2006-1962: DSP-BASED LOW-COST DIGITAL COMMUNICATIONS LABORATORY

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DSP-Based Low-Cost Digital Communications Laboratory

Introduction

Traditional undergraduate communications courses have focused on analog transmission schemes such as amplitude (AM) and frequency modulation (FM). Given the comparatively simple design of analog modulation circuitry, offering a laboratory component to the course is straightforward. In a typical laboratory session, students could construct and investigate the performance of AM or FM transmitters or receivers.

With the emergence of technology such as digital cellular telephony and wireline and wireless data communications, the emphasis has shifted from analog to digital modulation. Because of this shift, digital communications has become an important component to all levels of communications instruction. Due to the complexity of equipment that can emulate digital concepts, offering a hands-on laboratory component to support instruction is not as straightforward as in the analog case. Simulation based experiments, although a good supplement, are not a substitute for hands-on experimentation with real signals.

There are several approaches to offering a digital communications laboratory ranging from simulation packages teamed with standard instrumentation to dedicated custom hardware. In this paper, an approach that uses a vendor supplied low-cost DSP evaluation kit (the Texas Instruments TMS320C6713 DSP Starter Kit or 6713 DSK[1]) as an experimental platform is described. The use of this kit allows for a low-cost, highly flexible digital communications laboratory experience.

This report describes several digital communications experiments based upon the 6713 DSK. Results obtained from these experiments are presented as well.

Approaches

As stated earlier, it is desirable to go beyond simulation in a digital communications laboratory. A laboratory based on a vendor-supplied DSP kit is the approach to be discussed in this paper; however, it is useful to motivate the use of this method by briefly describing other approaches.

A popular approach is to use a software package such as National Instruments LabVIEW for the complex signal processing, along with data acquisition, teamed with either standard electronics instrumentation[2] or with general purpose signal generation[3]. This method is similar in spirit to the DSP approach although not quite as cost effective or as flexible. Another method is to use special, high-end test equipment, together with vendor-specific instrumentation software[4,5]. This setup can be used to offer a wide range of experiments, for both digital communications and other areas. The drawback of this approach is the prohibitively high equipment cost. A comprehensive, turnkey approach is the use communications modules specifically designed for educational purposes, such as the Feedback system[6] or the TIMS system**Error! Reference source not found.**. Unfortunately, these systems are quite expensive. Finally, there is the approach that uses custom hardware. Using gate arrays, there are labs that require students to build modules[8] or use modules already assembled[9]. Lastly, students can implement custom hardware modules or even look to fabricating custom ICs[10]. Clearly, this approach is not very flexible and in the case of student built hardware, quite ambitious.

Lastly is the vendor-supplied DSP development kit approach. There are several vendors in addition to Texas Instruments offering such kits, with a range of processing capabilities and on-board peripherals, such as Analog Devices[11] and Freescale[12]. The 6713 DSK was chosen since it had the right combination of features (see below), development environment, support and low cost.

There are several advantages to using a DSP kit for a digital communications laboratory. First and foremost is cost; the kits are comparatively inexpensive and they can be used for laboratory instruction in any related signals and systems course. Secondly is flexibility; applications are written in high level language, thus a wide range of algorithms can be developed. Another advantage is community support - there are many applications that are freely distributed. Furthermore, students like working with the kits. If so desired, it is possible to allow them to develop the code for the experiments. Students view working with the DSP kits as valuable experience beyond learning the communications concepts.

Quite commonly, the DSP kits have been primarily used for labs that teach DSP concepts. Only recently have these kits emerged as a vehicle for digital communications instruction. Recent examples include the implementation PCM speech processing[10], Frequency Shift Keying (FSK) and Differential Phase Shift Keying (DPSK)[13] and as a general purpose communications lab tool that includes a digital modulator for Binary Phase Shift Keying (BPSK), Quaternary Phase Shift Keying (QPSK), 8-PSK, 16-PSK, 8-level Quaternary Amplitude Modulation (8-QAM), and 16-QAM schemes[14]. With this last example, users may choose pulse shape, baseband or carrier transmission (0 to 24 kHz) and impairment (I/Q imbalance/skew and noise/interferer). Contained within this paper are several digital communications experiments that augment the DSP based communications features mentioned above.

DSP Starter Kit Features

The 6713 DSK is a turnkey DSP module, offering high-end computational capability, on-board peripherals and memory, JTAG based emulation, full-featured IDE and an extensive knowledge base, all at a relatively low cost. The board architecture is shown in Figure 1[15]. Board hardware features include:

- A Texas Instruments floating point TMS320C6713 DSP operating at 225 MHz. With an architecture employing a Very Long Instruction Word (VLIW) and eight computational units, the DSP provides up to 1800 MIPS/1350 MFLOPS, which is usually more than sufficient for single-channel communications applications. Included is 256 kB of internal RAM and 2 Multi-channel Buffered Serial Ports (McBSPs).

- A 24-bit stereo codec, software configurable for sampling rates of 8 ksps up to 96 ksps. The codecs interface to the McBSPs, delivering an interleaved stereo audio stream. The board includes 4 audio jacks (2 input, 2 output) for connecting signals.
- 16 Mbytes of off-DSP SDRAM. Memory management features are extensive with this platform; however, most communications experiments typically can reside fully in on-chip memory, simplifying the need for advanced memory management.
- 4 user accessible LEDs and DIP switches. These devices are easily controlled in software, providing a simple way to provide user interaction.
- JTAG emulation through on-board JTAG emulator with USB host. Thus, the interface to the host machine is via a USB connection.
- Standard expansion connectors for daughter card use. Typical applications include higher performance codecs or other types of interfaces.

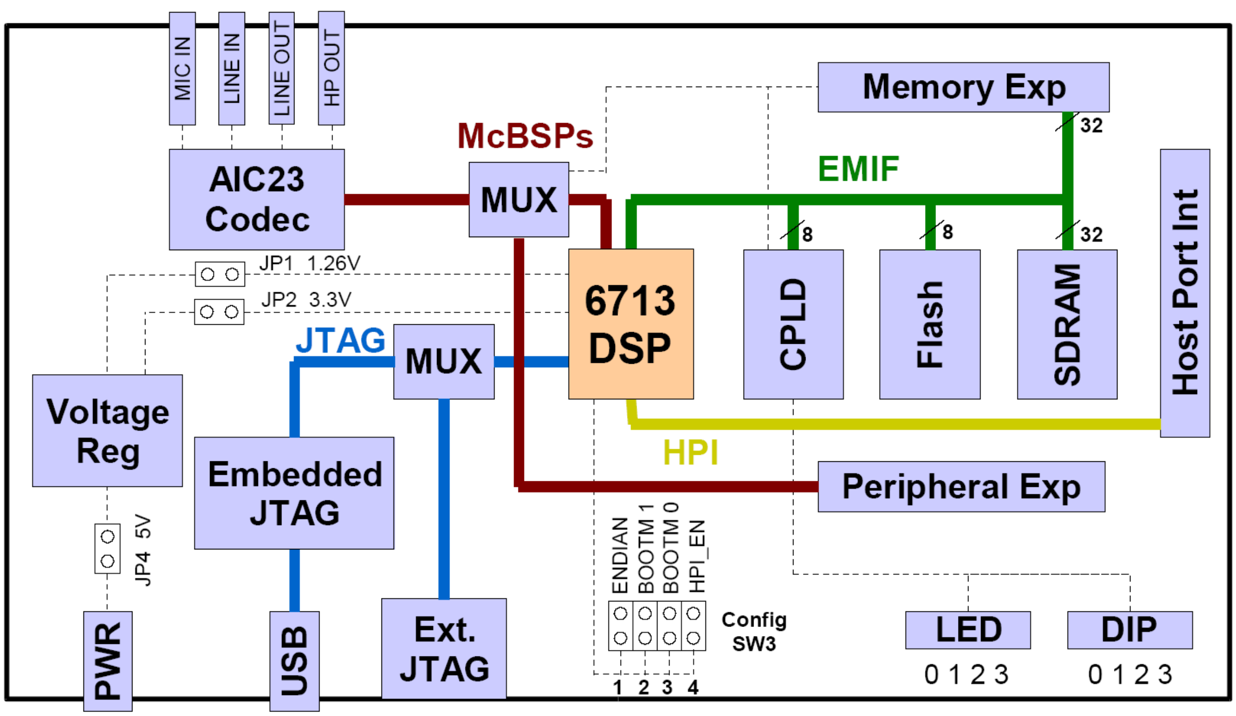

Figure 1. Texas Instruments/Spectrum Digital 6713 DSK Architecture

The 6713 DSK comes bundled with the Code Composer Studio (CCS) C/C++ IDE. The compiler is designed to be highly optimized such that the performance penalty for high-level programming is minimal. For the DSK, CCS supports all of the standard features but is only operational in emulation mode (i.e., with the hardware connected). The major features of CCS include:

- Extensive documentation, training and community support. Despite the support, the learning curve for CCS is steep and non-trivial. The difficulty in developing applications is the major hurdle in the DSP based approach.

- Full-featured debugging including various types of conditional execution and breakpoints, data watch windows and performance analysis such as CPU loading and thread execution via the DSP resident DSP/BIOS kernel.
- Source code-level libraries to support the DSP (Chip Support Libraries or CSL) and the peripherals (Board Support Libraries or BSL). These routines are used for codec configuration and simple operations such as reading a DIP switch or setting an LED.
- Real-time Data Exchange (RTDX) allowing for PC host-based file I/O, graphing and other analysis tools. This feature has proven useful for communications experiments, such as signal plotting, FFT analysis and constellation diagram plotting.
- General Extension Language (GEL) scripts. These C-like scripts are loaded at run-time, and allow the user to execute commands on the target or the host. Of particular use is the ability to add a slide bar or radio button for user control of the experiment. For example, a slide bar is used to change the level of the additive noise impairment.

Experiments

The setup for the 6713 DSK experiments is given in Figure 2. The host PC runs CCS and communicates to the DSK via the USB connection. Stimulus for the DSK card is generated by a function generator or some audio source (such as a CD player) for certain experiments. The DSK output may connect to a circuit breadboard (for simple channel impairments such as an op-amp based LPF) and then be displayed on a general purpose oscilloscope (or audio speaker for certain experiments).

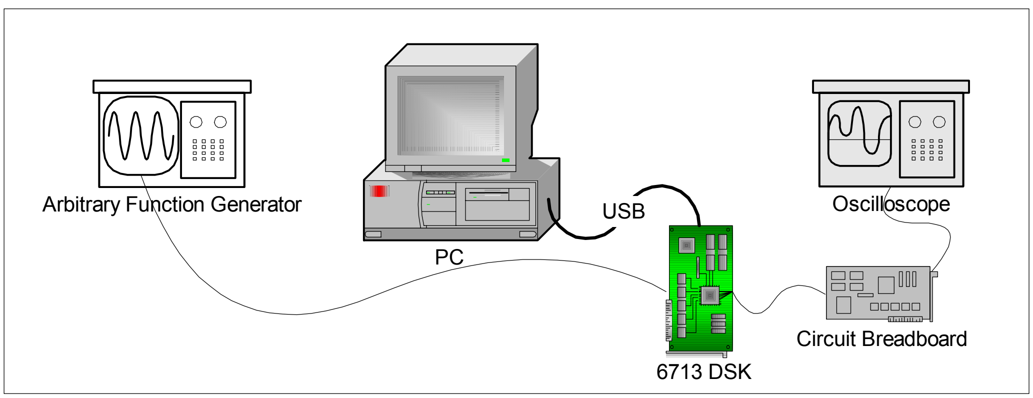

Figure 2. 6713 DSK Experiment Setup

The list of experiments discussed in this report is given in Table 1. Each experiment is described in the following sub-sections.

Uniform and Non-Uniform Quantization

This experiment is intended to demonstrate the benefits of non-uniform quantization. The concept is that the input signal is applied to both R and L inputs. This signal is sampled by the codec (a low sampling frequency of 8 kHz is used), resulting in a DSP internal 12-bit linear representation. The 12-bit samples are then converted into 8-bit representations, using different methods for R and L. The uniformly quantized channel (channel R) simply loses 4 bits of resolution. The non-uniformly quantized channel (channel L) is effectively mapped using the μ-law companding curve with $\mu = 255$, and then unmapped back to a 12-bit representation. In so doing, the R and L output buffer contain 12-bit samples that have one of 256 possible values, with the R having uniform jumps and the L having non-uniform jumps. Note that the L channel will have finer jumps at lower values, larger steps at higher values.

Table 1. 6713 DSK Communications Experiments

Experiment	Description	Equipment
Uniform and Non-Uniform Quantization	An audio signal is quantized and reconstructed using both uniform and non-uniform schemes and compared.	▪ TI 6713 DSK ▪ PC running CCS ▪ Audio I/O device
Delta Modulation	The tradeoff between slope overload and granular noise is investigated by varying the step size for a delta modulation scheme	▪ TI 6713 DSK ▪ PC running CCS ▪ Function generator ▪ Oscilloscope
Pulse Shaping	Eye diagrams are generated for two pulse shaping schemes. A low-pass channel is simulated, and ISI performance is compared between schemes.	▪ TI 6713 DSK ▪ PC running CCS ▪ Oscilloscope ▪ LPF
Characterizing Noise	Histograms for a uniformly distributed random variable and a Normally distributed random variable are generated and compared in real-time.	▪ TI 6713 DSK ▪ PC running CCS ▪ Audio Output Device
Constellation Diagrams	A 16-QAM constellation is drawn in real-time as the additive noise is varied. BER is updated and displayed.	▪ TI 6713 DSK ▪ PC running CCS

In the first part of the experiment, students apply a low frequency square wave to the DSK L and R inputs, and vary the peak-to-peak voltage (maximum voltage is 5 $V_{p\text{-}p}$). The DSK output is connected to the oscilloscope, with L and R to channels 1 and 2, respectively. As the input level is varied, there are ranges where no output change is seen, contrasted with certain input level thresholds where a jump in output level is observed. From these observations, a plot of voltage out versus voltage in generates the companding curve for the uniform and non-uniform methods.

The second part of the experiment is subjective. The codec sample frequency is increased to 48 kHz. The students apply a music signal (mono) to the DSP R and L channels simultaneously (it is usually required to add some gain to boost the signal level). The audio quality of the L versus R output is then subjectively evaluated, presumably with the L channel being judged superior, if still unacceptable (recall the 8-bit representation).

Whereas this experiment succeeded in demonstrating the desired concept, there were issues yet to be resolved. In the first part, when observing the square wave output, there was significant ringing at the square wave edges. Furthermore, the finer regions for the non-uniform quantizer were difficult to differentiate from each other.

Delta Modulation

This experiment is intended to demonstrate Delta Modulation (DM). The tradeoff between slope overload (inability to track variation in a signal) and granular noise (oscillation between levels for a slowly varying signal) is investigated as the step size is varied.

The DM step size is set via the DIP switches on the DSK, resulting in 16 permutations, with all switches "off" being the smallest step size and all four switches "on" the largest (the actual numeric values assigned to each setting depends on the range of input signal - given the 24 bit input, empirically the step size range was chosen to be 4 to 124). The same signal is applied to both R and L channels. The program simply copies the L input to the L output, while the R signal is delta modulated. Simply stated, the algorithm is the following: if the input is greater than the previous output, add the step size to the previous output; if not, decrease the previous output by the step size; in either case, the modified previous output becomes the current output. For this experiment, the codec sample rate was set to 48 kHz, and the students were advised to use signals well below 24 kHz.

Both DSK output channels are displayed on the scope, one being the target signal, and the other the delta modulated approximation. The students attempted to find the optimum step size to best track the input (slew rate) and minimize granular noise while experimenting with different input signals such as sinusoids, triangle waves and square waves at different amplitudes and frequencies.

This lab was successful in demonstrating how DM works. Due to what appeared to be codec related ringing effects, it was not always clear as to what step size was actually optimal.

Pulse Shaping

This experiment is intended to demonstrate the importance of pulse shape in minimizing ISI for baseband digital communications (principle applies to bandpass digital communications as well). The DSP generates a random baseband digital stream, using either a raised cosine pulse template or a simple square pulse (user selectable via a DIP switch) on the R output channel. On the L output channel, a triggering signal at the pulse rate is generated. Together, these signals are applied to the oscilloscope, with the L channel supplied the scope trigger. The R signal is displayed, and the oscilloscopes persistence is set to its maximum value (infinite for most digital scopes), resulting in an eye pattern. In order to see the benefit of using the raised cosine pulse, the R signal is first processed by a LPF before being applied to the oscilloscope. The user measures the separation distance for both pulse templates, seeing the advantage to the raised cosine. A final feature is included which allows a four level signal, again selected with a DIP switch, underlining the importance of minimizing ISI.

In order to implement this algorithm on the DSP, the baud rate was a sub-multiple of the DSP codec sample rate. The pulse was stored in memory, effectively at the higher DSP sample rate.

The random data was generated at the slower rate, and convolved with the pulse template, zero padding in between data points.

This experiment was quite successful; however, the eye pattern traces were fuzzier than expected, even without the lowpass channel impairment. It was speculated that the triggering signal contained jitter that manifested itself as variance in the eye pattern.

Characterizing Noise

Understanding probability concepts is fundamental in the study of communications. In this exercise, students learn about how the distribution effects how the noise sounds. The DSP generates Gaussian (or Normally) distributed noise on the L channel, and uniformly distributed noise on the R. The students listen to the two noise types, and display each on the oscilloscope. For scopes equipped with an FFT module, the spectrum can be measured. Using scope math operations, the students average many instances of the noise waveforms, resulting in the periodogram, where it is seen that both noises are approximately flat, nearly up to the sampling frequency.

Additionally, the DSP code calculates histograms for the two noise types. The students use the features of CCS to plot these histograms as they are generated, resulting in graphs that look like those of Figure 3. Shown in the figure is the evolution of the histogram as more and more samples are collected, with the left plot the initial estimate, the right the more refined estimate, and the uniform noise histograms plotted above the Gaussian histograms.

This exercise was successful in demonstrating some basic noise concepts. For the histogram plots, the students could vary the number of bins and note the effect (this change required a rebuild and reload). Finally, this experiment helped clarify the difference between density functions and spectral distribution.

Constellation Diagrams

This experiment is intended to demonstrate how constellation diagrams infer the performance of a digital communication system. A 16-QAM signal is generated and output to both R and L channels for display on the oscilloscope. Furthermore, using the plotting features of the 6713 DSK, the constellation diagram for this signal is drawn in real-time. Using a GEL script, a slider bar is available to allow the student to increase or decrease the additive noise. The DSP implements a slicer function, and compares the decoded symbols with the generated symbols. Finally, a watch window is created to keep track of the number of samples, the number of errors and the resulting symbol error rate. An example of a typical display after processing 6000 samples is given in Figure 4.

Visually, the students can see that as they increase the noise, the signal clusters spread out, making the decision more likely to be in error. Simultaneously, they see the effect on error count and error rate. The noise bar setting, in this case 5, does not indicate actual SNR; however, the symbol error rate does. Students were asked to set the noise to a high level (above 8) and let the experiment run for a long time. From the symbol error rate, the approximate SNR can be determined. With the information that each click on the slider bar represented a 1 dB change in noise power, students should be able to extrapolate the symbol error rate for any noise setting.

They were asked to predict and verify the symbol error rate for several noise bar settings (note that the program must be reloaded to reset the error rate counter for each new noise setting).

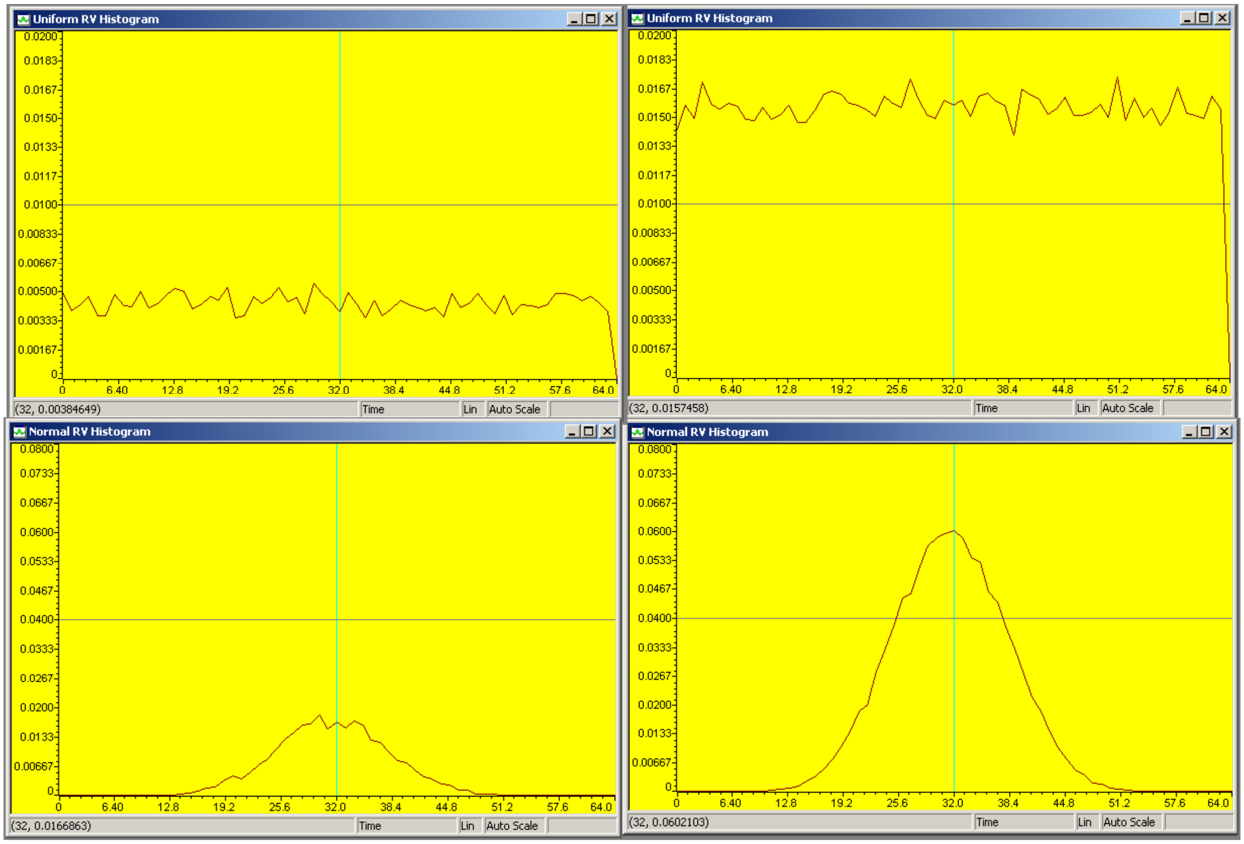

Figure 3. Evolution of Uniform and Normal Random Variable Histograms

This experiment was very successful in demonstrating the relationship between SNR and error rate. In principle, this experiment is very close to simulation except for the fact that plots and control are updated in seemingly real time. Future enhancement will include the implementation of a true QAM receiver, although this will involve solving symbol synchronization issues.

Conclusion

The vendor-supplied DSP board approach has been shown to be a low-cost, highly flexible approach to offering a digital communications laboratory. Students are enthusiastic about using this platform to learn about digital communications, and it has uses in several other courses as well. The major drawback to its use is the steep learning curve, although this is somewhat mitigated with training, documentation and community support. Finally, several enhancements and improvements are planned for the labs.

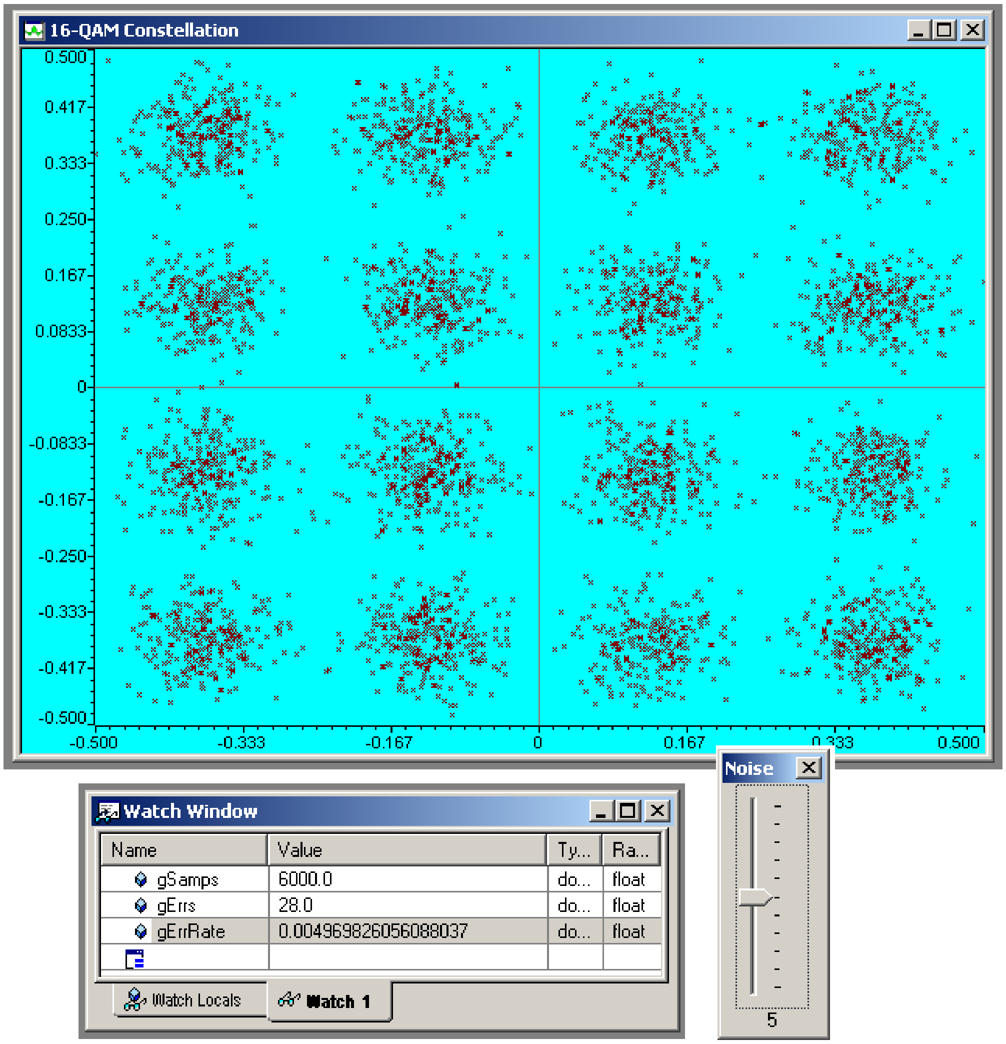

Figure 4. Constellation Experiment Display

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