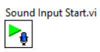
Fateme chngizian 9332682

**describe new block

Part1

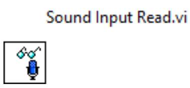
 Sound Input Configure VI

Configures a sound input device to acquire data and send the data to the buffer. Use the Sound Input Read VI to read the data.



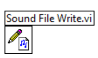
Sound Input Start VI

Starts data acquisition from the device. This VI is necessary only if Sound Input Stop has previously been called.



Sound Input Read VI

Reads data from a sound input device.



Sound File Write VI

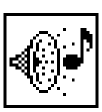
Writes data from a waveform or an array of waveforms to a .wav file.

 Sound File Close VI

Closes a .wav file.

 Simple Error Handler VI

Indicates whether an error occurred. If an error occurred, this VI returns a description of the error and optionally displays a dialog box.

 Beep VI

Causes the system to issue an audible tone.

 File Dialog Express VI

Displays a dialog box with which you can specify the path to a file or directory.

You can use this dialog box to select existing files or directories or to select a location and name for a new file or directory.

Wait (ms) Function

Waits the specified number of milliseconds and returns the value of the millisecond timer. Wiring a value of 0 to the **milliseconds to wait** input.

2.abcd

RMS VI

Computes the root mean square (rms) of the input sequence **X**.

AC & DC Estimator VI

Estimates the AC and DC levels of the input **Signal**.

Basic Averaged DC-RMS VI

Calculates the DC and RMS values of an input waveform or array of waveforms. This VI is similar to the Averaged DC-RMS VI, but this VI returns only one **DC value** and one **RMS value** per input waveform.

Convert from Dynamic Data Express VI

Converts the dynamic data type to numeric, Boolean, waveform, and array data types for use with other VIs and functions.

Get Waveform Components (Analog Waveform) Function

Returns the analog waveform you specify. You specify components by clicking on the center of the output terminal and selecting the component you want.

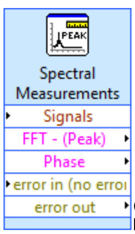
t0 returns the trigger time of the waveform.

dt returns the time interval in seconds between data points in the waveform.

Y returns the data values of the waveform.

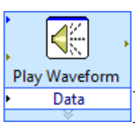
FFT Spectrum (Mag-Phase) VI

Computes the averaged FFT spectrum of **time signal**. This VI returns the FFT results as **magnitude** and **phase**.



Spectral Measurements Express VI

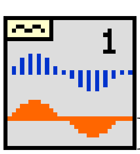
Performs FFT-based spectral measurements, such as the averaged magnitude spectrum, power spectrum, and phase spectrum on a signal.



Play Waveform Express VI

Plays data from the sound output device using finite sampling. This Express VI automatically configures an output task and clears the task after the output completes.

2.e

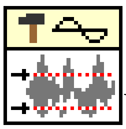


Resample Waveforms (single shot) VI

Resamples input waveforms or data according to the user-defined **t0** and **dt** values.

dt is the user-defined sampling interval for **resampled waveform out**.
t0 is the user-defined start time value for **resampled waveform out**.

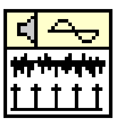
2.g



Y[i]=Clip{X[i]} VI

Clips the elements of **Input Array** to within the bounds specified by **upper limit** and **lower limit**.

2.h

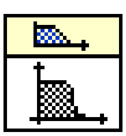


Periodic Random Noise VI

Generates an array containing periodic random noise (PRN).

samples is the number of samples of the **periodic random noise**. The default is 128.
spectral amplitude is the magnitude of the frequency domain components of the **periodic random noise**

2.i



Butterworth Filter VI

Generates a digital Butterworth filter by calling the Butterworth Coefficients VI.

filter type specifies the passband of the filter.

0 Lowpass
1 Highpass
2 Bandpass
3 Bandstop

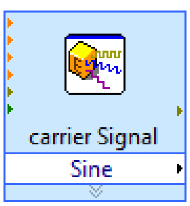
X is the input signal to filter.

sampling freq: fs is the frequency in Hz at which you want to sample **X** and must be greater than 0. The default is 1.0 Hz. If **sampling freq: fs** is less than or equal to 0, this VI sets **Filtered X** to an empty array and returns an error.

high cutoff freq: fh is the high cutoff frequency in Hz. The default is 0.45 Hz. The VI ignores this parameter when **filter type** is 0 (Lowpass) or 1 (Highpass). When **filter type** is 2 (Bandpass) or 3 (Bandstop), **high cutoff freq: fh** must be greater than **low cutoff freq: fl** and observe the Nyquist criterion.

low cutoff freq: fl is the low cutoff frequency in Hz and must observe the Nyquist criterion. The default is 0.125 Hz. If **low cutoff freq: fl** is less than or equal to 0 or greater than half the value of **sampling freq: fs**, the VI sets **Filtered X** to an empty array and returns an error. When **filter type** is 2 (Bandpass) or 3 (Bandstop), **low cutoff freq: fl** must be less than **high cutoff freq: fh**.

2.j



Simulate Signal Express VI

Simulates a sine wave, square wave, triangle wave, sawtooth wave, or noise signal.