Characterizing Mixed Signal DSP Designs with SigLab (Part 1)

Getting the ultimate performance from a mixed signal design usually requires bench time and honest-to-goodness test equipment. Simulation of a DSP design can provide answers to many questions, but not all. Eventually hardware must be built and tested. The SigLab 50-21 Dynamic Signal Analyzer is a flexible, fast, and accurate measurement tool that is perfectly suited to this task. This application note will demonstrate some simple, but important, measurements, which are the first steps in characterizing a mixed signal DSP design.

Overview

Analog Signals in a Digital World

Many, but not all, DSP designs interface to the "real world" of analog signals. Some examples of this include:

- Modems
- CD players
- Hearing Aids
- Speaker Phones
- PC Sound Cards
- Cellular Telephones
- Digital Audio Recorders
- Noise Canceling Headsets
- Communication Transceivers
- Active Noise Cancellation Devices

Each of the above examples contains at least one audio-bandwidth analog input and/or output. Analog circuitry and data conversion devices must co-exist in the digital world of the DSP subsystem. Achieving satisfactory analog performance in this digital environment and addressing the limitations and characteristics of the sampling and data conversion process, are the initial stumbling blocks of many DSP designs. Design considerations such as the sophistication of the DSP algorithms or "number of bits" are not relevant until the issues of:

- Noise
- Aliasing and Imaging
- Distortion
- Frequency and Phase Response pertaining to the mixed signal components are quantified

This application note is the first in a series. The series will cover some of the measurements and techniques used to quantify these issues in the development of a DSP based communication transceiver[1].

Measurements

The intent of the measurements is to verify that the conversion devices and associated analog circuitry (op-amps, analog switches, filters etc.) are operating within their specifications. Problems frequently arise from digital noise, PCB layout, power supply noise and other real-world design flaws.

The SigLab model 50-21 Dynamic Signal Analyzer is used to characterize the analog I/O performance of a Motorola DSP56L811 Evaluation Module (EVM). The measurements are typical of those one would make on any audio bandwidth mixed signal DSP design. A codec is provided on this particular EVM, but the measurements apply equally well to all forms of data conversion technology.

The DSP Evaluation Board

DSP "evaluation boards" are readily available from all DSP chip suppliers at very reasonable cost. These boards typically come with an analog I/O subsystem, peripherals, and software in addition to the DSP chip. The Motorola DSP56L811 Evaluation Module was chosen since it can be used to nicely illustrate the measurements typically required in a mixed signal design.

The EVM has the following attributes:

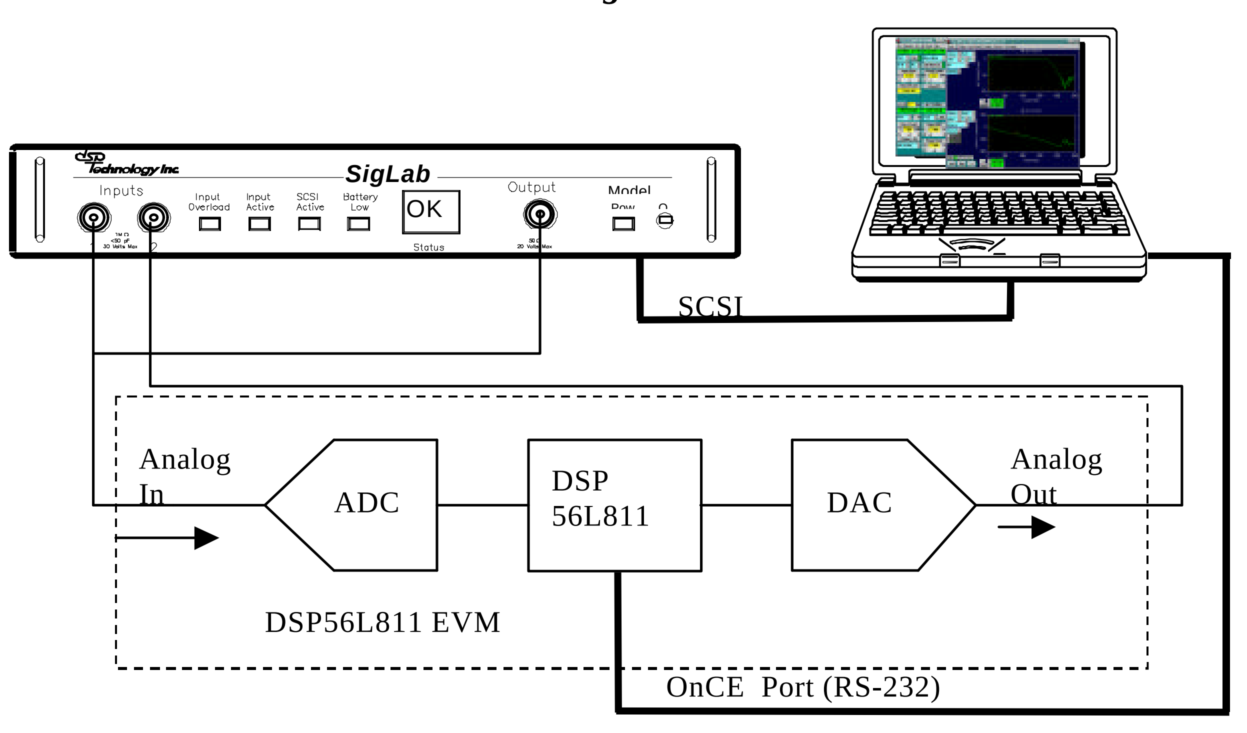
- A DSP core processor
- Analog (codec) interface
- General purpose MCU extensions
- Synchronous Serial Interface
- Serial and General purpose I/O ports
- On chip timers
- JTAG OnCE™ Windows debugger
- 32Kx16 SRAM external data mem
- 32Kx16 SRAM external program mem
- 32Kx8 Flash program (stand alone)
- 2 70 pin I/O connectors for I/O

System Configuration

Figure 1 shows the interconnections between the host PC, EVM, and the SigLab 50-21 Dynamic Signal Analyzer. The EVM is connected to the host PC via RS-232. Downloading code as well as debugging control is done through this serial interface.

The EVM comes with a "13-Bit Linear PCM Codec-Filter" providing a single channel of analog I/O. This device is primarily used in telephony applications. It is represented by the ADC and DAC blocks in Figure 1. The analog output channel of SigLab provides a variety of stimuli to the EVM ADC. This excitation signal is monitored by SigLab's analog input channel 1. The response of the EVM DAC is measured by input channel 2. The code executing in the DSP chip is the moral equivalent of a "wire" in that the input data stream from the ADC is simply passed to the output DAC with no "processing" whatsoever.

Figure 1



Noise

Mixed signal designs naturally are more noise prone than pure analog designs. A true RMS voltmeter teamed with an oscilloscope can provide a general idea of the overall noise level, but they do not provide the valuable diagnostic information that can be obtained with a spectrum analyzer. The spectrum analyzer is an excellent tool for measuring both broadband noise level as well as any narrow-band noise components. It provides the true RMS capability of a simple meter and goes one important step further by enabling the RMS level to be measured over a precisely specified frequency band.

For the noise measurement, the analog input to the EVM is terminated with a 600 ohm resistor. The output noise spectrum is shown in Figure 2. The spectrum analyzer reveals broadband noise over the 3.4 kHz pass band along with numerous narrow band components. The broadband noise is most likely due to switched capacitor filters internal to the codec and the quantization noise of the converters.

Using the analyzer's display expansion feature, the noise level is measured to be 130 uVrms in the 200 to 3500 Hz range. The specification sheet refers all analog measurements to a full scale of 0.436Vrms. Therefore, the ratio of available signal power in a sinewave to noise power, in dB, is given by:

$SNR = 20\log\left(\frac{0.436}{130\times10^{-6}}\right) = 70.5$ which is in good agreement with the 70 dB of "Idle channel noise" stated in the converter's specifications.

The clearly visible narrowband signals are at precise multiples of the 7812.5 Hz sampling clock. It is difficult to determine exactly where these signals originate. They could possibly come from the switched capacitor filter used for output DAC image reduction or they may simply be from the DAC clock coupling to the analog circuits. In any event, these components are below the idle channel noise level and are therefore not a great concern.

Output Subsystem Images

Images are artifacts in the analog output signal resulting from the process of reconstructing the digital data sequence into a continuous time waveform. For an input signal with a frequency component at f_{in} there will be energy present in the analog output at frequencies of $n * f_s \pm f_{in}$ where f_s is the sampling frequency and

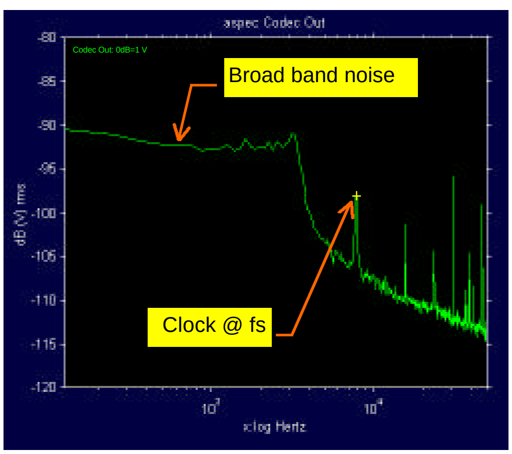


Figure 2 - Output Noise Spectrum of EVM

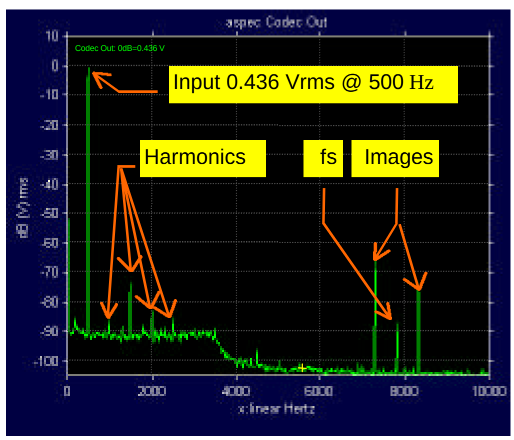


Figure 3 - Distortion, Noise and Images

$-\infty < n < \infty$. A 500 Hz sine wave at 0.436 Vrms is applied to the input of the codec. Figure 3 clearly shows the desired 500 Hz output as well as the first set of images (n==1) present on either side of the sampling clock (f_s).

The images are considerably reduced in magnitude with repsect to the desired 500 Hz output. There are two mechanisms responsible for this reduction. First, the ouput DAC does not produce impulses. Its value is held constant between samples. This "holding" produces a $\sin(x)/x$ rolloff.

Secondly, there is an on chip switched capacitor image reduction filter. By measuring the image level and accounting for the $\sin(x)/x$ term, the attenuation provided by this filter can be determined, albeit at a single frequncy.

The amount of rejection due to the $\sin(x)/x$ mechanism is given by:

$$image = \sin(p \cdot f_{image}/f_s)/(p \cdot f_{image}/f)$$

For a 500 Hz input, the first image frequency will be at 7812.5-500 = 7312.5 Hz. Therefore, the image magnitude will be

$$\sin(p \cdot 7312.5/7812.5)/(p \cdot 7312.5/7812.5)$$
$$= 0.0679$$

which, expressed in dB, is –23.36 dB. Using the spectrum analyzer display cursor (not shown), the first image image at 7312.5 Hz is measured to be at –65.2 dB with respect to the 500 Hz input. This indicates that the extra image attenuation provided by the switched capacitor filter is about 42 dB at the 7312.5 Hz image frequency which is within the expected range.

Harmonic Distortion

Along with the images and noise, Figure 3 also shows harmonic distortion. The third harmonic is seen to be –73 dB down from the fundamental, quite low for a telephone grade codec. This distortion represents the total of both the input and output conversion distortions. To measure the level of all the distortion and noise terms, the display is expanded over the 600 to 3500 Hz range. Over this display range, the RMS level is found to be 180uVrms. Assuming again the 0.436 Vrms maximum signal level, the signal/(noise+distortion) level in dB is:

$$S/(N+D) = 20\log\left(\frac{0.436}{180 \times 10^{-6}}\right) = 67$$

This is also in good agreement with the spec, which states a "typical" of 60 dB.

Intermodulation Distortion

An intermodulation distortion measurement is another method of characterizing performance. Figure 4 shows two input tones at 3100 and 3200 Hz. Odd order non-linearities produce sum and difference frequencies around the two input tones. The plot shows the worst of these components to be about 75 dB down from the amplitude of a single tone. IMD measurements are referred to the peak envelope power, which is 6 dB higher than either of the tones. Therefore, the IMD is 81 dB below the peak envelope power in the 2 tone input signal.

Even order non-linearities produce low frequency components at multiples of the frequency difference between the tones. These even order IMD terms are just visible above the noise.

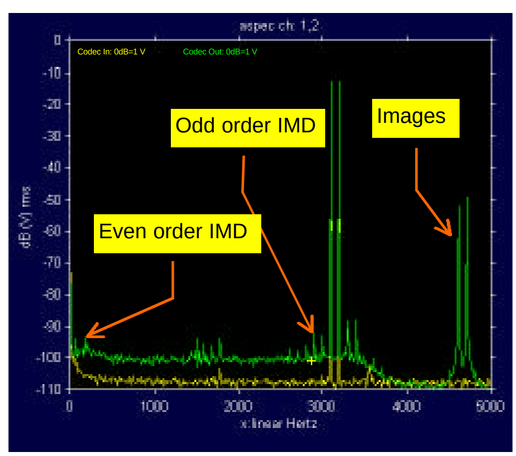


Figure 4 - Intermodulation Distortion, and Images

For reference, the spectrum of the two-tone input signal to the codec is plotted with the yellow trace (lower trace in pdf). This demonstrates the low distortion of SigLab's input and output subsystem. These IMD results are quite respectable for a codec.

Aliasing

Aliasing is due to the input sampling process. Signals with frequency components beyond $f_s / 2$ will "fold back" and appear as an "alias". The "fold back point" is at $f_s / 2$ and this frquency is often referred to as the "Nyquist" frequency. Given the sampling rate of 7812.5 Hz, the Nyquist frequency is 3906.25 Hz. The codec, as with any rational sampled data system, has an alias filter before the sampling operation takes place. This filter attenuates frequencies above aproximatly 3400 Hz. As is the case with any filter, there is a finite transition band and an ultimate stopband attenuation. The smaller the transition band and the larger the attenuation, all other things being constant, the better the filter. The codec uses a 5 pole switched capacitor filter for alias suppression.

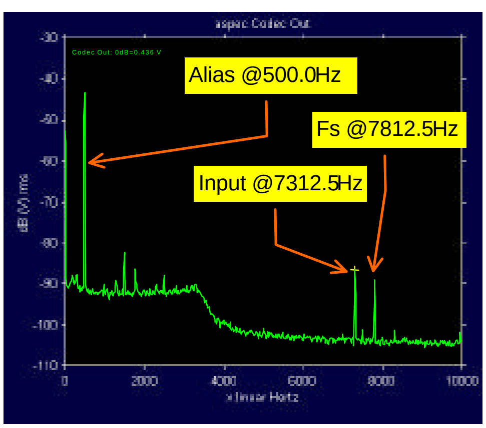


Figure 5 - Alias at 7812.5-7312.5=500 Hz

Figure 5 shows the codec output spectrum with an input frequency of 7312.5 Hz at a 0.436 Vrms (0 dB) level. The alias at 500 Hz is seen to be at a –38 dB. The attenuation of the alias (38 dB) is provided by the switched capacitor filter. Notice that a component is present at the sampling clock frequency, along with the highly attenuated input signal at 7312.5 Hz. The 38 dB alias attenation is in rough agreement with the 32 dB worst case number in the spec sheet.

The input signal component, at 7312.5 Hz, present in the output at the -87 dB level, is attenuated by both the input anti-alias filter, and the output anti-image filter.

Frequency Response

Since alias and image filters are present in the codec, the frequency and phase responses of these filters are of interest. SigLab has 2 primary methods of making this measurement. The quickest and easiest method to setup and use is the broad band fft based technique. This technique provides an unbiased measurement by taking the ratio of the cross-spectrum between the excitation and response and dividing this quantity by

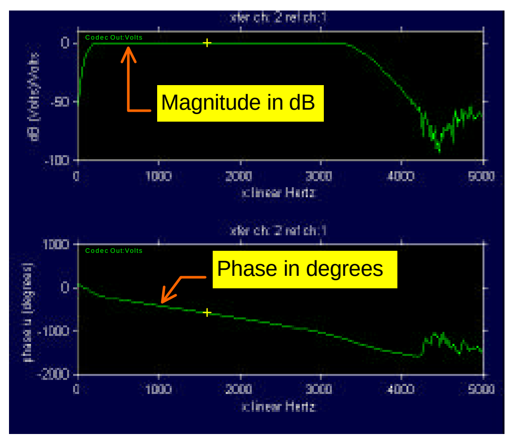


Figure 6 – Codec Frequency and Phase Response

the auto-spectrum of the excitation. Figure 6 shows the frequency and phase response of the overall analog I/O subsystem. The upper plot shows the magnitude of the system. Note that the codec has an input high pass filter that can be seen in the low frequency portion of the measurement. The codec's output subsystem corrects for the intrinsic $\sin(x) / x$ rolloff over the pass band region. This provides a good “flat” response in the 300 to 3300 Hz range. The phase response of the system is shown on the lower plot. The analog I/O subsystem’s “group delay” may be calculated from the phase measurement.

Figure 7 shows the group delay over the 500 to 3000 Hz range. The delay minimum occurs at approximately mid-band and can be seen to be about 0.8 ms. The codec specs indicate that the group delay should be approximately 0.67 ms. There is, therefore, a discrepancy of about 130 us. The sampling period is $\frac{1}{f_s} = 128us$, and this excess group delay turns out to be due to the 1 sample pipeline in the DSP code!

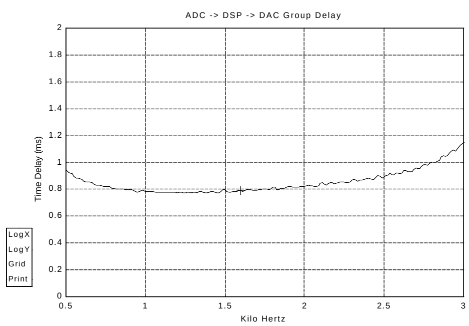


Figure 7 – Group Delay in Milli-seconds

The second transfer function measurement technique also uses the cross-spectrum and auto-spectrum calculations. However, the measurement is made using a sine wave excitation a single frequency point at a time. This method is often referred to as “swept-sine” but in reality, its implementation makes it “stepped-sine”.

The broadband technique has the advantage of speed, since the transfer function measurement is made over all the specified frequency points simultaneously. The disadvantage to this method is that the excitation energy must also be spread over the entire analysis range. When systems are noisy, or there are *aliases and/or images*, the ultimate measurement dynamic range is limited by these spurious components. This can be seen in Figure 7 where the measurement of the stop-band at higher frequencies gets very noisy.

The swept sine technique has superior immunity to these spurious responses for two primary reasons:

1. single frequency excitation improves SNR
2. narrow band tracking filters (within SigLab)

Figure 8 shows the measurement results using the swept sine measurement. The stop-band measurement has been improved and the -70 dB response level is clear. Again, this measurement represents the overall system response. The coherence plot indicates that the measurement is trustworthy up to about 6000 Hz.

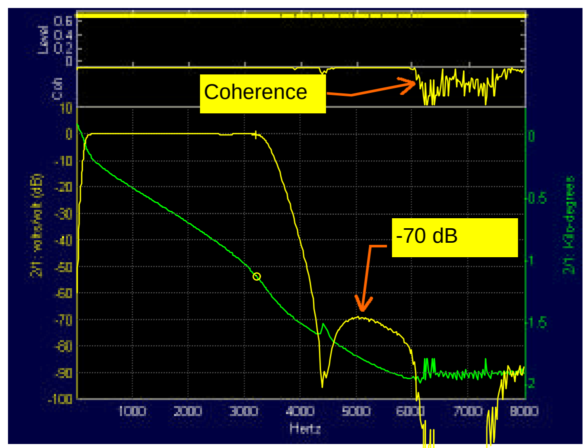


Figure 8 – Swept-sine Transfer Function Measurement

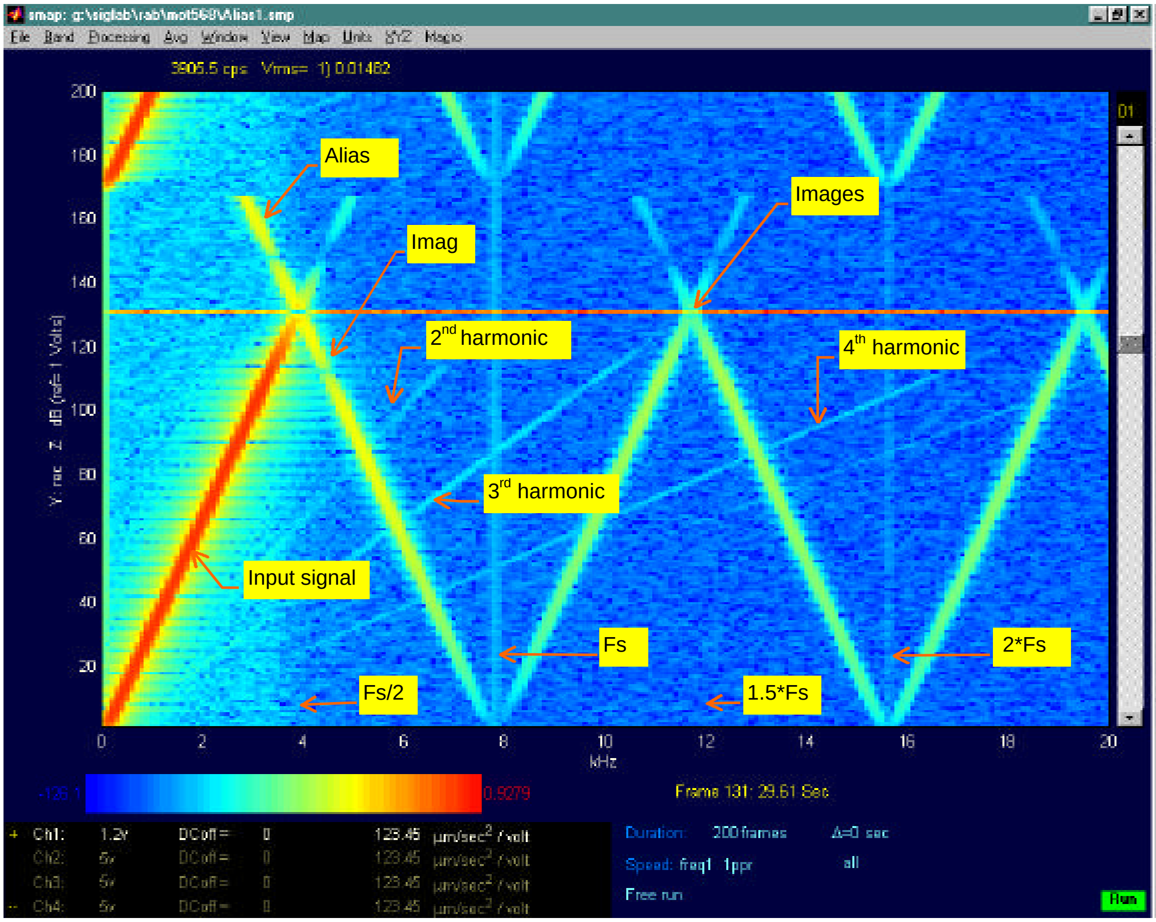


Figure 9 – Spectral Map Display of Analog I/O Subsystem Showing Noise (f<3000 Hz), Harmonics, Images, Aliases, and Sampling Clock Feedthrough.

A "Dynamic View" of Harmonics, Aliases, and Images

A spectral map can provide a dynamic view of how harmonics, aliases, and images vary as a function of input frequency.

SigLab's output source is setup to generate a slowly sweeping sine wave. It starts at 200 Hz and sweeps up to 6000 Hz. While the input signal is sweeping, 200 spectra are acquired and displayed in a spectral map format. This format maps magnitude to color, and displays the measurements in a visually appealing and illuminating way.

Figure 9 shows the results of this analysis. Points of interest are called out on the graphic. It is interesting to note that noise, distortion, images, and aliasing (when the frequency of the input goes beyond the Nyquist frequency), are all clearly visible. The red horizontal line is a cursor.

Figure 10 shows the same measurement data in a "waterfall" format. This format provides a better feel for the relative amplitudes of the spurious components and how they vary with frequency.

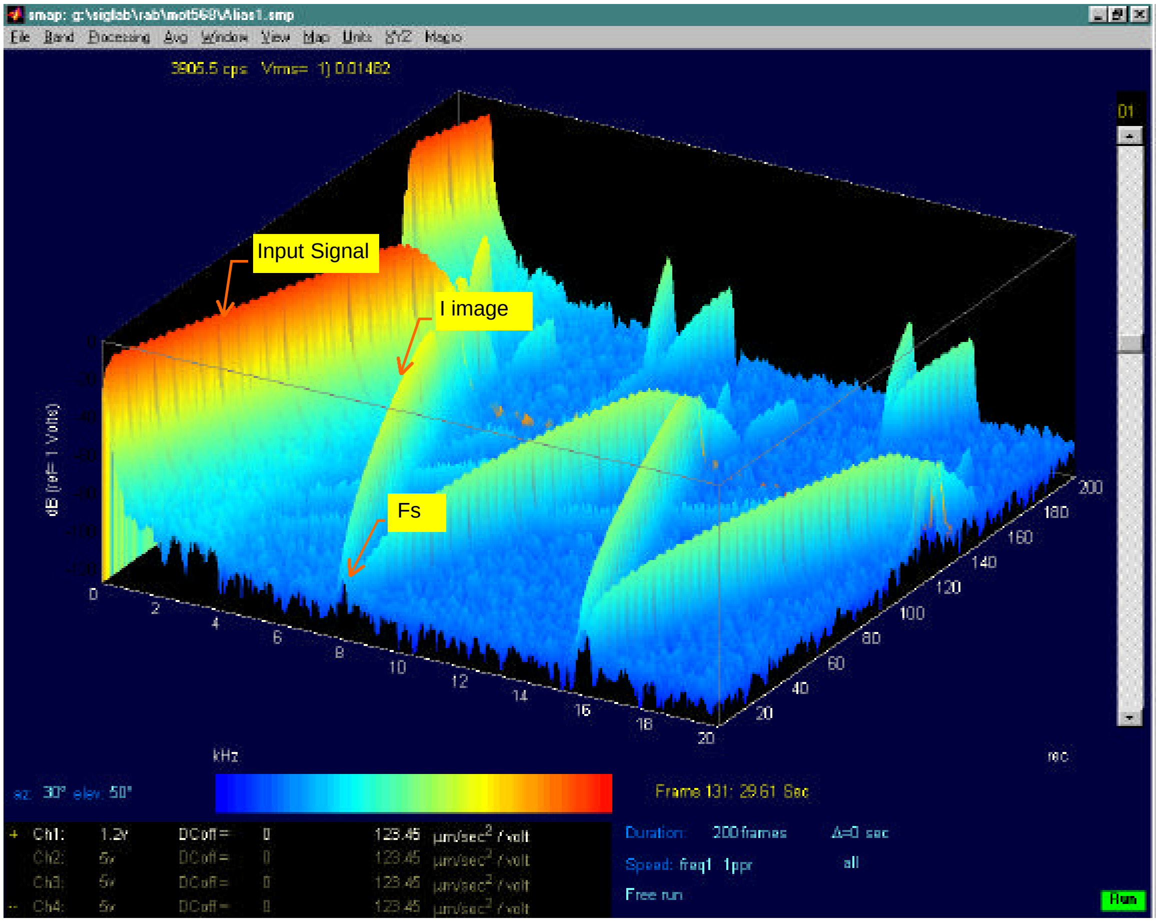


Figure 10 – "Waterfall" Display of Analog I/O Subsystem Showing Noise (f<3000 Hz), Harmonics, Images, Aliases, and Sampling Clock Feedthrough.

Conclusion

Simulation is unquestionably a valuable tool for DSP design. The overall performance of any mixed signal design, however, must ultimately be quantified by "real" measurements. This note has shown some of the measurements that can, and should, be made on a DSP design.

The measurements indicate that the codec supplied with the EVM appears to meets all its published specs. It's performance therefore not compromised and no significant improvements are possible. Although the codec no doubt does a good job for its intended market (telephony), the lack of aggressive aliasing filters is a concern. Although the alias / image characteristics could be improved by external analog filters, it is easier to use sigma-delta conversion technology. The second installment of this application note series will provide a similar set of analog I/O measurements, but using sigma-delta converters.

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[1] This design is being conducted as a case study.