Demo Application: *Response Tracker*

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1 Introduction

This document describes a LabView application, which is used to demonstrate the LabView interface to the Signal Ranger DSP board. This application loads and launches a real time DSP code, and then allows the user to interract with it. This user interraction is in the form of *Reads* of DSP variables and vectors, and *Writes* of parameters that act on the real time processing.

2 What it does

This application uses the Signal Ranger board to generate a pseudo-random, white reference signal on one of its analog output, send it to an unknown or variable linear “system under test”, read the signal from the output of the system, and through a Least Mean Square optimization, continuously calculate the impulse response of the unknown system. This impulse response is modeled as an FIR filter. An FFT of the impulse response produces the transfer function that is also plotted in the main window.

An unknown linear system must be connected between the analog audio output and analog audio input of the DSP board. The system can be as simple as a patch cable between input and output, or the Analog Interface Circuit (AIC) can be configured in *analog loopback*, which connects its input to its output internally.

The optimized response includes the gains, delays and anti-aliasing filters of the AICs. It is identified between the signals generated and acquired at the digital level.

Note: The coefficients of the FIR model filter saturate at –1 and +1. The overall gain of the estimated system must be such that no coefficient goes over these bounds. Usually, for broadband systems, this means that the gain should be below -6db. If it is not, and some filter coefficients saturate, the precision of the identification will suffer. In such a case, the following actions can be taken:

- *The **GAIN (analog output)** control can be adjusted to introduce some attenuation.*
- *The **GAIN (analog input)** control can be adjusted to present some attenuation.*
- *In the last resort, the gain of the estimated system should be lowered.*

3 What is Real Time, what is not

For this application the generation of the white reference signal, the acquisition of the signal on the AIC input, and the LMS algorithm are implemented in real time on the DSP.

All user interactions, including the display of impulse and frequency responses, and various warning conditions, as well as actions and parameters from the user, are implemented on the PC and not synchronized with the DSP processing.

4 Running the demo

To run the demo, you must have a Signal Ranger board, powered and connected to the USB port of a PC. Additionally, you can have an unknown system (filter) connected between the analog output and input of the DSP board (minimally a simple cable, connecting output to input).

The following files must be in the same directory, from which you will execute the demo:

- **SRangerRespTracker.llb** LabView library containing the Vis of the ResponseTracker demo.

- **SRanger.llb** LabView library containing the user Vis of the LabView interface.
- **SRangerU.llb** LabView library containing the utility Vis of the LabView interface.
- **SRKernel.out** COFF file of the Signal Ranger kernel.
- **demo_sr.out** COFF file of the ResponseTracker demo.
- **SRanger.dll** Low-level board communications dll.
- **ResponseTracker.exe** Windows executable of the demo (if you do not have LabView installed on your PC).

If you do have LabView installed on your PC Simply double click on the SRangerRespTracker.llb icon. If you don't, use the "ResponseTracker.exe" executable that has been built for the demo.

You are then presented with the following front panel:

Note: Fonts might not appear correctly if the font size in the Windows display settings is not set to "small fonts".

To run the application, simply click on the arrow at the top left of the window.

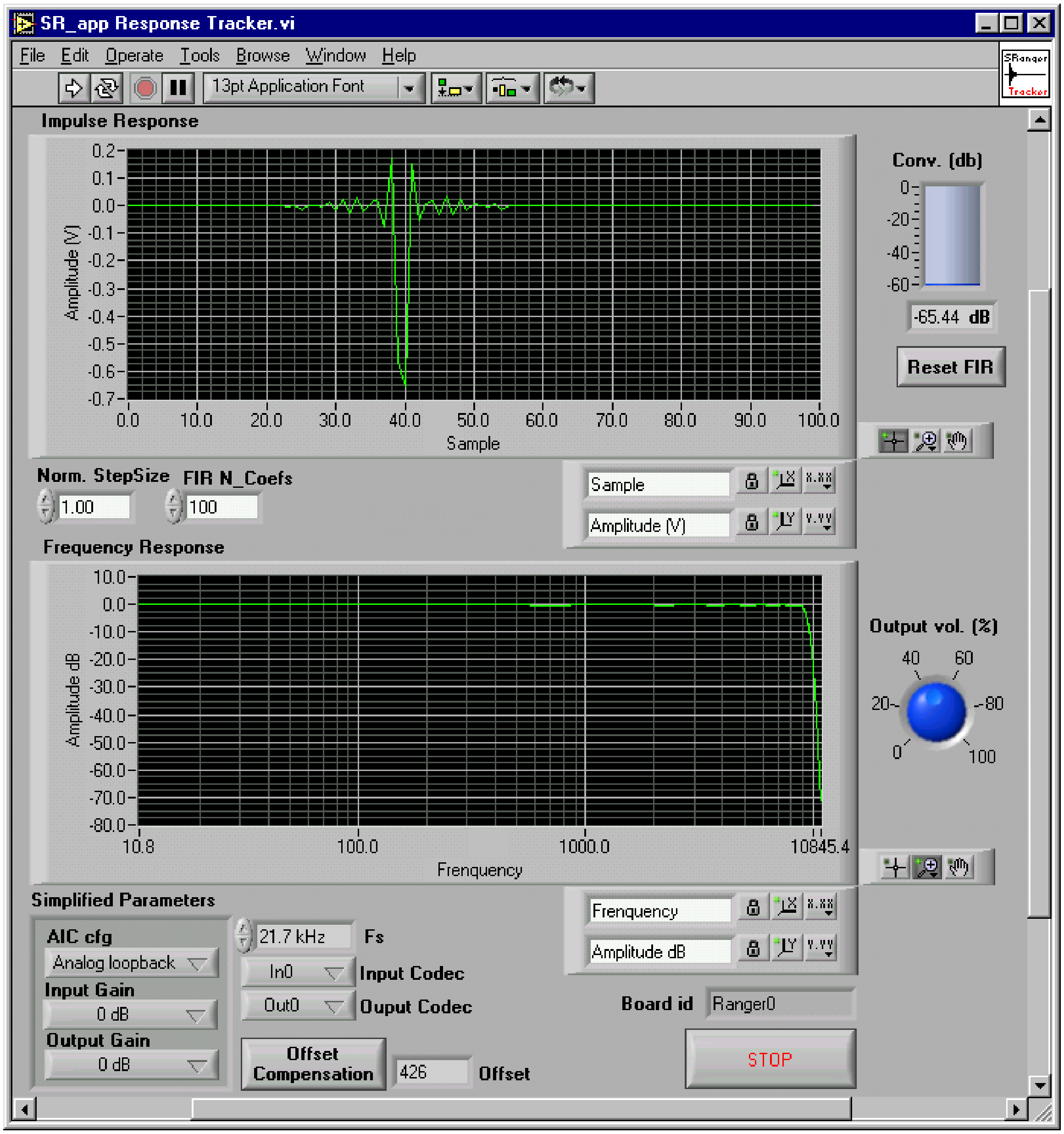

Figure 1 Front panel of the demo application

5 User interface

All controls except the **Board id** control can be operated while the application is running and have a dynamic action on the real time process executing on the DSP.

- **Board id** Selects which Signal Ranger board to use for the test. If only one is connected to the USB port, "Ranger0" should be used. If not, change the index to match the desired board.
- **Simplified parameters** Controls for the configuration of the Analog Interface Circuits.
  - **AIC cfg** Input/Output configuration:
    - Default: routes the In and Out signals to the RCA connectors

- Analog loopback: puts the selected AIC in analog loopback
- Digital loopback: puts the selected AIC in digital loopback
- **Input Gain**: Sets the input gain from –36db to +24db in 3db steps
- **Output Gain**: Sets the output gain from –36db to +24db in 3db steps

- **Fs** Sets the sampling frequency, as well as the cutoff frequency of the anti-aliasing filters (0.45Fs).
- **Input AIC** Selects an AIC (from 0 to 7) from which to sample the input signal.
- **Output AIC** Selects an AIC (from 0 to 7) to which to route the white test signal.
- **Impulse response** Displays the impulse response as it is optimized. The x-axis is in number of samples. The y-axis is in normalized coefficient values. The dynamic range for the coefficients is –1 -> +1. This display is updated as fast as the PC allows.
- **Frequency response** Displays the Fourier Transform of the impulse response. The x-axis is in Hz. The y-axis is in db. This display is updated once every 10 updates of the impulse response.
- **Buttons**
  - **Stop** Stops the application.
  - **Reset FIR** Resets the impulse response being optimized. Use it if the optimization diverges.
  - **Offset compensation** The DC offset of the analog input is measured and thereafter subtracted from the input. This control triggers the following operations:
    - Mutes the output.
    - Measures the input for a short period of time.
    - Averages the measured DC offset of the input over that period of time.
    - Resumes the generation of the white reference signal on the output, and use the new average value as the offset compensation on the input (subtract the offset from the acquired samples).

Note : For a best estimation of the low frequency components of the system, it is useful to do an offset compensation procedure. Since the offset can vary with temperature, it is best to repeat this procedure as long as the AIC has not reached its equilibrium temperature).

- **Offset** Indicates the averaged value of the input offset of the AIC, after the offset calibration has completed. This value is expressed in 16 bits, 2's complement notation.
- **Normalized Stepsize** Controls the Step size for the LMS optimization. A value over 2 makes the optimization diverge. A value of 1gives the fastest convergence. Use it for tracking rapidly varying systems. A smaller value gives a somewhat less noisy estimate, but takes a longer time to converge. A value over 1 gives no benefit (will converge slower, and give a more noisy estimate than a value of 1). A value of 0 freezes the esimation. Due to its possible saturation in fixed-point representation, the actual step size may saturate to a value less than what is requested. In such a case, the displayed value will change just after being modified by the user, to reflect its actual saturated value.

*Note: The actual (non normalized) step size sent to the DSP, is also influenced by the **FIR N_Coefs**, and the **Output vol. (%)** controls. Modifying these controls may saturate the actual step size, which will be reflected by a change in the **Normalized Stepsize** display.*

*Note: For small numbers of coefficients (below 50), and for a value of **Normalized StepSize** above 1.5, the LMS may diverge. This may seem unexpected, since the value of the StepSize is not actually over 2. This is due to imperfections in the pseudo random sequence that is used as the excitation signal. This sequence is about 18000 samples long, and becomes « less random » at rollover. At these moments (every approximately 18000 samples), the LMS is more likely to diverge under the above conditions. Please note that there is no performance advantage in operating the LMS with a Step Size above 1.*

- **Conv (db)** This is a level-style indicator, which indicates the level of convergence of the LMS. The convergence is the ratio (in db) of the residual signal (real signal at the output of the system minus signal out of the filter model) to the real signal out of the system. Values around 0db indicate that the residual is as large as the signal out of the real system. It is indicative of a poor convergence. Values around –50db or lower indicate that the amplitude of the error signal is at least 300 times smaller than the amplitude of the modeled signal. It is indicative of a very good convergence.
- **FIR N_Coefs** This control adjusts the length of the optimized filter. It is expressed in number of samples. The number of coefficients is limited by the available processing power (number of CPU cycles between 2 samples). It is roughly inversely proportional to the sampling frequency. For instance, the limit is 719 samples at 21.7KHz.
- **Output vol. (%)** This control adjusts the amplitude of the reference signal which is generated at the output of the AIC.
- **Input Saturation** This indicator is located below the **Output vol. (%)** control. It will light up when the input signal acquired by the AIC is above 30000, or below –30000 (about 90% of the input dynamic range). If this is the case, the following steps can be taken:
    - The reference signal can be lowered by using the **Output vol. (%)** (this does not modify the gain of the estimated system)
    - The **Output Gain** control can be adjusted to introduce an attenuation if this is not already the case (this modifies the overall gain of the estimated system).
    - The **Input Gain** control can be adjusted to present an attenuation (this modifies the overall gain of the estimated system).
    - In the last resort, the gain of the estimated system must be lowered.