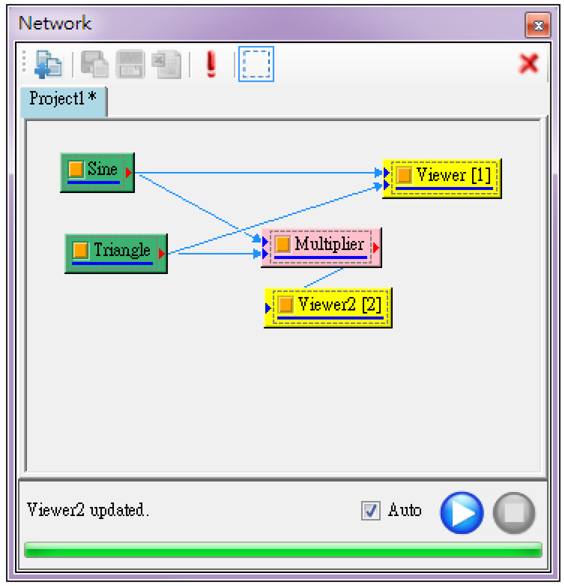
1. Use **`Source→Sine Wave`** and **`Source→Triangle Wave`** to generate a sine wave and a triangle wave. Change the *`SignalFreq`* of the triangle wave to 5 and use the **`Viewer→Channel Viewer`** to observe the original wave.

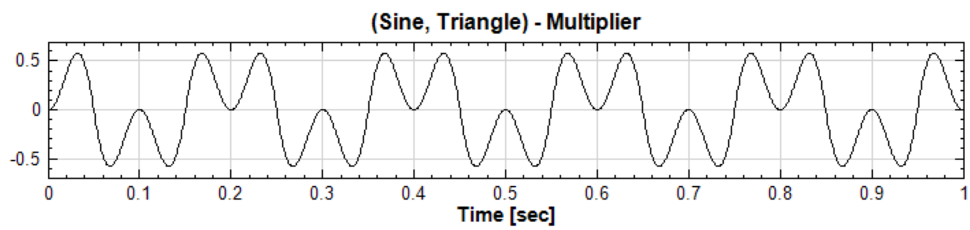




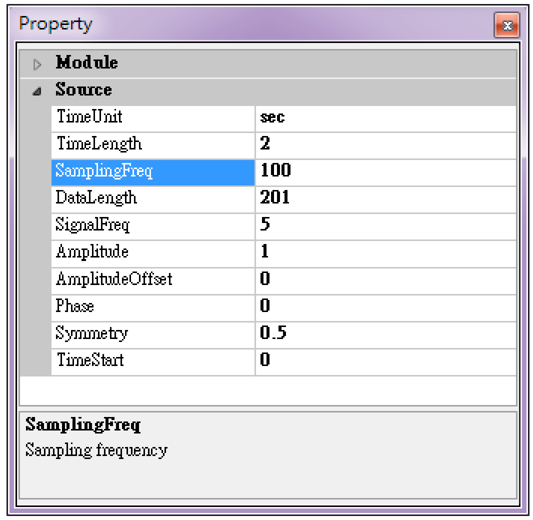


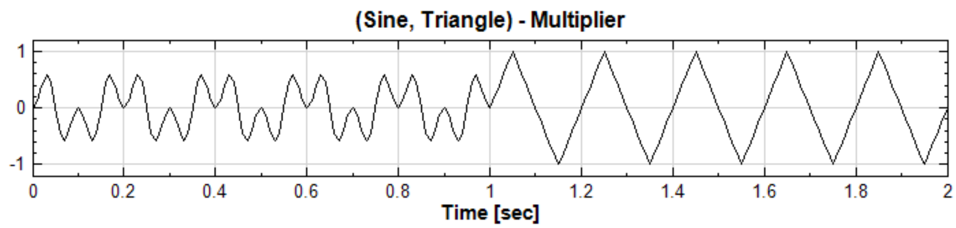
2. Multiply these two signals using **Compute→Mathematics→Multiplier**. The output signals are shown as below.





3. Like **`Mixer`**, the **`Multiplier`** function allows the sampling frequency and time length of the two signals to be different. The sampling frequency of the output signal is identical to the minimum sampling frequency in the input ones. On the time axis, the overlapping parts of the input signals are multiplied together while the other part is intact. Change the *`SamplingFreq`* of the triangular wave to 100, *`TimeLength`* to 2. Then, in the output signal, the signal frequency will become 100 and the time length will become 2 seconds.





Related Functions

`Channel Switch, Mixer, Channel Viewer, Source`

5.1.3.6 Normalize

The signal data is divided by different normalization values.

Introduction

Let the signal source be $X = \{x_0, x_1, \cdots, x_{n-1}\}$, and its standard deviation be σ. The value, σ, is the normalization value. The normalized signal is $Y = \left\{\frac{x_0}{\sigma}, \frac{x_1}{\sigma}, \cdots, \frac{x_{n-1}}{\sigma}\right\}$.

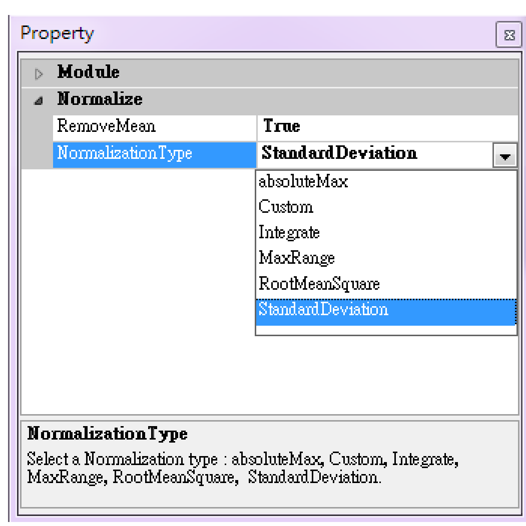
Properties

This module accepts input of Signal (which could be a real number, single channel/multi-channel, Regular) and Audio (which could be a real number, single channel/multi-channel, Regular).

The property, `Normalization Type`, in the **`Normalize`** component includes 6 types. Selecting the property `absoluteMax` will use the maximum value of the signal to normalize. Selecting `Custom` will show another option, `CustomNormalizationValue`, where you can specify the normalization value. Select `Integrate` and the normalization value is the integral of the signal by the trapezoidal rule. Select `MaxRange` and the normalization value is the maximum subtracted by the minimum of the signal. Select `RootMeanSquare` and the normalization value is the following equation below

$$\mathrm{RMS} = \sqrt{\frac{x_0^2 + x_1^2 + \cdots + x_{N-1}^2}{N}}$$

where $X = \{x_0, x_1, \cdots, x_{N-1}\}$ is the source signal



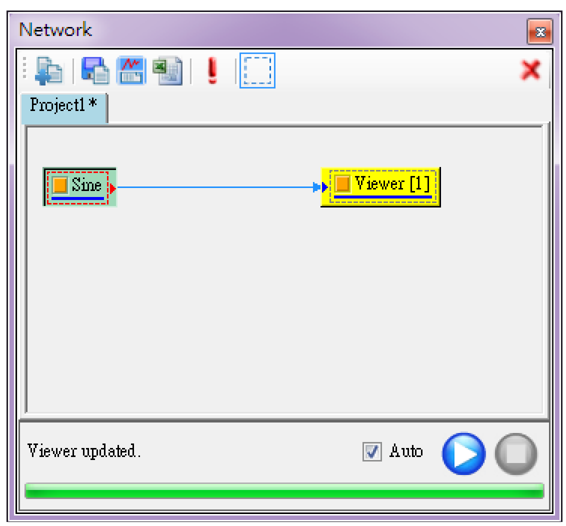
{Normalize} Property Name	Property Definition	Default Value
RemoveMean	Setting to True removes the mean value of signal. Otherwise, False.	True
Normalization Type	There are six normalization types which consist of *absoluteMax*, *Custom*, *Integrate*, *MaxRange*, *RootMeanSquare*, and *StandardDeviation*.	*StandardDeviation*

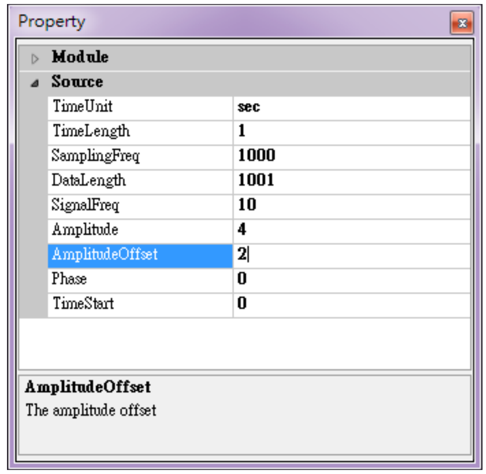
Variable Name	Variable Definition	Default Value
CustomNormalizationValue	Specify the custom normalization value	1

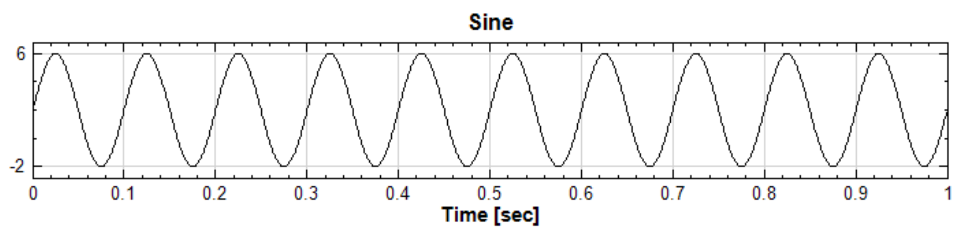
Example

Create a signal, which is shifted with y-axis, and enlarge the amplitude. Then, use the **Normalize** component to remove the shift and normalization.

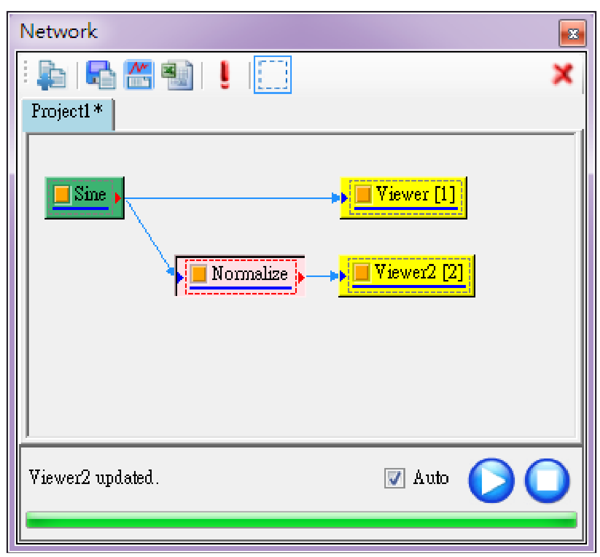
1. Create a sine wave using **Source→Sine Wave**, and change the property *Amplitude* to 4 and *AmplitudeOffset* to 2. Connect a viewer component using **Viewer→Channel Viewer** and observe the graph.

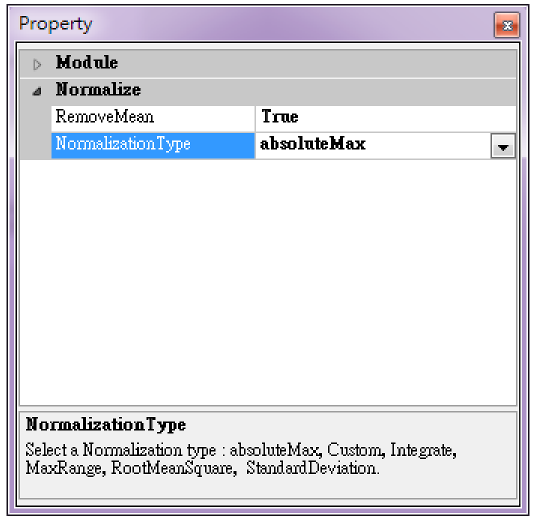


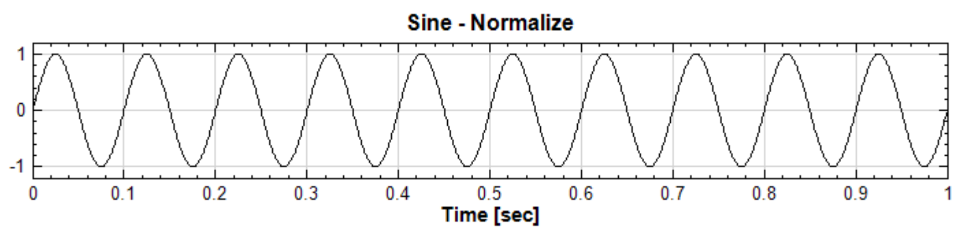




2. Connect the signal to **Compute→Math→Normalize** and set *RemoveMean* to *True*. Select *absoluteMax* from the *NormalizationType* field, and connect a viewer component to observe the signal. The amplitude of signal is normalized to range $[-1, 1]$.







Related Functions

`Source, Channel Viewer`

5.1.3.7 Remove DC

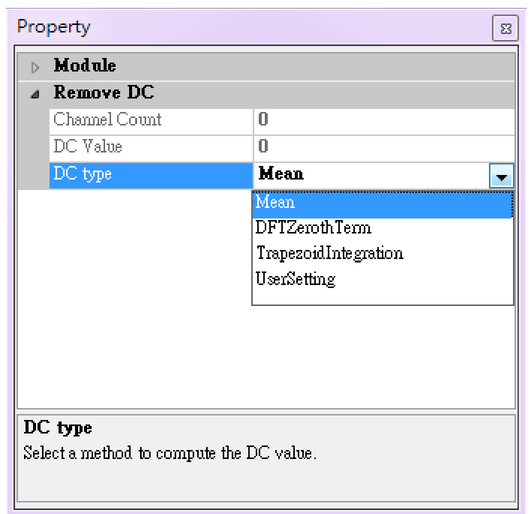
Remove the signal direct current component, i.e. remove the signal shift along the Y-axis.

Introduction

Let the signal source be $X = \{x_0, x_1, \cdots, x_{N-1}\}$ with $\bar{x}$, i.e. DC. Here after, $X' = \{x_0 - \bar{x}, x_1 - \bar{x}, \cdots, x_{N-1} - \bar{x}\}$ is said to be **Remove DC**.

Properties

This module accepts input of Signal (which could be a real number, single channel, Regular) and Audio (which could be a real number, single channel, Regular).
The property is of DC type which includes four types of calculation methods to compute the shift along the Y-axis, where the default option is *Mean*. The detailed meaning of these methods is given in the table below.



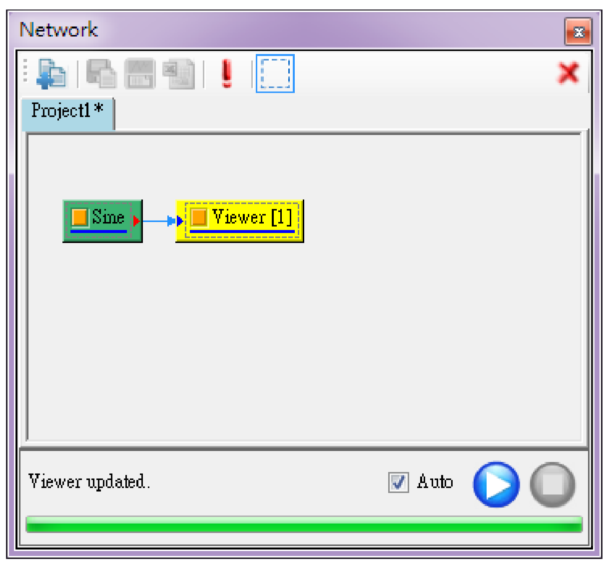
{Remove DC} Property Name	Property Definition	Default Value
Channel Count	The input channel count	0
DC Value	The input DC value	0
DC type	There are four types of DC which consist of *Mean*, *DFTZerothTerm*, *TrapezoidIntegration*, and *UserSetting*.	*Mean*

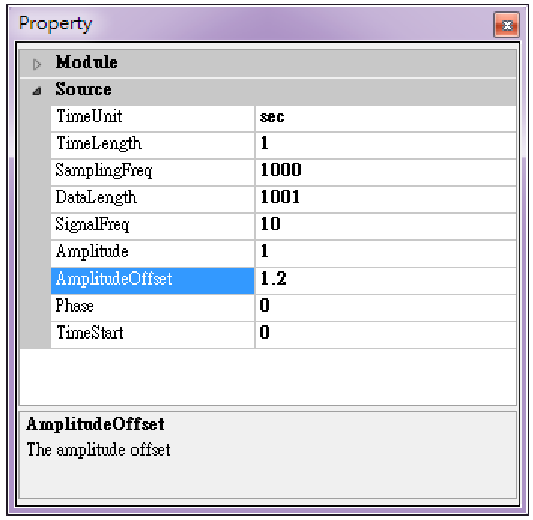
{DC type} Variable Property	Property Definition
`Mean`	To calculate the arithmetic average
`DFTZerothTerm`	After performing Fourier Transform on the original data, define X-axis as zero point which has the value $DC \equiv \frac{a_0}{2} = \int_{-\infty}^{\infty} f(t)e^{-i\omega t}dt\Big|_{\omega=0}$
`TrapezoidIntegration`	Divide the result of a Trapezoid Integration by the total number of points. The result is the DC.
`UserSetting`	The users can set the desired shift value manually

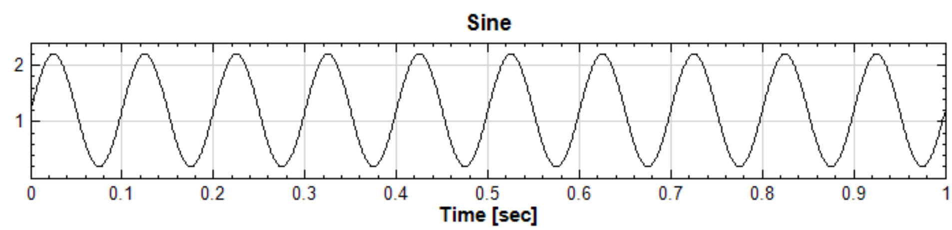
Example

Create a sine wave which is shifted along Y-axis and then use **RemoveDC** to remove the shift.

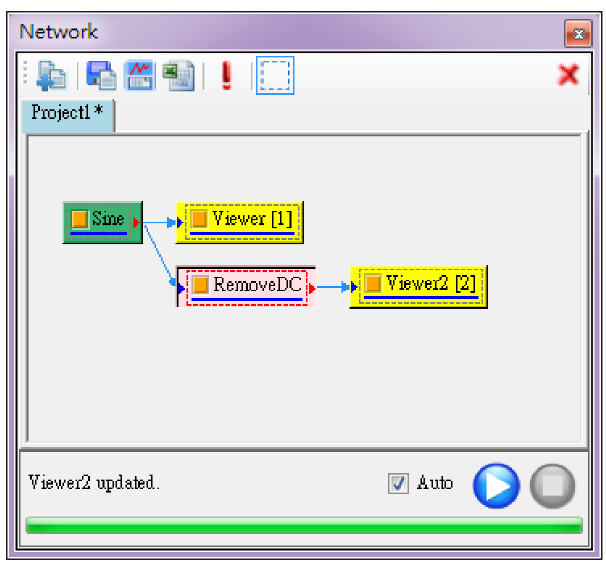
1. Create a sine wave using the **Source→Sine Wave** and then adjust the *AmplitudeOffset* to 1.2 to shift the signal along the Y-axis in positive direction for 1.2 unit. Next, use the **Viewer→Channel Viewer** to observe.

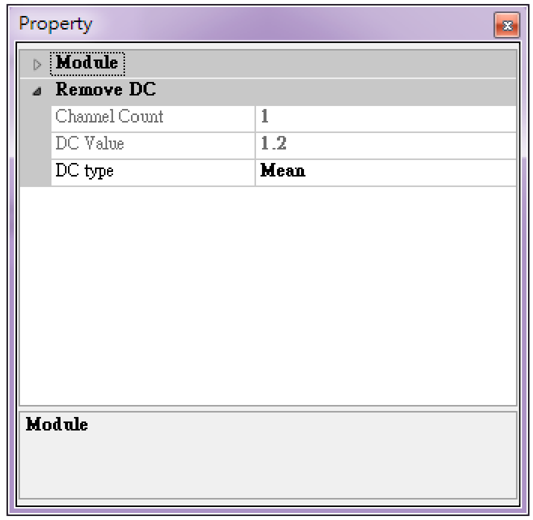


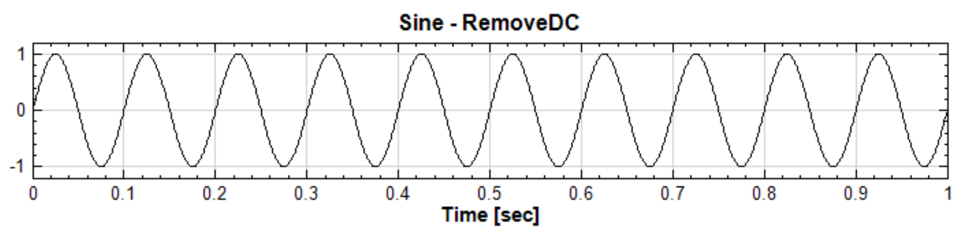




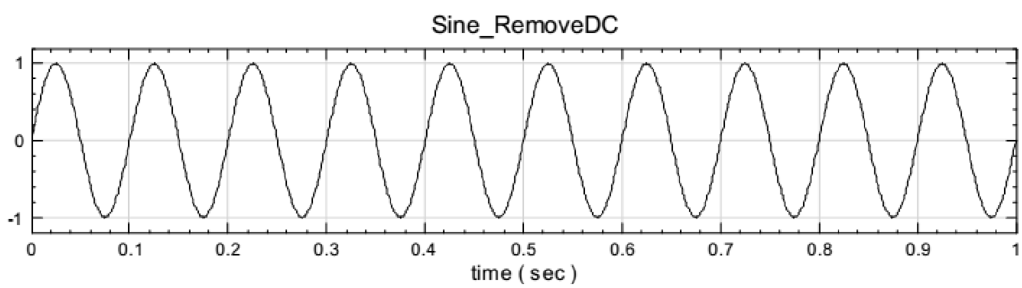
2. Connect the original signal to **Compute→Mathematics→RemoveDC** and set the method as *Mean* in the property *DCType*. It can be seen that the shift is removed.



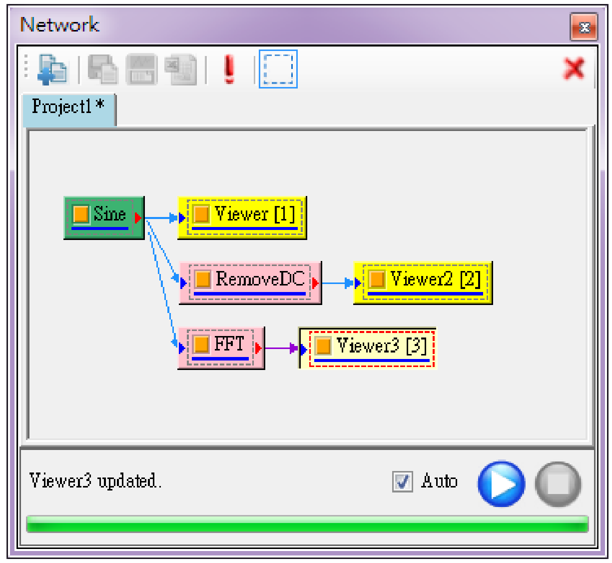


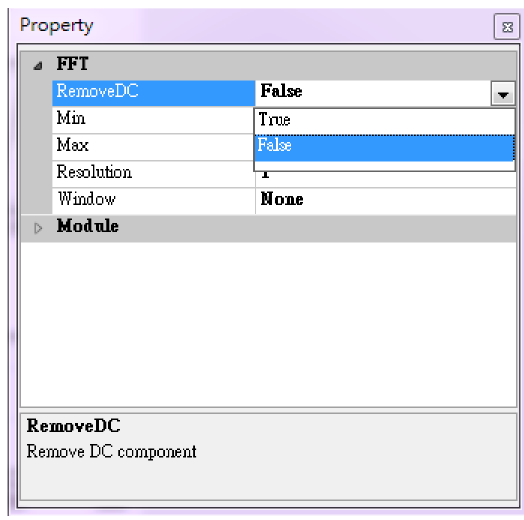


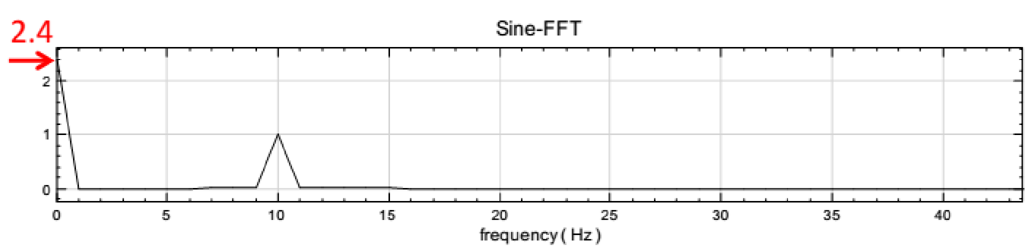
3. The DC Type can also be changed, e.g. *`DFTZerothTerm`*. However, in this example, the result would be identical.



4. Connect the signal to **`Compute→Transform→Fourier Transform`** to perform Fourier Transform. Without the **`RemoveDC`** in Fourier Transform, it can be seen that the amplitude at 0Hz is 2 times of 1.2.







For horizontal shifting along time axis, please reference the **`Channel`** → **`TimeShift`** module.

Related Functions

`Source, Fourier Transform, TimeShift`

4.1.3.8 RMS

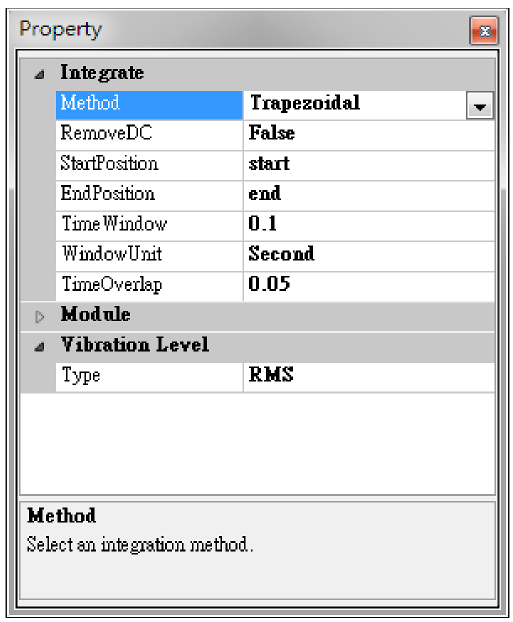
Root Mean Square (abbreviated RMS or rms), is a statistical measure of the magnitude of a varying quantity or a measurement of energy of an input signal.

Introduction

A signal can be expressed as $X = \{x_0, x_1, \cdots, x_{N-1}\}$, and the RMS is given by the formula below

$$x_{RMS} = \sqrt{\frac{(x_0^2 + x_1^2 + \cdots + x_{N-1}^2)}{N}}$$

Properties



{Integrate } Property Name	Property Definition	Default Value
`Method`	Set the method for the numerical integral to either `Trapezoidal` or `Simpson`.	`Trapezoidal`
`RemoveDC`	Choose whether to remove the DC or not.	True
`StartPosition`	Enters the value of the start position of the input data	The original start time of the input data
`EndPosition`	Enters the value of the end of the input data	The original end time of the input data

TimeWindow	Set the value to decide the window size rolling time.	0.1
WindowUnit	Set the unit as second or sample, of the window and overlap.	second
TimeOverlap	Set the time overlap while rolling. **Note**: time overlap must be less than time window	0.05

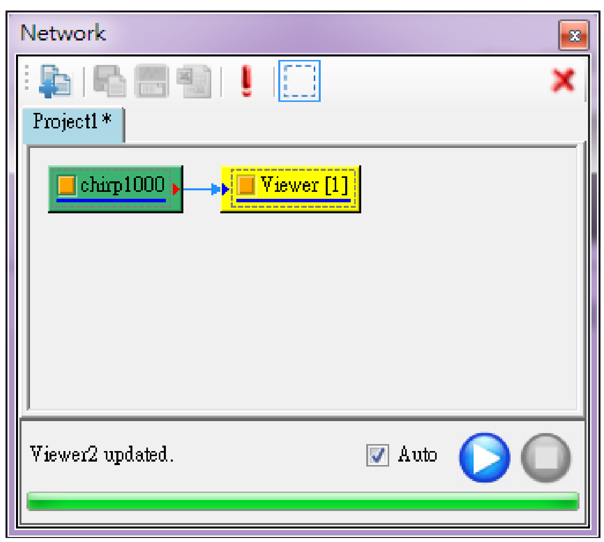
{Vibration Level} Property Name	Property Definition	Default Value
Type	Specify the vibration level by using *RMS*, *Peak*, or *PeaktoPeak*	*RMS*

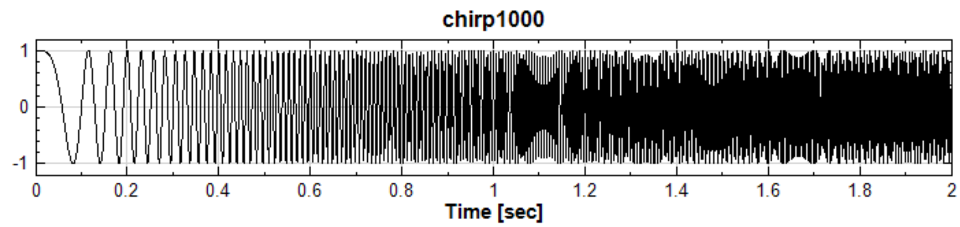
Variable Name	Property Definition
RMS	The average energy of the input signal within an interval
Peak	The Peak value of a sine wave is about $\sqrt{2}$ times the RMS value.
PeaktoPeak	The PeaktoPeak value of a sine wave is about 2.8 (2xPeak) times the RMS value.

Example

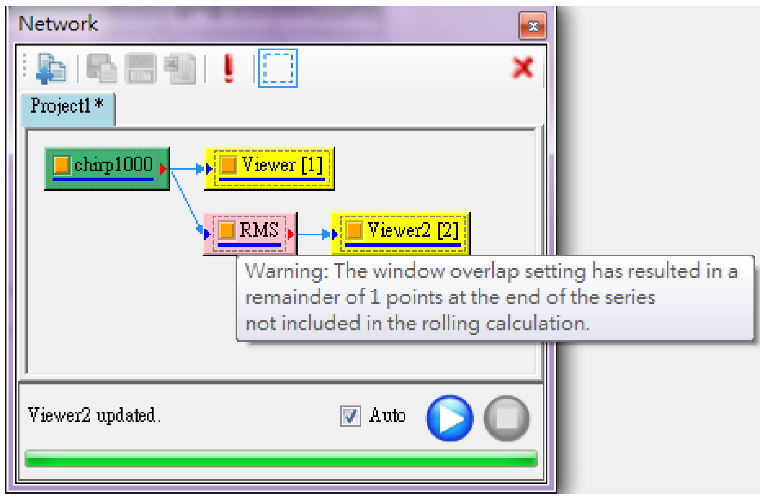
Open the file chirp1000.tfa in C:\Program Files\AnCAD\Visual Signal\demo\Basic and observe the differences between each vibration level.

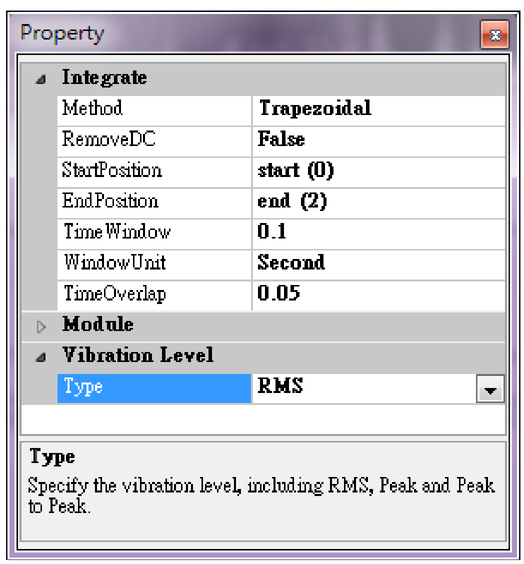
1. Open chirp1000.tfa in C:\Program Files\AnCAD\Visual Signal\demo\Basic using **`Source→Open Data`**. Then, connect chrip1000 component to a **`Viewer→Channel Viewer`**.

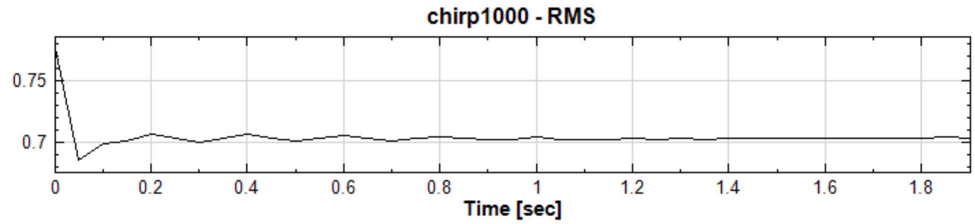




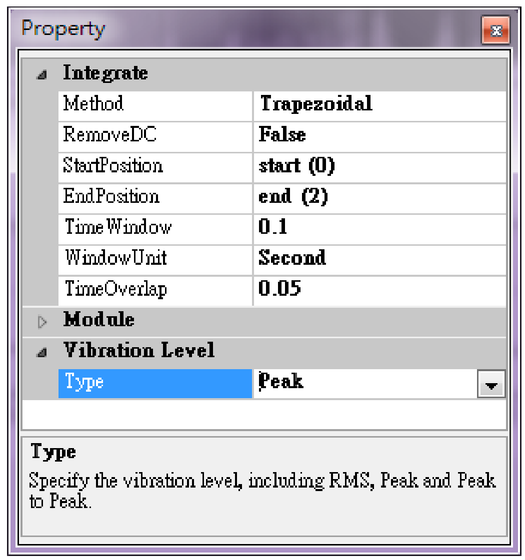
2. Connect the chrip1000 component to **Compute→Mathematics→RMS**. Use all the default properties of **RMS** component. Now, the **RMS** component will show a warning sign. The details of the message can be seen by moving the mouse cursor close the warning sign.

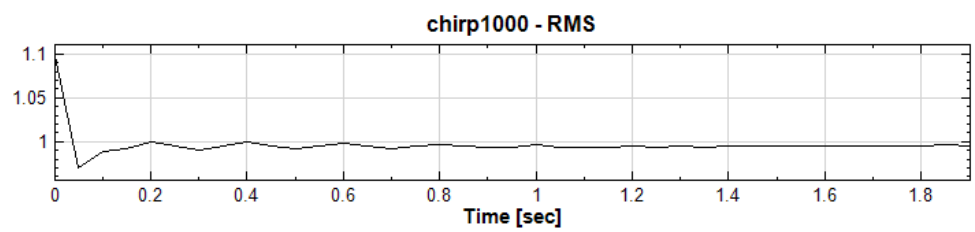




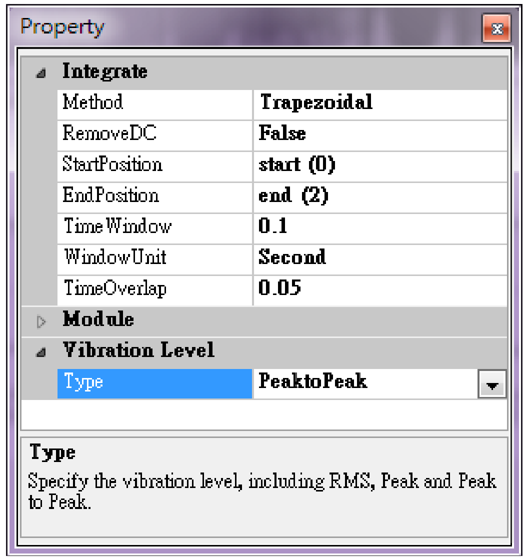


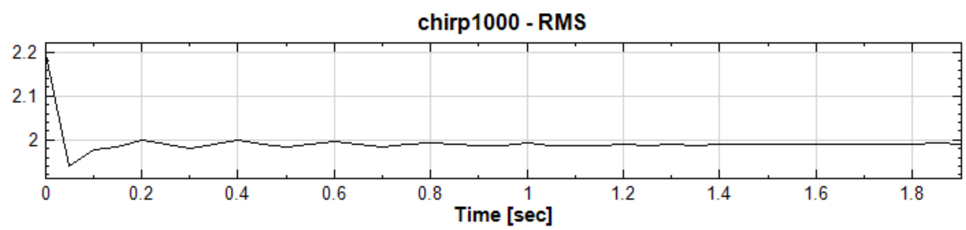
3. Set the `Type` of `Vibration Level` to `Peak`. The amplitude is about $\sqrt{2}$ times the value before setting *RMS* to `Peak`.



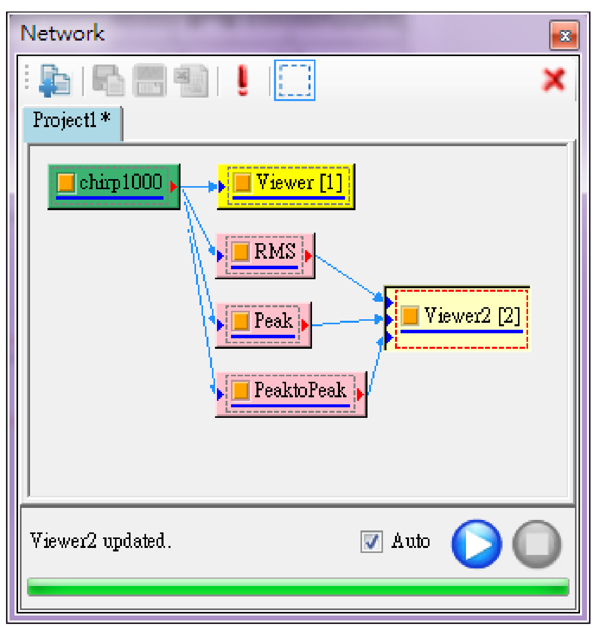


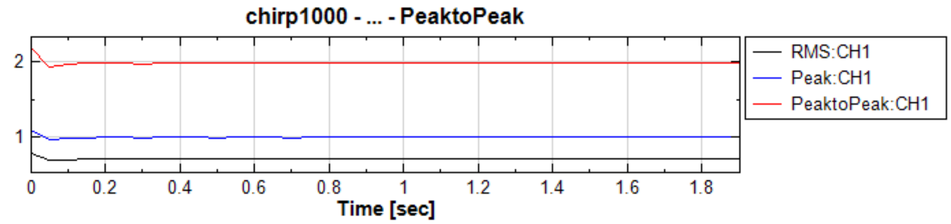
4. Follow step 3, but instead set `Peak` to `PeaktoPeak`. The amplitude is about 2 times the `Peak` value.





5. Let's create three **RMS** components and set the *Type* of *Vibration Level* property to *RMS*, *Peak*, and *PeaktoPeak* respectively. Connect all **RMS** components to the same **Channel Viewer**. The differences will then become more apparent.





Related Functions

Channel Viewer

4.1.4 Time-Frequency Analysis (TFA)

This module provides calculation of time-frequency analysis.

4.1.4.1 Short Term Fourier Transform

Short-Term Fourier Transform (STFT) is a mathematical transform related to Fourier Transform, which is used to calculate the instantaneous frequency, amplitude and phase of signals.

Introduction

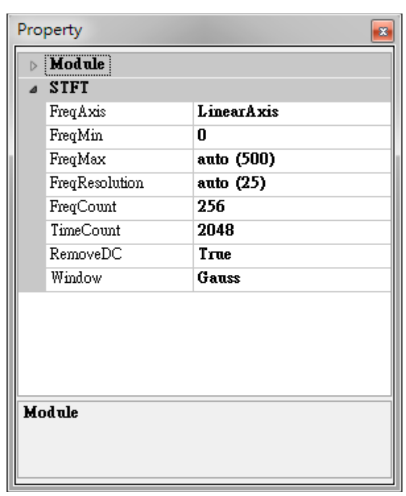
Use continuous-time function as an example, a function could multiply a time window function which is not zero, perform one-dimensional Fourier Transform, and then shift this window function along the time axis to get a series of Fourier Transform results which can be arranged to form a two-dimensional result. Mathematically, such an operation could be written as

$$\mathrm{STFT}[x(t)] \equiv X(\tau,\omega) = \int_{-\infty}^{\infty} x(t)\omega(t-\tau)e^{-i\omega t}\,dt$$

Where $\omega(t-\tau)$ is the window function, $x(t)$ is the signal to be transformed. Essentially, $X(t,\omega)$ is a complex function obtained by performing Fourier Transform on $x(t)\omega(t-\tau)$, which represents the amplitude and phase of the input signal in time and frequency space.

Properties

This module accepts input of Signal (which could be a real number, single channel, Regular) and Audio (which could be a real number, single channel, Regular). The output format is complex and signal-channel spectra data.

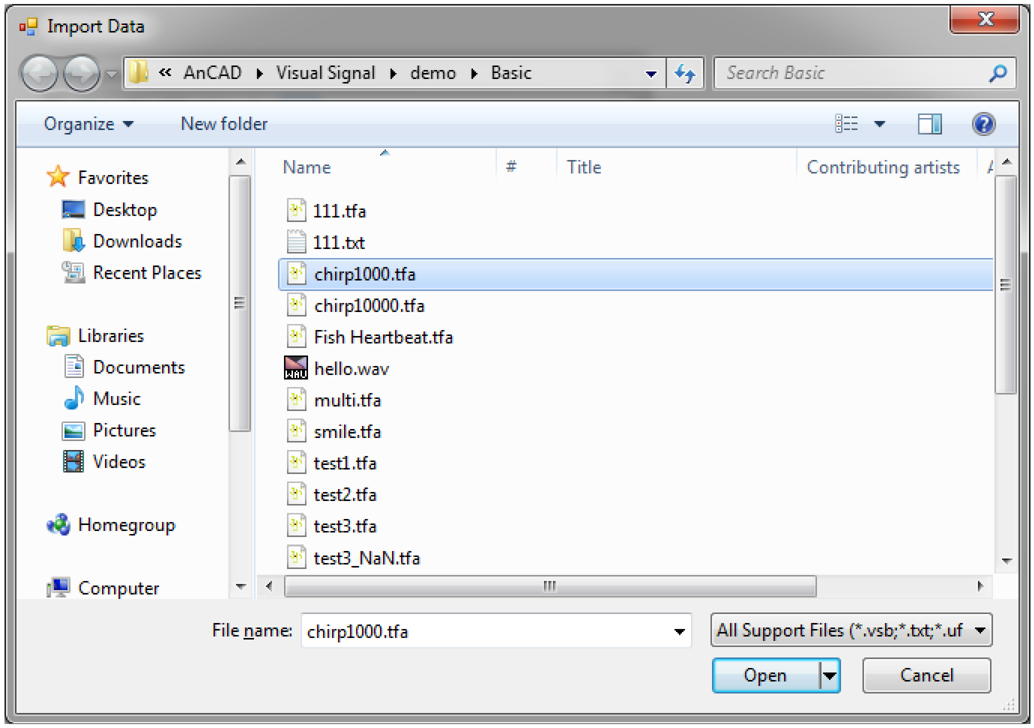


Property Name	Property Definition	Default Value
FreqAxis	The frequency axis could be LinearAxis (Linear measurement) or LogAxis (Logarithmic measurement). LogAxis are mostly used in audio analysis	LinearAxis
FreqMin; FreqMax	To define the frequency boundary for frequency plotting	0; 0.5*(Sample Frequency)
FreqResolution	To define the range of the window function. It would affect the size of the window function. The smaller this value, the smaller the window function	(Sample Frequency) / 40
FreqCount	The number of discrete lattice in frequency	256
TimeCount	The number of discrete lattice in time	2048
RemoveDC	Use to choose whether to remove the DC or not before STFT	True
Window	To select a different window function in STFT. For the definitions of window functions, please reference to Fourier Transform	Gaussian

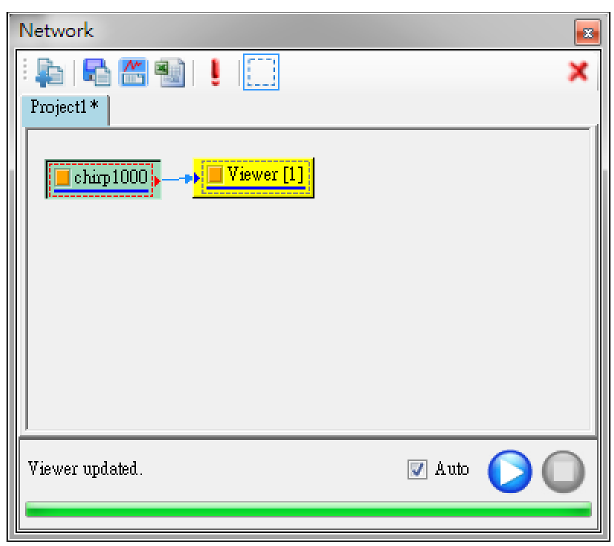
Example

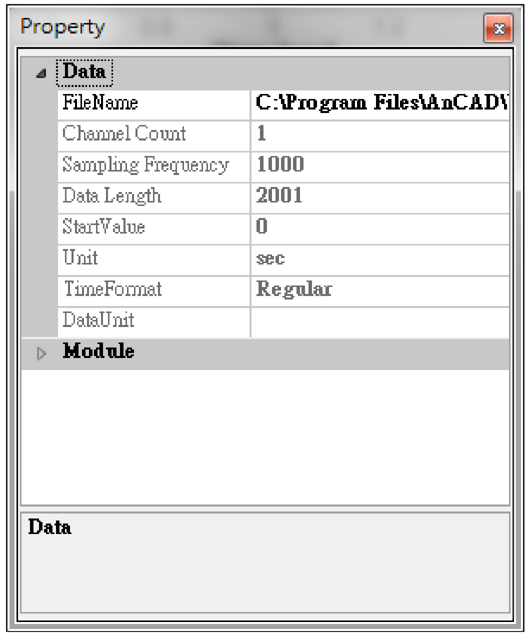
In the example below, use a Chirp signal as input, and then use Visual Signal to perform time-frequency analysis. It can be seen that a frequency which varies linearly with time.

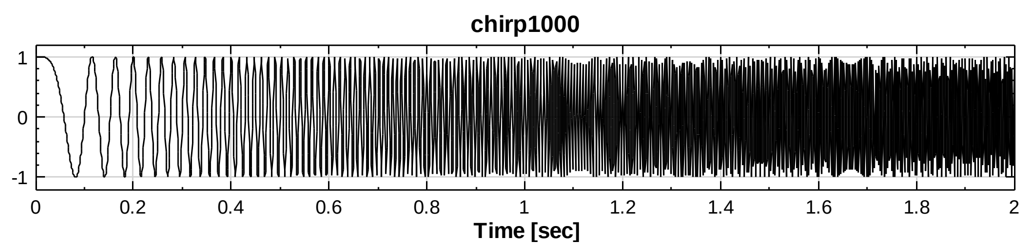
1. Press the  in the Network tools, or use **`Source→Import data`** from file to read a signal file, chirp1000.tfa, in the installation directory (the default directory is C:\Program Files\AnCAD\Visual Signal\data)
   Note: File locations will be different depending on platform (x86 or x64) or the installation path you selected.



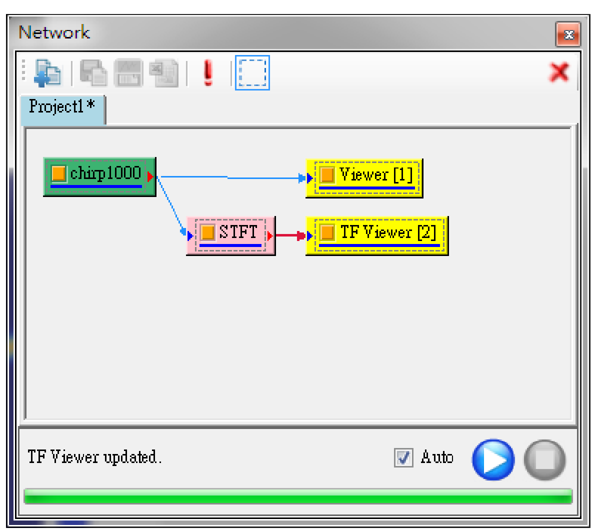
2. Click on the Chirp_1000 component, whose Properties show that the number of channels (`Channel Count`) is 1 and the `Sampling Frequency` is 1000Hz. Next, use **`Viewer`→`Channel Viewer`** to plot this signal. It can be seen that the signal frequency increases with the time increasing.

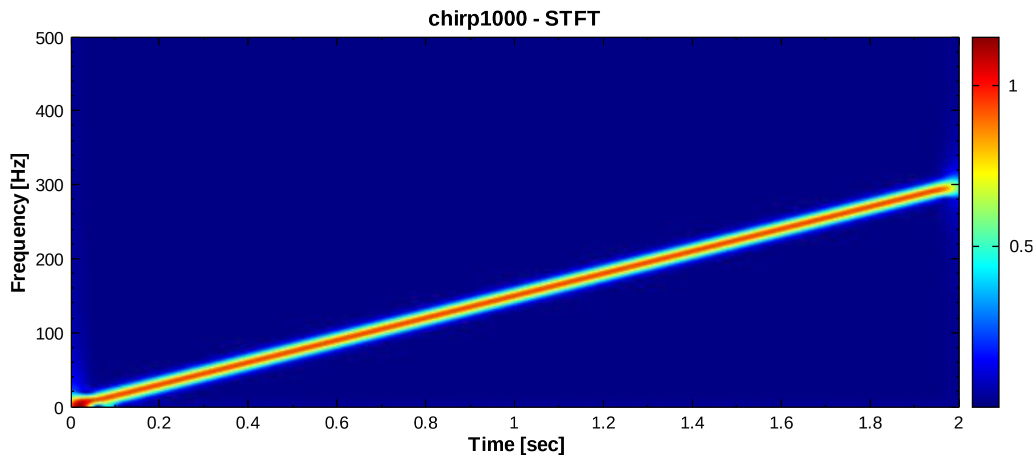




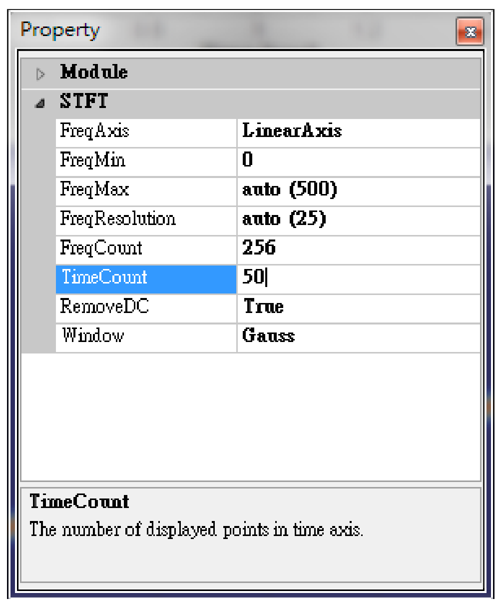


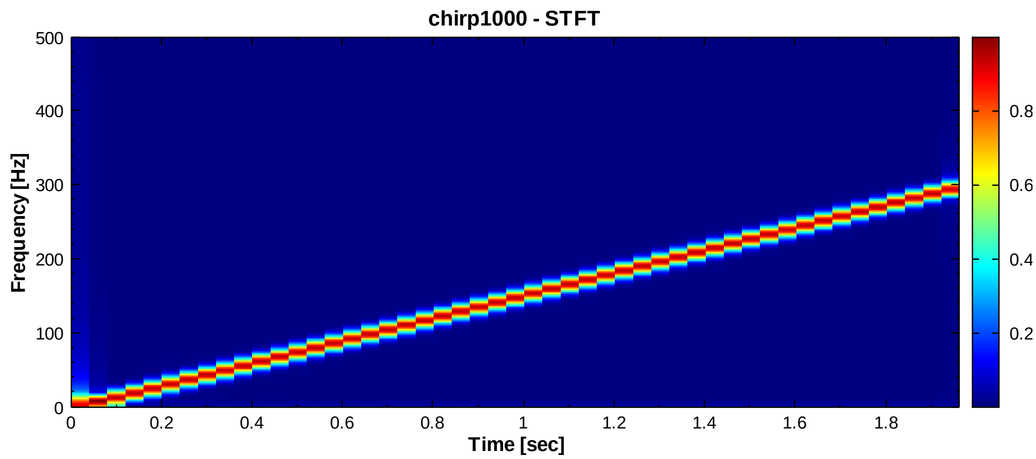
3. Select **Compute→TFA→Short Term Fourier Transform** to perform STFT on this signal and use **Viewer→Time Frequency Viewer** to plot the result. Observing the time-frequency diagram, it can be seen that the signal frequency varies lineally with time. From the result, the frequency at a given time point is available.



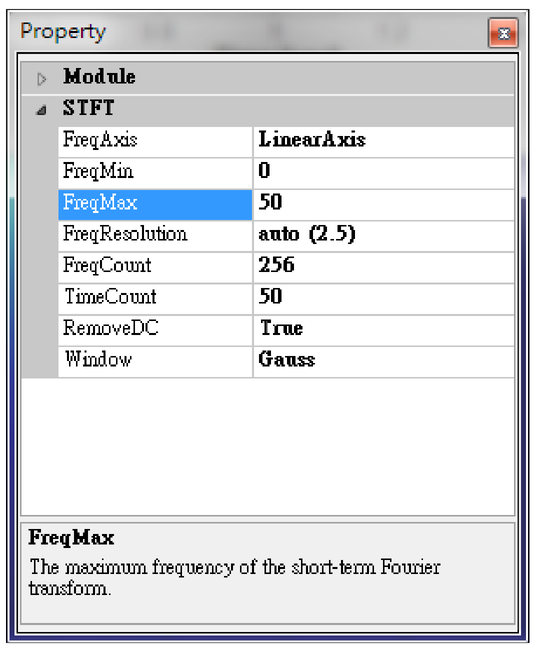


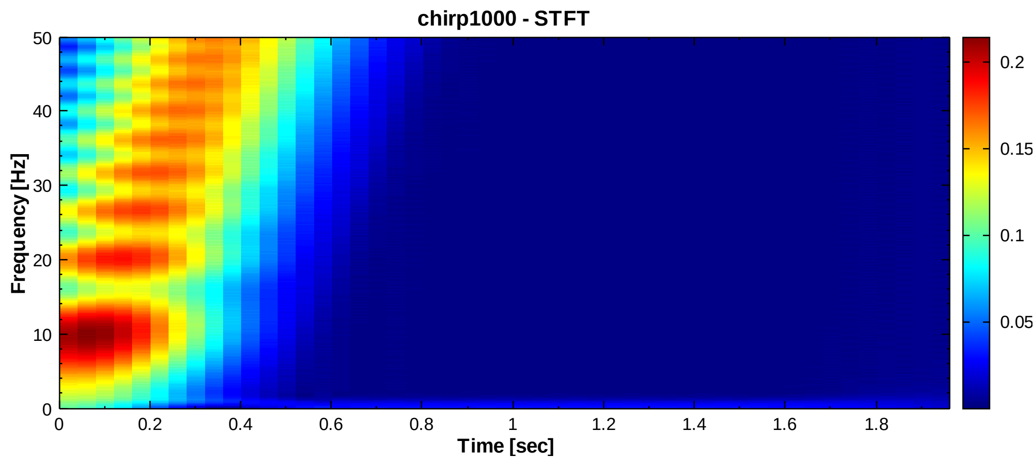
4. If the properties of `TimeCount` or `FreqCount` are changed, STFT would recalculate based on the re-set numbers of grids. Therefore, the result resolution and computation time would be affected. Change the `TimeCount` to 50, it can be seen that the computation runs faster while the resolution result becomes worse.





5. If the properties of *FreqMin* or *FreqMax* are changed, STFT would still calculate in the original frequency range. However, it will only output the result in the range defined by the properties and would not affect the computation time. Changing the *FreqMax* to 50 will not decrease the computation time.





Related Functions

`Fourier Transform`

Reference

A Wavelet Tour of Signal Processing (2nd Edition)

4.1.5 Transform

This module provides Fourier transforms for signal processing

4.1.5.1 Fourier Transform and Inverse Fourier Transform

Fourier Transform converts a time signal to a frequency signal for checking the frequency and amplitude distribution in the signal. The frequency signal could be converted back to time signal by Inverse Fourier Transform. This method is widely used in communication, voice signal, system analysis, and other scientific fields.

Introduction

Let $X = \{x_0, x_1, x_2, \cdots, x_{N-1}\}$be a N-length time signal, x_N be the n[th] signal, $0 \le \mathrm{n} \le \mathrm{N} - 1$, the discrete Fourier Transform of Signal X is defined as a N-length series,

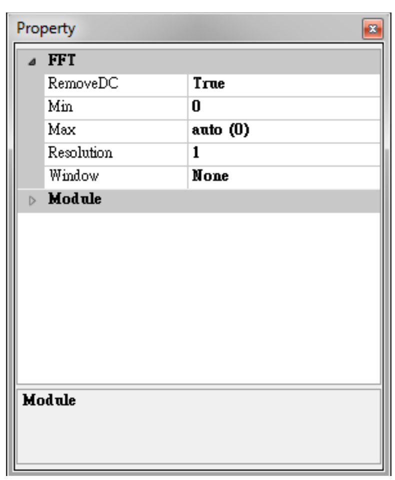
$$F(x_k) = \frac{1}{N}\sum_{n=0}^{N-1} x_n e^{-i\frac{2\pi}{N}kn}, \qquad 0 \le k \le N-1$$

The Inverse Fourier Transform is defined as follows,

$$x_n = N\sum_{k=0}^{N-1} X_k e^{i\frac{2\pi}{N}kn}, \qquad 0 \le k \le N-1$$

Properties

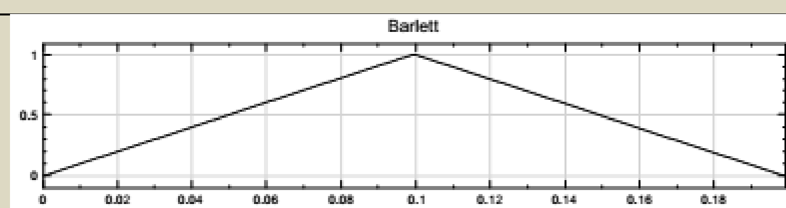
This module accepts input of Signal (which could be a real number, single channel or multi-channel, Regular) and Audio (which could be a real number, single channel or multi-channel, Regular). The definition of properties and corresponding setting are given below.

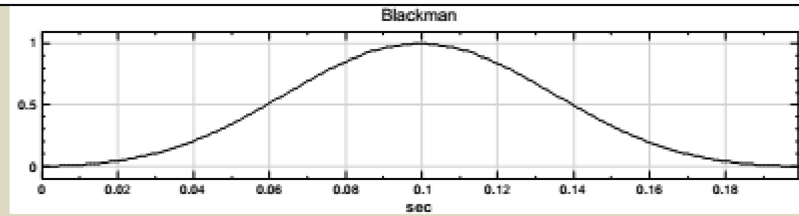
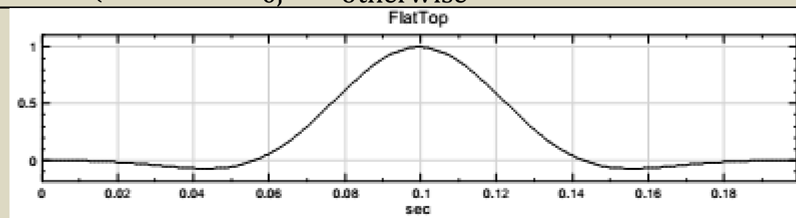
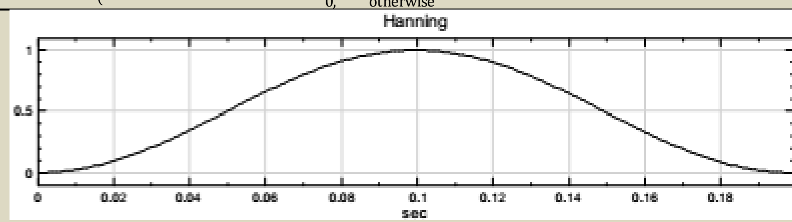
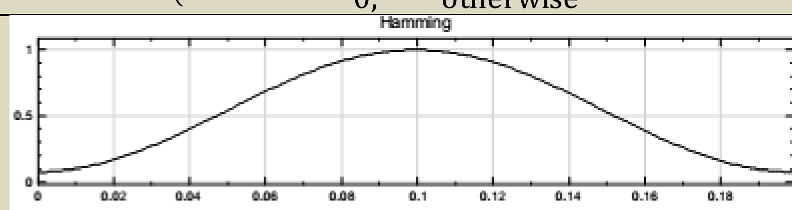
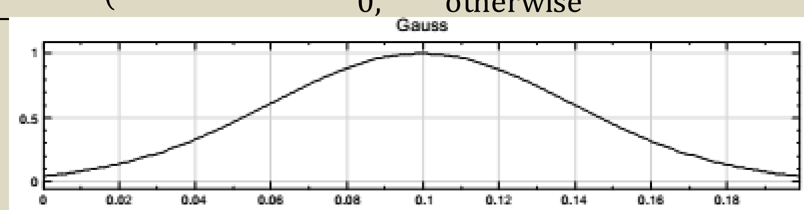


The property of RemoveDC is used to remove the average of the signal. The properties of Min and Max define the frequency range of FFT. The Property of Resolution is used to duplicate the signal to double the single length, k, of Fourier Transform, for better spectrum resolution. In the Window properties, there are 6 common window functions which can be used to smooth discrete signal and therefore remedy the numerical error caused by boundary effects.

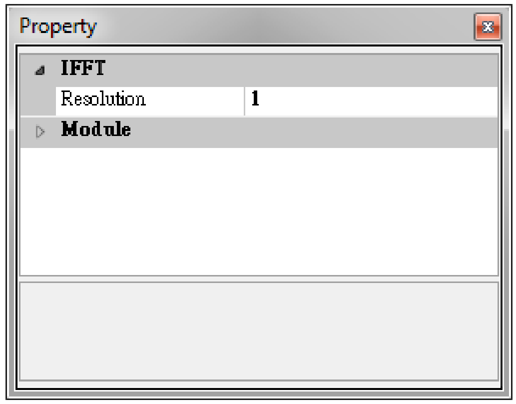
Property Name	Property Definition	Default Value
`RemoveDC`	To remove the shift along the y-axis, making the signal average zero	True
`Min`	To set the lower frequency boundary of the Fourier Transform	0
`Max`	To set the upper frequency boundary of the Fourier Transform, which varies based on the input signal	auto
`Resolution`	To adjust the Fourier Transform resolution. The approach is to multiply the input data point with the Resolution for increasing the transform resolution, then use Cherp Z Transform to obtain high-resolution Fourier Transform	1
`Window`	Use window function to reduce the leakage effect on the transform. The window functions include 6 types: Barlett, Blackman, Flat Top, Hanning, Hamming, and Gauss, whose definitions are given below.	none

Window Function

Window Function	Definition and Diagram
`Barlett`	 $w[n] = \begin{cases} \frac{2n}{N-1}, & 0 \le n \le \frac{N-1}{2} \\ 2 - \frac{2n}{N-1}, & \frac{N-1}{2} \le n \le N-1 \\ 0, & \text{otherwise} \end{cases}$

`Blackman`	 $w[n]=\begin{cases}\frac{1-\alpha}{2}-\frac{1}{2}\cos\left(\frac{2\pi}{N-1}n\right)+\frac{\alpha}{2}\cos\left(\frac{4\pi}{N-1}n\right), 0\le \mathrm{n}\le \mathrm{N}-1\\ 0, \quad \text{otherwise}\end{cases}$
`FlatTop`	 $w[n]=\begin{cases}1-1.93\cos\left(\frac{2\pi n}{N-1}\right)+1.29\cos\left(\frac{4\pi n}{N-1}\right)-0.388\cos\left(\frac{6\pi n}{N-1}\right)+0.032\cos\left(\frac{8\pi n}{N-1}\right), 0\le n\le N-1\\ 0, \quad \text{otherwise}\end{cases}$
`Hanning`	 $w[n]=\begin{cases}0.5-0.5\cos\left(\frac{2\pi n}{N-1}\right), & 0\le n\le N-1\\ 0, & \text{otherwise}\end{cases}$
`Hamming`	 $w[n]=\begin{cases}0.53836-0.46164\cos\left(\frac{2\pi n}{N-1}\right), & 0\le n\le N-1\\ 0, & \text{otherwise}\end{cases}$
`Gauss`	 $w[n]=e^{-\frac{1}{2}\left(\frac{n-\frac{N-1}{2}}{\sigma\frac{N-1}{2}}\right)^{2}}, \qquad \sigma\le 0.5$

The properties of Inverse Fourier Transform are given as follows.

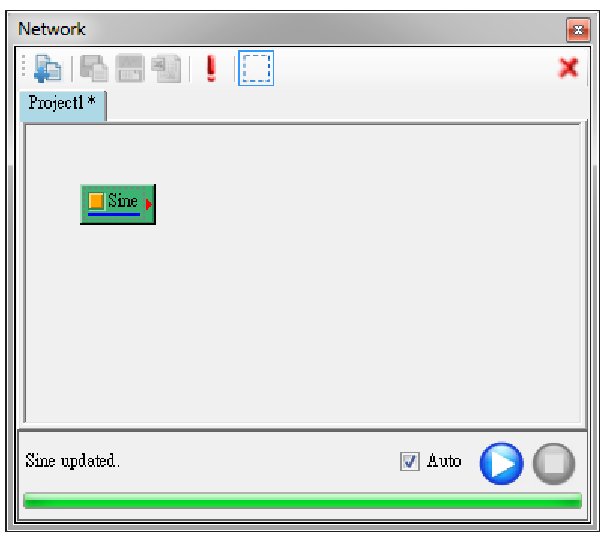


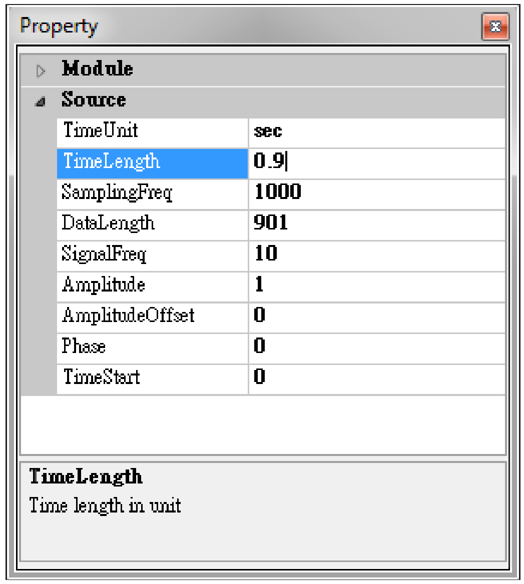
The property of Inverse Fourier Transform is Resolution, which has identical meaning as that in Fourier Transform. The number of signals in Inverse Fourier Transform would be twice as many as the Resolution value.

Example

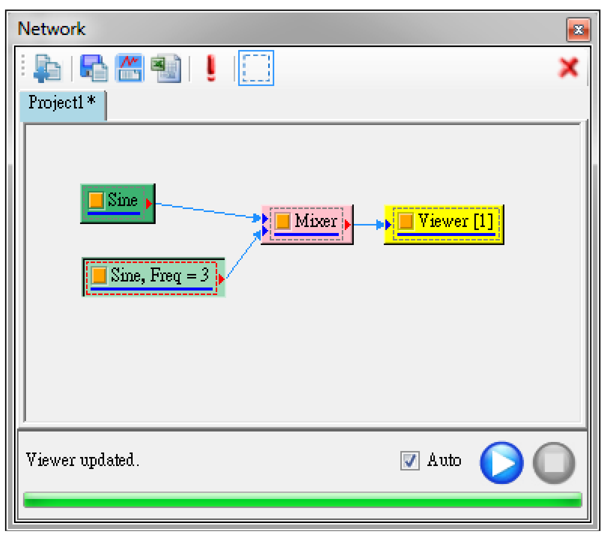
This example uses the Mixer function to generate a combined signal of two sine waves, and then perform Fourier Transformations to the new signal.

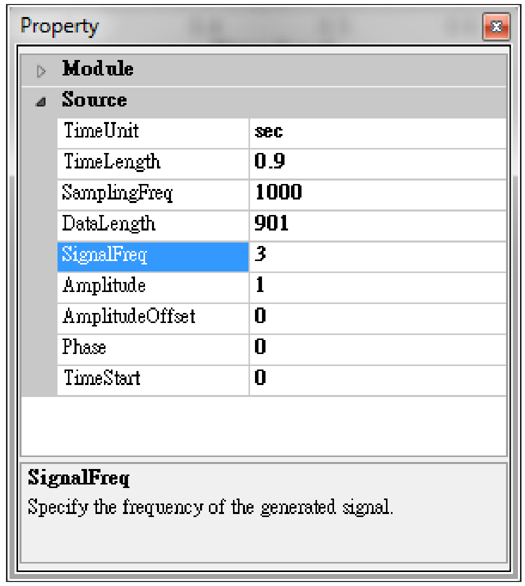
1. In the **Network Window**, use **`Source`→`Sine`** to create a sine wave. In the Properties window, change the Name filed to Sine, freq=10. The default value of Signal frequency is 10 Hz. Change *`TimeLength`* field to 0.9 sec.

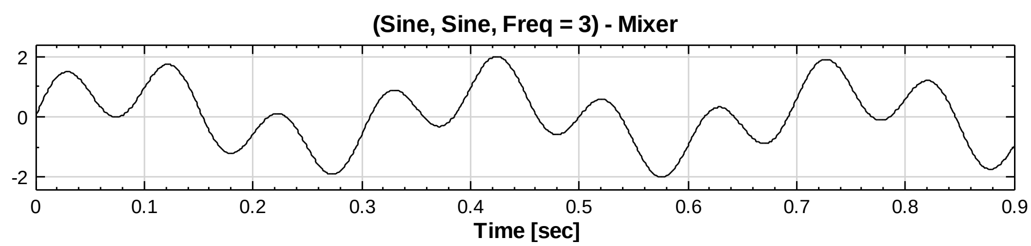




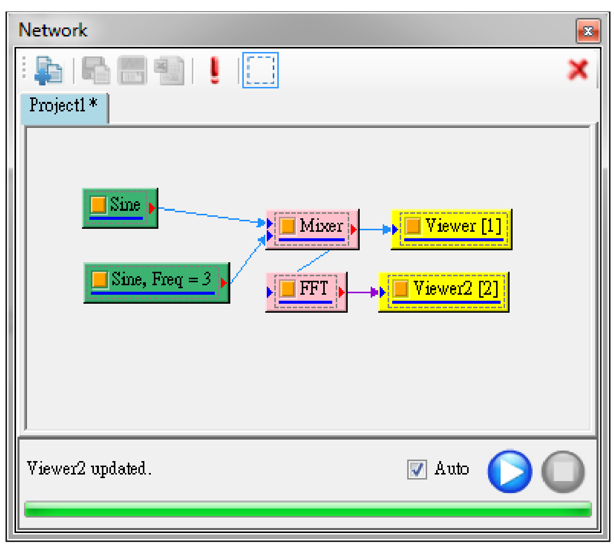
2. Create another sine wave and set the `Signal Frequency` to 3, and `TimeLength` to 0.9 second. Add **`Compute`**→**`Mathematics`**→**`Mixer`** to combine these two signals and use **`View`**→**`Channel Viewer`** to plot the output.

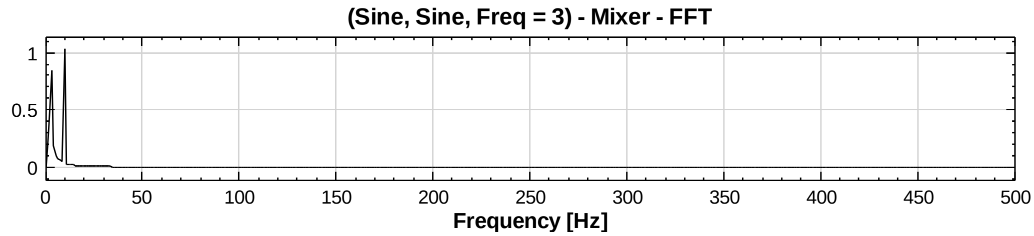




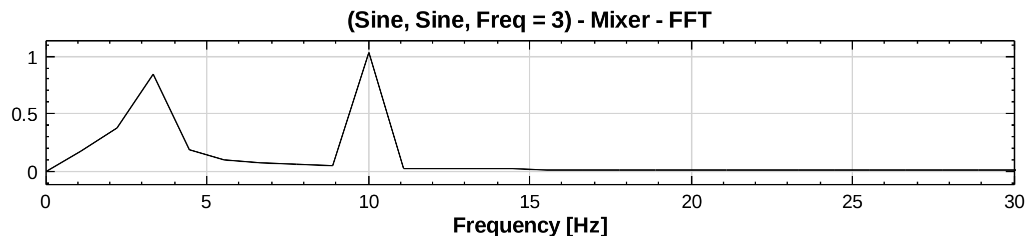


3. On the **Mixer** component, select **Compute** → **Transform** → **Fourier Transform** to perform FFT and then use **Channel Viewer** to plot the spectrum in the left window.



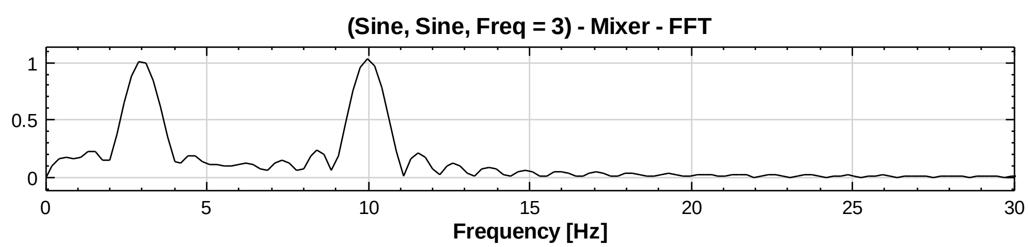


Because most frequencies are less than 20Hz and the default x-max is 500 in the properties of Viewer, set this field to 30 for better observation.

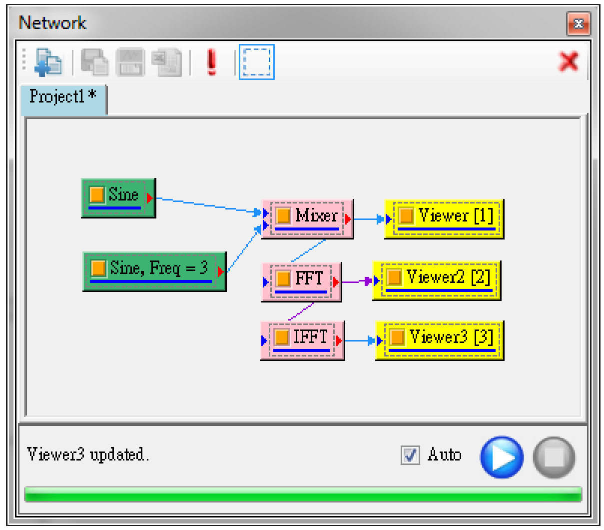


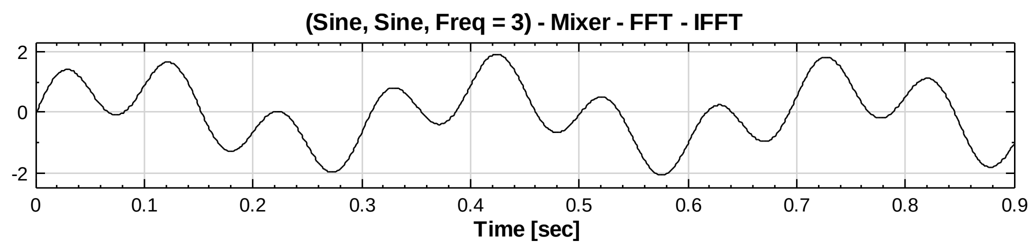
4. In the spectrum diagram generated by FFT, frequency components mainly concentrate at 10Hz and 3 Hz. However, the magnitude around 3Hz is underestimated due to the low frequency. This could be enhanced by changing the resolution. Click on the FFT icon, change the Properties/Resolution to 5 to obtain a new result.

   It is shown that the spectrum around 3 Hz has been improved significantly after changing the resolution. Note that increasing the resolution would result in multiplication of output data length in FFT. In this example, the input data length is 901 and the output length of FFT is 451 when the resolution is 1. After changing the resolution to 5, the output length would increase 5 times to 2255.



5. Change the FFT resolution back to 1, right click the FFT icon to select **`Compute`** → **`Transform`** → **`Inverse Fourier Transform`**, then use **`Channel Viewer`** to view the result. The result is the same as the original signal.





Related Functions

ShortTerm Fourier Transform

Reference

http://en.wikipedia.org/wiki/Fourier_transform

4.2 Format Conversion of Signal Flow Object

4.2.1 Convert from Spectra

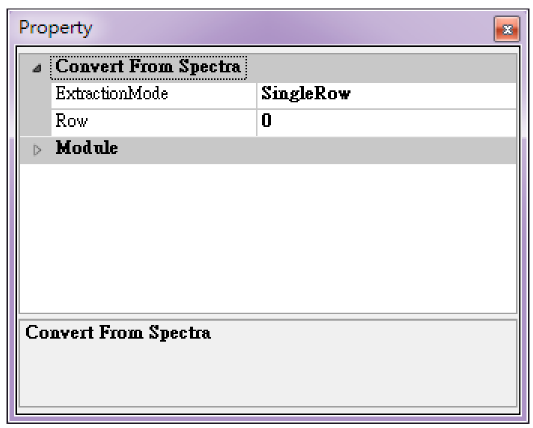
Convert Spectra data to single/multi-channel time series or single/multi-channel frequency distribution signal.

Introduction

We can extract one row or all rows from the Spectra, which is the amplitude time series at a fixed frequency. We can also extract one column or all columns from the Spectra, which is the frequency distribution at a fixed time point.

Properties

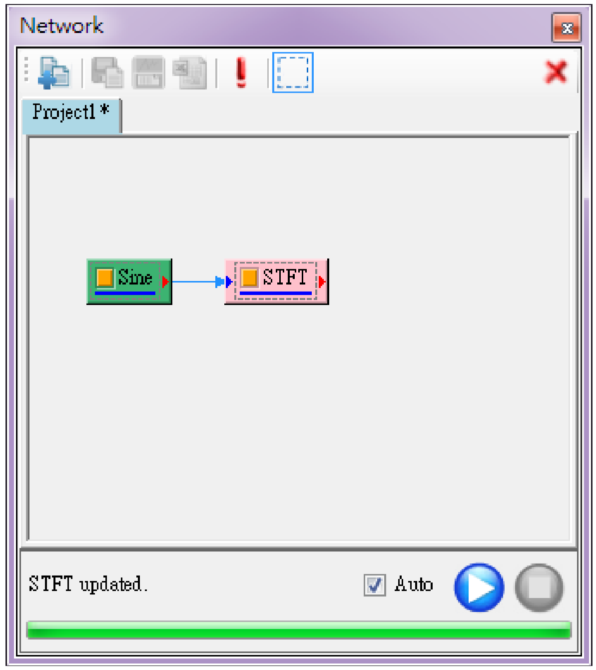
This module accepts spectra with real, complex, single channel, or multi-channel data.

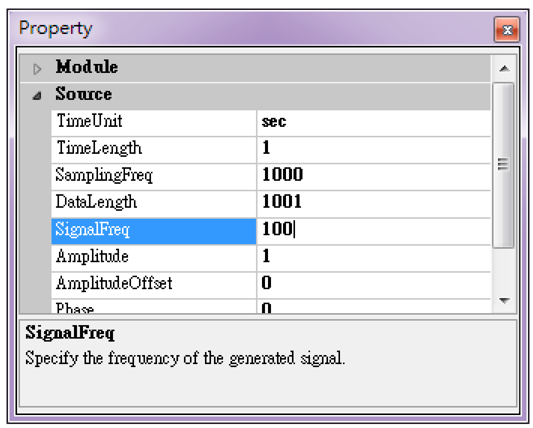


{Convert From Spectra} Property Name	Property Definition	Default Value
`ExtractionMode`	Extract Row or Column, options: `MultiChannelRows`, `MultiChannelColumns`, `SingleRow`, and `SingleColumn`	`SingleRow`
`Row`	Set which row to extract.	0
`Column`	Set which column to extract.	0

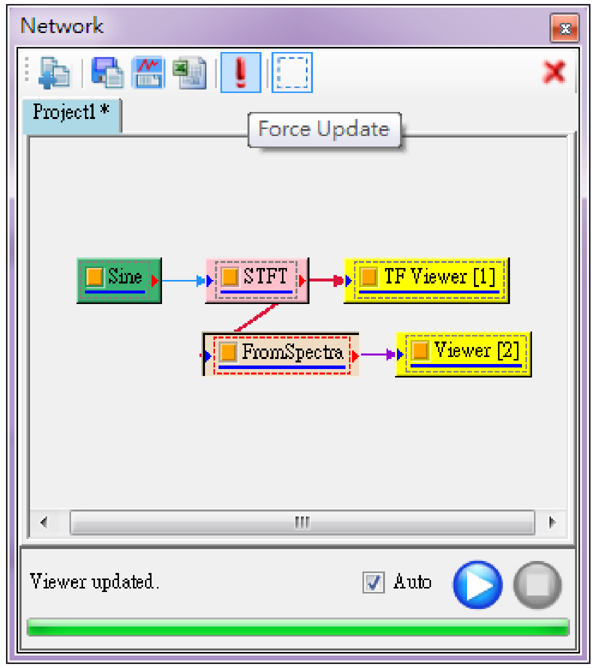
Example

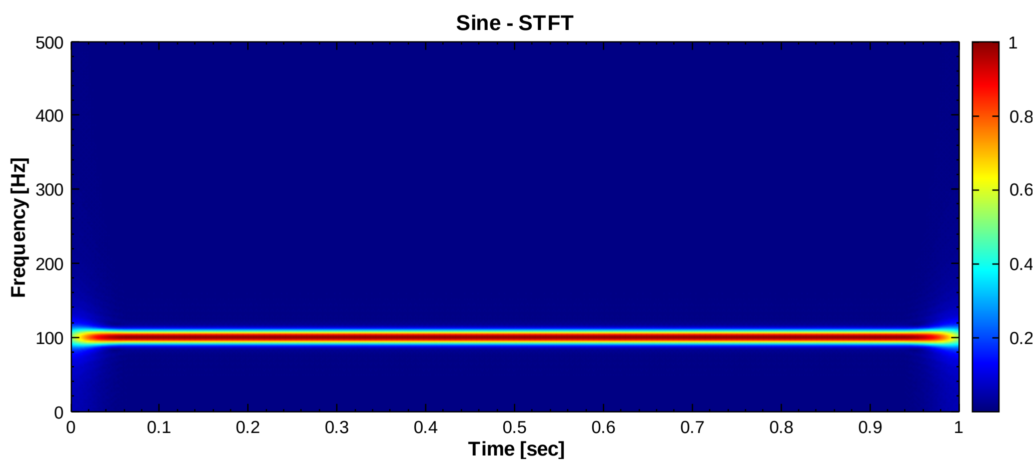
1. Use **Source→Sine Wave** to create an input signal and connect the signal to **Compute→TFA→ShortTerm Fourier Transform**. Use all the default property settings.

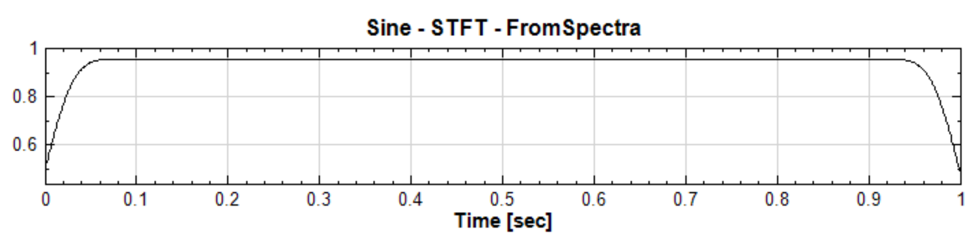




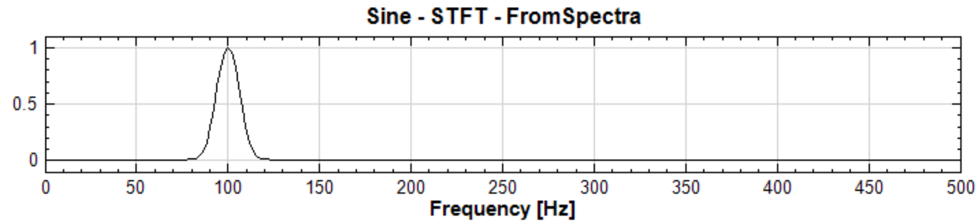
2. Connect **`STFT`** component to **`Conversion→Convert from Spectra`** and set *Row* to 50 in the **`Convert from Spectra`** properties, then display the 50th row data using **`Channel Viewer`**.







3. In **`Convert from Spectra`** properties, set *`ExtractionMode`* to *`SingleColumn`* and set *`Column`* to 100. Show the frequency distribution of the 100th column using **`Channel Viewer`**.



Related Functions

ShortTerm Fourier Transform

4.2.2 Map to Real

This function converts a Complex Signal to a Specific Real Signal.

Introduction

Let $X^{(c)} = \left\{x_0^{(c)}, x_1^{(c)}, \cdots, x_{N-1}^{(c)}\right\}$ and $Y^{(c)} = \left\{y_0^{(c)}, y_1^{(c)}, \cdots, y_{N-1}^{(c)}\right\}$ be the real part and the imaginary part of a complex signal, respectively; where c represents the c^{th} channel. The output signal $Z^{(c)}$ can be used to calculate real signals. There are 6 types as shown below.

$$z_j^{(c)} = \sqrt{\left(x_j^{(c)}\right)^2 + \left(y_j^{(c)}\right)^2}\, e^{i\theta} = Ae^{i\theta}$$

Magnitude: A

Phase: θ

Real Part: $x_j^{(c)}$

Imaginary Part: $y_j^{(c)}$

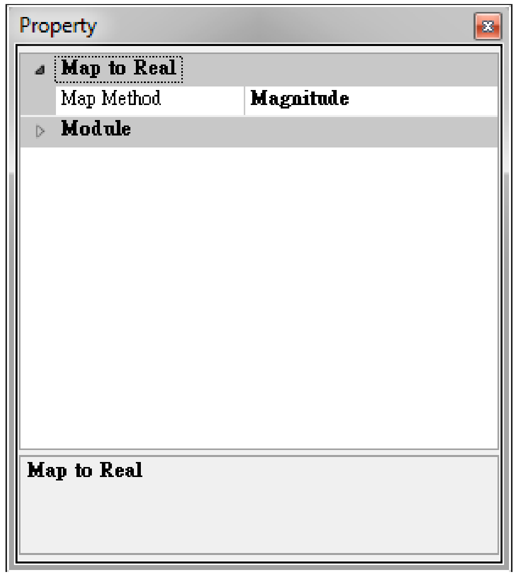
Gain: $20 \times \log_{10}\left(\frac{A}{Gain_{ref}}\right)$, $Gain_{ref}$ is the Gain reference

Power Spectrum: A^2

Properties

This module accepts input of Signal (which could be a complex number, single channel or multi-channel, Regular or Indexed), Audio (which could be a complex number, single channel or multi-channel, Regular), Numeric (which could be a complex number, single channel or multi-channel, Regular or Indexed) and Spectra (which could be a complex number, single channel or multi-channel, Regular). The output format is identical to the input signal except that it is a real number.

The property `Map Method` is how to convert the complex function, with a default value of Real Part, i.e., the real part of the input signal. Imaginary Part represents the imaginary part, Magnitude is the absolute value of the complex signal, Phase denotes the phase, Gain is used to set Gain Reference for Power Gain calculation, and Power Spectrum is the power of Magnitude.

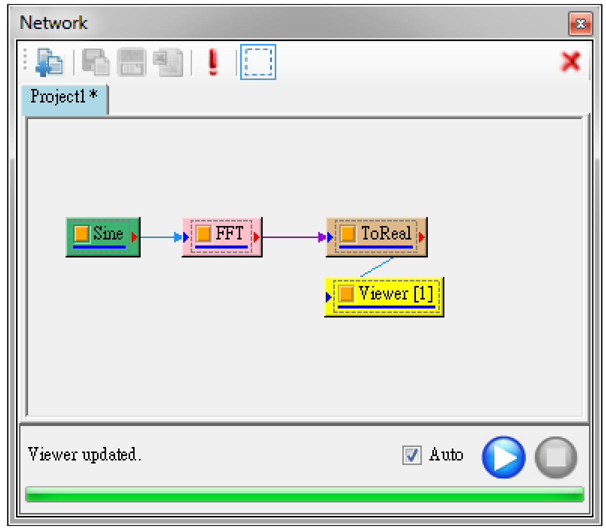


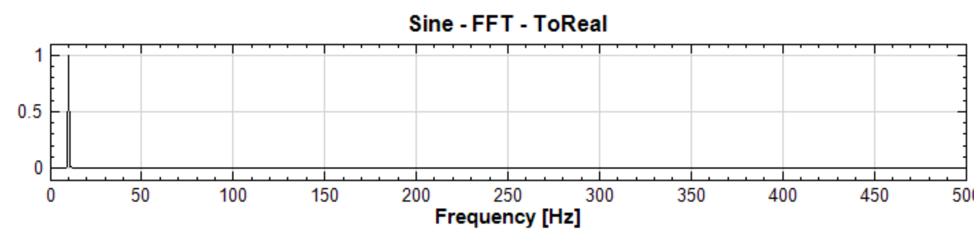
{Map to Real} Property Name	Property Definition	Default Value
MapMethod	Select the signal type which the complex signal should be converted to. The method options are *Magnitude*, *Phase*, *RealPart*, *ImagPart*, *Gain*, and *Powerspectrum*	*Magnitude*
Unwrap phase	Set True if phase is to be unwrapped. Only appear in *Phase* method	False
GainReference	Set the gain reference. Only appear in *Gain* method	1

Example

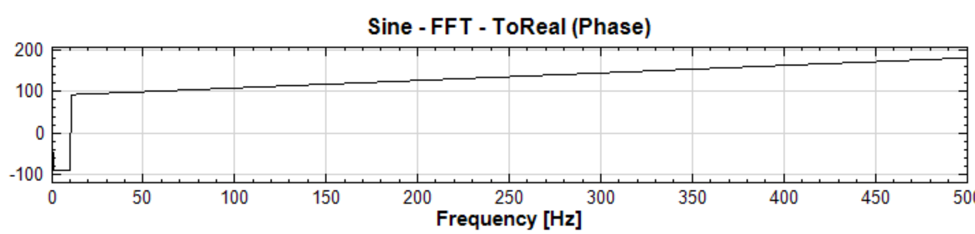
Map to real two sine waves with the sampling frequency of 1000Hz, length of 1 second and amplitude of 1 that are used as input signals.

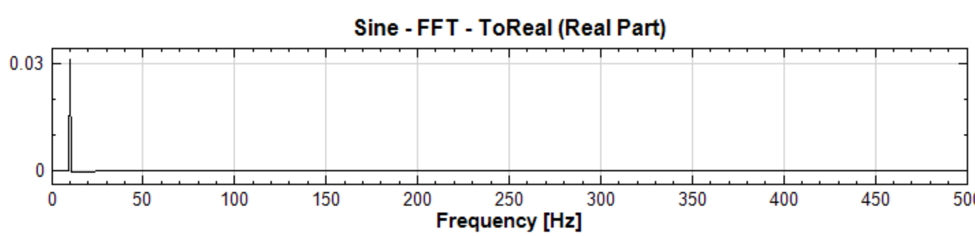
1. In the **Network Window**, use **Source→Sine Wave** to create a sine wave and perform a Fourier Transform using **Compute→Transform→Fourier Transform**. Then, use **Compute→Transform→Map To Real** to choose which kind of conversions of a complex signal to use and use **Viewer→ Channel Viewer** to plot the result.

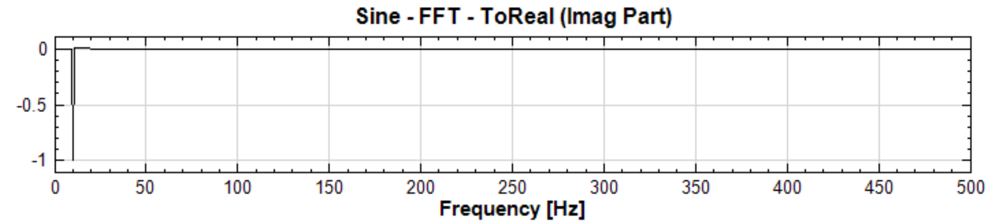


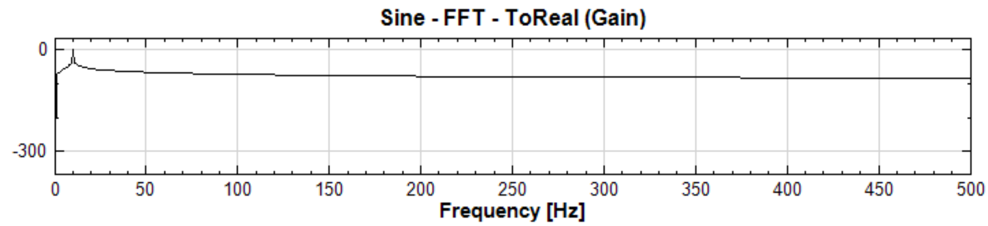


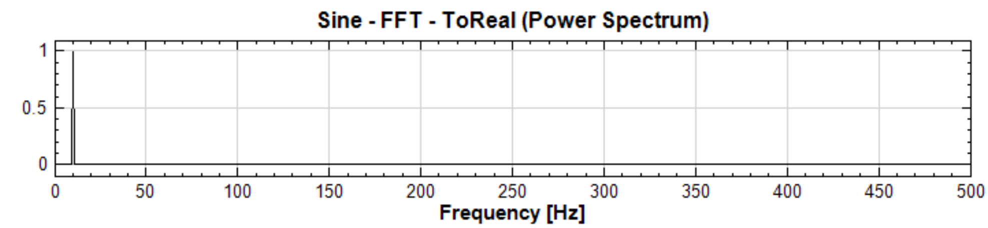
2. Change `Map Method` to each of the methods, and observe the differences between them.











Related Functions

`Source, Channel Viewer`

4.2.3 Merge to Multi-Channel

This component merges several single-channel signals into a multi-channel signal.

Introduction

Let x_j be a signal whose time-axis is j, y_k to be a signal whose time-axis is k, then the merged signals z_m is

$$z_m = [x_m, y_m]$$

$$\text{where} \begin{cases} m = j, \text{Reference Input} = x \\ m = k, \text{Reference Input} = y \end{cases}$$

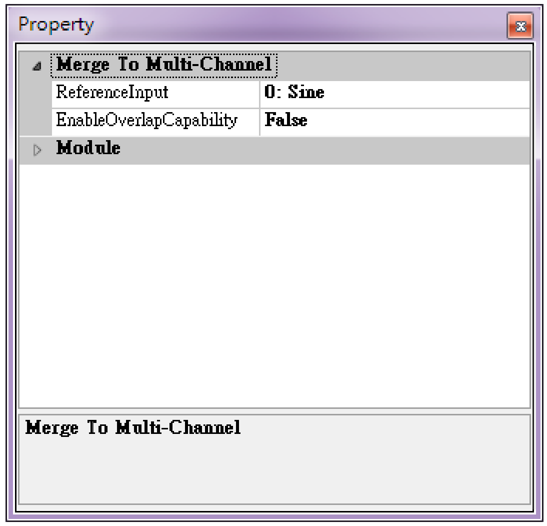
m represents the signal time-axis. If the `Reference Input` is set to x, the time-axis of the output signal z is identical to the one in x. i.e. $m = j$, and the time-axis of y would be replaced by the time-axis of x. Note that this module replaces the coordinates of the time-axis and more attention is needed for the length of input signals.

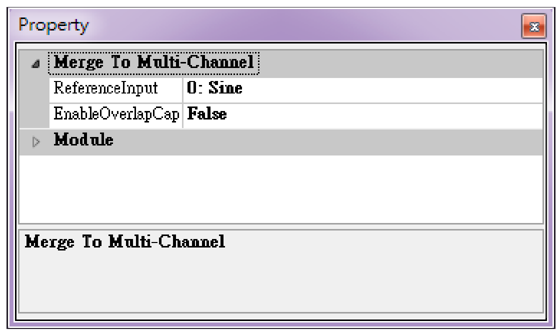
Properties

This module accepts input of Signal (which could be a real number or complex number, single channel, Regular or Indexed), Audio (which could be a real number or complex number, single channel, Regular), Numeric (which could be a real number or complex number, single channel, Regular or Indexed).

In this module, `Reference Input` selects an input signal used as reference of the number of channels and time-axis of the output signal. The default value is 0 which means that the output references the 1st input signal. The time-axis settings of other input signals would be copied directly from the the 1st signal. The principle of copying time-axis setting is that the time points of missing data are filled with 0 and time points of exceeding the time reference are discarded.

In order to avoid the operation confusion, it is recommended to use signals with identical settings, such as SamplingFreq, time starting point, and Time Length. Definitions and default values of properties are given below.





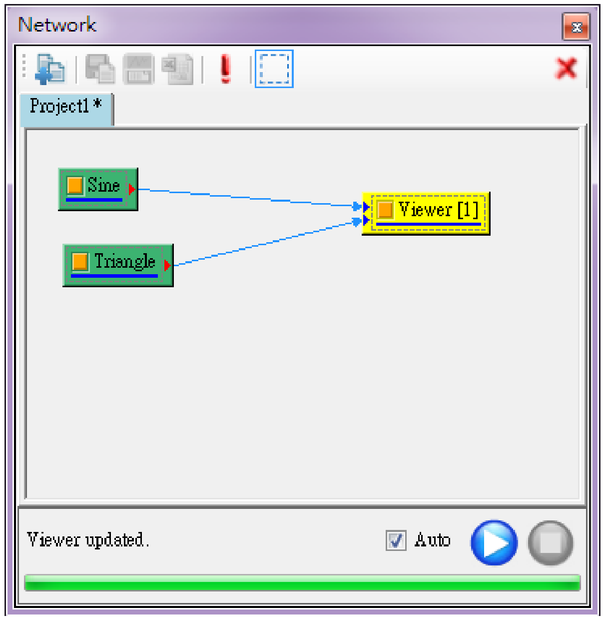
{Merge To Multi-Channel} Property Name	Property Definition	Default Value
`ReferenceInput`	To set the reference signal, its time axis is used as the time axis of the output signal	The 1st input signal
`EnableOverlapCapability`	Specify True to use AND or OR functions, or False not to use them. False uses the time axis of ReferenceInput for output and ignores different startTime in the inputs.	False

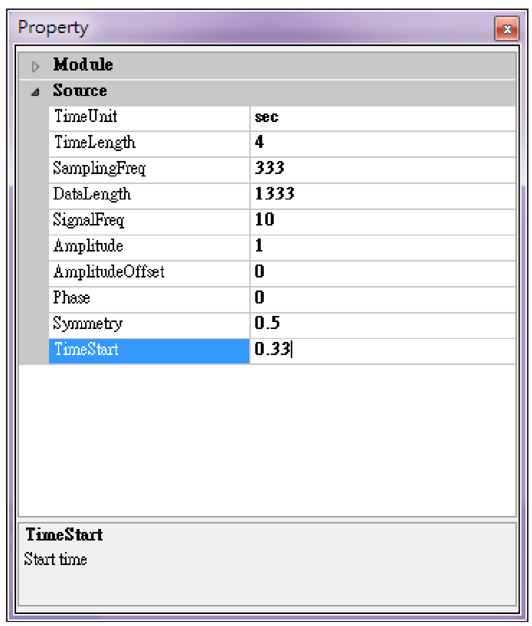
Example

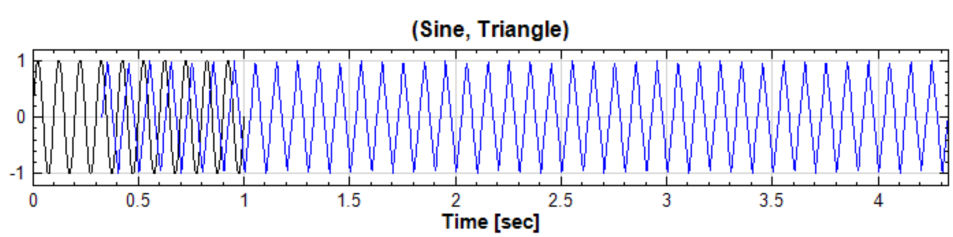
This module accepts Regular, Indexed input signals. The examples show the operation of input signals with identical and different time-axis settings.

1. Use **Source→Sine Wave** to generate a Sine Wave and use **Source→ Triangle Wave** to generate a triangular signal. Change the *SamplingFreq*

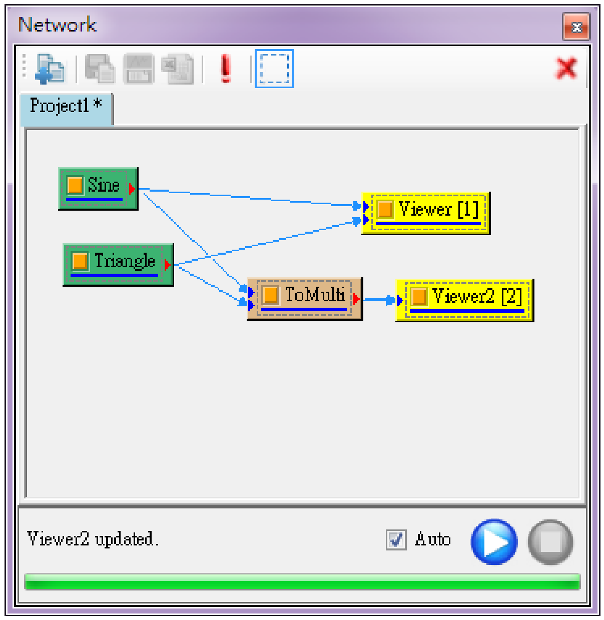
of Triangle to 333, `TimeStart` to 0.33, and `TimeLength` to 4 seconds. Finally, link these two signals to **`Viewer`→`Channel Viewer`** to plot figures.

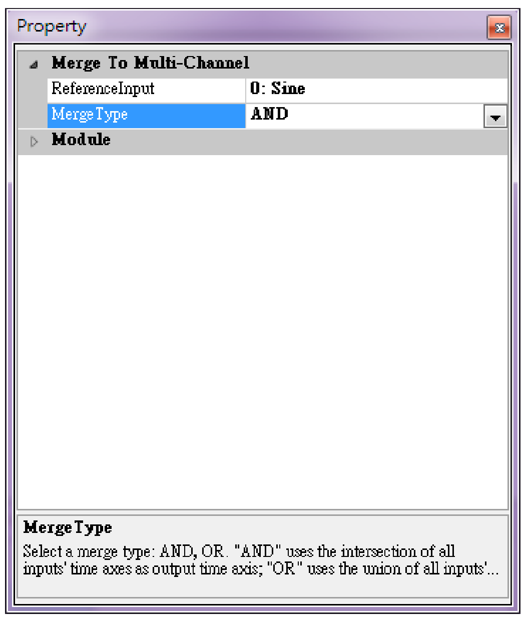


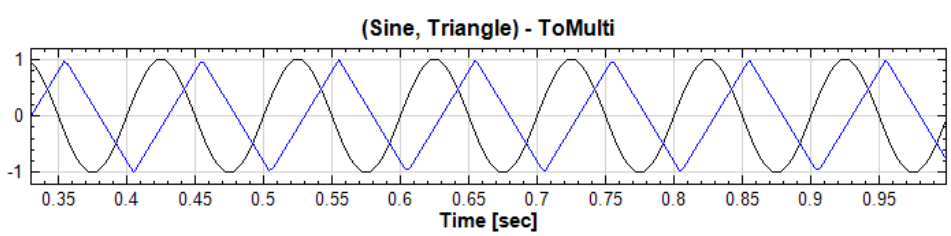




2. Connect these two signals to **Conversion→Merge to Multi-Channel** and use **Channel Viewer** to plot figures.



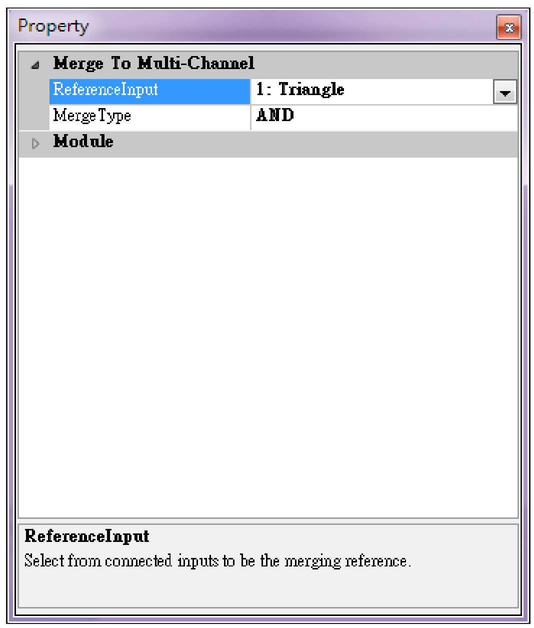


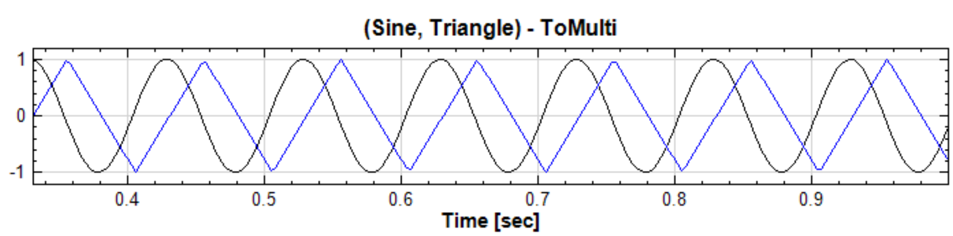


Click on the **ToMulti**. Because its *ReferenceInput* is set as Sine, the time-axis setting of the output is identical to those in the Sine Wave, i.e., the time starting point is 0, sampling frequency is 1000Hz, time length is 1 second, and the number of data point is 10001. The contents of Sine are copied to the CH1 in the **ToMulti** completely.

For the input signal of Triangle, the original time axis would be replaced by the time-axis of Sine. The number of data point in Sine is 1001 while it is 1333 in Triangle. This module would place the first 1001 data points of Triangle to the CH2 in the output signal and delete all other.

3. Change the *ReferenceInput* to 1:Triangle, the time-axis setting of the output is identical to those in Triangle. Because there are 1333 data points in Triangle while there are only 1001 data points in Sine, the CH1 of the output signal is filled with 0 in the corresponding points which have no data in Sine.

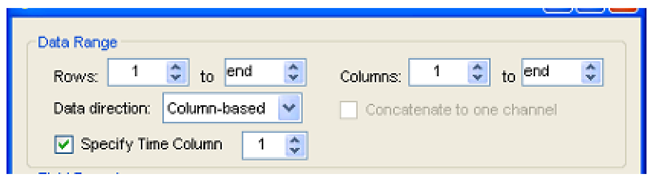




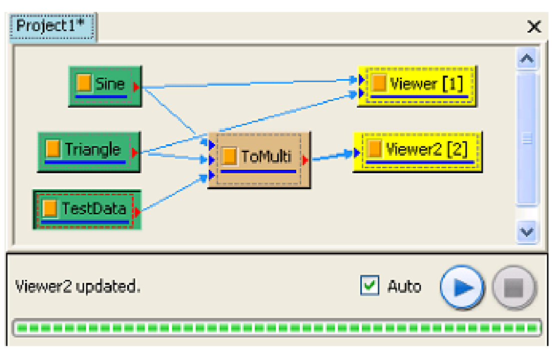
4. Next, read in a set of signals in Indexed format. First, create a simple data set as shown in the figure below, where the 1st column is time and the 2nd column is data.

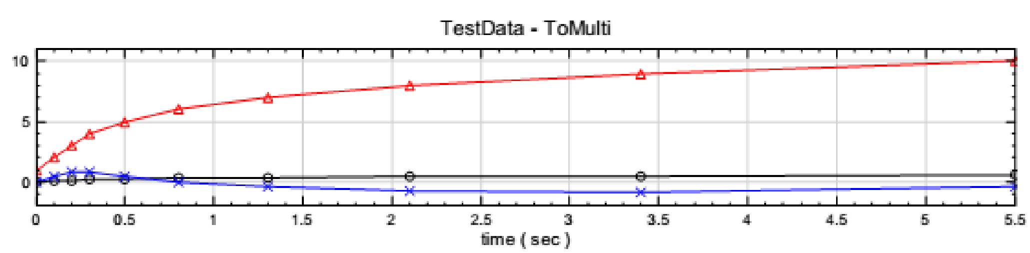
0	1
0.1	2
0.2	3
0.3	4
0.5	5
0.8	6
1.3	7
2.1	8
3.4	9
5.5	10

5. Next, press in the Network tools or use **Source→Open Data** from file to read in this data file, TestData.txt. In **Text Importer**, check **Specify Time Column** and then press the confirm button.



6. Not only does **ToMulti** accept signals in formats of Regular and Indexed, it also accept input signal of mixed Regular and Indexed. Drag **TestData** to **ToMulti** and change the *ReferenceInput* to 2:TestData, the original Regular format in the time-axis of Sine and Triangle signals is replaced by the time-axis of TestData. This is clearer when observing the output of **Channel Viewer**.





Related Functions

Noise, Sine, Channel Viewer

4.2.4 Convert to Audio

This component changes the type of signal data from Signal to Audio.

Introduction

The output data format of **`Convert to Audio`** follows the Microsoft Wave Format. The Microsoft Wave Format contains 3 data blocks: RIFF, FMT, and DATA. The details are given as follows.

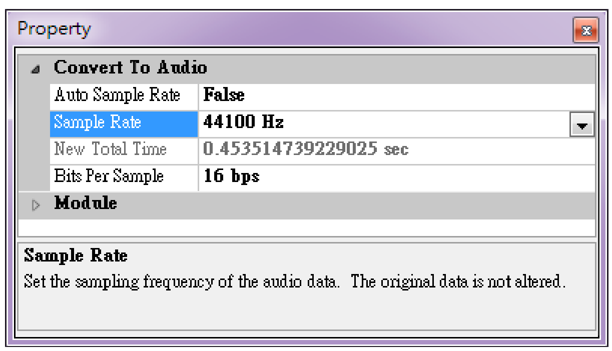
RIFF: defines the file format, file size and other information. The format is WAVE.

FMT: contains the related properties of audio signal such as code type, sampling frequency, number of audio channels, byte rate etc.

DATA: The original data which contains audio information.

Properties

Acceptable input data sources are: real, single channel or multi-channel, Regular signal or audio signal. Note that this component only accepts double channels for multi-channel. The output format is real, single channel or double channel, Regular audio signal. Available properties are **Sample rate** and **Bits per sample**.



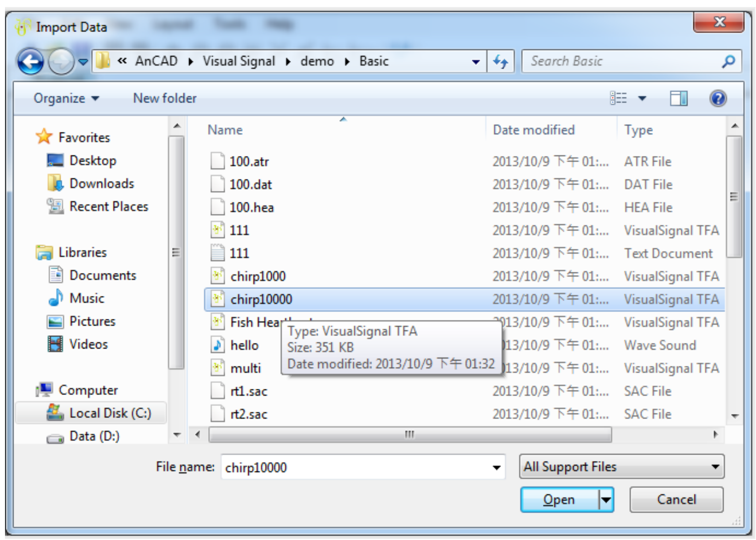
Property Name	Property Definition	Default Value
`Auto Sample Rate`	When set to true, the sample rate will be set automatically.	True
`Sample Rate`	The number of sampling points in every second. It affects the resolution of voice frequency. The available options are 1000,2000, 4000, 8000, 11025, 22050, 44100,48000, and 96000Hz.	44100
`New Total Time`	If the sample rate is changed, the new total time will be changed to	None

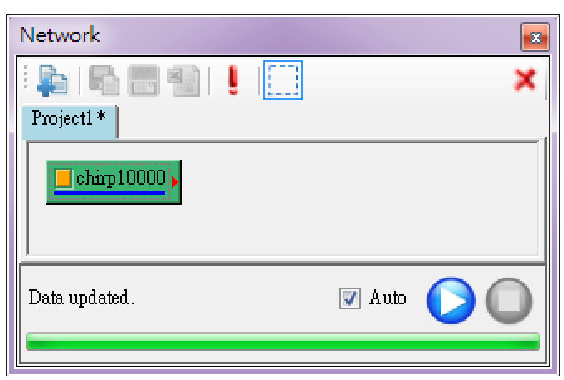
	the correct time corresponding to the new sample rate.	
`Bits Per Sample`	Define the value of every saved data which could affect the resolution of sound intensity. The available options are 8, 16, 24, and 32 bps.	16

Example

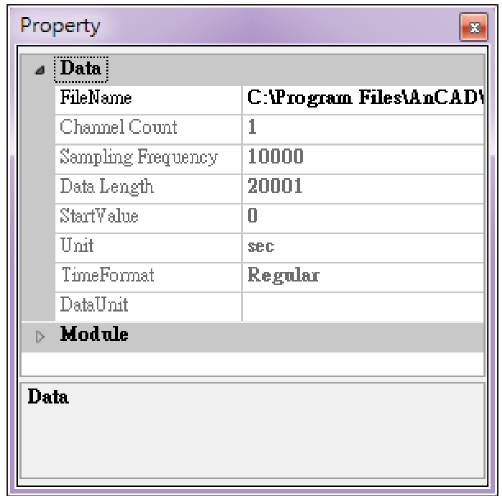
Convert the signal data file "Chirp1000.tfa" to audio signal using the **`Convert To Audio`** function.

1. Press in the Network tools, or use **`Source→Import data from File`** to read the signal file, chirp1000.tfa, in the data folder of installation directory which has a default location C:\Program Files\AnCAD\Visual Signal\demo\Basic.
   Note: File locations will be different depending on platform (x86 or x64) or the installation path you selected.

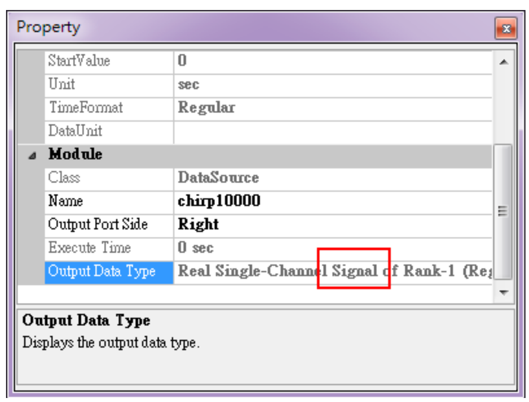




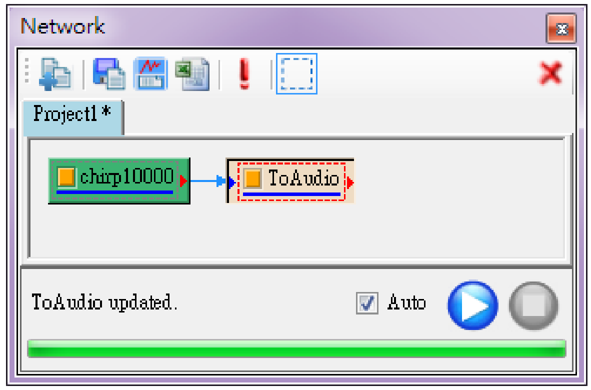
In Properties, it shows that the signal has the `SamplingFrequency` of 10000, the `DataLength` of 20001 and the `Unit` is seconds.



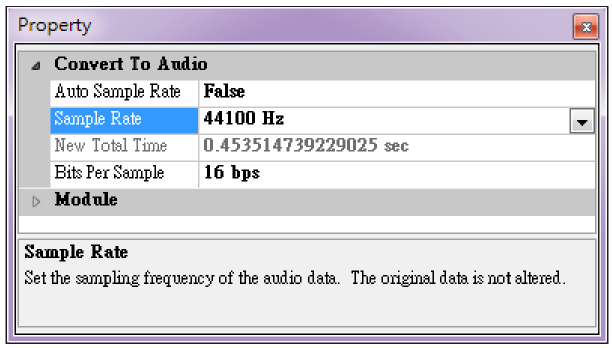
In addition, open the Module type in the Properties, where the `OutputDataType` shows the signal format and type of this module output. Since the `OutputDataType` is Real Single-Channel Signal of Rank-1(Regular) Data, the data type of Chirp10000 is a signal. Refer to the introduction of Properties in Chapter one for more details.



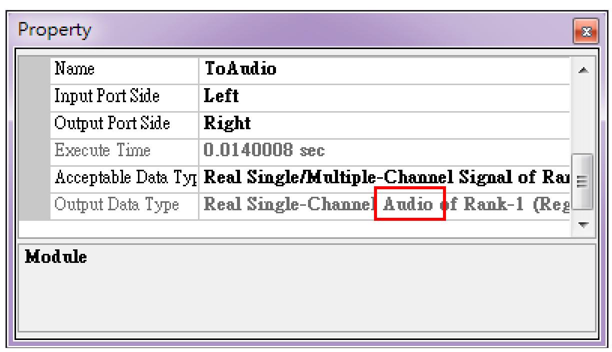
2. From Chirp10000 component, select **`Conversion`→`Convert To Audio`** directly.



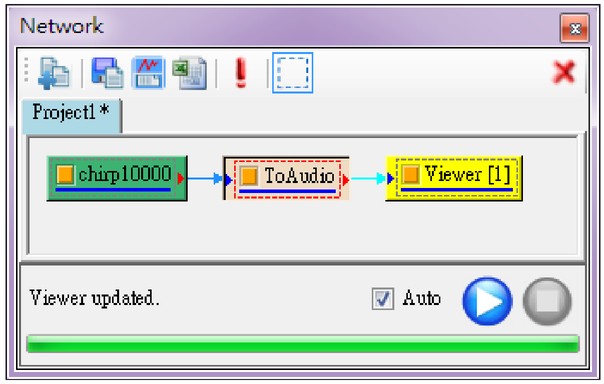
In Properties, it can be seen that the `Sample Rate` = 441000Hz and `Bits Per Sample` = 16 bps. These two properties can be changed in the drop-down menu.

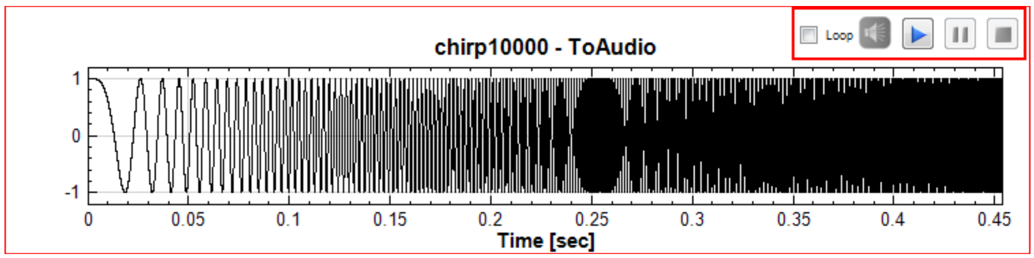


Then check Module in Properties and check to see if the `OutputDataType` has been changed to Audio.



3. **`Connect Viewer→Channel Viewer`** to the **`ToAudio`** component, the tool at the top right of the Viewer could be used to play this audio signal.





Related Functions

Channel Viewer

References

Microsoft Wave Format:
https://ccrma.stanford.edu/courses/422/projects/WaveFormat/

4.2.5 Convert to Regular

This function changes the time-axis setting of a signal from Indexed data to Regular data.

Introduction

When reading text files such as .txt and .csv etc. with **Import data from file**, if a row or a column in the data contains time-axis coordinates, it is recommended to use the `Specify Time column/row` in the **Text Importer**. As such, the data format is marked as Indexed and the sampling periods are assumed to be different. However, most modules require the signal format to be Regular. This module can be used to convert Indexed signals to Regular signals.

In Indexed format, it is assumed that the data points on time-axis are discrete and the intervals are uneven. Therefore, there exists a corresponding time coordinate for every data point. Let the input signal be $X^{(1)} = \left\{x_0^{(1)}, x_1^{(1)}, \cdots, x_{N-1}^{(1)}\right\}$ and the signal time-axis is defined as,

$$T^{(1)} = \left\{t_0^{(1)}, t_1^{(1)}, \cdots, t_{N-1}^{(1)}\right\}$$

`Convert to Regular` performs re-sampling on the signal above to convert the time-axis of Indexed signal into Regular which is discrete and equidistant.

$$t_j^{(0)} = t_0^{(0)} + j \times \Delta t^{(0)}, \qquad 1 \le j \le M-1$$

Where $t_j^{(0)}$ is the time-axis after **`Convert to Regular`** processing, $\Delta t^{(0)}$ is the output sampling period, M is the number of the output data points. The output signal $X^{(0)}$ is obtained by the formula below.

$$X^{(0)} = \left\{x_0^{(0)}, x_1^{(0)}, \cdots, x_{M-1}^{(0)}\right\}$$

Two types of calculations, FillGap and RemoveGap, can be used to convert Indexed data to Regular data. The details are given below.

FillGap:

FillGap can preserve the signal time characteristics and add values at the locations where the time intervals are too large. In calculation, it can detect the minimum sample period, $\Delta t_{min}^{(1)}$, in the input signals, and then use it to perform re-sampling on the signals, $\Delta t^{(0)} = \Delta t_{min}^{(1)}$. The re-sampling methods are the same as the ones in the Resample component with 7 methods. Users can also set sampling period manually.

However some constraints may apply. For consistency of input and output signals, the sampling period $\Delta t^{(0)}$ must be less than or equal to 1.5 times of $\Delta t_{min}^{(1)}$. The computation logics of re-sampling are briefly introduced as follows.

If $t_{i+1}^{(1)} - t_i^{(1)} > 1.5 \times \mathrm{Min}\left(\Delta t^{(0)}\right)$, the following methods are used to calculate the filling data, $x_j^{(0)}$, between $x_j^{(1)}$ and $x_{j+1}^{(1)}$.

> `Fix`: Use fixed value as filling-value
>
> `Prev`: Use value of the preceding point as filling-value.
>
> `Next`: Use value of the subsequent point as filling-value.
>
> `Linear Interpolation`: Use the preceding and subsequent points to perform Linear Interpolation.
>
> `Spline Interpolation`: Use Spline to fill values.
>
> `Monotonic Cubic Spline`: This is a e-degree interpolation with damping. It has better performance than Spline in the case of processing a signal with a large slope like square waves because it can avoid large vibrations.
>
> `No Fill`: No additional value is added, NaN.

If $t_{i+1}^{(1)} - t_i^{(1)} > 1.5 \times \mathrm{Min}\left(\Delta t^{(0)}\right)$, the value which is corresponding to the input signal of $x_{j+1}^{(1)}$ would be output directly to the corresponding position, $X^{(0)}$ in the output signal.

RemoveFillGap:

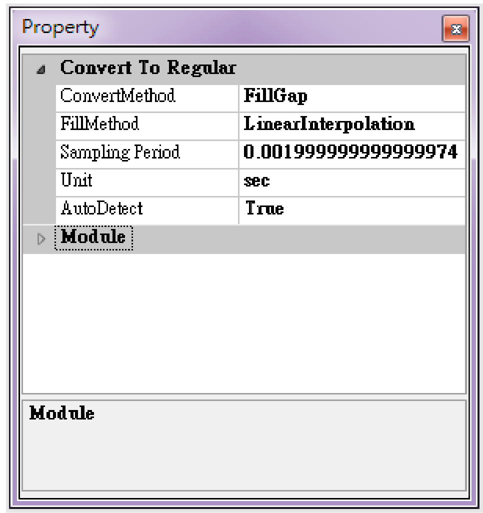
RemoveGap discards the time-axis $T^{(1)}$ of the input signal, uses the starting time $t_0^{(1)}$ and the minimum sampling period $\Delta t_{min}^{(1)}$ to re-calculate the time-axis $T^{(0)}$ of the output signal, and replaces the time-axis $T^{(1)}$ with $T^{(0)}$.
The formula for $T^{(0)}$ is

$$t_j^{(0)} = t_0^{(1)} + j \times \Delta t^{(0)}, \qquad 1 \leq j \leq N - 1$$

Therefore, the output signal from RemoveGap has the same number of data points as the input signal. However, uneven intervals in the input signal are changed to even intervals.

Properties

This module accepts input of Signal (which could be a real number or complex number, single channel or multi-channel, Indexed). The outputs are Regular signals which are real or complex, single channel or multiple channels.



Property of `AutoDetect` provides the option to set the `Sampling Period` of the output signal manually. If it is set as True, this module would detect the minimum sampling period of the input signals automatically and use it as the Sampling Period. If AutoDetect is set as False, user can set the Sampling Period manually. Because a large sampling period would cause discrepancies between the output signal and the original one, the manual setting of the Sampling period must be less than 1.5 times of the sampling period obtained by AutoDetect.

Property of ConvertMethod allows users to select FillGap or RemoveGap to calculate the time-axis of the input signal. If FillGap is selected, a new property of FillMethod is provided for value-filling methods, which are explained below.

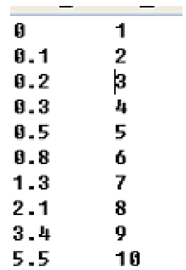
Property Name	Property Definition	Default Value
`Convert Method`	FillGap: Use to conduct signal re-sampling by value filling RemoveGap: Directly change the time-axis of the original signal. Use the time starting point and the Sampling period to re-arrange data time	FillGap
`FillMethod`	When ConvertMethod = FillGap, different data-filling methods can be selected. FixedValue: Use NullValue as the fixed filling data	LinearInterpolation

	PrevValue: Previous value NextValue: Next Value LinearInterpolation: Linear interpolation SplineInterpolation: Use Spline Curve to calculate the difference Monotonic Cubic: This is a 3-degree interpolation with damping. It has better performance than Spline in case of processing signal with large slope like square wave. NoFill: The value in this location is Null. No value is added.	
`Sampling Period`	To show or set the sampling period, $\Delta t^{(0)}$ of the output signal. When AutoDetect is set to True, it shows the minimum sampling period detected of the input signals, i.e. $\Delta t^{(0)} = \Delta t_{min}^{(1)}$ When AudioDetect is set to False, besides showing $\Delta t_{min}^{(1)}$ it can also be used to set the sampling period $\Delta t^{(0)}$.	$\Delta t^{(0)} = \mathrm{Min}\left(\Delta t^{(1)}\right)$
`Unit`	To show or set the sampling time unit of output signal. When AutoDetect is set to True, show the signal time unit detected. When AutoDetect is set to False, besides showing the signal time unit, set the signal time unit by using Sampling Period together.	Based on input signals
`AutoDetect`	Determine whether to detect Sampling Period and Unit automatically	True
`NullValue`	If ConvertMethod is set as FillGap and FillMethod set as FixedValue, this property would be provided to set the fixed value for data-filling	0

Example

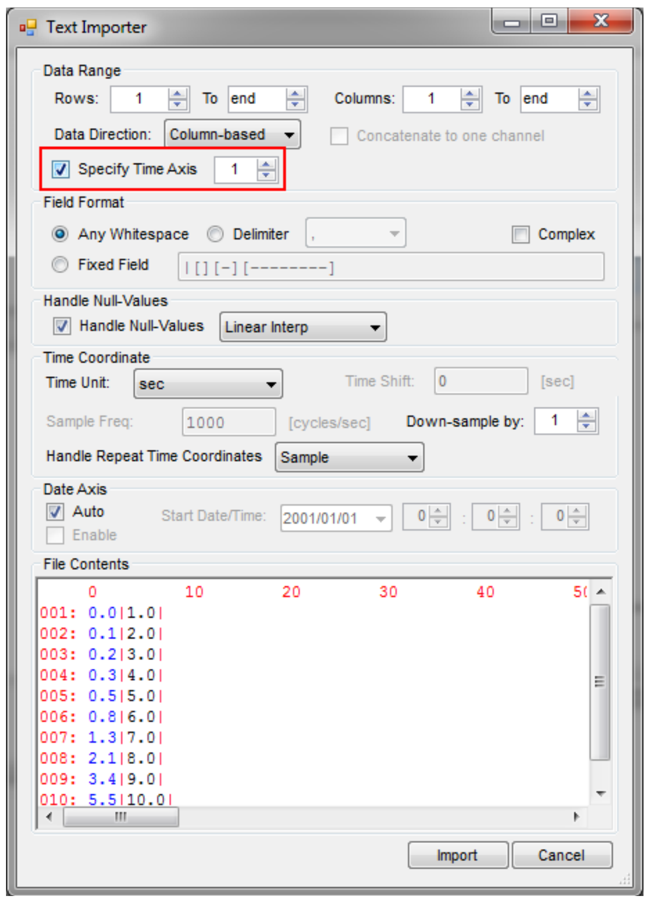
Read a set of signal data that is in the Indexed format.

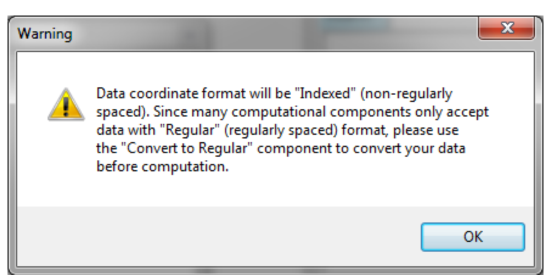
1. First, generate one set of simple data as shown below, where the first column is time while the second column is data.



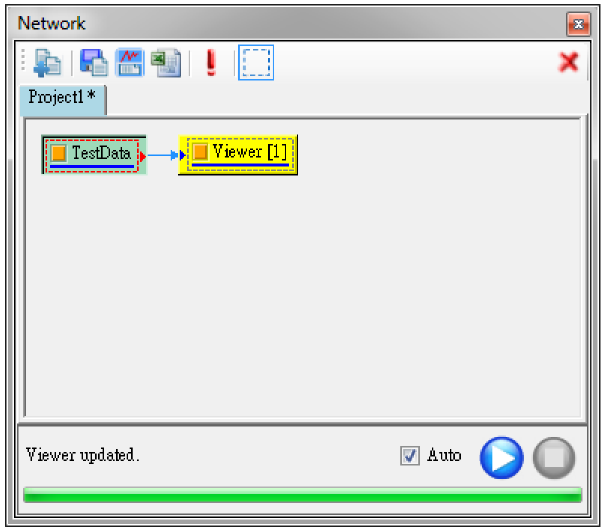
0	1
0.1	2
0.2	3
0.3	4
0.5	5
0.8	6
1.3	7
2.1	8
3.4	9
5.5	10

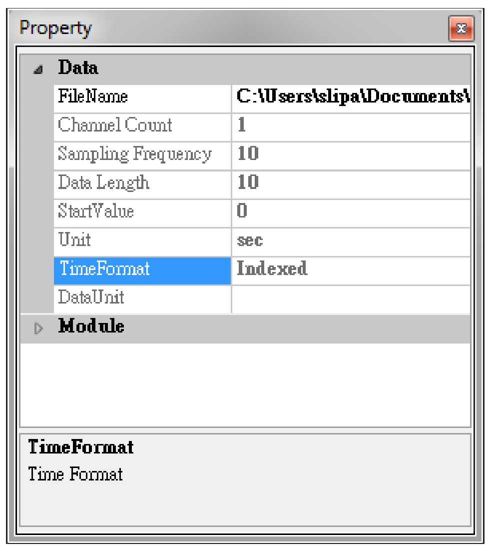
2. Then, press the in the Network tools or use **Source→Open Data** to read the signal file, TestData.txt. Check the `Specify Time Column` in **Text Importer** and then press OK.

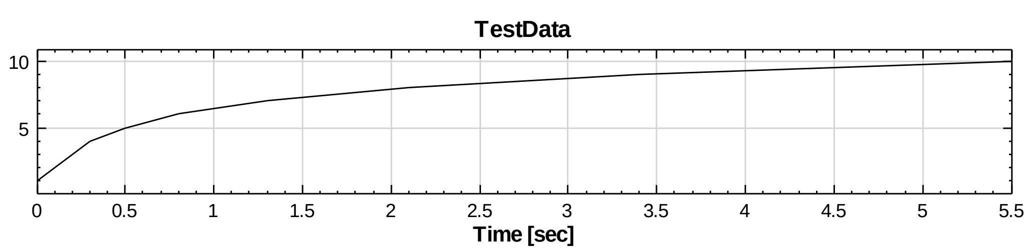




3. After reading the signal, use **`Viewer`→`Channel Viewer`** to plot figures. And select the **`TestData`** component to verify the *`OutputDataType`* in Properties/Module. It can be seen that the time-axis format is Indexed.

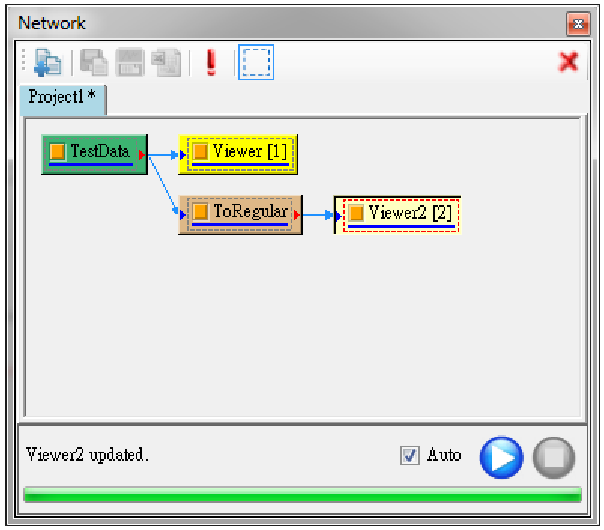


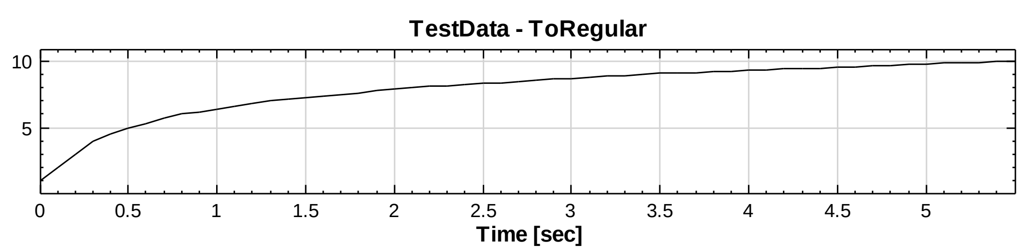




4. Connect **`ToRegular`** to the **`TestData`** component to convert the signal into an evenly sampled data. Then use **`Channel Viewer`** to plot the result. In the Properties of **`ToRegular`**, it shows that the default method is *`RemoveGap`*.

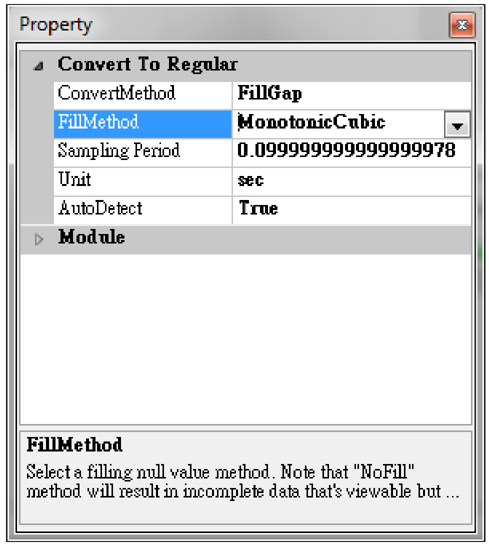
The `Sampling Period` detects the minimum sample period and this value is used for re-sampling. Therefore, the `Sampling Period` is 0.1 second and the total time length is $0.1 \times 9 = 0.9$ second.

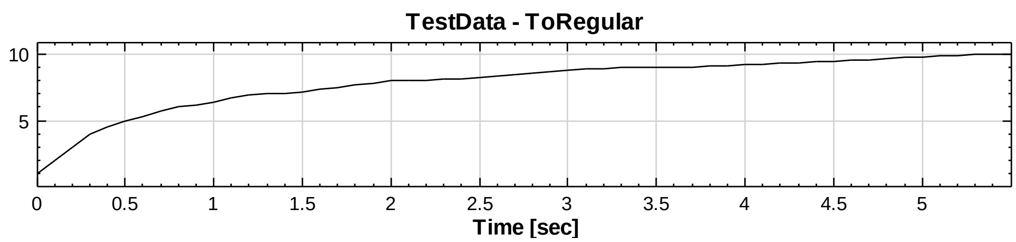




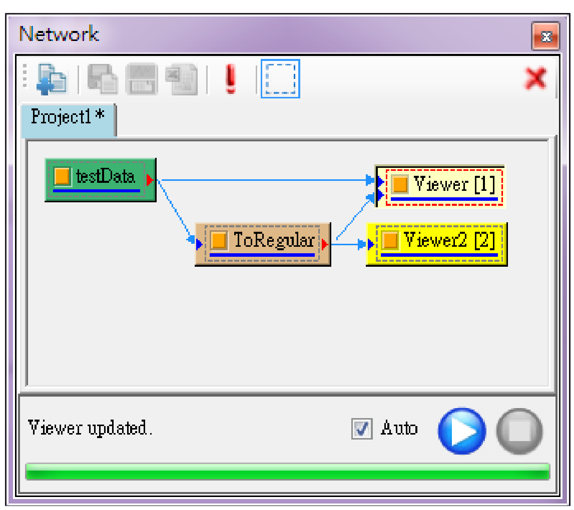
5. Change the `ConvertMethod` in `ToRegular` to FillGap, the `FillMethod` to Monotonic Cubic. The output result is shown as below. It shows that the

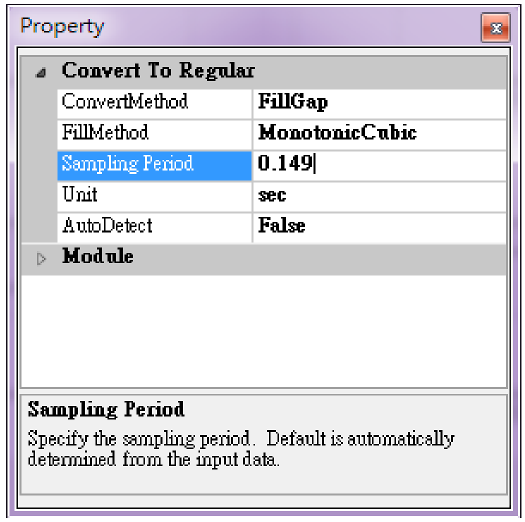
FillGap preserves the time-axis definition of the original signal and the signal time-axis is changed to even interval of an approximate 0.1 second sampling frequency.

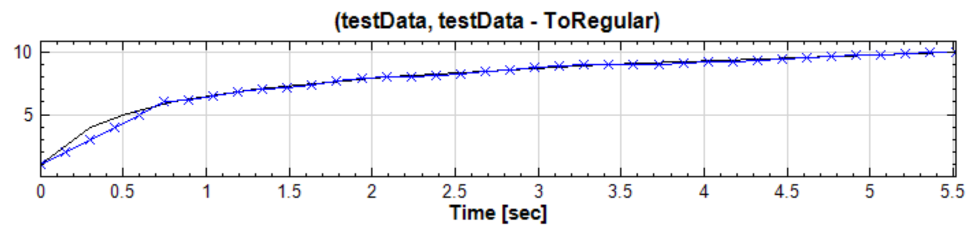




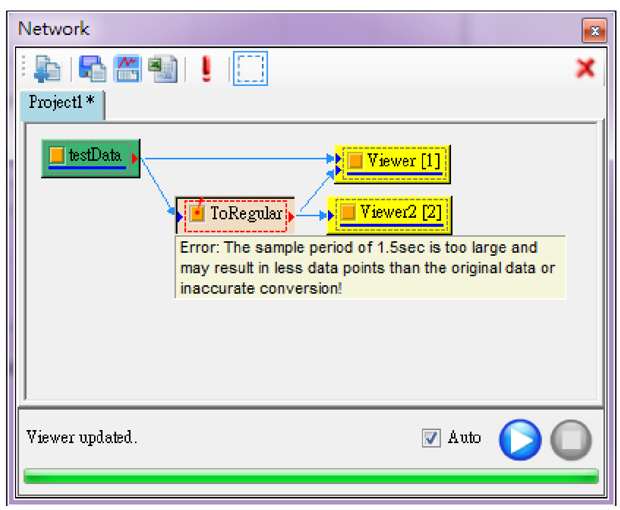
6. **`ToReguar`** allows the users to fine tune the signal sampling frequency. First, set *`AutoDetect`* to False, and then change the *`SamplingPeriod`* to 0.149. Drag the output result to Viewer[1] and compare with the original signal. The black curve is the original signal and the blue curve with 「-x」 is the **`ToRegular`** signal. It shows that the signal with the larger sampling frequency is distorted.







7. Next, try to change the `Sampling Period` to 0.15. An error message will pop up saying that entering a value which is bigger than 1.5 times of the minimum sampling frequency of the input signal is not allowed.



Related Functions

`Convert to Audio, Fill Null Value, Resample`

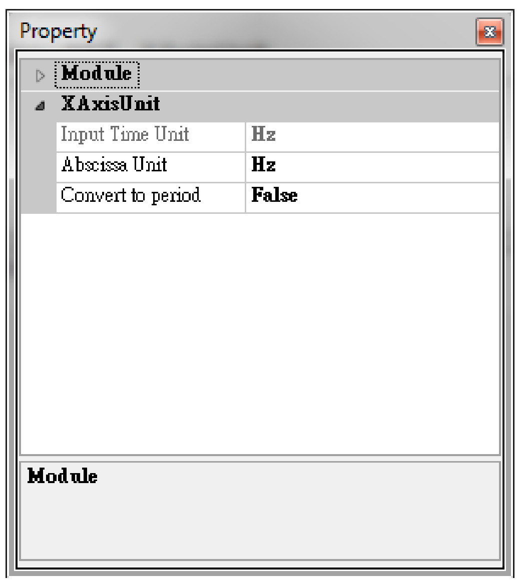
4.2.6 Change X Axis Unit

After reading in the signal data, the time unit of the data usually needs to be changed. In this case, Change X-Axis Unit can be used to convert time directly. In addition to time unit conversion, this module can also convert the spectrum-axis, i.e. the X-axis, from frequency to period.

Properties

This module accepts input of Signal (which could be a real number or complex number, single channel or multi-channel, Regular) and Audio (which could be a real number or complex number, single channel or multi-channel, Regular). The output formats are real, complex, single channel/multiple channel Regular signal and audio signal. If the property of `Convert` to period is changed to True, the format of output signal would be changed from Regular to Indexed, because the data in x-axis are not separated with equal interval anymore.

Property of *`Abscissa Unit`* shows the unit of x-axis to be converted to. The default is in second. Changing the *`Abscissa Unit`* can trigger the unit conversion on x-axis for the input data. The explanation of the unit is given in the table below.



Property Name	Property Definition	Default Value
`Convert to period`	When the unit of x-axis is frequency, this option can convert the unit of frequency to the unit of period on the x-axis	False
`ps`	Picosecond	10^{-12} second
`ns`	Nanosecond	10^{-9} second
`us`	Microsecond	10^{-6} second

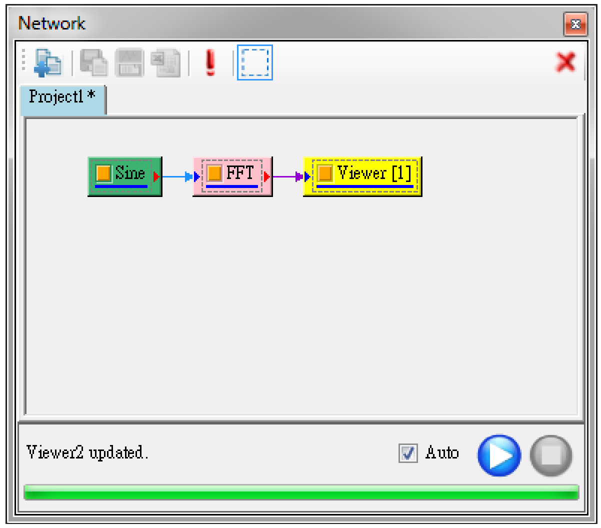
ms	Millisecond	10^{-3} second
sec	Second	1 second
min	Minute	60 seconds
hour	Hour	60 minutes
day	Day	24 hours
week	Week	7 days
month	Month	30 days
year	year	365 days

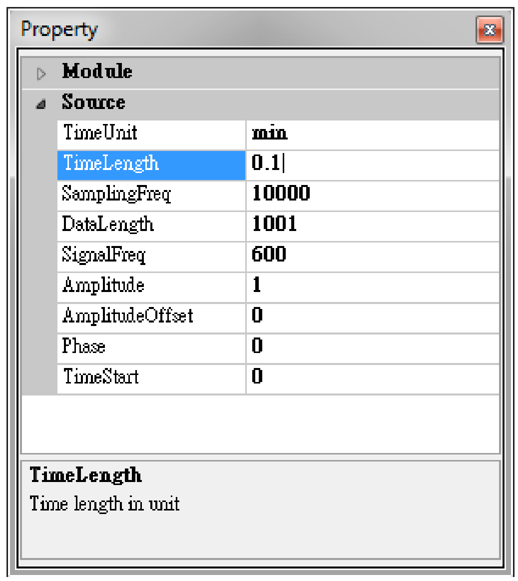
Property Name	Property Definition
THz	Terahertz Cycles per 10^{-12} second
GHz	Gigahertz Cycles per 10^{-9} second
MHz	Megahertz Cycles per 10^{-6} second
KHz	Kilohertz Cycles per 10^{-3} second
Hz	Hertz Cycles per second
Cycles_per_min	Cycles per minute
Cycles_per_hour	Cycles per hour
Cycles_per_day	Cycles per day
Cycles_per_week	Cycles per week
Cycles_per_month	Cycles per month
Cycles_per_year	Cycles per year

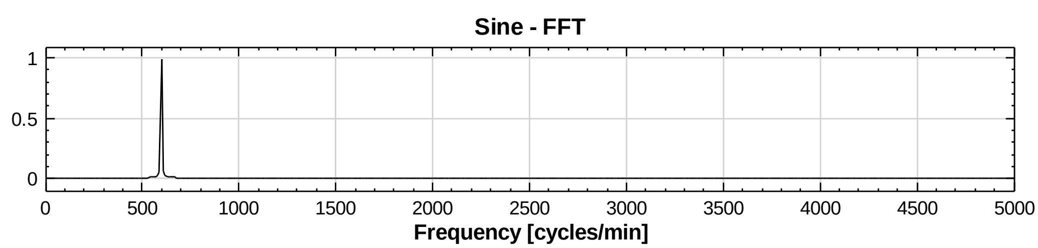
Example

1. Select **Source→Sine Wave** to create a sine wave with default signal frequency of 10Hz, sampling frequency of 1000Hz, and length of 1 second. Then change the *TimeUnit* to minute, *SamplingFreq* to 10000, *SignalFreq* to 600 for obtaining a signal whose x-axis unit is in minute and signal frequency is 10Hz.

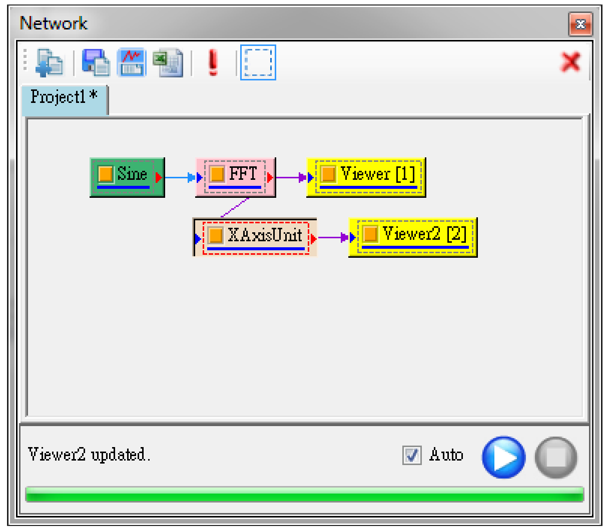
   Connect the signal to **Compute→Transform→Fourier Transform** to perform an FFT calculation, and then connect the Viewer to show the curve, where the x-axis is in frequency and the unit is in cycles per minute.

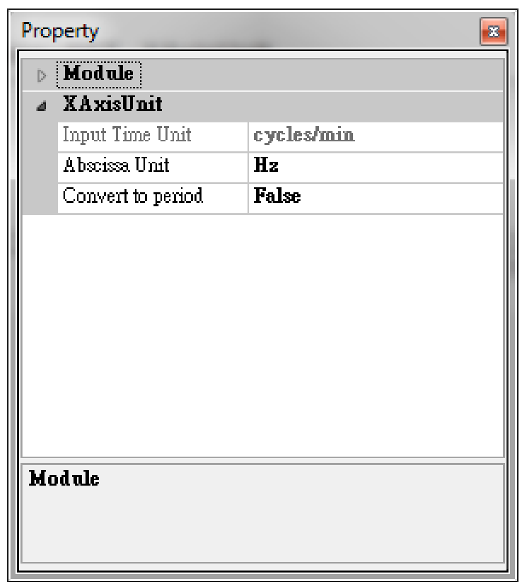


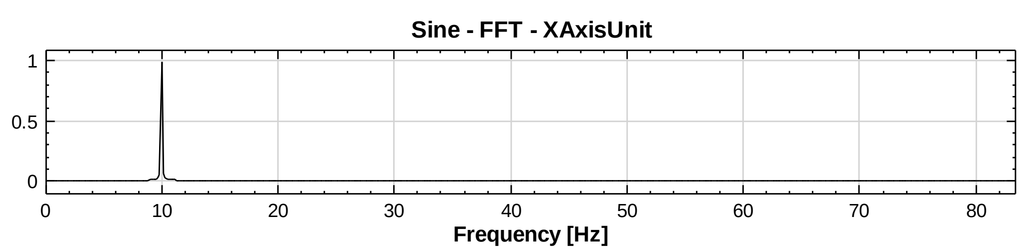




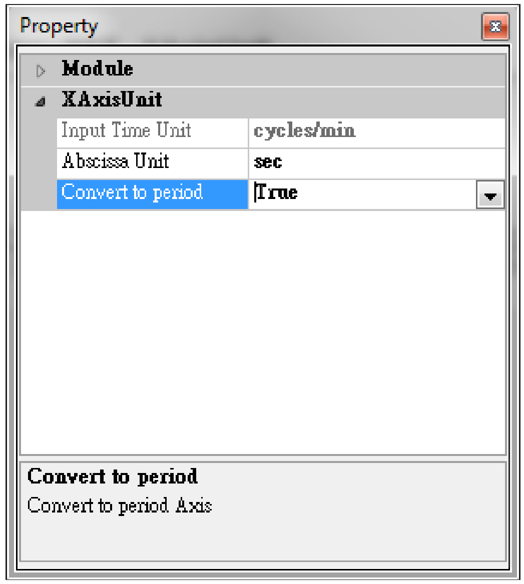
From the result, the frequency is 600 cycles per minute and not in Hz. Connect **`Change X Axis Unit`** to the output of **`FFT`**. Change the *`Properties/Abscissa`* unit and use **`Channel Viewer`** to show the result. Now the x-axis has changed to Hz and the values on x-axis has also changed to second automatically.

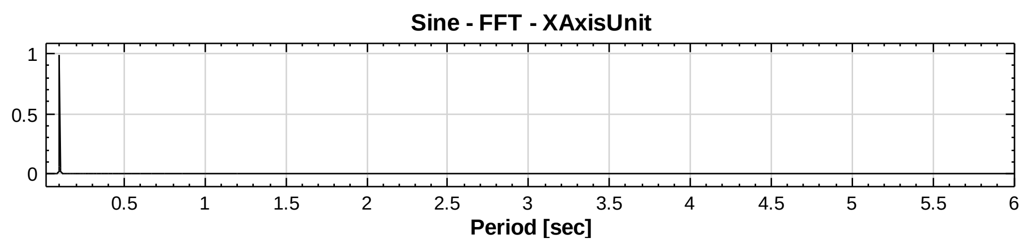






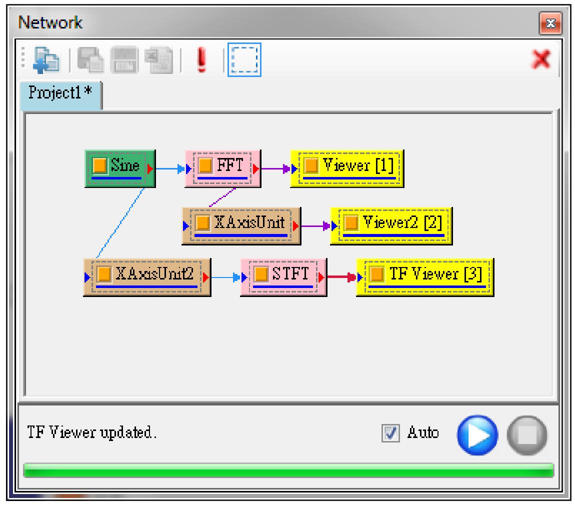
2. In addition, the `Convert` to period can be set to `True`. This converts the x-axis from frequency to period, as shown in the figure below. The x-axis is converted to Time and the unit is in seconds, i.e. the period corresponding to Hz.

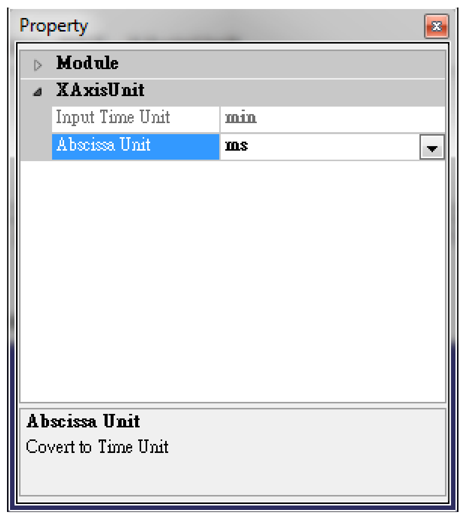


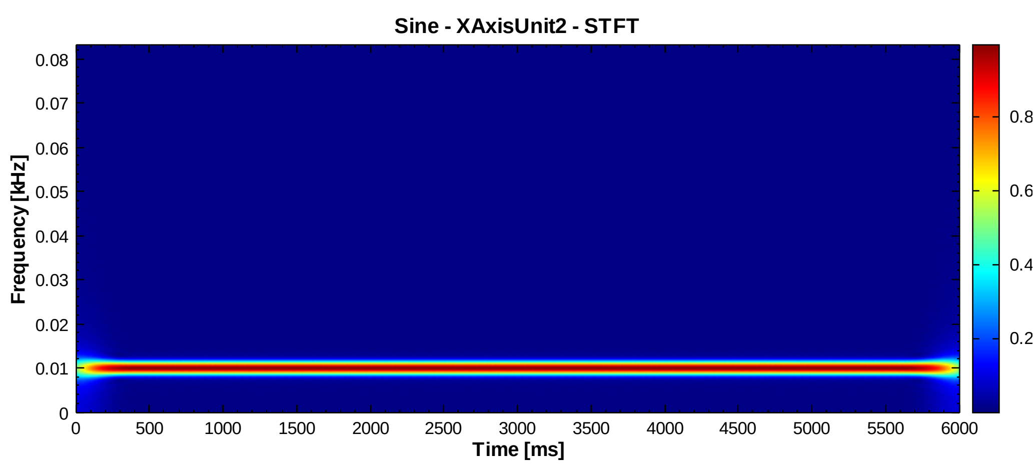


The time-frequency analysis module (TFA) is not able to pass the result to Change X Axis Unit for frequency unit modification directly, since the frequency is located on the y-axis for the time-frequency diagram. However, the conversion can be achieved by changing the unit of x-axis first and then performing time-frequency analysis.

3. Connect **Change X axis Unit** to the **Sine** component, change the *Properties/Abscissa* unit to msec, use **Compute → TFA → ShortTerm Fourier Transform** to perform time-frequency analysis, and then use **Viewer→Time-Frequency Viewer** to plot the result. It shows that the y-axis, i.e. the frequency axis, has been changed to KHz, i.e. 1/msec.







Related Functions

Import Data from File, Viewer, Fourier Transform

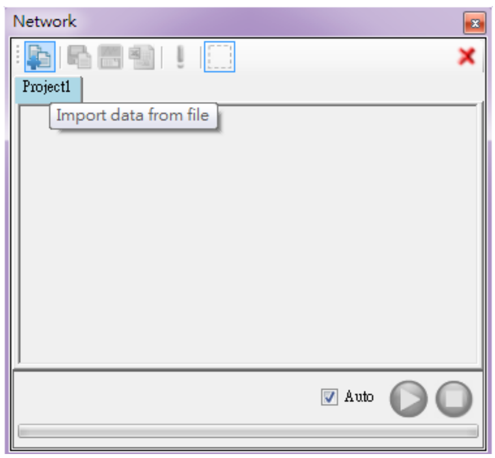
4.3 Source Of Signal Flow Object

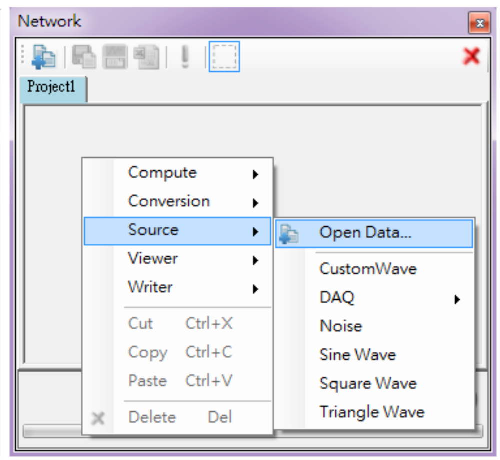
4.3.1 Open Data

Open a file that is to be used in Visual Signal.

Properties

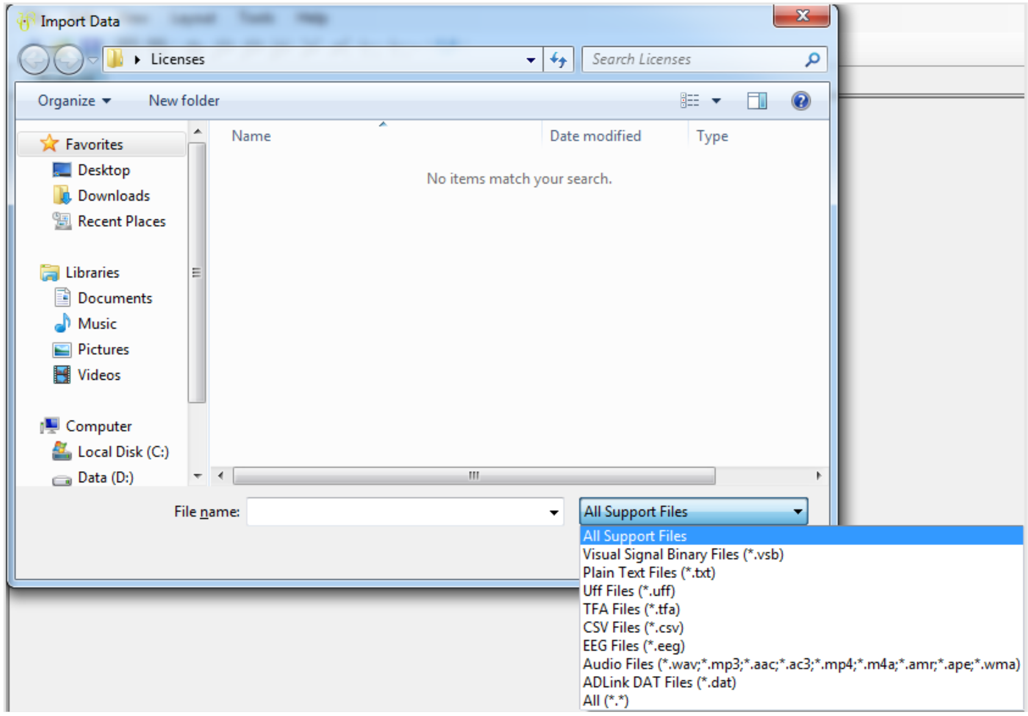
There are two methods to open a data file. The first method is to click on the **Import data from file** button and select the file you want to load into Visual Signal. The second method is to right mouse click the **Network Window** and the **Network Window** menu will pop up. From the menu, select **`Source→Open Data`** to select a file to be loaded into Visual Signal.

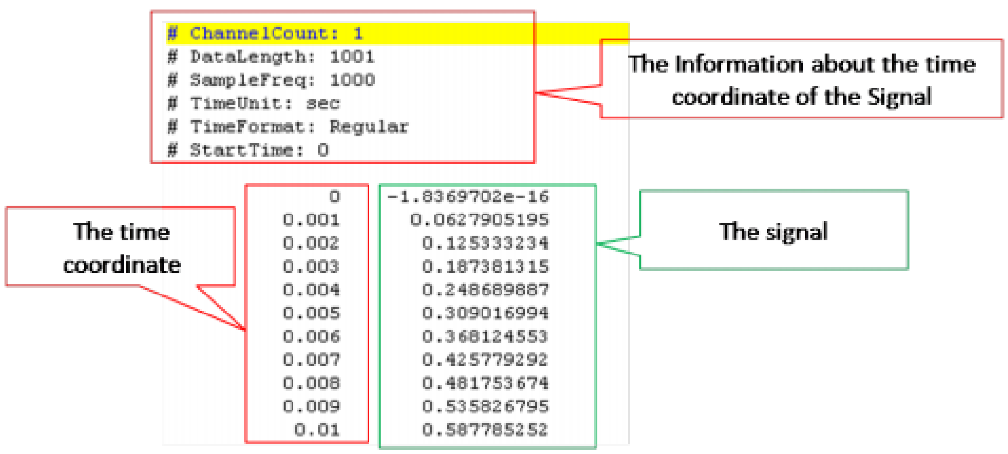




Supported File Types

Using either of above two methods, a browser window is opened for displaying all supported files. If you click on the file type drop down menu, a list of supported file types will be shown. The supported file types include: .txt, .csv, .tfa, .wav, .mp3 and special file types like .eeg for EEG file and .tfa for Visual Signal. When you open a .tfa file, the lines which begin with "#" contain information about the detailed aspects of the data and other lines are data signals. So a .tfa file not only has signals, but also has metadata information.

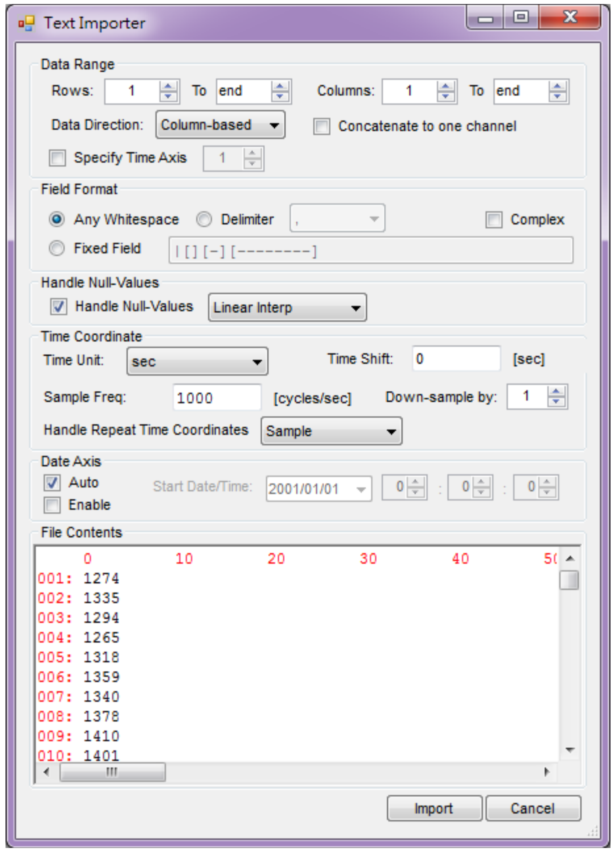




When opening any other types of files such as .csv and .txt, the Text Importer will open.

4.3.1.1 Text Importer

Opening .csv and .txt file types will open a **Text Importer**. The **Text Importer** is a complicated importer that allows you to specify options for importing .csv and .txt files.



The fields in **Data Range** are explained in the table below.

Property Name	Property Definition	Default Value
Rows	Enter the range of rows to be read.	1 to End
Columns	Enter the range of columns to be read	1 to End
Data Direction	Determine the way to read the data, either row based or column based	Column-based
Concatenate to one channel	Determine if the data is to be displayed in one channel or multiple channels (uncheck)	Unchecked
Specify Time	Determines if the time information already exists in the signal data. Check to select the column representing the time information. (**Note:** After checking the box the data will be displayed in the Index format).	Unchecked

In Field Format there are three options to select: **Any Whitespace**, **Delimiter** and **Fixed Field**. User can select the **Any Whitespace** option to separate each data by the white spaced character (most .txt files go by this method). User can select the **Delimiter** option and choose from the drop down menu to separate each data either by "," character or the TAB character. User can select the **Fixed Field** option to customize how the data is to be read. The "|" character allows one character to be read from each row to form a channel, "[ ]" character allows two characters to be read from each row to form a channel and "[-]" character allows three characters to be read from each row to form a channel. To read more than three characters into a channel, just add one "-" into "[-]" which becomes "[--]" to read four characters into a channel.

Another option is a check-box for complex data if the data being imported is of complex type. Checking the complex box will have the importer merge the data to a complex type. There needs to be at least two columns for the complex data, one is the real part of the complex, while the other is the imaginary part of the complex.

Example

If " | [--] [] " was entered in the **Fixed field** to read from a row with the numbers "123456789", then "1" is included in the first channel, "2345" is included in the second channel, "67" is included in the third channel, "8" and "9" are disregarded.

NULL Value Handle option allows user to choose a method to fill in missing values such as NULL or NaN. Currently methods are **Fixed value**, **Prev value**, **Next value**, **Linear Interp**, **Spline Interp** and **MonotonicCubic** (TIPS: For more information on filling in missing values, please look up on `Resample` in Chapter 3.1.7).

In the **Time Coordinates** field, there are a few options that can be selected. The property of **Time Coordinates** are listed below

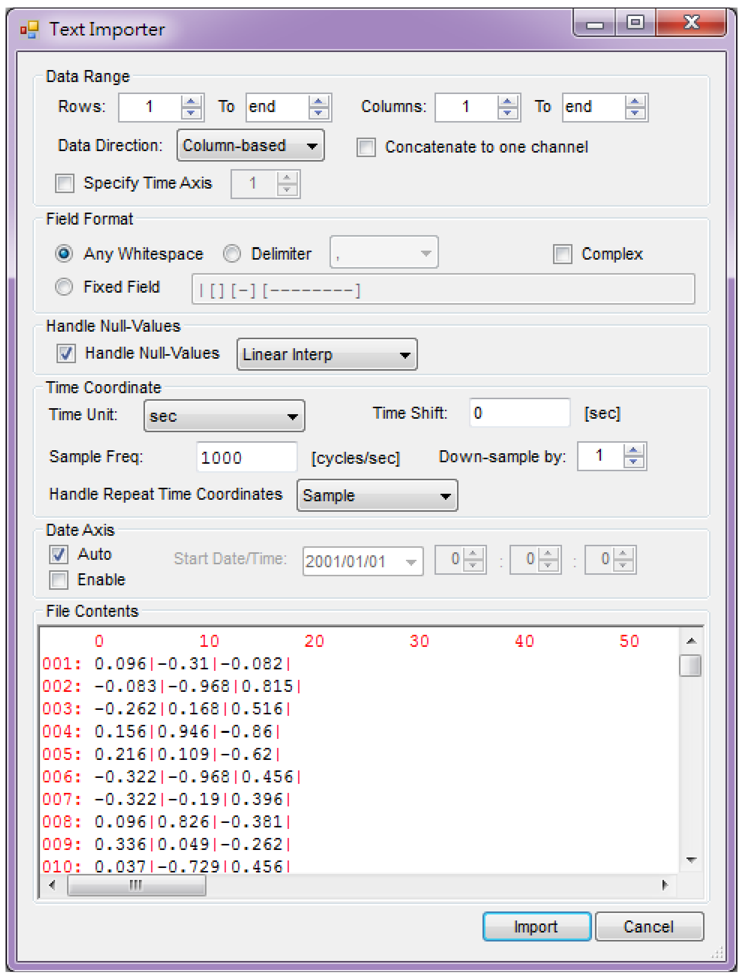
Property Name	Property Definition	Default Value
Time Unit	Select the time unit from picosecond, nanosecond, microsecond, millisecond, sec, min, hour, day, week, month (30 days) ,and year (365 days)	sec
Time Shift	Set the starting time of the data	0
Sample Frequency	Set the Sample Frequency	1000
Down-sample by	Set the Down-Sample rate. With every increment of the value, the sample data is reduced to save time during calculation. **Note:** The Sampling Frequency value will be automatically recalculated depending on the	Unchecked

	down-sample value. E.g. Sampling Frequency=1000 with Down-sample=2 will result in creating an imported Source component with Sampling Frequency=500	
Handle Repeat Time Coordinates	Sets how to handle repeating time coordinates.	Sample

Examples

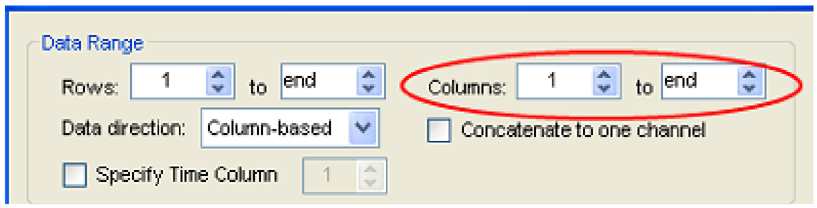
1. Import a Multi-Channel Signal Data

Load a multi-channel signal data file. The data is separated by white space character and there are three groups of data (one column is one channel.)

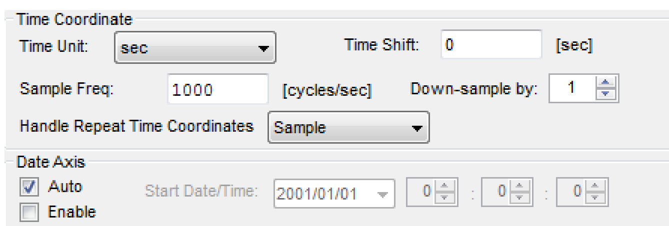


1. Click on  Import data from file button in the **Network Window** Toolbar or open it from right mouse clicking on the **Network Window** to open up the **Network Window** menu and select **`Source→Open Data`**.

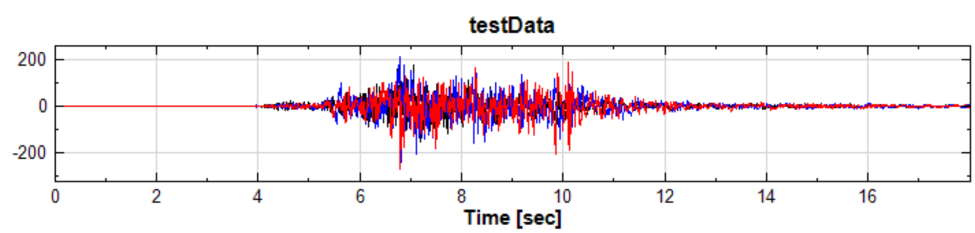
   The **Text Importer** will pop up when you have selected the text file to import (Assume the multi-channel text file was selected as describe above). If you want to import all three data columns then leave the Column option as 1 to end.
   Note: Set the Column option as 2 to 2, if you only want to import the second data column.



2. Because the imported data does not contain any time information, the **Time Unit** will be set to **sec** and **Sample Freq** set as 1000.

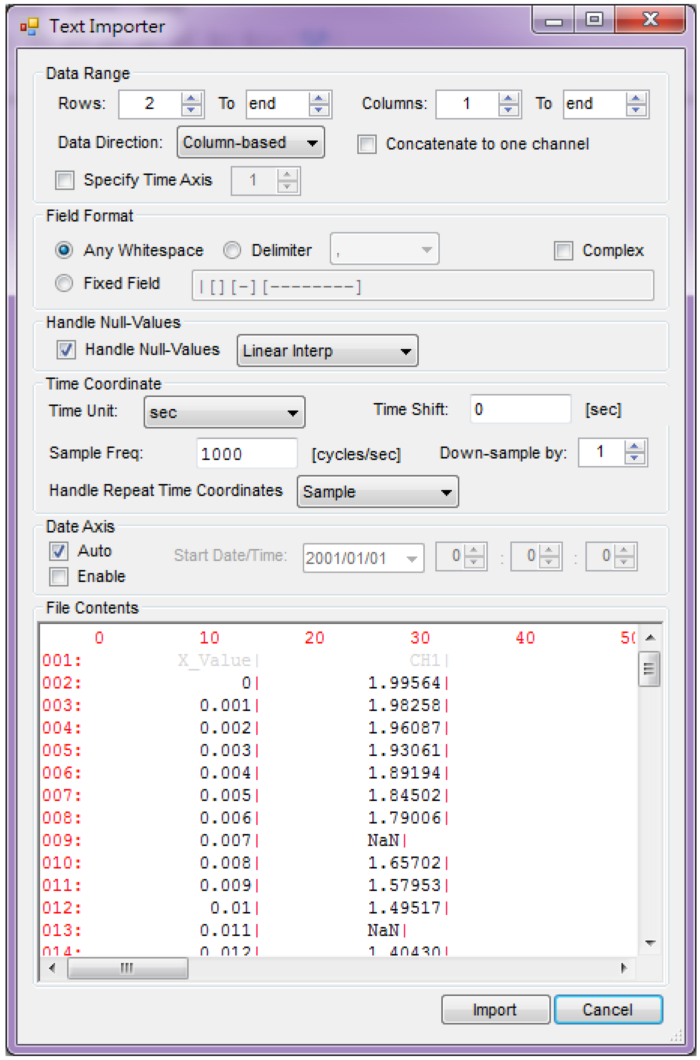


3. Click on the **Import** button to import the data

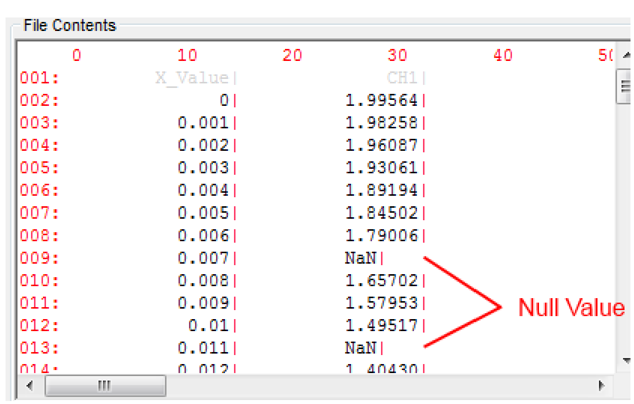


2. Import a data file which has time information and some missing data values.

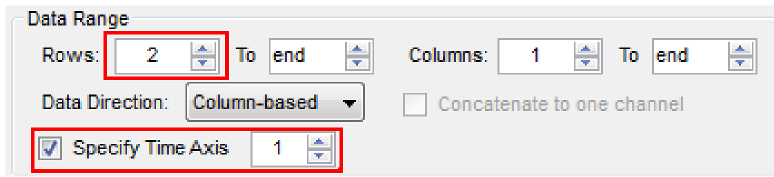
This example demonstrates how to import a data file which has time information included but contains some missing data values.



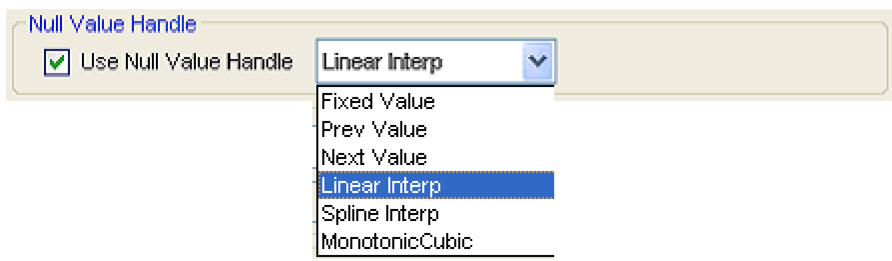
1. The data to be imported has to first be understood. You can see that there is NaN (missing data value) in 009 and 013 of the CH1 column. The first column is the X_Value (time) and the second column is the CH1 data value. **Text Importer** will be configured to import this data properly into Visual Signal.



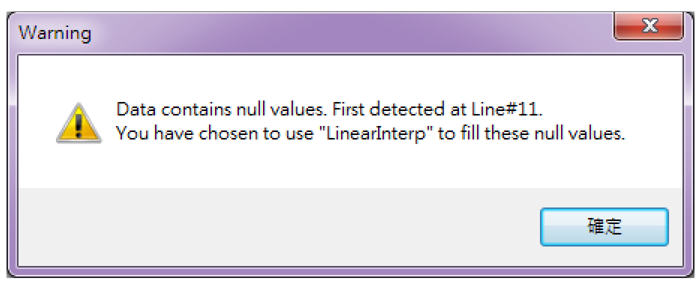
2. The first row in the data contains the titles for the two columns. So in the Rows option under **Data Range**, the **Rows** should begin from 2 (the second row is where the data values begin). Because the data contains time information, check the **Specify Time Column** option under **Data Range** and select 1 (first column of the data is the time information).



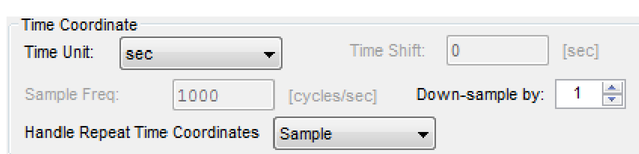
3. The **Field Format** does not need to be edited because the data is separated by **Any Whitespace** (which is the default selection). Check Use **NULL Value Handle** option under **NULL Value Handle** and select **Linear Interp** calculation from the drop down menu to fill in the NaN values (missing value).



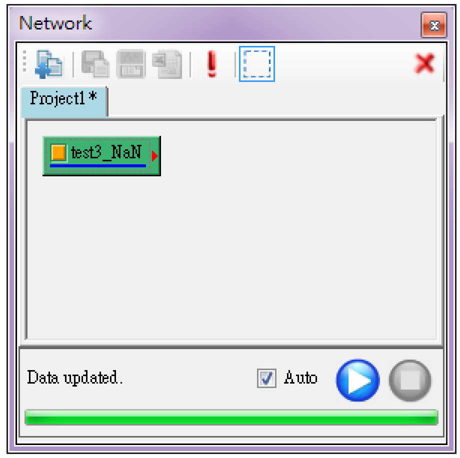
If there are *NULL* or *NaN* values in the data but **Use NULL Value Handle** option isn't checked, then the following warning message will appear.



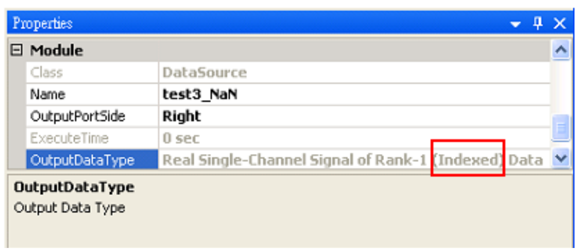
4. Although the time information exists in the data, you still need to set the unit of the time information. Click on the **Time Unit** option under **Time Coordinate** and select **sec** from the drop down menu.



5. Click on the **Import** button once the configurations are done.



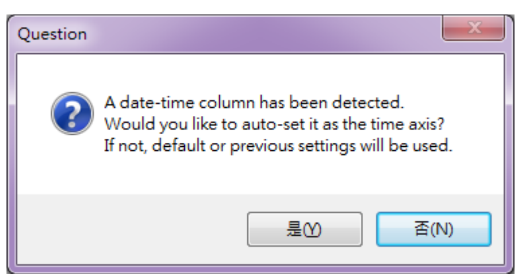
If the signal data is to be calculated further, it needs to connect to **`Conversion`**→ **`Convert To Regular`** because **Specify Time Column** is in the Indexed format. It needs to convert it to regular format.

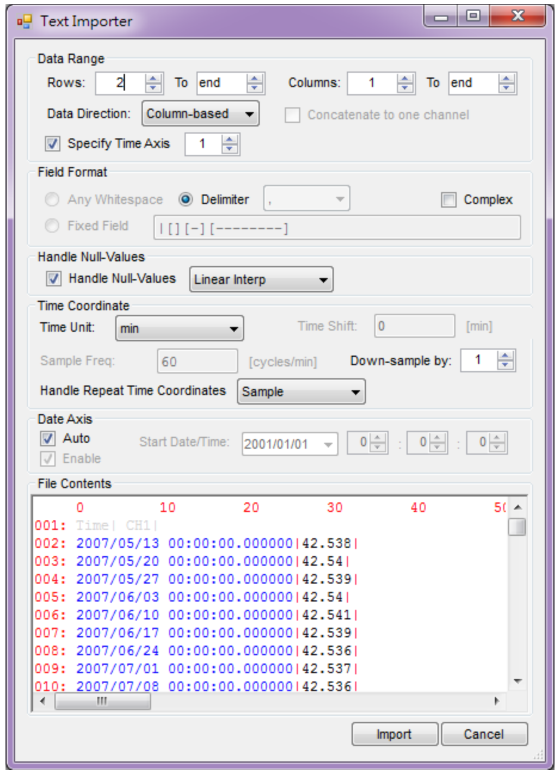


4.3.1.2 Import csv file format

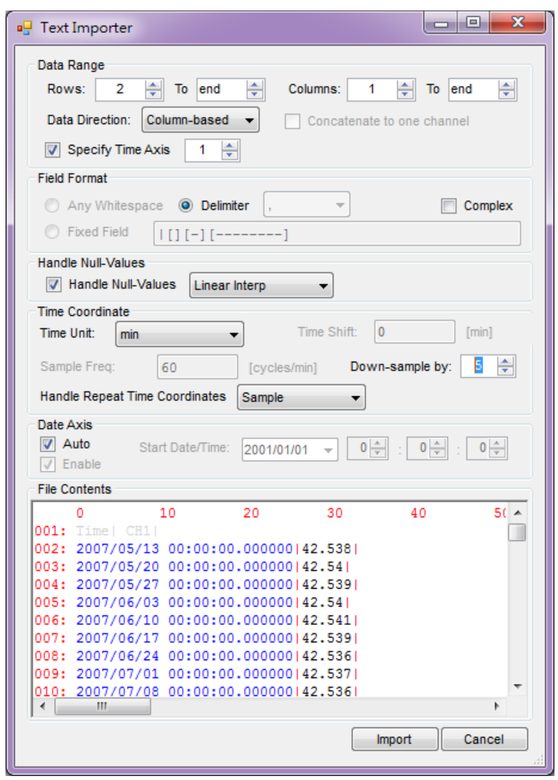
The data in the csv file format is separated by a "," comma character. The first row in the data contains the titles of the columns, so the data has to begin from the second row.

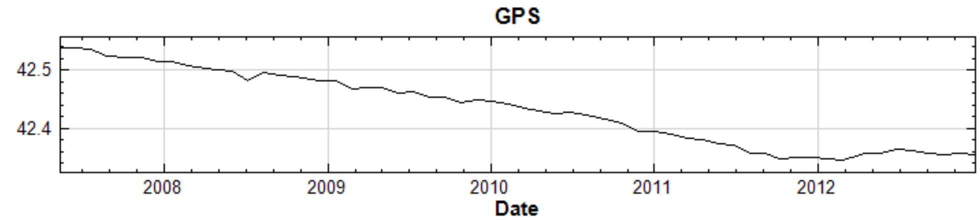
Text Importer will automatically detect the inputted file format. When a csv file format is loaded, the **Delimiter** function will be checked. **Text Importer** can also detect the data-time format. If a date-time axis is found, a **Question window** will pop up like the figure below, giving you the option to have the time axis be automatically set for you.





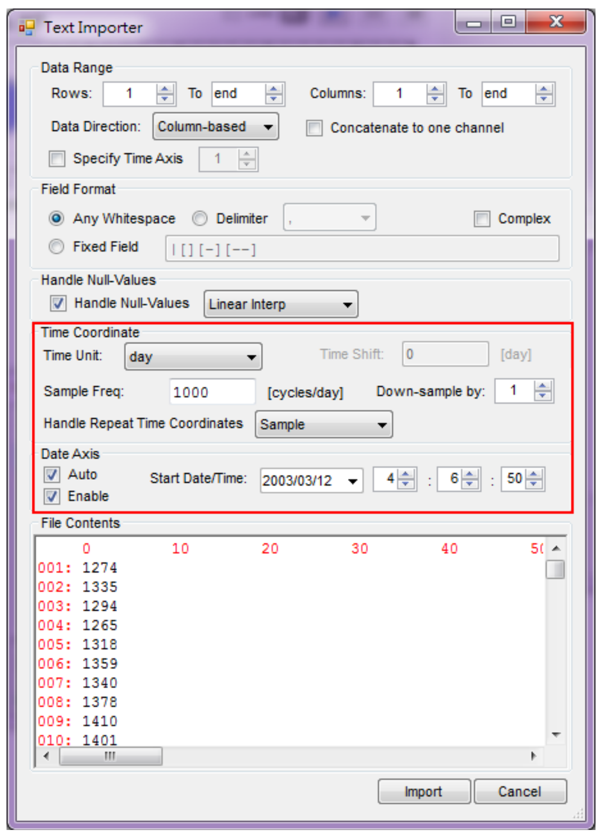
You can increase the **Down-Sample by** number to 5 if the data is too large, which means that for every 5 data points only 1 will be read.

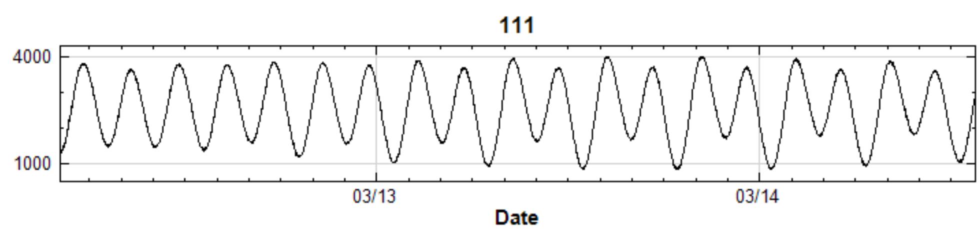




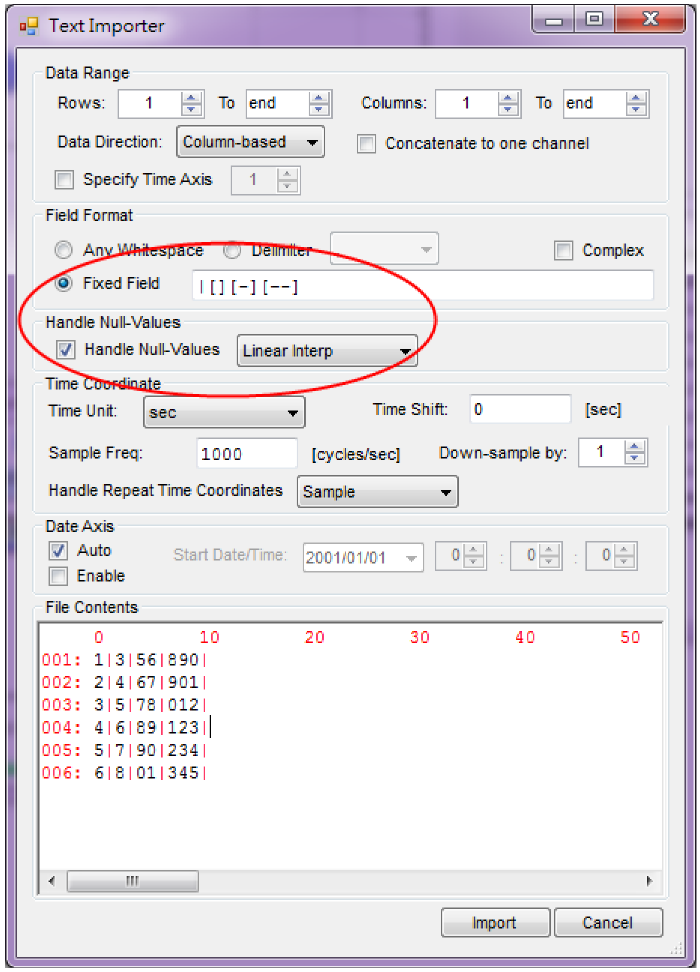
Note: It is not advised to have decimal numbers within the time of the data. E.g. 2005/3/18 15:05:35:.01242

1. If you wish to import a data with date and time and it is not in the csv format then you will have to configure the **Data Axis** and **Time Coordinate** options. In this example, **Time Coordinate** is set as day and **Sample Frequency** is set as 1000. **Data Axis** is Enabled and the date and time is set as 2003/03/12 4(hour):6(minute):50(seconds).





2. If all rows of the imported data have the same character length, you can use customize **Fixed field** to read in the data.

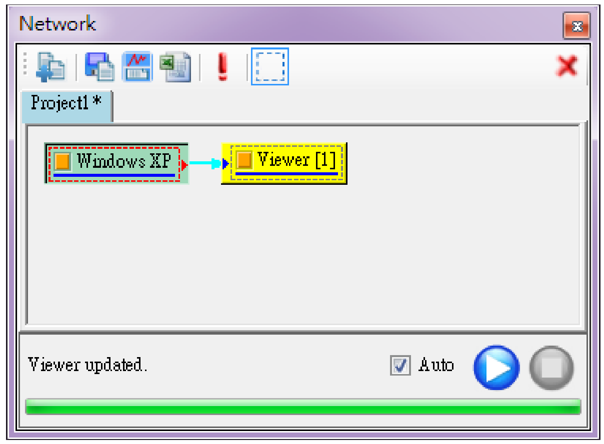


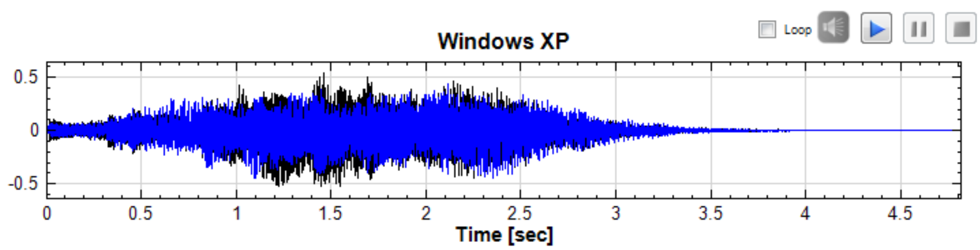
The data imported are placed in four channels: the first channel contains one character, the second channel contains two characters, the third channel contains three characters, and the fourth channel contains four characters. Under **Data Viewer**, the X values are based on the **Sample Frequency** of 1000Hz. So every data value is read at 0.001 increments.

Index	X Value	CH1	CH2	CH3
0	0	1	23	456
1	0.001	2	34	567
2	0.002	3	45	678
3	0.003	4	56	789
4	0.004	5	67	890
5	0.005	6	78	901

4.3.1.3 Import wav or mp3 file format

If the file imported is .wav or .mp3, these two types of sound formats will directly create a source component in the **Network Window** and will not open any importers.





Property Name	Property Definition	Default Value
Data Range: Contains the options to set the range for the data		
Rows	Enter the range of rows to be read	1 to End
Columns	Enter the range of columns to be read	1 to End
Data Direction	Determine the way to read the data, either row based or column based	Column-based
Concatenate to one channel	Determine if the data is to be displayed in one channel or multiple channels (uncheck)	Unchecked
Specify Time Column	Determine if the time information already exists in the signal data. Check to select the column representing the time information. (**Note:** After checking the box the data will be displayed in the Indexed format).	Unchecked
Field Format: Contains the options to set how data values are read		

Any Whitespace	Separate the data values by the white space character	Yes
Delimiter	Separate the data values by the comma or the TAB character	No
Fixed Field	Customize your own rules to read the data values	No
Null Value Handle: Contains the options to deal with *NULL* or *NaN* values (missing values)		
NULLFilledMethod	Check the calculation method to fill in the missing value	Linear Interp.
Time Coordinate: Contains the options to set the date and time		
Time Unit	Select the time unit from psec, nsec, msec, sec, minute, hour, day, week, month (30 days) and year (365 days)	sec
Time Shift	Set the starting time of the data	0
Sample Frequency	Set the Sample Frequency	1000
Down-sample by	Set the Down-Sample rate. With every increment of the value, the sample data will be shortened to save time during calculation. (**Note:** The Sampling Frequency value will be automatically recalculated depending on the down-sample value. E.g. Sampling Frequency = 1000 with Down-sample = 2 will result in creating an imported Source component with Sampling Frequency = 500).	1
Date Axis		
Enable	Select to enable the date and time option	Unchecked→Disabled
Start Date/Time	Set the Date and Time for the data values	2001/01/01 0:0:0

Related Functions

`Channel Viewer, Fill NULL Value, Resample`

4.3.2 Noise

The Noise function has the ability to create seven different types of noise signal waves.

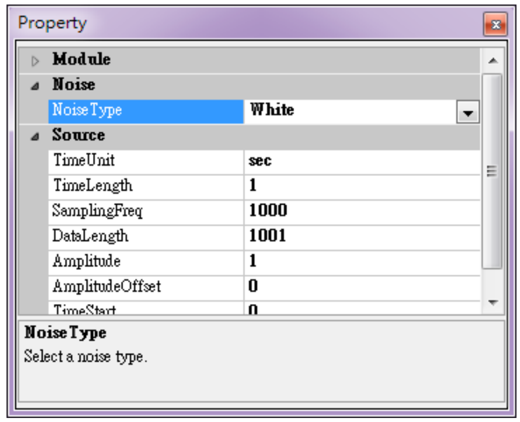
Introduction

Below are the following descriptions for each noise definition:

Noise	Equation	Description
White	$E[x_{\mathrm{white}}] = 0$ $E[f_{\mathrm{white}}(t) f_{\mathrm{white}}(t-\tau)] = \delta(\tau)$	The noise that has a wide range of frequencies of uniform intensity, where *E* is expected value. It has an autocorrelation which can be represented by a Delta function over the relevant space dimensions.
Gaussian	$x_x = \frac{1}{\sigma\sqrt{2\pi}} e^{-\frac{(t-\bar{t})^2}{2\sigma^2}}$	Gaussian noise is noise that has a probability density function (abbreviated pdf) of the normal distribution (also known as Gaussian distribution).
Speckle	-	Speckle-type noise, its amplitude is either zero or one, it is controlled by the probability P.
Pink	$F\left[x_{\mathrm{pink}}(t)\right]^2 \propto \frac{1}{f}$	Pink noise or $1/f$ noise is a signal or process with a frequency spectrum such that the power spectral density is proportional to the reciprocal of the frequency
Brown	$F[x_{\mathrm{brown}}(t)]^2 \propto \frac{1}{f^2}$	Brownian noise is the kind of signal noise produced by Brownian motion hence its alternative name of random walk noise
Blue	$F[x_{\mathrm{blue}}(t)]^2 \propto f$	Blue noise's power density increased 3 dB per octave with increasing frequency (density proportional to f) over a finite frequency range
Violet	$F[x_{\mathrm{violet}}(t)]^2 \propto f^2$	Violet noise's power density increases 6 dB per octave with

		increasing frequency (density proportional to f^2) over a finite frequency range

Properties



{Source} Property Name	Property Definition	Default Value
`TimeUnit`	Set the time in `ps`, `ns`, `us`, `ms`, `sec`, `min`, `hour`, `day`, `week`, `month` ,or `year`	`sec`
`TimeLength`	Set the value of time selected in `TimeUnit`	1
`SamplingFreq`	Set the number of Sampling frequency (the amount of data values to be sampled)	1000
`DataLength`	Set the length of the data (`SamplingFreq` × `TimeLength` + 1)	1001
`Amplitude`	Set the maximum displacement of a periodic wave	1
`AmplitudeOffSet`	Set the amplitude offset	0
`TimeStart`	Set the start time for the data	0

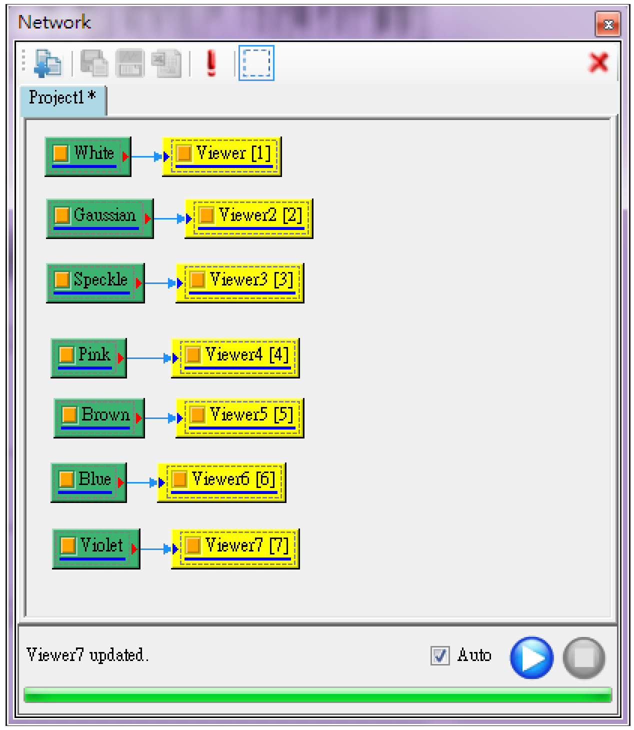
There are two more variable options in Gaussian Noise and Speckle Noise

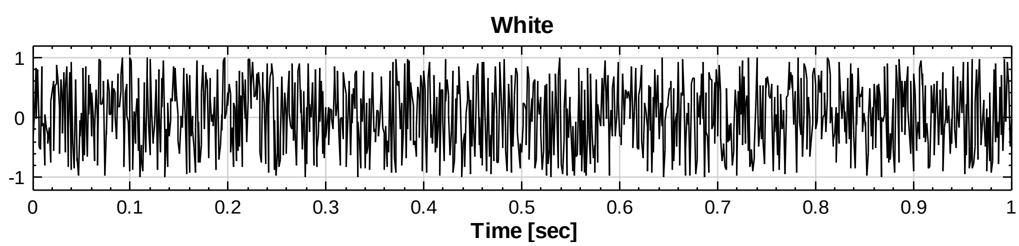
{Noise} Property Name	Property Definition	Default Value
`Sigma (Gaussian)`	Set the standard deviation σ for Gaussian Noise	1
`Probability (Speckle)`	Set the probability of occurrence for Speckle Noise.	0.005

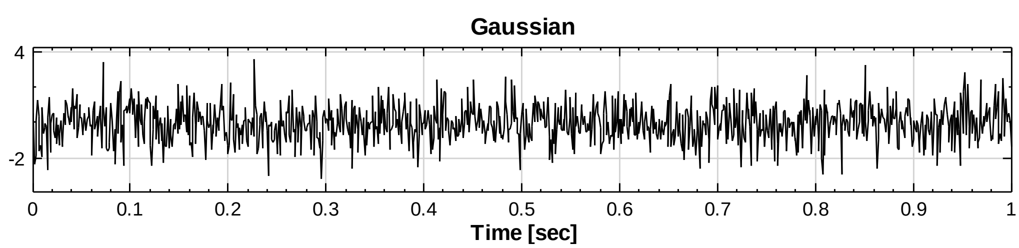
Example

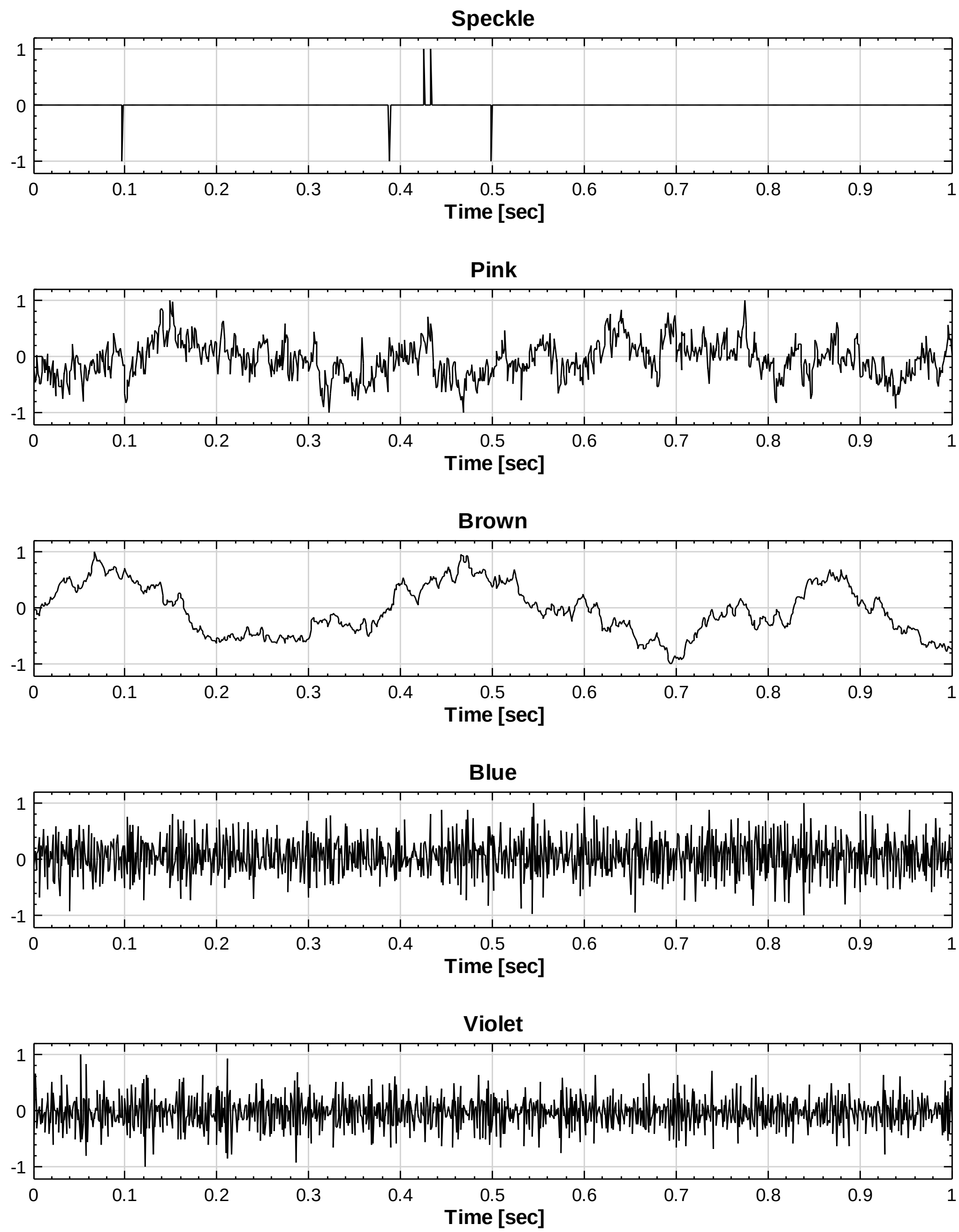
Analyzing noise waves:

1. Create seven different types of noise through **Source→Noise** and connect each source component to a **Viewer→Channel Viewer**.

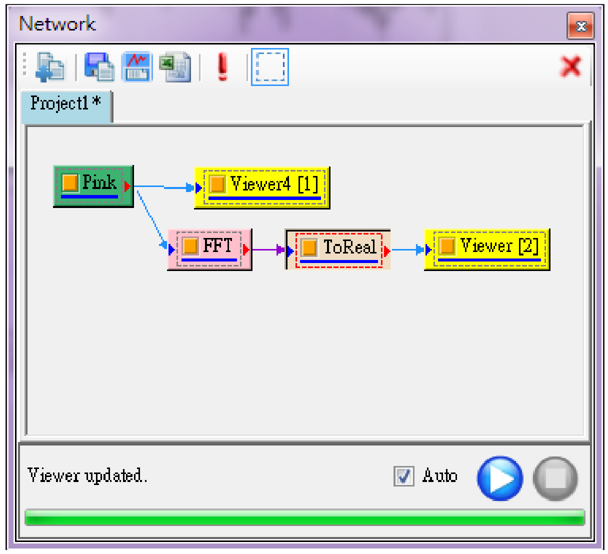


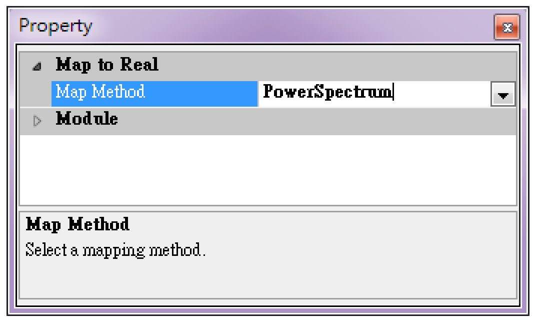


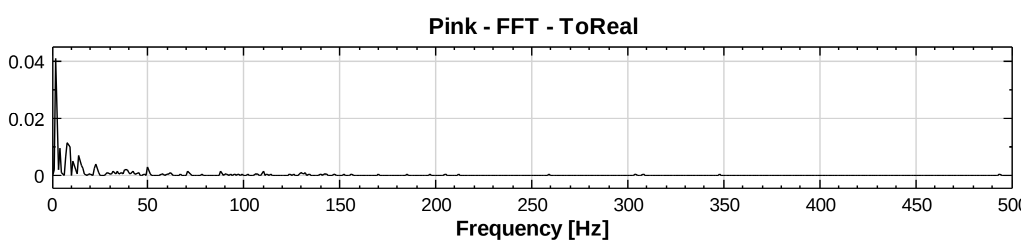




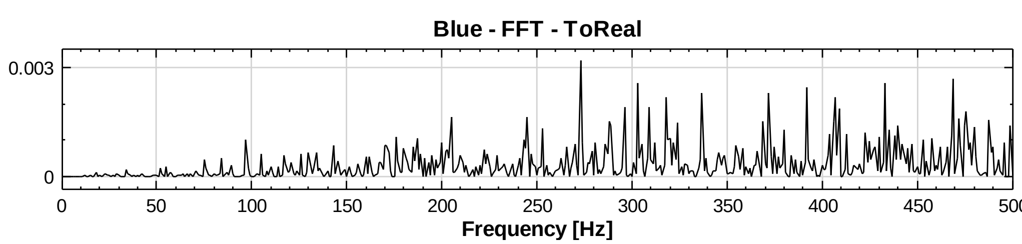
2. Connect the **`Noise`** component to **`Compute`**→**`Transform`**→**`FFT`** and connect the **`FFT`** component to **`Conversion`**→**`Map To Real`** and finally connect the result to a **`Channel Viewer`** component. Change the *`Map Method`* of the **`ToReal`** to *`PowerSpectrum`*. You can observe that the power spectrum density increases as the frequency decreases.





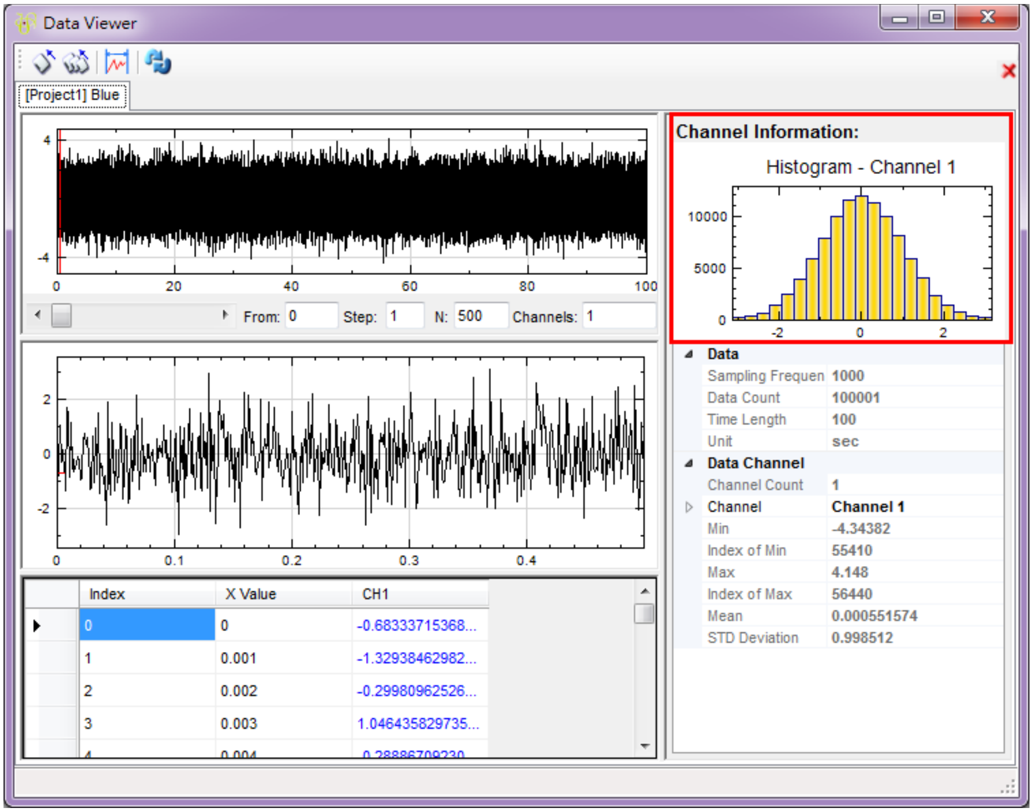


3. Repeat the steps to the other source signals. The Blue noise graph is shown below after the steps.



You can observe that the power spectrum density increases as frequency increases.

4. Set the `Noise Type` to `Gaussian` noise and set the `TimeLength` to 100 seconds and view the histogram with **Data Viewer**.



If you want to see example projects using the **`Noise`** function, open projects demo39 and demo40 in C:\Program Files\AnCAD\Visual Signal\demo\Basic.

Related Functions

`Channel Viewer, Fourier Transform, Map To Real`

Reference

http://en.wikipedia.org/wiki/Colors_of_noise
http://en.wikipedia.org/wiki/Gaussian_noise

4.3.3 Sine Wave

Explanation is given for the **Source→Sine Wave**.

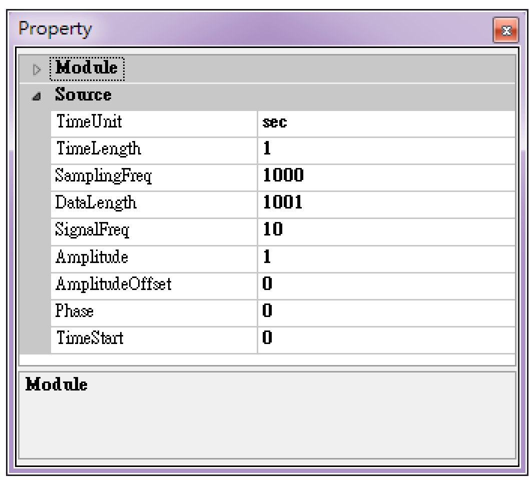
Introduction

Let t = time, N = length of the signal, $t_i = [t_0, t_1, \cdots, t_{N-1}]$ is the representation of the time coordinate and sine wave can be generated by

$$x_i = A \cdot \sin(\omega t_i + \delta) + V_0$$

Where A = amplitude, ω = angular frequency, δ = phase at t_0, V_0 =offset from X axis, and sampling frequency is defined as $f = \frac{\omega}{2\pi}$.

Properties



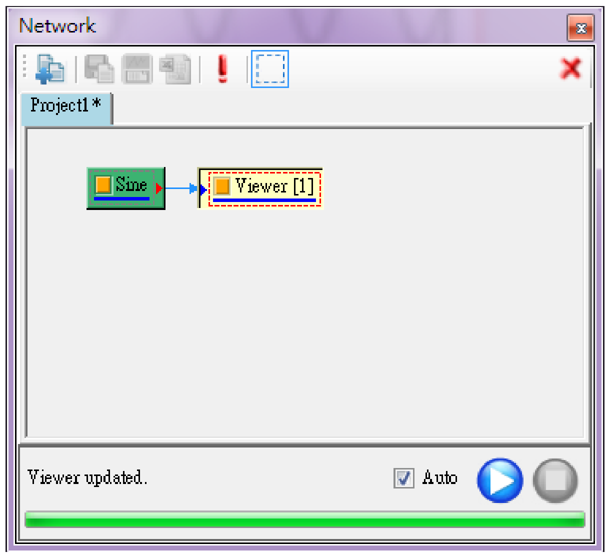
{Source} Property Name	Property Definition	Default Value
`TimeUnit`	Set the time in `ps`, `ns`, `us`, `ms`, `sec`, `min`, `hour`, `day`, `week`, `month` ,or `year`	`sec`
`TimeLength`	Set the value of time selected in `TimeUnit`	1
`SamplingFreq`	Set the number of Sampling Frequency (the amount of data values to be sampled)	1000
`DataLength`	Set the length to the data (`SamplingFreq` × `TimeLength` + 1)	1001
`SignalFreq`	Set the real signal frequency. The unit is in Hz.	10

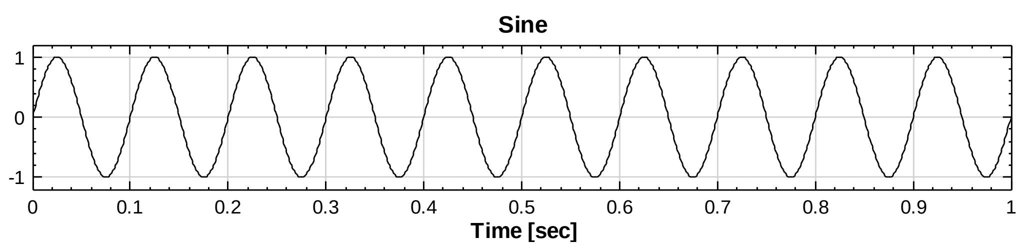
`Amplitude`	Set the maximum displacement of a periodic wave	1
`AmplitudeOffSet`	Set the amplitude offset	0
`Phase`	Set the `Phase` in degree. When the phase is non-zero, the entire waveform appears to be shifted in time with specified value	0°
`TimeStart`	Set the start time for the data	0

Example

Create a Sine wave

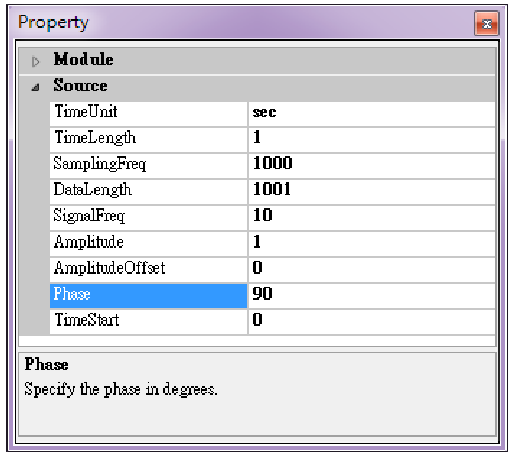
1. Create **`Source→Sine Wave`**.

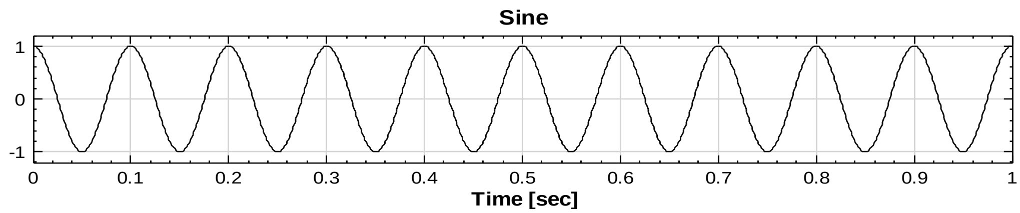




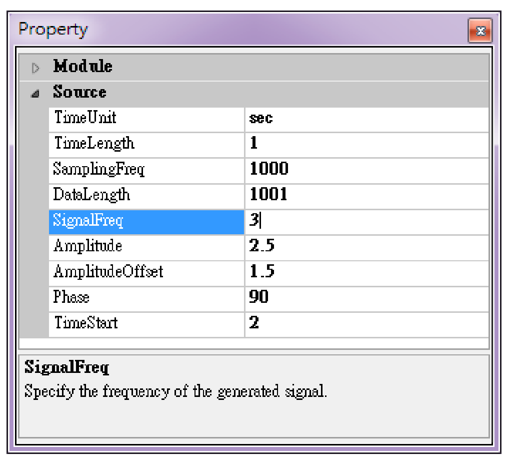
In this figure, the `SamplingFreq` is set to 1000 and `SignalFreq` is set to 10.

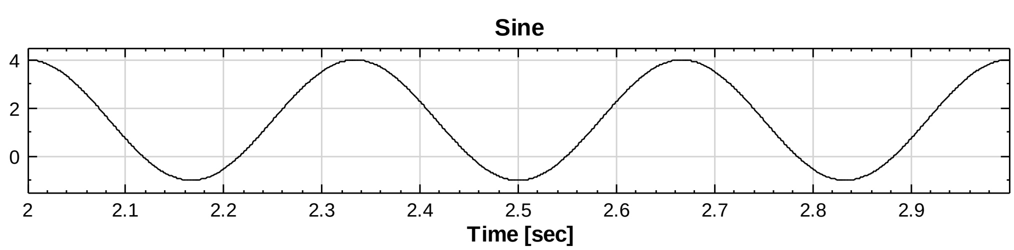
2. If you set the `Phase` to 90° then the sine wave will become a cosine save.





3. Set the `SignalFreq` to 3, `Amplitude` to 2.5, `AmplitudeOffSet` to 1.5, and `TimeStart` to 2, the graph is shown in the image below.





Related Functions

Channel Viewer

4.3.4 Square Wave

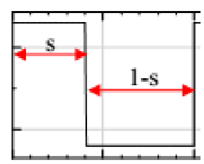
Explanation is given for the **Source→Square Wave**.

Introduction

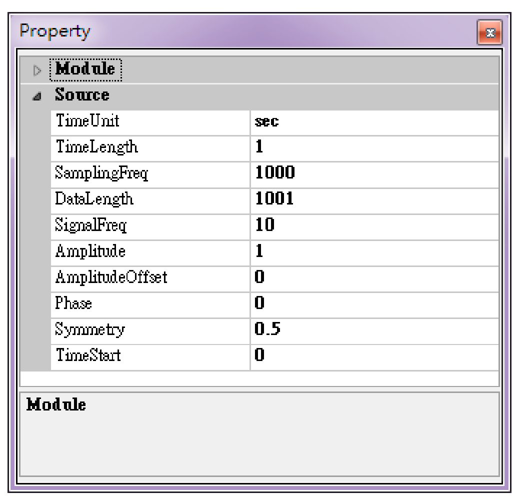
$$x_{\mathrm{sqr}}(t) = \begin{cases} A + V_0, & \delta \le t < \left(\frac{1}{f}\right)s + \delta \\ -A + V_0, & \left(\frac{1}{f}\right)s + \delta \le t < \left(\frac{1}{f}\right) + \delta \end{cases}$$

$$x_{\mathrm{sqr}}(t + T) = x_{\mathrm{sqr}}(t)$$

Where A = amplitude, f = sampling frequency, δ = phase at t_0, V_0 = offset from X axis, the ratio s is shown in the image below.



Properties



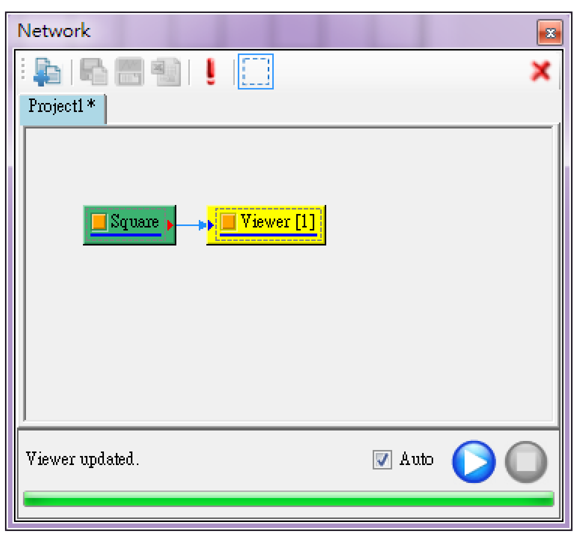
{Source} Property Name	Property Definition	Default Value
`TimeUnit`	Set the time in `ps`, `ns`, `us`, `ms`, `sec`, `min`, `hour`, `day`, `week`, `month` ,or `year`	`sec`
`TimeLength`	Set the value of time selected in `TimeUnit`	1

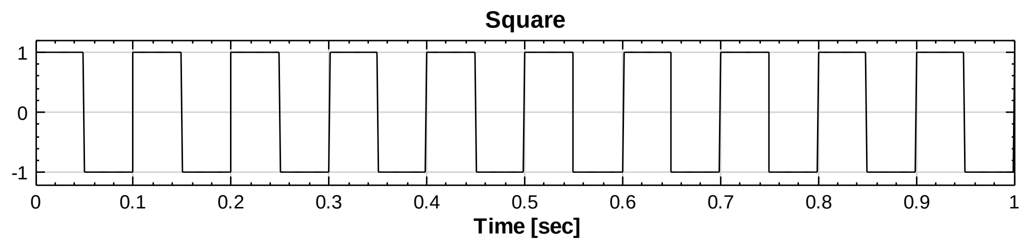
`SamplingFreq`	Set the number of Sampling Frequency (the amount of data values to be sampled)	1000
`DataLength`	Set the length o the data (`SamplingFreq` × `TimeLength` + 1)	1001
`SignalFreq`	Set the real signal frequency. The unit is in Hz.	10
`Amplitude`	Set the maximum displacement of a periodic wave	1
`AmplitudeOffSet`	Set the amplitude offset	0
`Phase`	Set the `Phase` in degree. When the phase is non-zero, the entire waveform appears to be shifted in time with specified value	0°
`Symmetry`	`Symmetry` set at 0.5 is equal symmetry where the left of the inflection point takes up 0.5 (half) of the period. E.g. `Symmetry` – 0.2 means that the left of the inflection point takes up only one-fifth of the period.	0.5
`TimeStart`	Set the start time for the data	0

Example

Create a square wave.

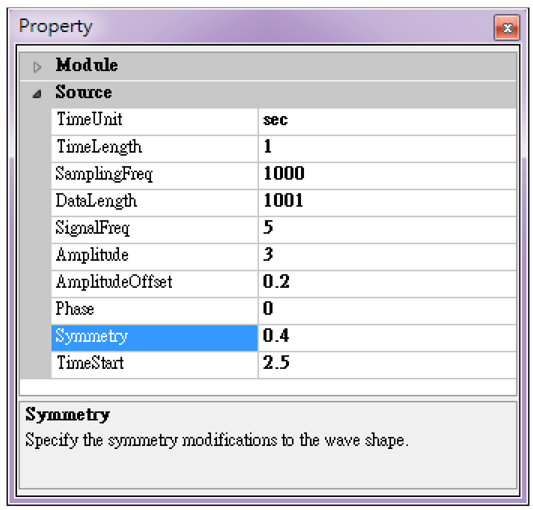
1. Create **`Source→Square Wave`**.

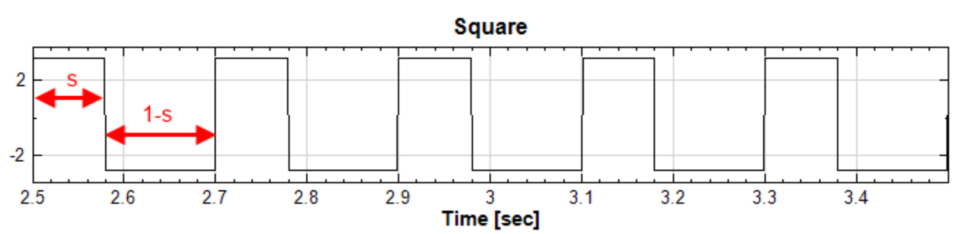




In this figure, the `SamplingFreq` is set to 1000 and `SignalFreq` is set to 10.

2. Set the *`SignalFreq`* to 5, *`Amplitude`* to 3, *`AmplitudeOffSet`* to 0.2, *`Phase`* to 0, *`Symmetry`* to 0.4, *`TimeStart`* to 2.5, the graph is shown in the image below.





If you want to see an example project using the **`Square`** function, open project demo14 in C:\Program Files\AnCAD\Visual Signal\demo\Basic.

Related Functions

`Channel Viewer`

4.3.5 Triangle Wave

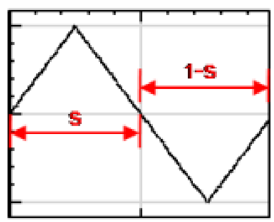
Explanation is given for the **Source→Triangle Wave**.

Introduction

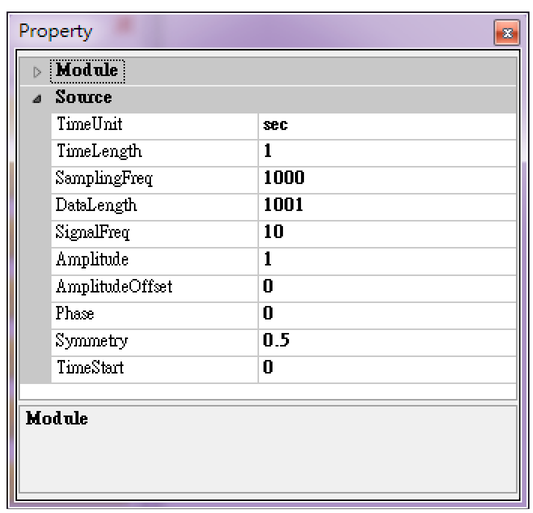
$$x_{tri}(t) = \begin{cases} \dfrac{Af}{s}t + V_0, & \theta \le t < \left(\dfrac{1}{f}\right)s + \theta \\ A\left[1 - \dfrac{f}{1-s}\left(t - \dfrac{s}{f}\right)\right] + V_0, & \left(\dfrac{1}{f}\right)s + \theta \le t \le \left(\dfrac{1}{f}\right) + \theta \end{cases}$$

$$x_{tri}(t+T) = x_{tri}(t)$$

Where A = amplitude, f = sampling frequency, θ = phase at t_0, V_0 = offset from X axis, the ratio s is shown in the image below.



Properties



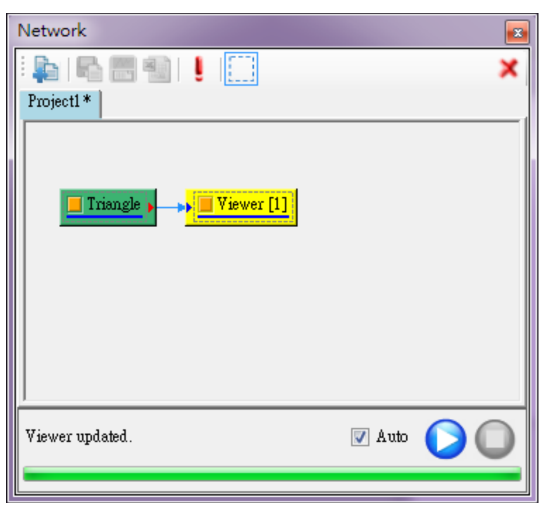
{Source} Property Name	Property Definition	Default Value
TimeUnit	Set the time in *ps*, *ns*, *us*, *ms*, *sec*, *min*, *hour*, *day*, *week*, *month* ,or *year*	*sec*

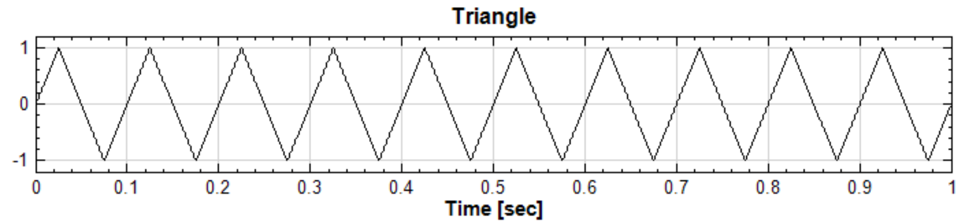
`TimeLength`	Set the value of time selected in `TimeUnit`	1
`SamplingFreq`	Set the number of Sampling Frequency (the amount of data values to be sampled)	1000
`DataLength`	Set the length o the data (`SamplingFreq` × `TimeLength` + 1)	1001
`SignalFreq`	Set the real signal frequency. The unit is in Hz.	10
`Amplitude`	Set the maximum displacement of a periodic wave	1
`AmplitudeOffSet`	Set the amplitude offset	0
`Phase`	Set the `Phase` in degree. When the phase is non-zero, the entire waveform appears to be shifted in time with specified value	0^o
`Symmetry`	`Symmetry` set at 0.5 is equal symmetry where the left of the inflection point takes up 0.5 (half) of the period. E.g. `Symmetry` – 0.2 means that the left of the inflection point takes up only one-fifth of the period.	0.5
`TimeStart`	Set the start time for the data	0

Example

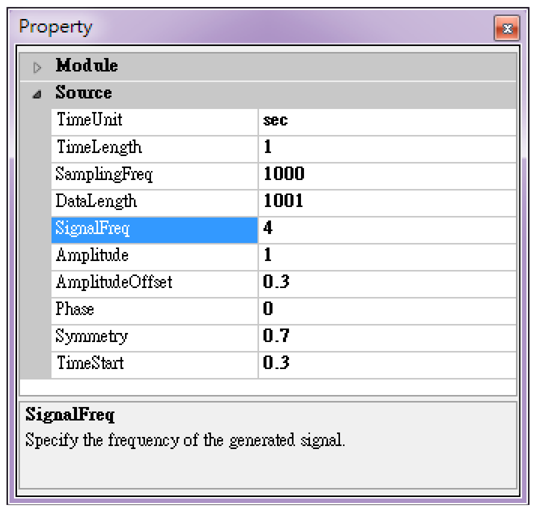
Create a Triangle wave.

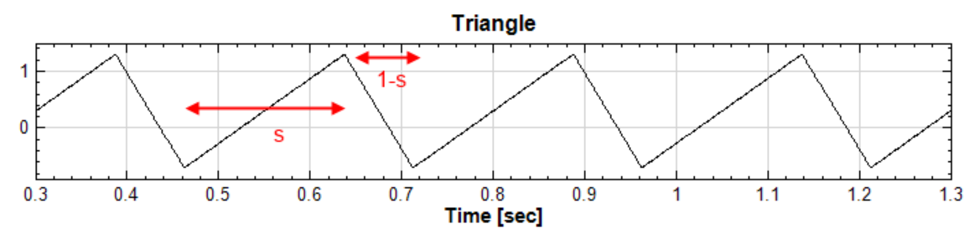
1. Create **`Source→Triangle Wave`**.





2. Set the `SignalFreq` to 4, `Amplitude` to 1, `AmplitudeOffSet` to 0.3, `Phase` to 0, `Symmetry` to 0.7 and `TimeStart` to 0.3, the graph is shown in the image below.





If you want to see an example project using the **`Triangle`** function, open project demo14 in C:\Program Files\AnCAD\Visual Signal\demo\Basic.

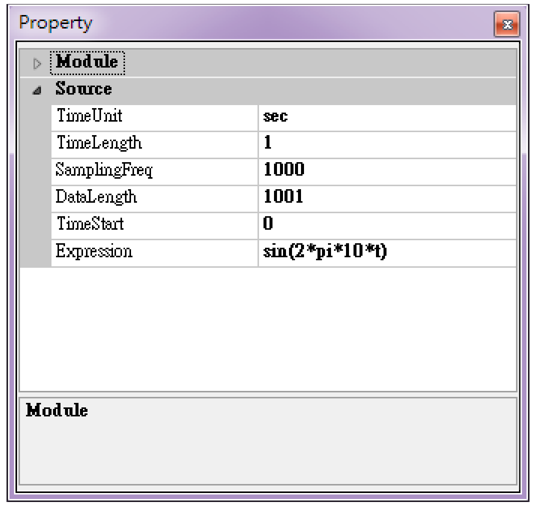
Related Functions

`Channel Viewer`

4.3.6 Custom Wave

The users can input equations to create signals via this module.

Properties



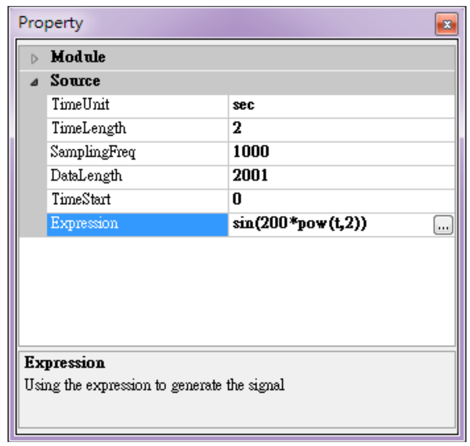
{Source} Property Name	Property Definition	Default Value
`TimeUnit`	Set the time in *ps*, *ns*, *us*, *ms*, *sec*, *min*, *hour*, *day*, *week*, *month* ,or *year*	*sec*
`TimeLength`	Set the value of time selected in `TimeUnit`	1
`SamplingFreq`	Set the number of Sampling Frequency (the amount of data values to be sampled)	1000
`DataLength`	Set the length to the data (`SamplingFreq` × `TimeLength` + 1)	1001
`TimeStart`	Set the start time for the data	0
`Expression`	Set the equations to calculate the signal	sin(2*pi*10*t)

Note: The expression area can use inputs of sin, cos, tan, exp, and asin etc. math functions, which are the same as functions in the f_n menu of the **Math** module. (Please also refer to reference of math functions for C# language). Note the expression of power, a^b is written as pow(a,b).

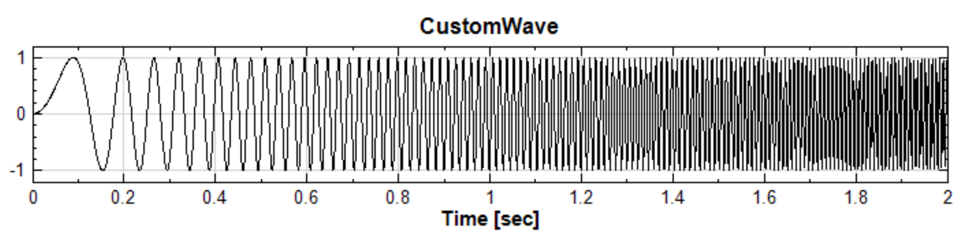
Example

Build a quasi-steady signal in which the time is a direct ratio with the frequency:

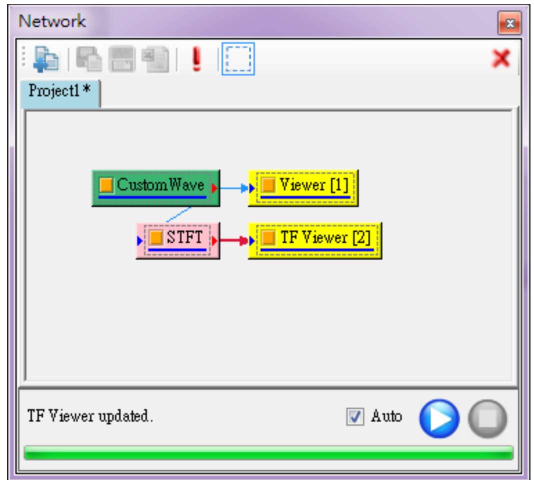
1. Under the menu of **Source→Custom Wave**, set the *TimeLength* to be 2. Then set the *Expression* property to 'sin(200*pow(t,2))'. The setting of the *Expression* is shown below:

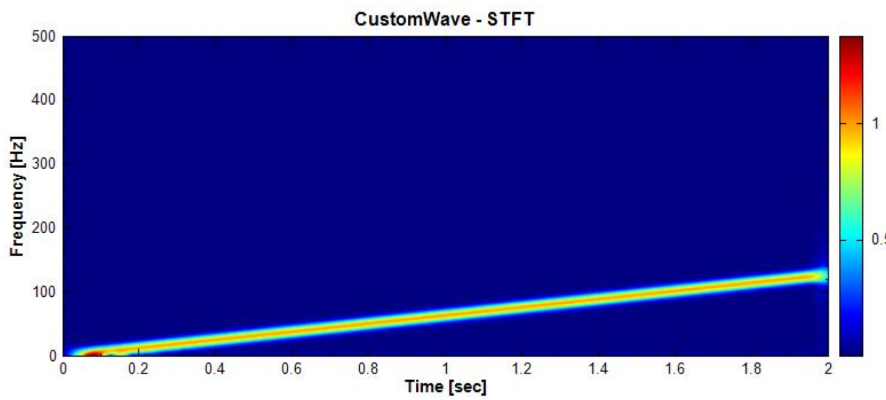


2. View the function with a **Channel Viewer**. The figure is shown below:

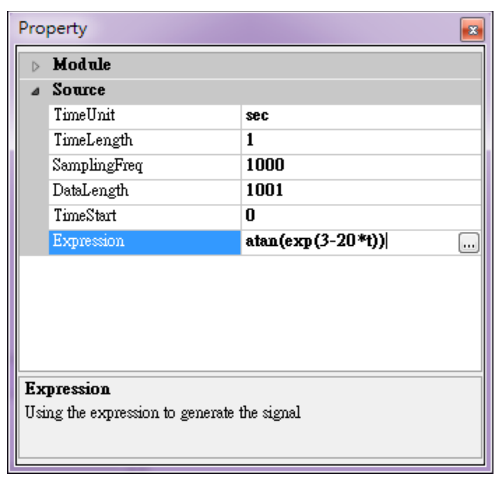


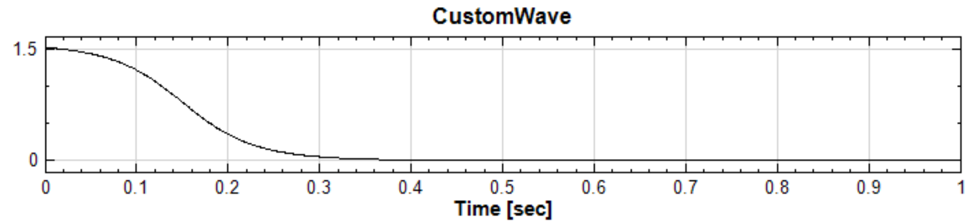
3. Then, validate that the frequency has direct ratio with time using the **ShortTerm Fourier Transform**. The setting and the frequency-time figure is shown as below:





The user can also build a wave of $a\tan^{-1}(e^{3-20t})$





If you want to see an example project using the **Custom Wave** function, open project demo14 in C:\Program Files\AnCAD\Visual Signal\demo\Basic

Related Functions

Channel Viewer, ShortTerm Fourier Transform, Math

References

http://msdn.microsoft.com/en-us/library/system.math_methods.aspx

4.4 Viewer Of Signal Flow Object

4.4.1 Channel Viewer

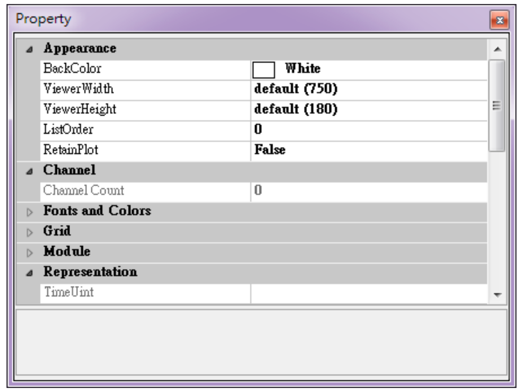
The purpose of the **Channel Viewer** is to convert signal data to be graphically displayed onto the **Visualization Window**. The graph will plot each signal data along the x-axis.

Properties

This module accepts input of Signal (which can be a real number or complex number, single channel or multi-channel, Regular or Indexed), Audio (which can be a real number or complex number, single channel or multi-channel, Regular). **Channel Viewer** can accept multiple input data sources.

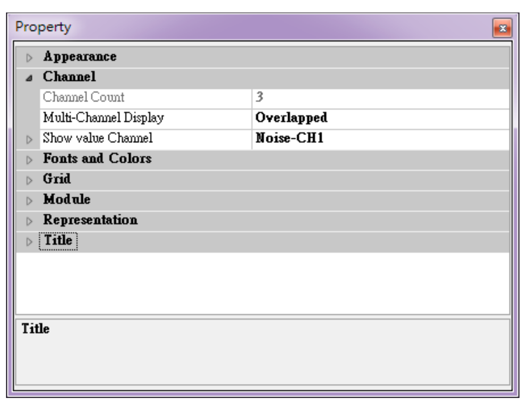
1. Appearance

The Appearance property contains the options to set the appearance of how the graph of the **Channel Viewer** will be shown on the **Visualization Window**.



{Appearance} Property Name	Property Definition	Default Value
BackColor	Set the background color of the graph displayed in the **Visualization Window**	White
ViewerWidth	Set the width of the graph in pixels	750
ViewerHeight	Set the height of the graph in pixels	180
ListOrder	Set the order of the graph to be shown on the **Visualization Window**	The default position on the **Visualization Window** is based on the order that the **Channel Viewer** is created
RetainPlot	Set True to retain previous plot	False

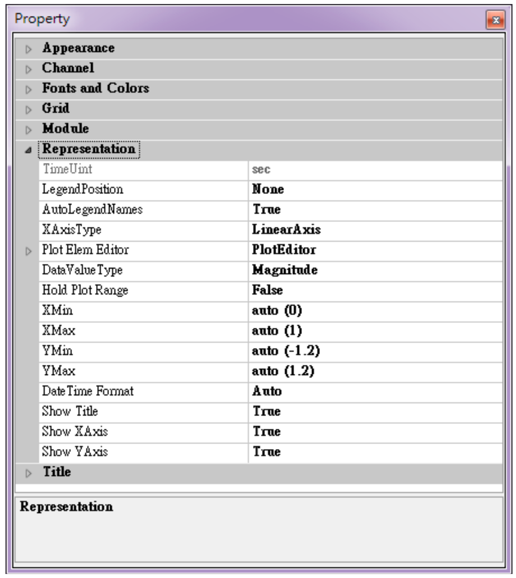
2. Channel



{Channel} Property Name	Property Definition	Default Value

`Channel Count`	Displays the number of input signals currently connected to the **`Channel Viewer`**	(Cannot be edited)
`Multi-Channel Display`	Select from the option `Overlapped` (to display the graphs of the input signals on the same graph overlapping each other) or `List` (to display the graphs on top of one another)	`Overlapped`
`Show value Channel`	When there are multiple inputs, select the channel (graph) from the drop down menu to use the **Show Value** button on the **Visualization Window** toolbar. When there are multiple inputs, knowing which graph shows what value can be rather difficult. So selecting a channel from the drop down menu, the user can specify the graph to perform the **Show Value** button (located on the **Visualization Window** toolbar)	Channel 1

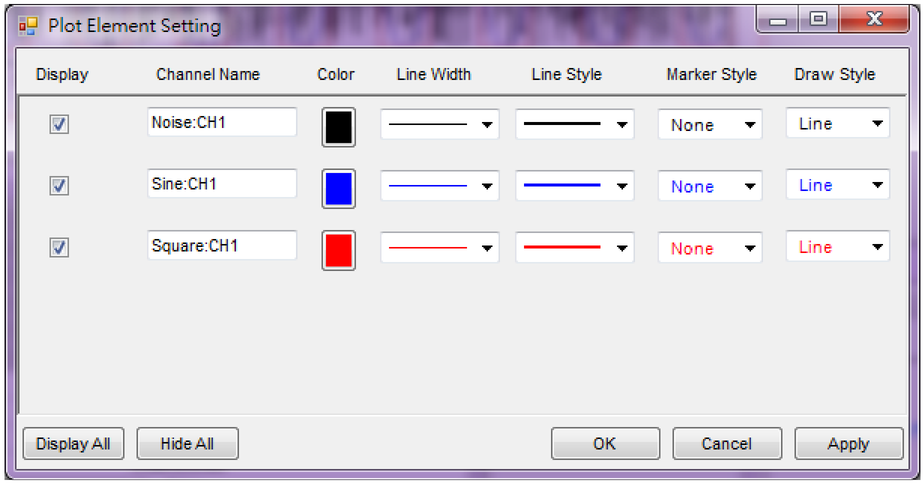
3. Representation



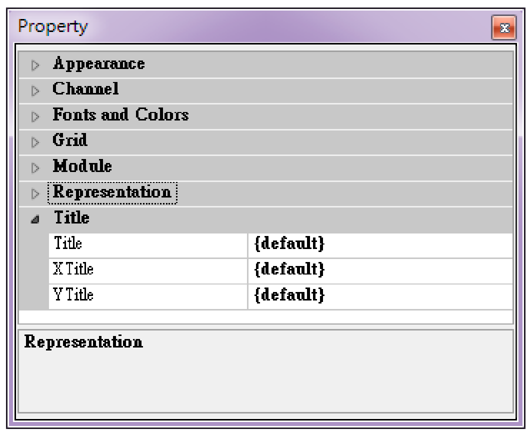
{Representation} Property Name	Property Definition	Default Value
`TimeUnit`	Displays the time unit of the data	Depends on the input signal data's time unit
`LegendPosition`	Select the position: `None`, `TopLeft`, `BottomLeft`, `TopRight`, `BottomRight` and `RightOutSide` to display the legend on the graph. Or select `Custom` to position the legend via drag-and-drop.	`None`
`AutoLegendNames`	Set True to automatically retrieve legend names; otherwise they are taken from `Plot Elem Editor`	True
`XAxisType`	Select the representation of the x-axis, choose between `LinearAxis` and `LogAxis`	`LinearAxis`
`Plot Elem Editor`	Click on the `PlotEditor` button next to the field to edit how the graph is displayed, from the line color, line thickness, dot representation, etc. **Note**: User will need to click on the Plot Elem Editor field for the button to appear or double-click the **`Channel Viewer`**	default
`DataValueType`	Select different ways to display the y-axis from a selection of	`Magnitude`

	`Magnitude`, `Phase`, `RealPart`, `ImagPart`, `Gain` and `PowerSpectrum`. Normally this option is used for spectrum data. When there are multiple channels in the signal.	
`GainReference`	If `DataValueType` is set as `Gain`, this option field will appear. See also **`Map To Real`**	1
`Hold Plot Range`	When `Hold Plot Range` is set as True, after resizing, moving and zooming into the graph, the calculation done will still be based on the original range.	False
`Xmin`	Set the minimum value of the x-axis	auto
`Xmax`	Set the maximum value of the x-axis	auto
`Ymin`	Set the minimum value of the y-axis	auto
`Ymax`	Set the maximum value of the y-axis	auto
`Date Time Format`	Set a date-time format. There are `Auto`, `WeekdayOnly`, `MonthOnly`, `YearOnly`, `YearMonth`, `YearMonthDay`, and `Custom` six options.	`Auto`
`FormatString`	Specify a custom date-time format string	yyyy/MM/dd
`Show Title`	Select True to show the title on the graph and False to hide the title	True
`Show XAxis`	Select True to show the x-axis on the graph and False to hide the x-axis	True
`Show YAxis`	Select True to show the y-axis on the graph and False to hide the y-axis	True

Clicking on the `Plot Elem Editor` button will pop up the **Plot Element Setting** window. Check the **Display** tick box to show the signal on the graph (useful to determine which is which when there are multiple signals on one graph). You can change the **Channel Name**, Line **Color**, **Line Width**, **Line Style**, **Marker Style**, and **Draw Style** to improve the presentation of the graph and customize the looks according to your need.



4. Title



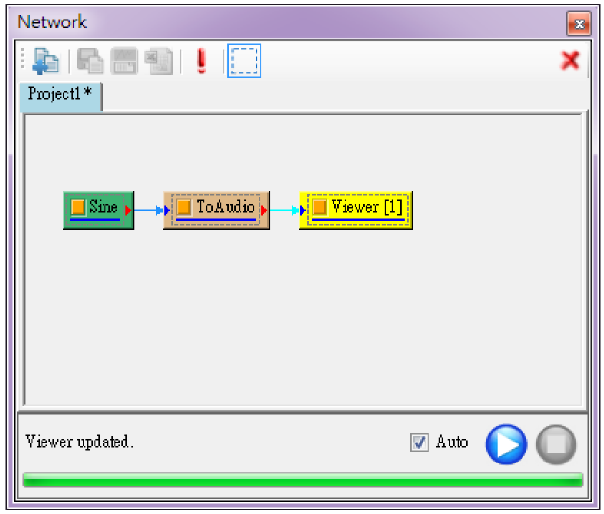
{Title} Property Name	Property Definition	Default Value
`Title`	Change the title of the graph	{default}
`XTitle`	Change the title of the x-axis	{default}
`YTitle`	Change the title of the y-axis	{default}

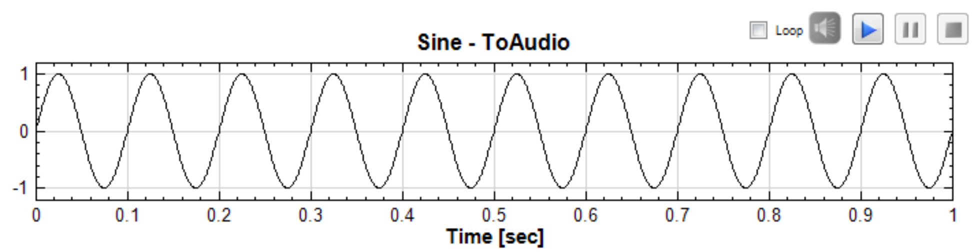
Example

A demonstration of how to use the audio player within a **`Channel Viewer`**.

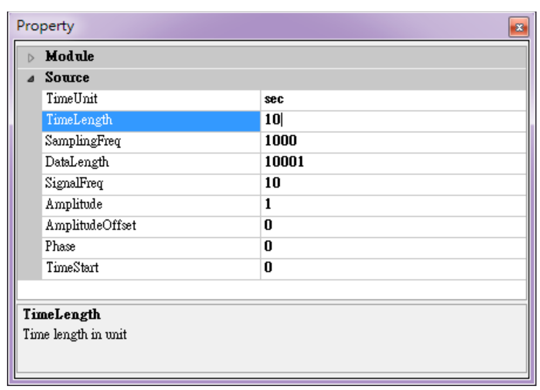
1. Create **`Source`** → **`Sine Wave`** and connect the **`Sine`** component to **`Conversion`**→**`Convert To Audio`** to turn the sine wave signal to an

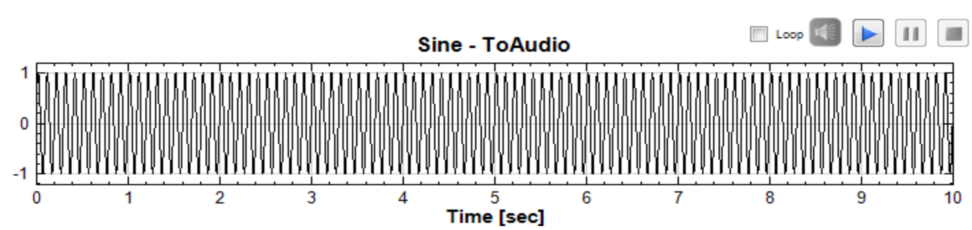
Audio file. Then connect the **ToAudio** to **Viewer→Channel Viewer** to display the graph and the audio playback on the **Visualization Window**.



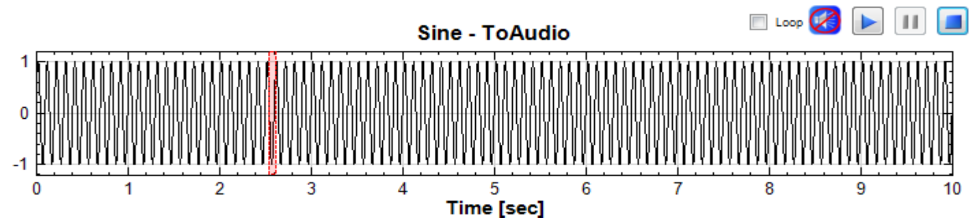


2. Change the *TimeLength* of the **Sine** component to 10.

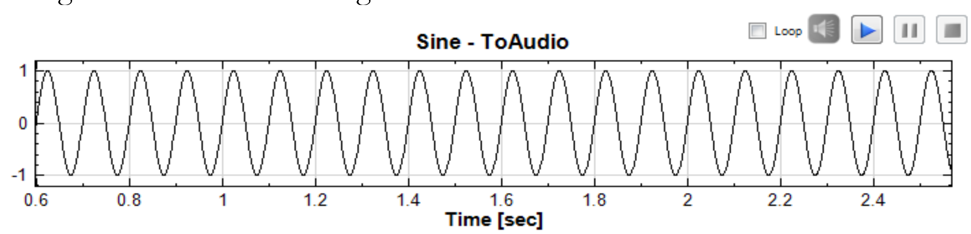




3. Click on the  audio play button on the top right corner of the graph and play the signal. A red line will run through the x-axis indicating the position of the audio currently being played.

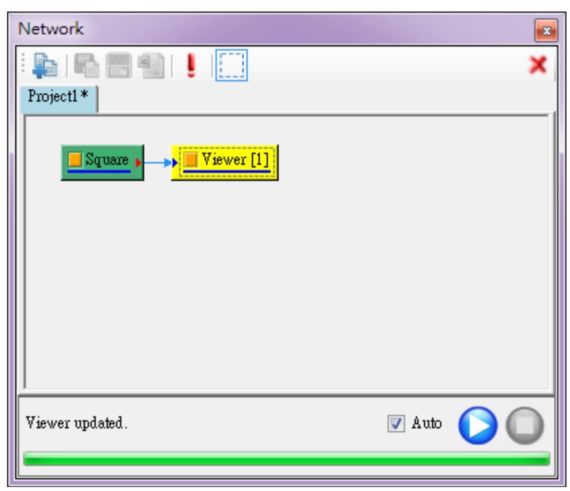


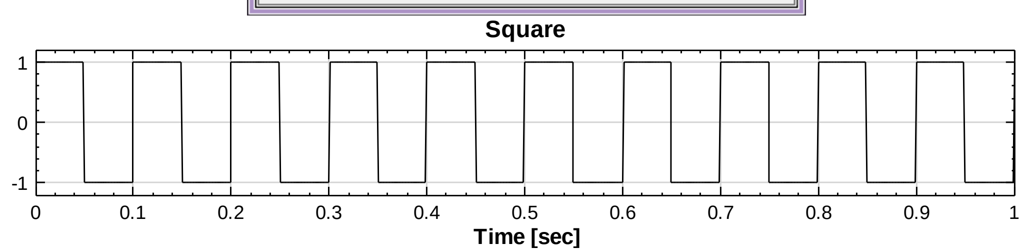
4. You can use the  **Zoom X** button off the **Visualization Window** toolbar to enlarge the area of the audio signal.



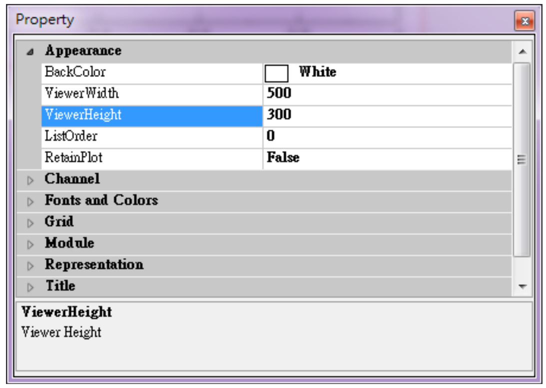
Below are some examples showing how to configure the other options in the **Properties Window**.

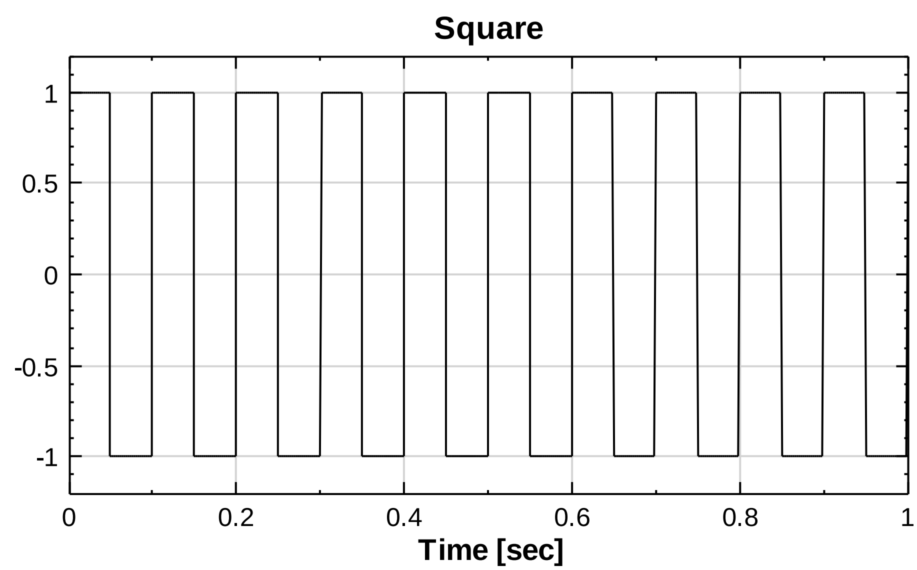
1. Create **`Source`→`Square Wave`** and connect it to **`Viewer`→`Channel Viewer`**.



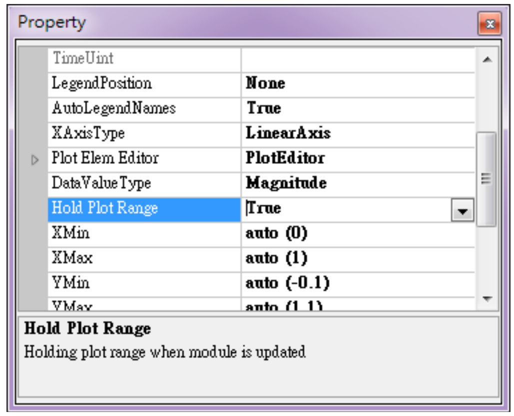


2. Change the `ViewerHeight` to 500 and `ViewerWidth` to 300 in the **`Channel Viewer`** properties.

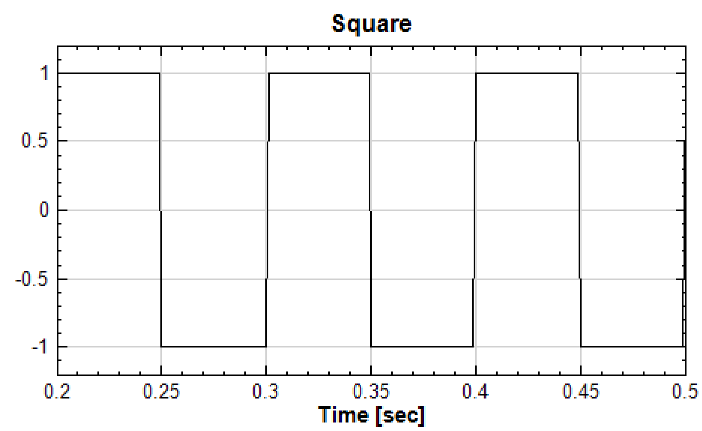




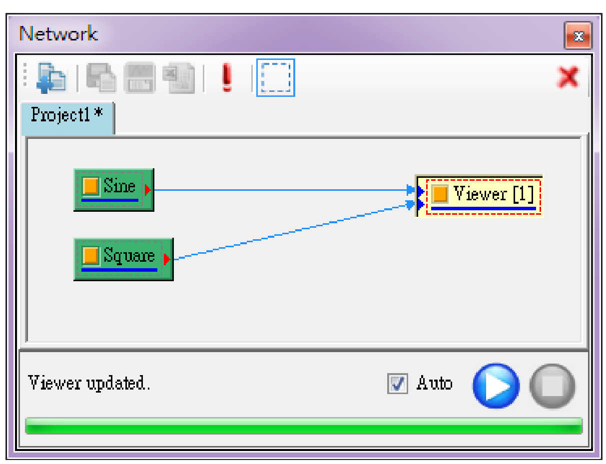
3. Use the **Zoom X**, **Zoom Y** or **Pan X** and **Pan Y** feature in the **Visualization Window** toolbar. If you want to maintain the current status, you can set the `Hold Plot Range` to True.

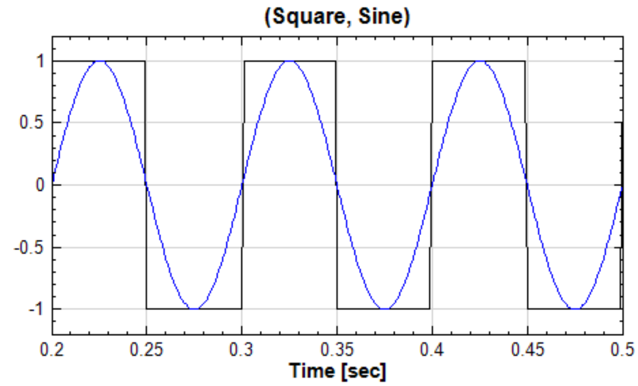


4. Use **Zoom X** and zoom in between 0.2 sec and 0.5 sec on the x-axis.

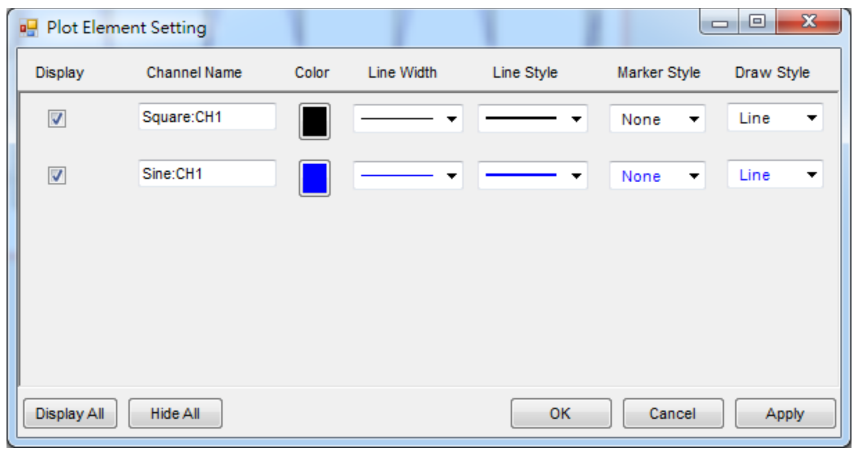


5. Create **`Source→Sine Wave`** and connect it to the same **`Channel Viewer`**. Because the **Auto** box is checked, the **`Channel Viewer`** will automatically update with the new sine wave graph. Since *`Hold Plot Range`* is set to True, the new update will not return the graph to the default position.

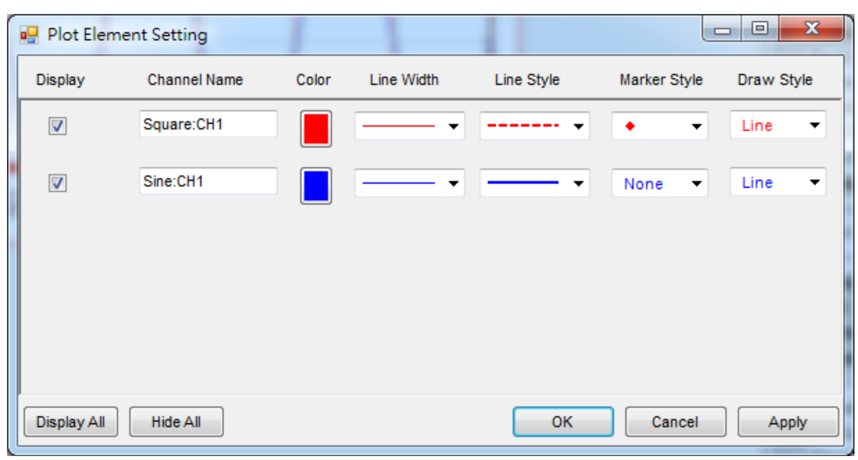


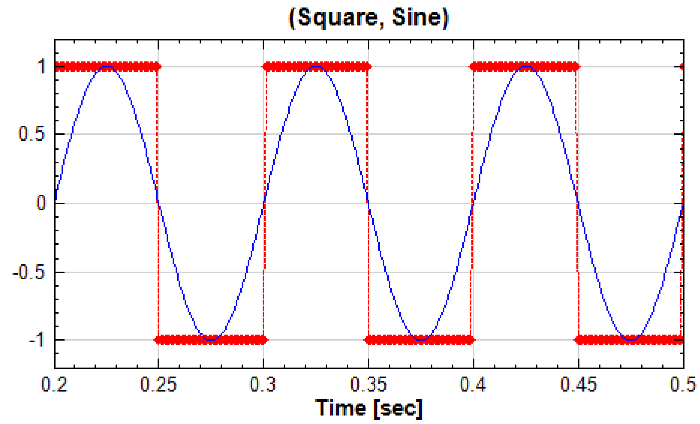


6. Click on the `Plot Elem Editor`, and then click on the button to open up the **Plot Element Setting** window.



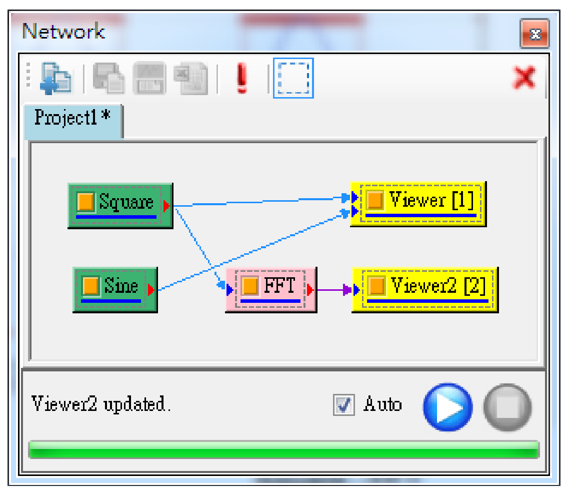
7. In the **Plot Element Setting** window, you can edit the display of all the input channel data on the graph. Set the Line **Color** of Square:CH1 to red, change the **Line Style** to dotted line and change the **Marker Style** to ◆ and click on the **Apply** button to see the change on the graph.

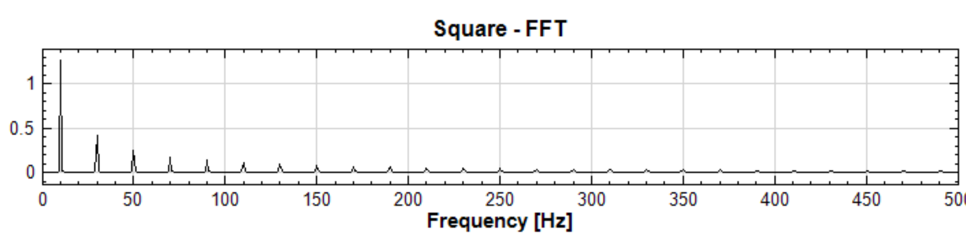




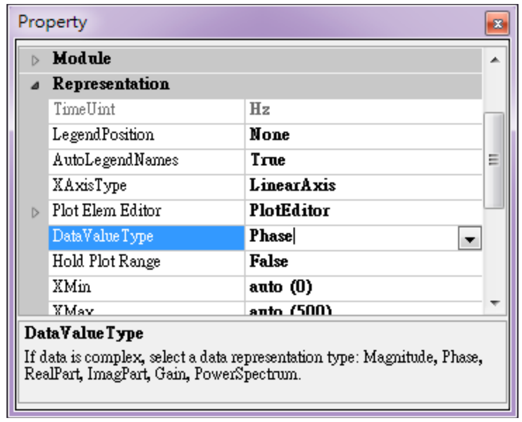
In the **`Channel Viewer`** you have the option to configure the *`DataValueType`* of the spectrum data to *`Magnitude`*, *`Phase`*, etc.

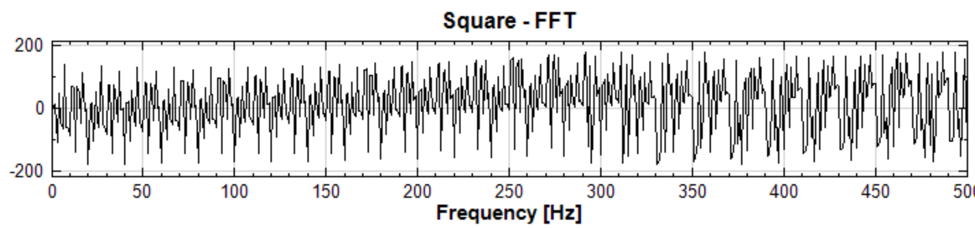
8. Continuing from the above example, connect the **`Square`** component to **`Compute→Transform→Fourier Transform`** and then connect it to a **`Channel Viewer`**.



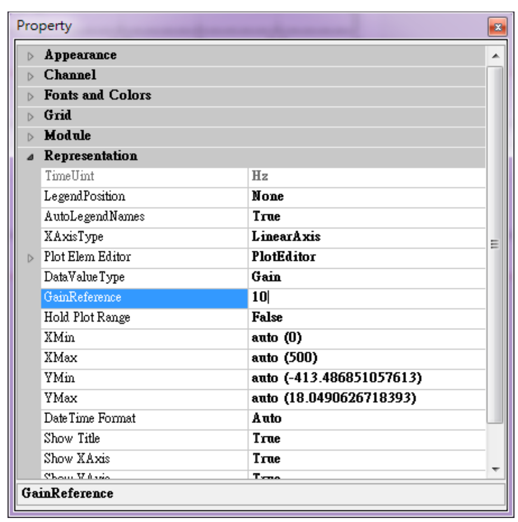


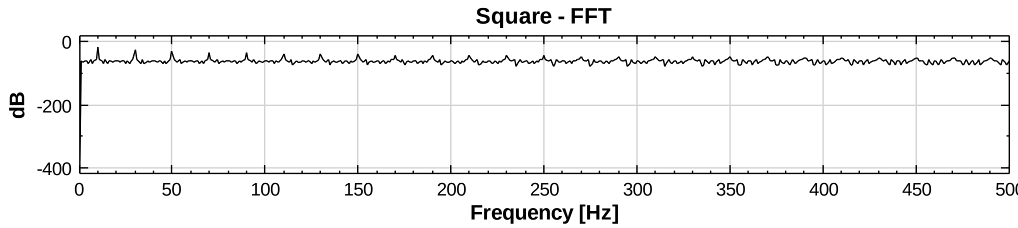
9. In the graph above, the x-axis is the frequency and the y-axis *`DataValueType`* is set as *`Magnitude`*. Now change the *`DataValueType`* to *`Phase`*, the y-axis will now represent the frequency of each phase.





Note: When `DataValueType` is changed to `Gain`, an additional option `GainReference` will appear and set the `GainReference` to 10. `Gain` is defined as $20 \times \log\left(\frac{A}{\mathrm{GainReference}}\right)$, unit is in dB, $\log$ is to the base of 10, A is the magnitude and the dominator is `GainReference`.





Related Functions

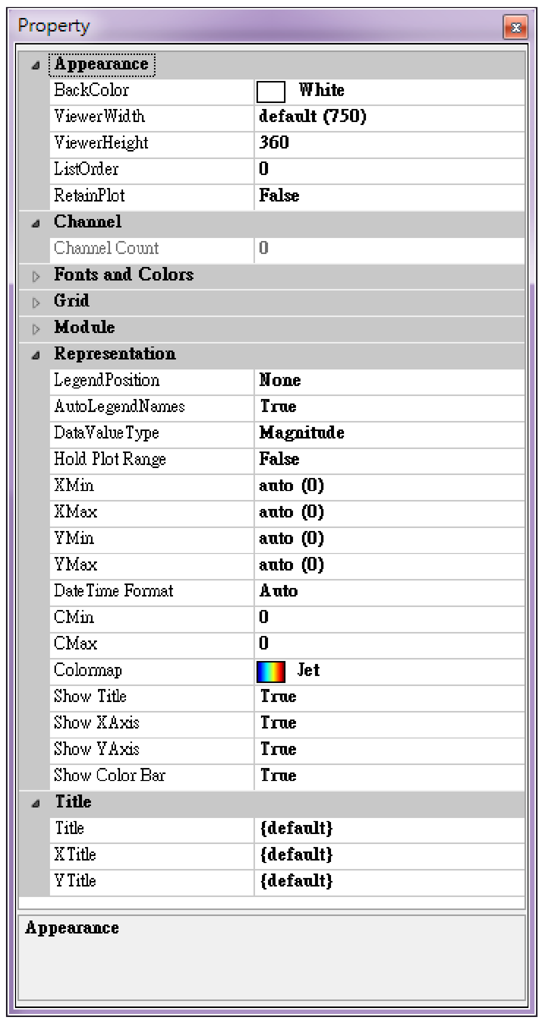
`Sine Wave, Square Wave, Fourier Transform, Map To Real`

4.4.2 Time-Frequency Viewer

Time-Frequency Viewer uses images to display three dimensional time-frequency signals (time, frequency and signal strength). The x-axis represents the time, the y-axis represents the frequency and the color represents the signal strength.

Properties

This module accepts input of Spectra (which could be a real number or complex number, single channel, Regular). **Time-frequency Viewer** and **Channel Viewer** are very similar with the difference being that there are more variable options for **Time-Frequency Viewer**.



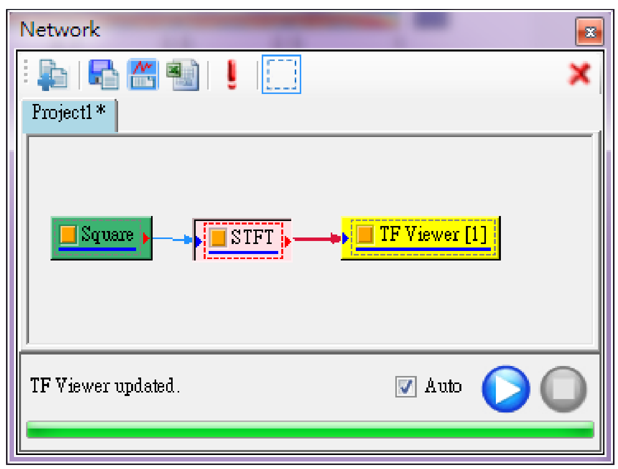
Property Name	Property Definition	Default Value
CMin	Set the minimum value of the time-frequency color	Auto
CMax	Set the maximum value of the	Auto

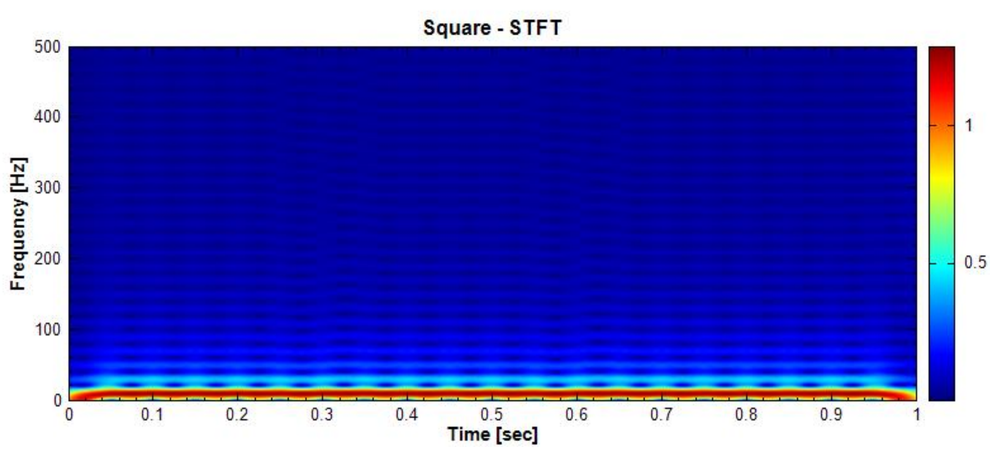
	time-frequency color	
Colormap	There are four types of color representations: *Jet*, *HSV*, *Rainbow* and *Gray*	*Jet*
Show Color Bar	Select whether or not to display the color bar at the right side of the graph	False

Example

Create a **Square Wave** component, connect it to a **ShortTerm Fourier Transform** component, and then connect it to a **Time-frequency Viewer** component. Then change some configurations to the **Time-Frequency Viewer**.

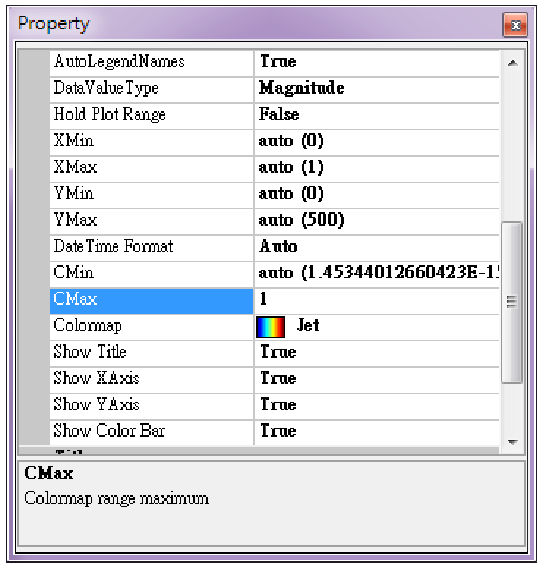
1. Create **Source→Square Wave** and connect it to **Compute→TFA→ShortTerm Fourier Transform** and then connect it to a **Viewer→Time-Frequency Viewer** component.

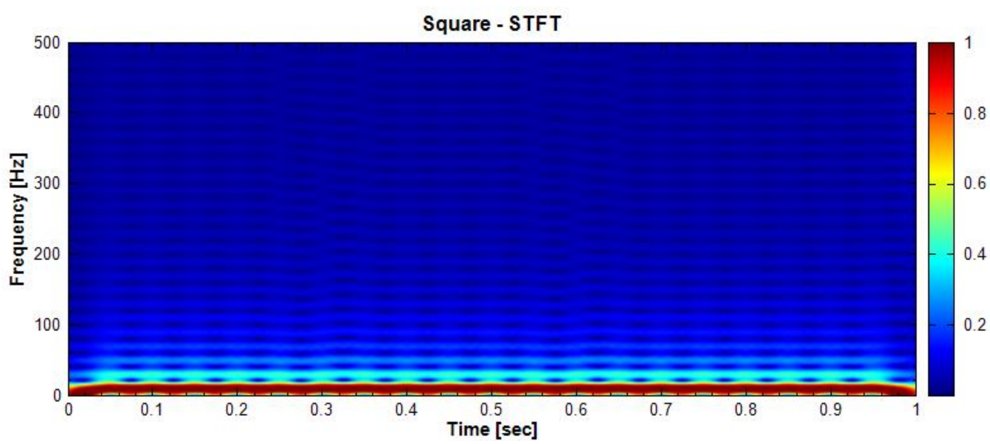




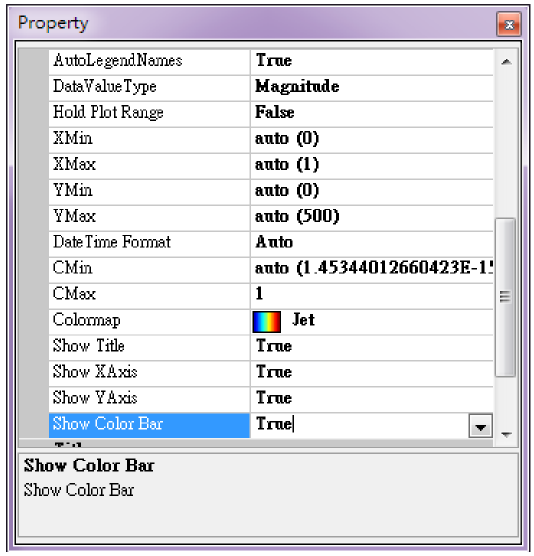
Set the `CMax` to `auto(1.28581059903091)`. This value is the maximum value of the signal strength and it is also the maximum color value on the color map. A user can set the value of the `CMax` variable to show the signal strength below this value. Since the colors on the color map keep the same, a better resolution of the signal strength can be presented if `CMax` becomes smaller.

Set `CMax` to 1, the graph uses this as the maximum color value for color map. All signal strength below 1 is remapped to the color map and the graph is redrawn to focus on the region that was unclear when `CMax` was 1.28581059903091.

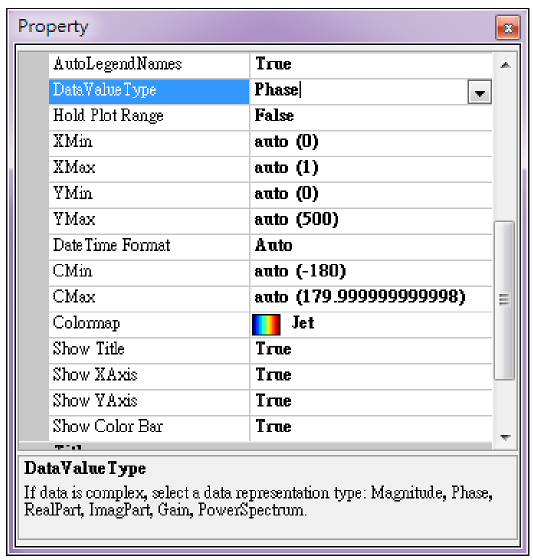


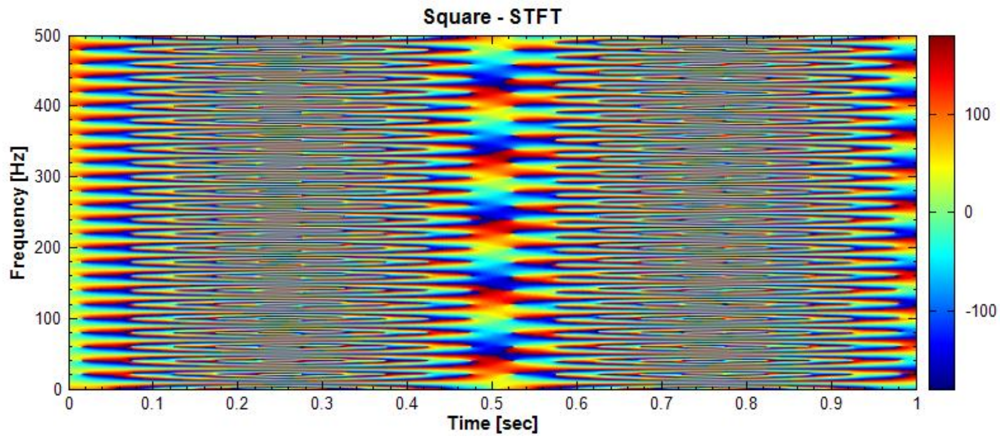


2. Set the `Show Color Bar` to `True` will display a color bar legend based on relationship between the color and its values.



3. Change the **`Time-Frequency Viewer`**'s *`DataValueType`* to *`Phase`* and the following image will be displayed.





Related Functions

Square Wave, ShortTerm Fourier Transform, Channel Viewer, Map to Real

4.4.3 XY Plot Viewer

Displays two signal data in one viewer, one corresponding to the x-axis and the other corresponding to the y-axis.

Introduction

`XY Plot` Viewer accepts three main signal data:

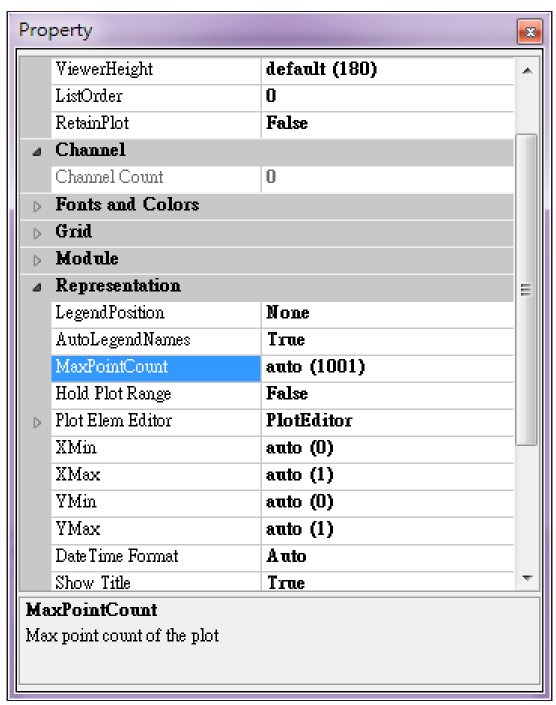
1. Two signal data, channel 1 is drawn on the x-axis and channel 2 is drawn on the y-axis and then the two signals are plotted on the graph.

2. Multi-Channel data with odd channels are drawn on the x-axis and even channels are drawn on the y-axis and then the channels are plotted on the graph.

3. A single channel with complex data, the real part is drawn on the x-axis and the imaginary part is drawn on the y-axis and the two values are plotted on the graph.

Properties

This module accepts input of Signal (which could be a real number or complex number, single channel or multi-channel, Regular or Indexed), Audio (which could be a real number or complex number, single channel or multi-channel, Regular). **`XY Plot`** Viewer and **`Channel Viewer`** are very similar with the difference being that there are more variable options for **`XY Plot`** Viewer.

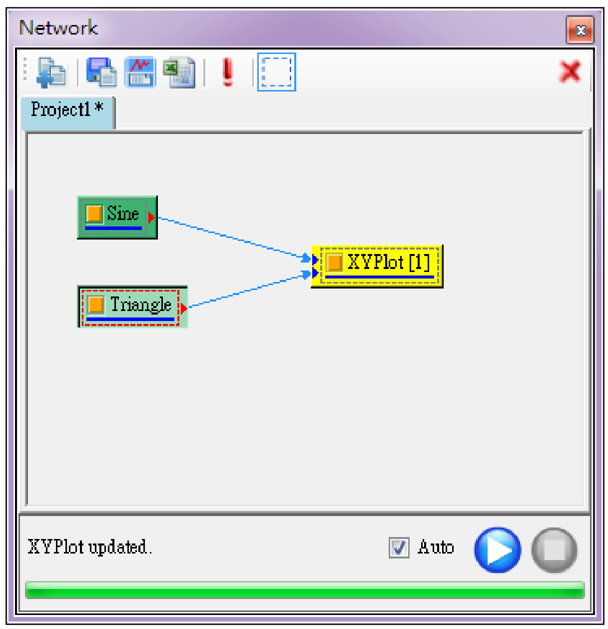


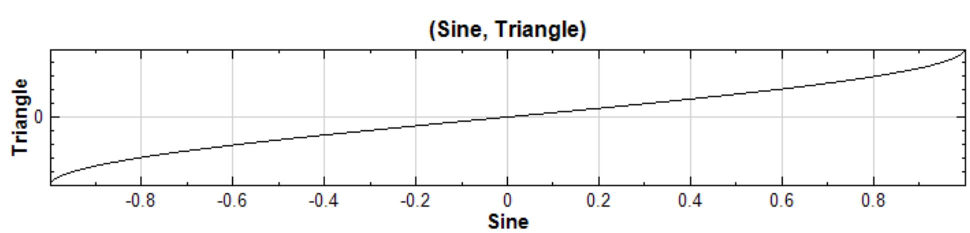
{Representation} Property Name	Property Definition	Default Value
MaxPointCount	The number of points to be drawn	1001

Example 1

Sine wave is drawn on the axis and triangle wave is drawn on the y-axis and then use the **XY Plot** Viewer to display the graph.

1. Create **Source→Sine Wave** and then create **Source→Triangle Wave** and connect both signal data to **Viewer→XY Plot Viewer**.





Related Functions

Sine Wave, Square Wave, Channel Viewer, Time-Frequency Viewer

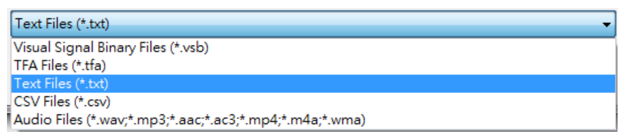
4.5 Writer For Signal Flow Object

4.5.1 Write Data & Export to Excel

The **Export Data** and **Export to Excel** functions allow you to export or save Visual Signal information into numerous types of file formats. Both of these functions can also be found in the **Network Window** toolbar.

Properties

Write Data can save data to five different file types: Visual Signal Binary Files (*.vsb), Time Frequency Analysis Files (*.tfa), Text Files (*.txt), Comma Separated Value Files (*.csv), and Audio Files (*.wav, *.mp3, *.aac, *.ac3, *.mp4, *.m4a, *.wma).



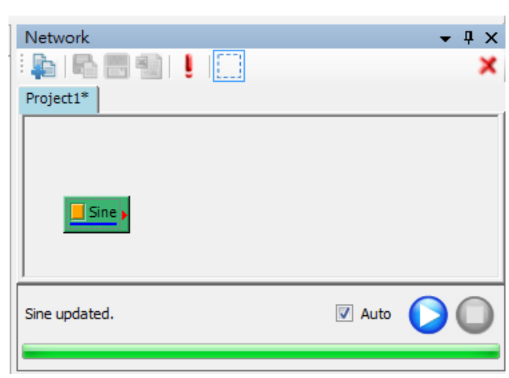
Export to Excel will export the data into Microsoft® Excel®. The first column will be the X values and from the second columns onwards represents the number of channels you have. Each row in the file contains the data values of the signal. Both **Export Data** and **Export to Excel** can save the information created by all Signal Flow Objects except the **Viewer** components.

Example

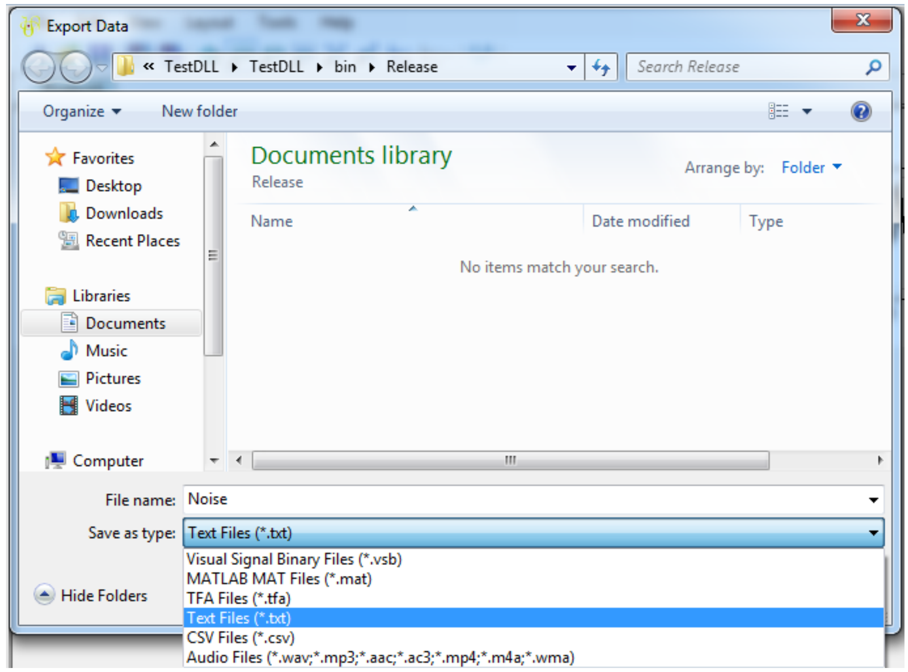
Note: The following examples use the **Save data to file** and **Export data to Excel** functions in the **Network Window** toolbar.

In the following examples, demonstrations are given on how to save a real signal, spectrum signal, spectra signal and numeric signal to a file.

1. Real Signal



Click on the Sine Wave component on the **Network Window** and then click on the  **Save data to file** button on the **Network Window** toolbar to save the information. After clicking on the **Save data to file** button an **Export Data Window** will appear, allowing the user to save different types of file formats.



If you click on the  **Export data to Excel** button, Microsoft® Excel® will automatically open with all the data transferred to a Microsoft® Excel® table. The X Value column stores the time information of the signal and the CH 1 column stores the data values. So if there is more than one channel, e.g. CH2, CH3 etc. then each channel will be listed in their own column.

Spectrum Signal

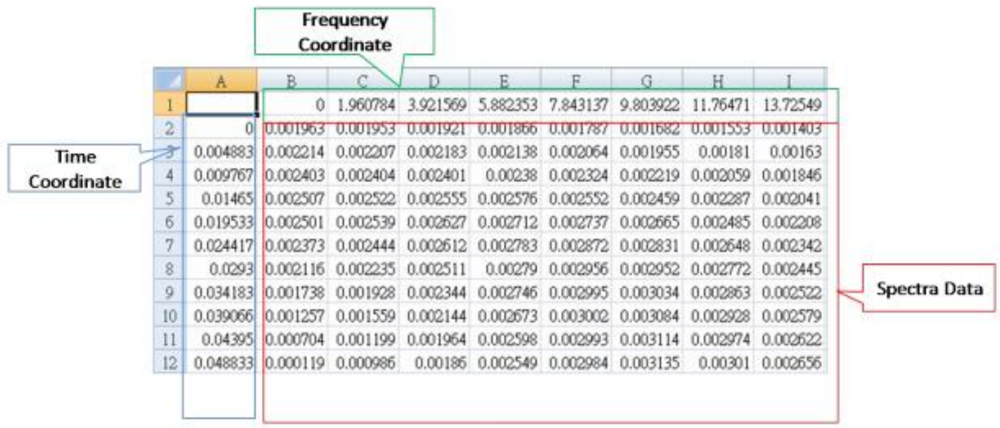
Export data to Excel and **Save data to file** on a spectrum signal will result in an output which looks something like this:

X Value	CH 1 - Real	CH 1 - Imag
0	2.875E-17	0
1	6.319E-07	-0.0002013
2	2.606E-06	-0.0004152
3	6.186E-06	-0.000657
4	1.191E-05	-0.0009488
5	2.084E-05	-0.0013279
6	3.515E-05	-0.0018665
7	5.999E-05	-0.0027303
⋮	⋮	⋮

The X Value column stores the time information of the signal and CH 1 – Real column stores the real part values, while CH1 – Imag column stores the imaginary part. If there is more than one channel, the information will be listed in the same way.
Visual Signal will view a spectrum signal as a multi-channel. The X Value will store the frequency and the rest will be the same as above.

2. Spectra

Export data to Excel with a Spectra signal will result in something like this:



	A	B	C	D	E	F	G	H	I
1		0	1.960784	3.921569	5.882353	7.843137	9.803922	11.76471	13.72549
2	0	0.001963	0.001953	0.001921	0.001866	0.001787	0.001682	0.001553	0.001403
3	0.004883	0.002214	0.002207	0.002183	0.002138	0.002064	0.001955	0.00181	0.00163
4	0.009767	0.002403	0.002404	0.002401	0.00238	0.002324	0.002219	0.002059	0.001846
5	0.01465	0.002507	0.002522	0.002555	0.002576	0.002552	0.002459	0.002287	0.002041
6	0.019533	0.002501	0.002539	0.002627	0.002712	0.002737	0.002665	0.002485	0.002208
7	0.024417	0.002373	0.002444	0.002612	0.002783	0.002872	0.002831	0.002648	0.002342
8	0.0293	0.002116	0.002235	0.002511	0.00279	0.002956	0.002952	0.002772	0.002445
9	0.034183	0.001738	0.001928	0.002344	0.002746	0.002995	0.003034	0.002863	0.002522
10	0.039066	0.001257	0.001559	0.002144	0.002673	0.003002	0.003084	0.002928	0.002579
11	0.04395	0.000704	0.001199	0.001964	0.002598	0.002993	0.003114	0.002974	0.002622
12	0.048833	0.000119	0.000986	0.00186	0.002549	0.002984	0.003135	0.00301	0.002656

The first column represents the time, the first row represents the frequency, and the data in between the first column and the first row represents the signal strength.

Related Functions

`Sine Wave, Data Writer`

4.5.2 Data Writer

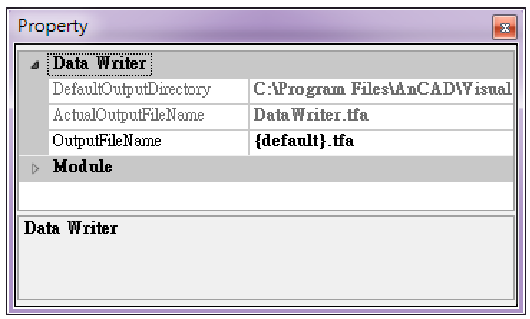
The **`Data Writer`** function allows you to save data to supported file formats.

Introduce

This is similar to the **Save data to file** function introduced in Section 4.5.1. The difference between **`Data Writer`** and **Save data to file** is that **`Data Writer`** is "auto" and **Save data to file** is "manual". **`Data Writer`** can be a component in the **Network Window** and will automatically save data to file when **Network Window** is updated. It will be saved to the folder you specified with the included file name and file format.

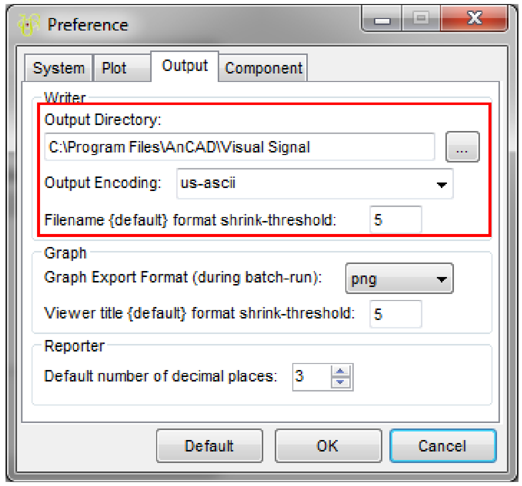
Properties

This module accepts all kind of data type. It is with three properties.



{Data Writer} Property Name	Property Definition	Default Value
`DefaultOutputDirectory`	This property shows the default output directory or folder	C:\Program Files\AnCAD\Visual Signal
`ActualOutputFileName`	Show the actual output file name and path	DataWriter.tfa
`OutputFileName`	Change the filename and specify the location to save to	{default}.tfa

`DefaultOutputDirectory` is unchangeable in **Property Window**. It only can be changed through **Tools | Preference | Output**, as shown in the figure below. You can also specify the encoding; the default value is us-ascii (ASCII).

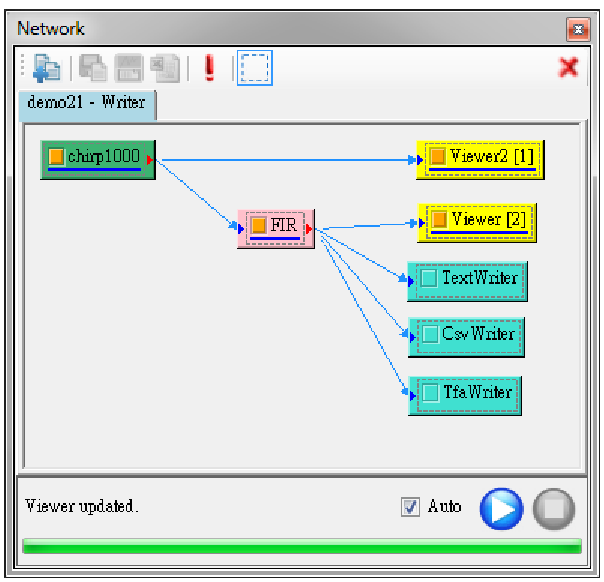


`ActualOutputFileName` only shows the full name, which is specified in `OutputFileName`. If the field, `OutputFileName`, is changed to another path that is different from `DefaultOutputDirectory`, it shows the full path in the `ActualOutputFileName` field.

`OutputFileName`: The file extension (file format) is also changeable, though only to formats Visual Signal supports.

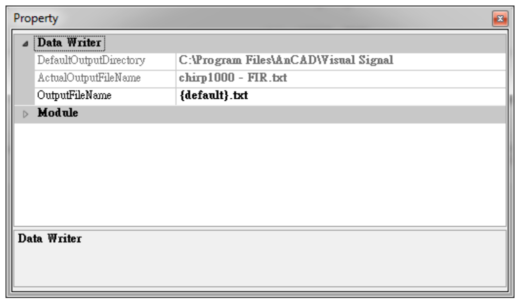
Example

1. Open demo21 – Writer.vsn in C:\Program Files\AnCAD\Visual Signal\demo\Basic. The network is shown below

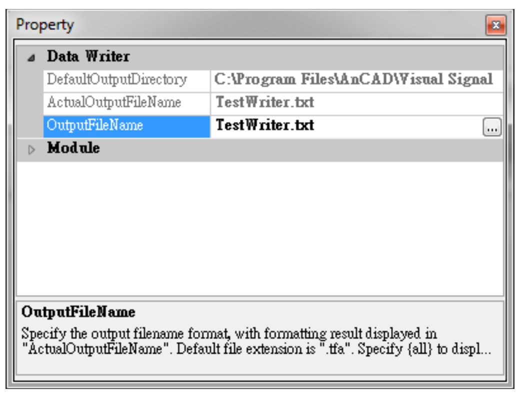


2. The default action of **`Data Writer`** is disabled. After all settings are set, it will write data to file immediately when you enable the component. In demo21, there

are three different formats, TextWriter, CsvWriter, and TfaWriter. Look into the TextWriter, the `DefaultOuputDirectory` is the default "C:\Program Files\AnCAD\Visual Signal." The `ActualOutputFileName` is the combination of previous components "chrip1000-FIR.txt." The filename "chrip1000-FIR" is the default value. It is also the default value of `OutputFileName` {default}.txt. But the difference between them is changeable in `OutputFileName` field.



3. Change the {default}.txt to TestWriter.txt. The `ActualOutputFileName` will be updated automatically after editing the `OutputFileName` field.



Related Functions

`Write Data, Export to Excel`