Flow Diagram

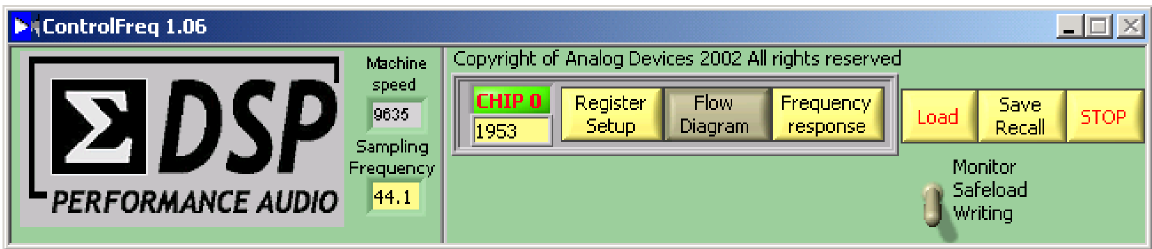

Figure 1. Flow Diagram, Home Window

Clicking on the Flow Diagram button brings up the main signal processing flow diagram and control panel (Figure 2). Here, all of the SigmaDSP parameters can be set and edited. All of the settings except for the dynamics processor curve are updated in real-time. This means that you'll hear the effect of sliders or knobs moving at the output as they are adjusted. The following descriptions will follow the signal processing flow from input to output (left to right).

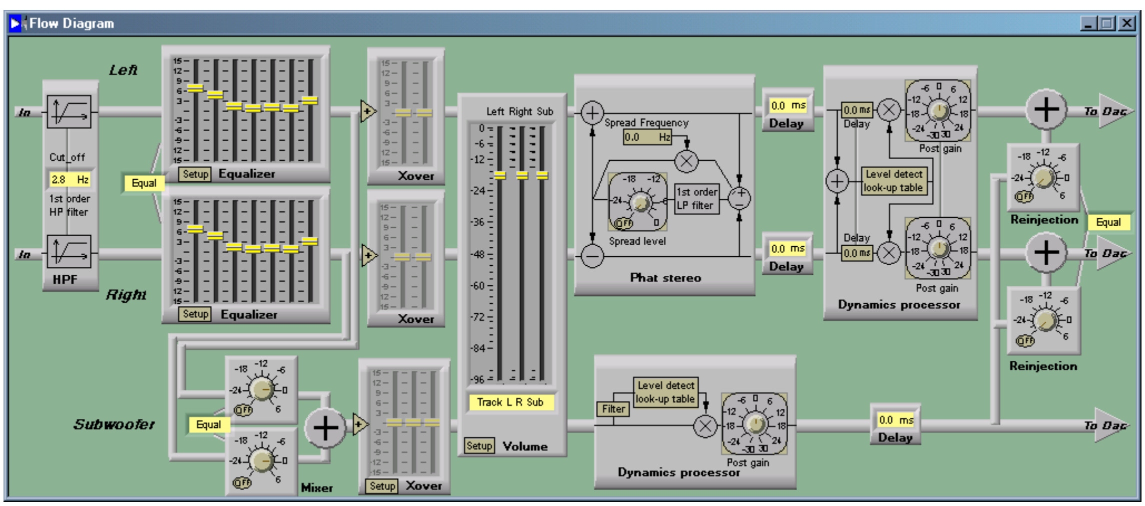

Figure 2. Flow Diagram Control Window

High-pass filter
Used to remove DC and very-low frequency components. It defaults to 2.8 Hz, but may be set to any frequency.

Equalization filter bank
The Filter Setup panel (Figure 3) controls the main equalizer's 7 biquad filters. Click on "setup" for each channel (left and right). At the top of each filter window, set the filter type, and then fill in the parameters for that filter. For a peaking filter, "boost" will control the boost/cut at the center frequency, while "gain" controls the overall gain of the filter. Setting the gain to more than 0 dB may cause some coefficients to

exceed 2.0, which will cause the coefficient values to "wrap" and may result in unstable filters. If this occurs, then an error window will advise you that the overall gain of the equalizer has been lowered to prevent coefficient wrap-around. It is advisable to compensate for this later (for example, by using post-compression gain). The effect of the filters can be seen directly in the frequency response window (selected from the main program panel).

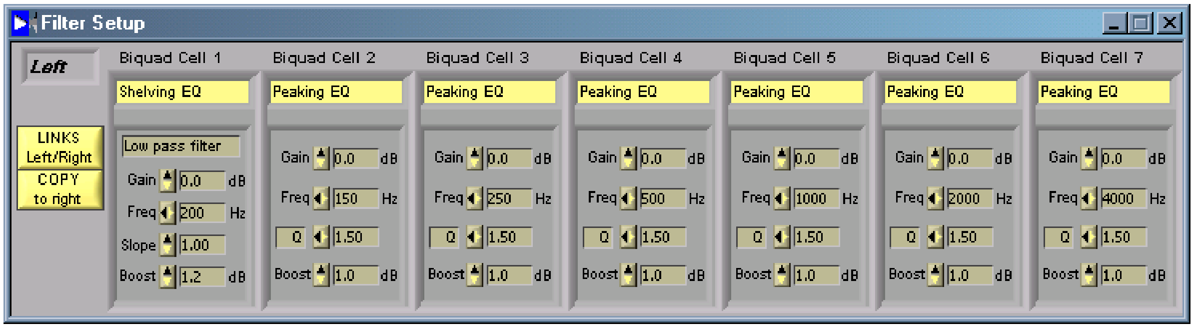

Figure 3. Filter Setup Window

The gain setting can be controlled with the sliders on the main parameters set-up screen, but the filter type, center/corner frequency, slope/Q, and boost/step settings can only be set in this window. The filter parameters may be dynamically changed by clicking on the up/down or left/right arrows next to the filter parameter that you want to adjust, and then moving the mouse outside the box while holding down the mouse button. This can be used to sweep frequency or boost/cut of a peaking filter, for example. The filter type can be set any of the following settings:

- Peaking
- Shelving – low pass or high pass
- General – low pass, high pass, band pass, notch
- Butterworth – low pass, high pass, band pass, band stop
- Bessel – low pass, high pass, band pass, band stop
- IIR coefficient – enter your own biquad filter coefficients

Note that the values for the IIR coefficients should be entered in the transfer function format:

$$H(Z) = \frac{(b_0 + b_1 Z^{-1} + b_2 Z^{-2})}{(1 + a_1 Z^{-1} + a_2 Z^{-2})}$$

ControlFreq will flip the signs of coefficients a1 and a2 before loading to the chip because the SigmaDSP uses inverted signs for these coefficients. The desired Q for a filter can be calculated with the equation Q=center frequency/bandwidth. A higher Q corresponds to a more narrow filter and a lower Q to a wider filter. Alternatively, the setting can be changed to directly select the bandwidth rather than Q.

The "Links Left/Right" button enables the program to simultaneously control both channels from the same panel. This is useful because in most applications the desired frequency response for both the right and left channels is the same.

Note that on each filter panel there is a button called "Copy to Left" or "Copy to Right." This will copy all filter parameters to the other channel, which helps to speed up the setup process.

If the left & right filters are set up the same way, then the sliders on the main equalization panel can be used to change the boost/cut values. If the "equal" box is selected, then both channels will be adjusted simultaneously. This can be used to operate the SigmaDSP in a graphic equalizer mode.

Subwoofer Mixer

This controls the gain of left and right signal going into the subwoofer mix. This value has a maximum of +6dB. If the "equal" box is checked, the left and right will be linked together. Values may be either entered in the box, or a rotary knob may be used. To bring up the large rotary knob and entry box, click on the outer edge of the box. The left and right mixer knobs can be set to always be equal, independent, track each other (increase or decrease with each other, but with an offset between each knob), or balance (inversely track each other). For a unity gain signal input to the subwoofer, set the gain to 0 dB.

Crossover filters

This controls the three crossover filter sections – left, right, and sub. Click on "setup" on the subwoofer crossover box, and a panel of filters will appear. There are two filters for each of the left and right channels, and three filters for the subwoofer. Note that if a subwoofer and the two-band compression feature will not be used then these crossover filters may be used as additional equalization filters for the two main channels.

The filter options are the same as described above in the equalization section. To set a 4^{th}-order Linkwitz-Reilly filter, set each main crossover filter to be a 2^{nd}-order Butterworth high-pass filter, and set two of the three subwoofer crossover filters to be a 2^{nd}-order Butterworth low-pass filter. This arrangement will add together to achieve flat amplitude response, and an allpass phase response. If only a 2^{nd}-order crossover is desired, one option is to use Bessel filters, which also adds to flat magnitude response and an allpass phase response. It is important to carefully design the crossover filters so that the overall amplitude response is still flat.

Volume control

The volume control adjusts the volume of all three channels. With the L=R=SUB option, all channels are adjusted together. If the Independent option is selected, all three volumes are adjusted separately. One of the track modes may be used to link channels together, but with an offset between them. When clicked, the setup button at the bottom of the volume box will allow the volume levels to be manually entered with a keyboard, and the slew rate of the sliders can be set. If the slew rate is set to 0, then the rate of the volume sliders is limited only by the mouse refresh rate.

Phat Stereo™

Phat Stereo is a stereo spreading algorithm that causes the apparent stereo image to shift outside the plane of the speakers. It is controlled by a spread level and spread frequency. The spread level may be adjusted up to +6dB; normally the effect starts to be heard at about –10 dB. The spread level may be entered either by entering a value in the box, or by bringing up the rotary knob by clicking on the upper right-hand corner of the box. The spread frequency controls the frequency range over which the algorithm will add an out-of-phase left minus right signal to the two main channels. This value will typically be set to between 500 Hz and 2 kHz.

Delay

This can be used to compensate for non-ideal speaker placement. Enter a value in the box up to 3.7ms. Since sound travels at approximately 1ft/ms, this can be used to compensate for speakers that are up to 3.7 ft off-axis from each other.

Main Dynamics Processor

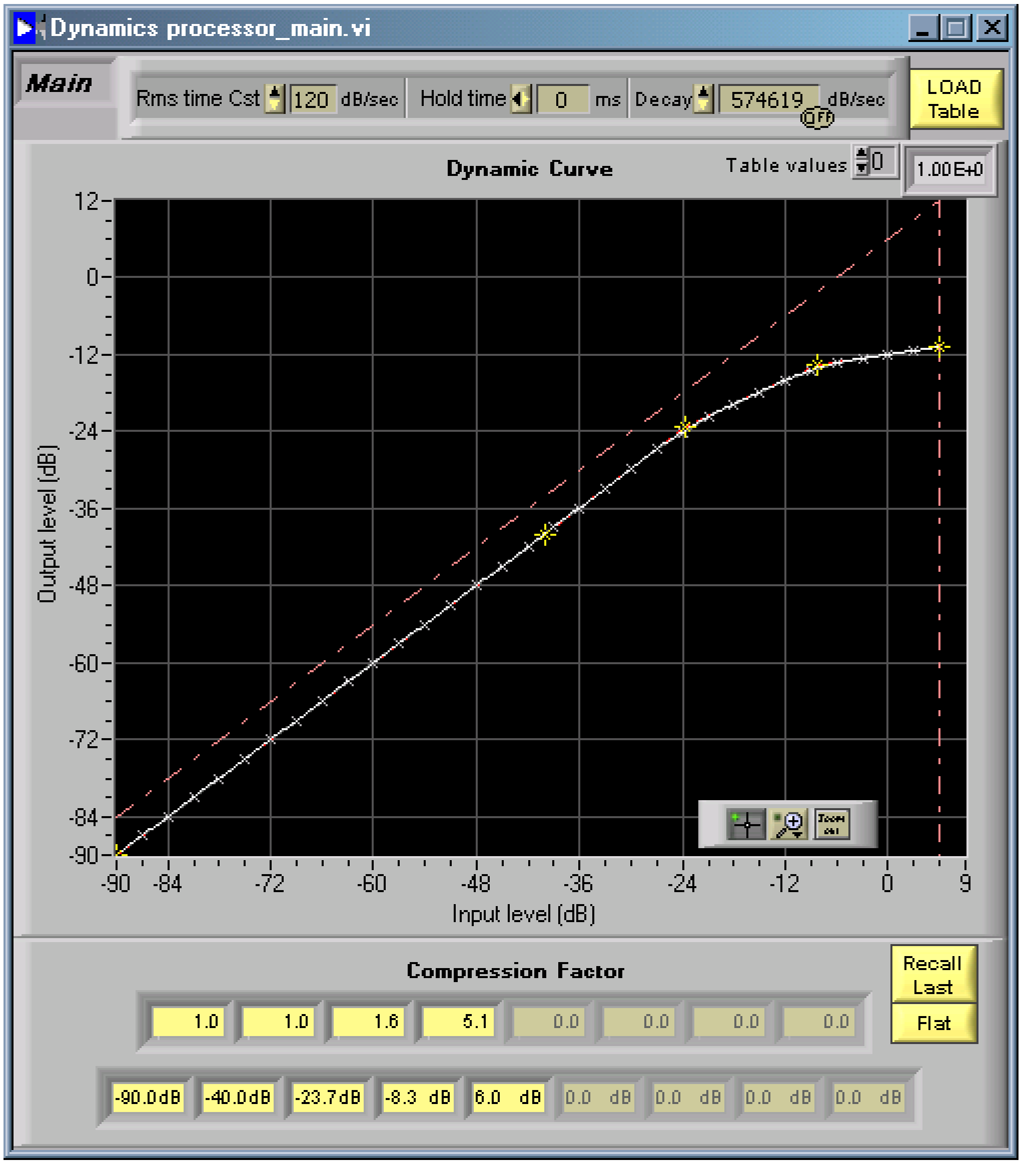

Figure 4. Dynamics Processor Control Window

This window controls the main dynamics processor (Figure 4) and is brought up by pressing the "Level Detect Look-Up Table" button. The graph displays the input/output curve of the compressor (input on the x-axis in dB, output on the y-axis in dB). The line in the window may be adjusted by selecting one of the points on the curve and dragging it with the mouse. The software performs a high-order interpolation between the points. Each point may be adjusted in both x and y directions. The compressor curve, unlike the rest of the SigmaDSP parameters that update in real-time, *does not take effect until the "Load Table" button is pressed*. Note that no point on the curve can exceed +6dB (represented by the diagonal dashed line in the graph window) because this will cause coefficients to wrap around. Higher values can be obtained by using the "post-compression gain" control after the dynamics processor, which in effect shifts the entire curve up and down. In general, compression curves should be free of sharp edges to get the best sound.

The volume control on the AD1954 will normally be set to about –15 dB or so, to allow for increased volume above the nominal level. If the music level is normally at about –15 dB, then together with the volume setting, the normal music level going in to the dynamics processor may be around –30 dB. Therefore, the compressor curve may be set to start gentle compression above -30 dB, with more severe compression occurring above –15dB. Then, when the volume is turned up, more of the music will enter the "compression area," allowing for a louder sound without distortion. To make up for the loss in maximum output level caused by using the compressor, the post-compressor-gain control (on the main parameters panel, after the dynamics processor) may be used.

On the bottom of the dynamics panel, there is a display of the slopes in each section of the dynamics curve, which can also be changed manually. The top row of numbers represents the slope, and the bottom row represents the break points on the input level axis. The slope values correspond to the slope between two adjacent breakpoints.

The "Recall" button will set the displayed curve back to the last curve that was loaded to the SigmaDSP. The "Flat" button sets the curve back to the default setting, with no compression/expansion. Remember that as when manually setting the curves, these do not take effect until the "Load" button is pressed.

On the top of the dynamics pane, the rms time constant, hold time and release rate are all adjustable. The rms time constant defaults to 120dB/sec, which is best for most situations. The hold time defaults to 0; this may be increased to improve low-frequency distortion while the compressor is operating. The release rate defaults to a very fast value, which allows the rms time constant to dominate. For example, if the rms time constant is set to 120 dB/sec, and the release rate is set to 200 dB/sec, the actual release rate will be dominated by the rms time constant, or 120 dB/sec. On the other hand, if the rms time constant is set to 120 dB/sec, and the release rate is set to 60dB/sec, then the release rate of the detector will be set to 60dB/sec.

Subwoofer dynamics processor.
This window operates in the same way as the main dynamics processor. By having independent dynamics processor for the low and high frequencies, audible "pumping" effects are avoided.

Sub Delay
Enter a value up to 2.3 ms. This can be used to compensate for positional errors between the subwoofer and main speakers.

Sub re-injection to Main channel control
This allows the subwoofer signal path to be added to the main (left & right) paths in situations where there is no subwoofer in the system. The benefit of two-band compression is still maintained using this signal flow, as the high and low frequencies are processed independently. Clicking on the outer edge of the box brings up the rotary knob and the data-entry window.

Frequency Response

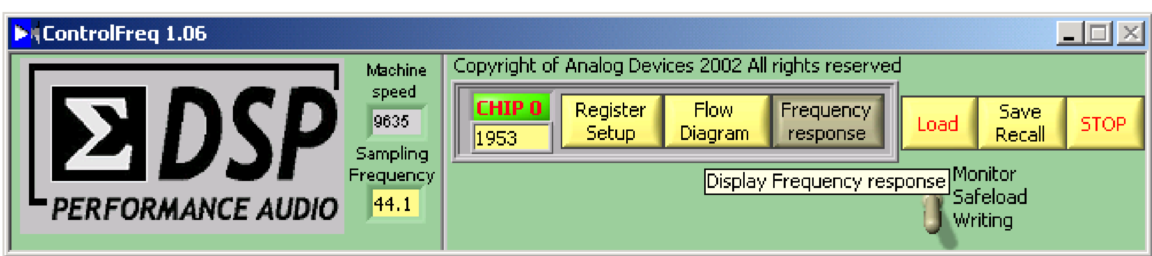

Figure 5. Frequency Response, Home Window

The frequency response window (Figure 6) displays the frequency response of the left equalizer (white), right equalizer (red), left crossover (purple), right crossover (magenta), subwoofer crossover (yellow), or compressor filter (green). The appropriate frequency responses can be selected in the box to the right of the display. Individual filters are shown as dashed lines, and the overall response of each bank of filters is shown as a solid line of the same color. Hold the Ctrl button to display multiple instances in the same window. Use the Zoom controls in the bottom right-hand corner of the plot to adjust the zoom levels.

The frequency response displayed is not the overall transfer function, but rather just the transfer function of the equalizers and crossover filters. This means that the sub re-injection is not taken into account in the frequency response displays.

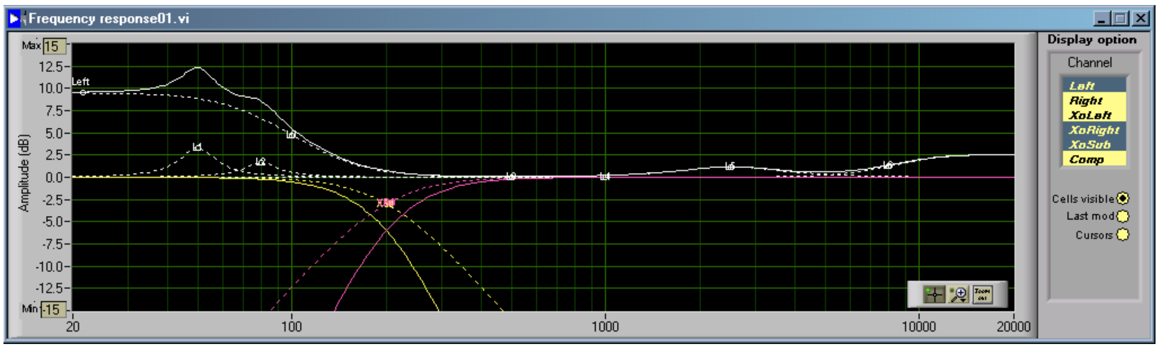

Figure 6. Frequency Response Plot

Load

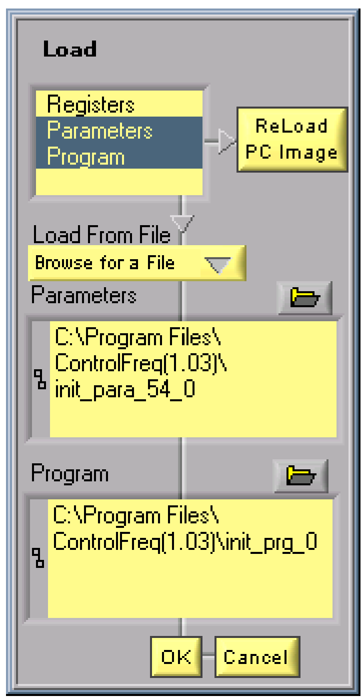

Figure 7. Load Window

The appropriate Program and Parameter .dat files are loaded to the AD1953/4 RAM using the Load window (Figure 7). The top box can be used to select Registers, Parameters, or a Program to load (hold *Shift* to make multiple selections). To bring up the program & parameter load window, click the Load button on the Home Window. In the Load window, select “Browse for a File” from the Load From drop-down box to select files created using SigmaComposer, a graphical compiler tool that can be used to reconfigure the signal flow of the SigmaDSP. Two file boxes will appear below – one for the parameters file and one for the program file. Browse for and select the appropriate .dat files (program_data.dat & spi_data.dat) created by SigmaComposer. The drop-down box can also select the init files, which are loaded when ControlFreq starts, and last file, which re-loads the last file that was loaded to the SigmaDSP. When the correct files have been selected, click "OK" and they will be loaded to the SigmaDSP chip.

While the program is loading, a blue progress bar will appear. If the bar turns green, then the load was successful; a red bar indicates that the load has failed. A failed load may be because the evaluation board is not powered, or the cable is not connected or is bad. Depending on the machine speed, a load can take anywhere from one to ten seconds.

The "Reload PC Image" button will reload the current GUI settings. This is a useful tool if the evaluation board has been powered-down or reset.

Save/Recall

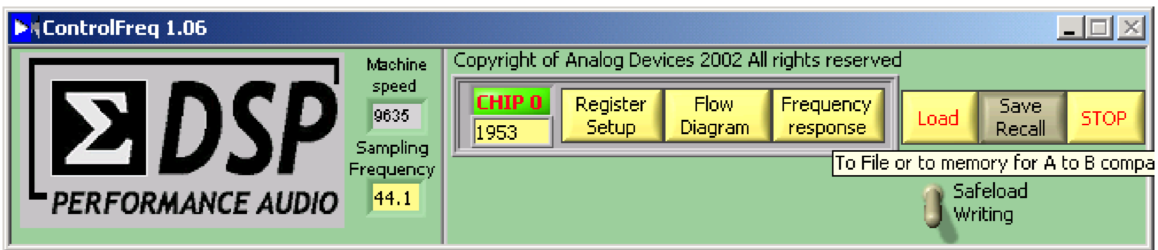

Figure 8. Save/Recall, Home Window

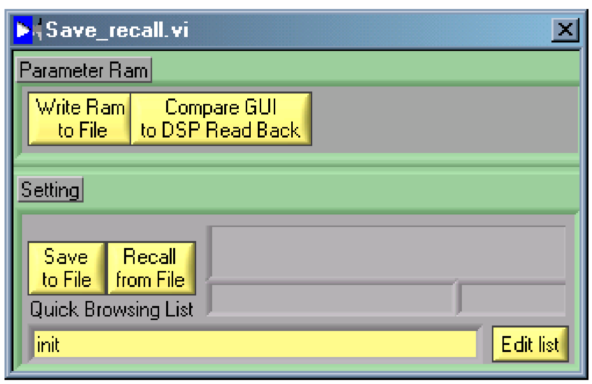

Figure 9. Save/Recall Control Window

The Save/Recall panel (Figure 9) allows entire setups to be saved and recalled from disk. To save a setup, click "Save to File," and select a file name. ControlFreq will automatically add either a "_53_0" or a "_54_0" suffix to the file name so that it can properly configure itself when the setup is recalled. To recall a previously-saved file, click "Recall from File." If you click on the Quick Browsing List, all recently saved files will appear in a pull-down menu. Different setups can be recalled by simply selecting the desired setup name.   The Edit List button allows the user to edit the Quick Browsing List by putting the files in a desired order, or delete setups from the Quick Browsing List. Deleting a file from this list will not delete the setups from the hard drive.

The Compare GUI to DSP Read Back button compares what is saved in the part's RAM to what is currently displayed in ControlFreq. When this is clicked, a window will pop up that displays all differences between the part and the GUI. This is useful after a reset and when using an external device, such as an EEPROM, to load the software.   The Write RAM to File button outputs the software content of the RAM in one of four selectable file formats (ASCII Decimal, ASCII Hex, ASCII binary, or byte stream).