Caller ID: A Project to Reinforce an Understanding of DSP-based Demodulation

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Abstract

This paper discusses one of the most popular projects in our real-time DSP course's, demodulating a telephone's caller ID signal. We will describe the use of a modified telephone coupler to allow students to access the caller ID signal and then discuss a number of the DSP methods and techniques that are required to successfully demodulate, decode, and display the caller ID information. This project has been exceedingly well received by both our undergraduate and graduate students and represents a DSP-based solution to a real world communications problem that every student has seen in use. This project involves almost every aspect of real-time signal processing implementation, but is not so overwhelming in scope that it cannot be completed in a reasonable amount of time.

1 Introduction

For several years now, we have been providing proven DSP teaching methodologies, hardware and software solutions, and DSP tools that have helped motivate students and faculty to implement DSP-based systems in real-time.[1–7] These efforts have emphasized the fact that DSP is much more than just a collection of theories and problem solving techniques. Students can easily be motivated to explore and implement DSP-based systems in an environment where they are limited only by their imagination. This process can be facilitated through real-time demonstration programs such as winDSK and winDSK6.[8]

All of our DSP-hardware courses involve a final project and the majority of our students migrate towards either audio-related or communications-related projects. This paper will discuss one of the more popular projects: demodulating a telephone line's caller ID signal.

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2 The Caller ID Signal

The caller ID signal is an asynchronous continuous-phase binary frequency shift keying (CP-BFSK) burst at 1200 bits per second which occurs between ring tones one and two of a properly provisioned handset of the plain old telephone system (POTS). The transmitted frequencies are either 2200 Hz (representing logic 0) or 1200 Hz (representing logic 1). The caller ID signal is composed of four parts:

1. A channel seizure signal which consists of 300 alternating zeros and ones.
2. A 180-bit sequence of all ones.
3. The actual caller ID message (variable length).
4. A one-byte message check sum.

The details of this BFSK burst can be found in.[9, 10]

3 Connecting to the Telephone System

To recover the caller ID signal, a telephone coupler must be attached to the telephone line. This coupler is attached in series between the incoming phone line and the existing telephone. We used a Comrex model TCB-1 telephone coupler[11] shown in Figure 1.

The telephone coupler's primary purpose is to allow a recording device (in our case, a PC's sound card) to monitor and (when desired) record telephone line activity (e.g., a conversation) without any damage being inflicted on the recording device by the telephone line ring tones. Ring tone voltages are approximately 100 volts and may cause significant damage to the PC's sound card if it is not adequately protected. The existing telephone coupler performs this conversation recording task flawlessly.

Unfortunately, as soon as the telephone coupler is attached to the telephone line and activated, the telephone is effectively "off hook." This is equivalent to picking up the telephone handset. When the telephone goes "off hook," all caller ID signaling is *immediately* terminated. This fact should help explain why, when you pick up the handset you never hear caller ID signaling, and why no data is displayed on a caller ID device if you pickup the telephone handset (answer the phone) *immediately* after the first ring tone.

The telephone system senses that the customer's telephone is "off hook" by monitoring the DC current being supplied to the customer's telephone. As shown in Figure 2, a slight modification to the coupler will block this DC current from flowing. This will allow the telephone coupler to monitor telephone line activity without the system sensing an apparent "off hook" condition.

With a fully operational telephone coupler at our disposal, we then recorded a number of caller ID signals as monoaural waveforms, using 16 bits per sample, and a sample frequency of 44,100 samples (i.e., 44.1 ksamples) per second. These recording parameters allowed the signals to be saved as a wave file for subsequent playback or distribution on an audio CD. With copies of these recorded signals now available to all of our students, multiple telephone couplers are no longer needed, nor is any coordination required to ensure that an incoming telephone signal is available during algorithmic development.

Figure 1: Photograph of the modified Comrex model TCB-1 telephone coupler (cover removed). The audio cable (1/4-inch plug to 1/8-inch plug) shown in the foreground is used to connect the telephone coupler to a PC's sound card.

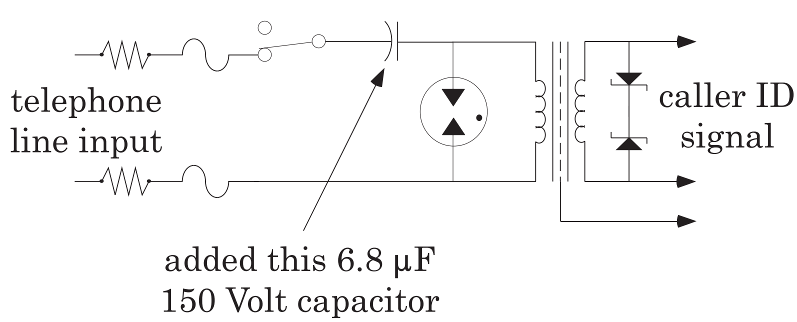

Figure 2: Schematic diagram of the modified telephone coupler, with the added capacitor shown.

4 DSP-based Implementation of the Caller ID System

The block diagram of the overall DSP-based system is shown in Figure 3, with the caller ID signal input coming from the Comrex coupler. The functionality of each of the blocks shown in Figure 3 is implemented on a Texas Instruments TMS320C6711 or TMS320C6713 DSK. A host PC is required to develop the algorithms needed to perform these operations, to control the DSK, and to act as the display device for the caller ID information. A more detailed diagram of the FSK demodulator block from Figure 3 is shown in Figure 4.[12,13]

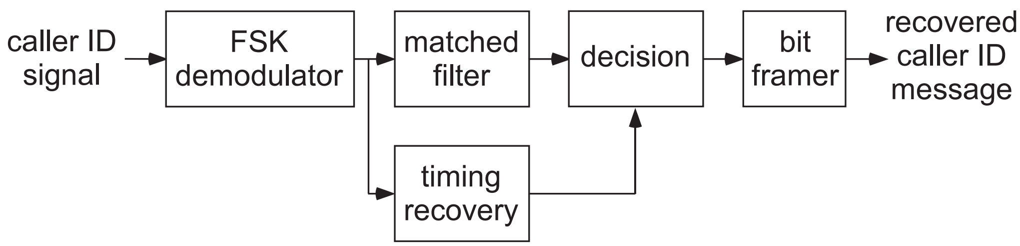

Figure 3: Block diagram of the DSP-based caller ID system.

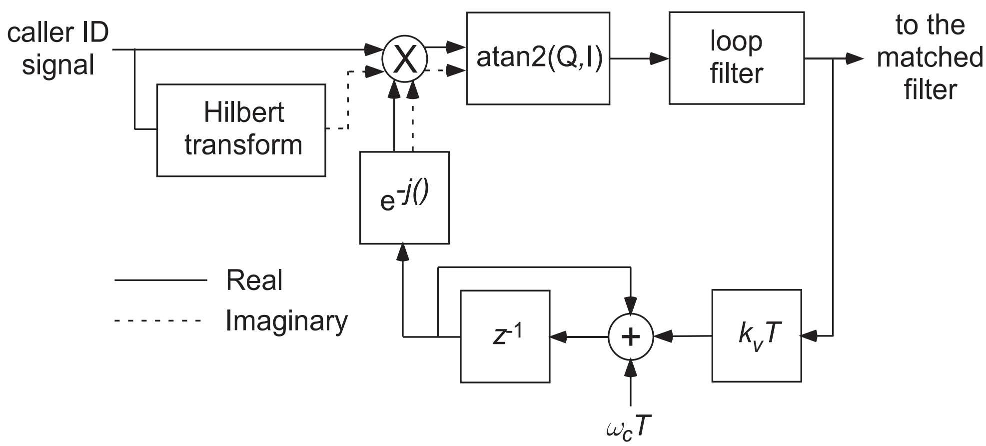

Figure 4: Block diagram of the DSP-based FSK demodulator.

Finally, a more detailed diagram of one of the many possible implementations of the FSK demodulator's loop filter from Figure 4 is shown in Figure 5. This loop filter structure is of particular interest since a high performance lowpass filter with variable gain can be implemented with only 2 coefficients. The frequency response of this first order infinite impulse response (IIR) filter is shown in Figure 6.

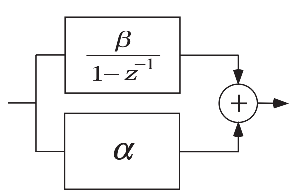

Figure 5: Block diagram of the FSK demodulator's loop filter.

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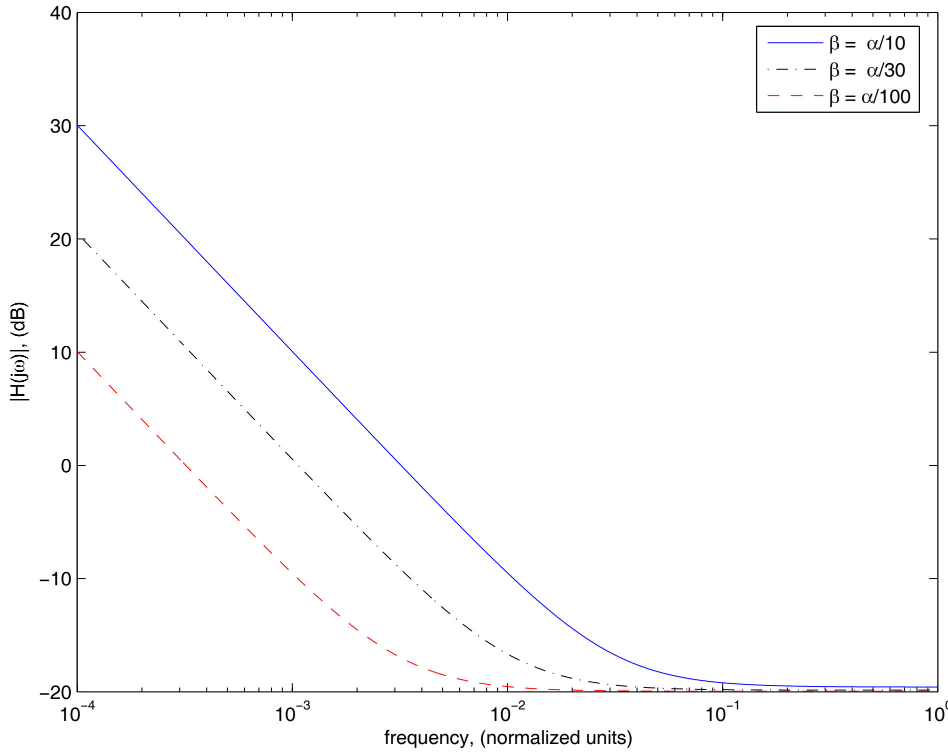

Figure 6: Frequency response of the FSK demodulator's loop filter ($\alpha = 0.1$).

5 But Why Use Caller ID?

Primarily due to cost constraints, we have limited the majority of our real-time DSP hardware projects to monaural or stereo signals in the audio band. These limitations allow audio frequency interfaces/daughterboards (analog to digital converters (ADC's), digital to analog converts (DAC's), and coder decoders (codec's)[14]) to be used with affordable digital signal processing (DSP) starter kits (DSKs).[15,16] A Texas Instruments TMS320C6713 DSK is shown in Figure 7.

The U.S. Naval Academy, the University of Wyoming, and Johns Hopkins University (EPP) all use a thematic approach to their DSP hardware courses. Specifically, these courses consist of:

week 1 Course introduction

week 2 Introduction to sampling and Code Composer Studio (CCS)

week 3 Sampling projects

week 4 FIR filters

week 5 FIR filter projects

week 6 IIR filters

week 7 IIR filter projects

week 8 Signal generation

week 9 Signal generation projects

week 10 Frame-based signal processing

week 11 Frame-based signal processing projects

weeks 12–15 Final project and presentation time

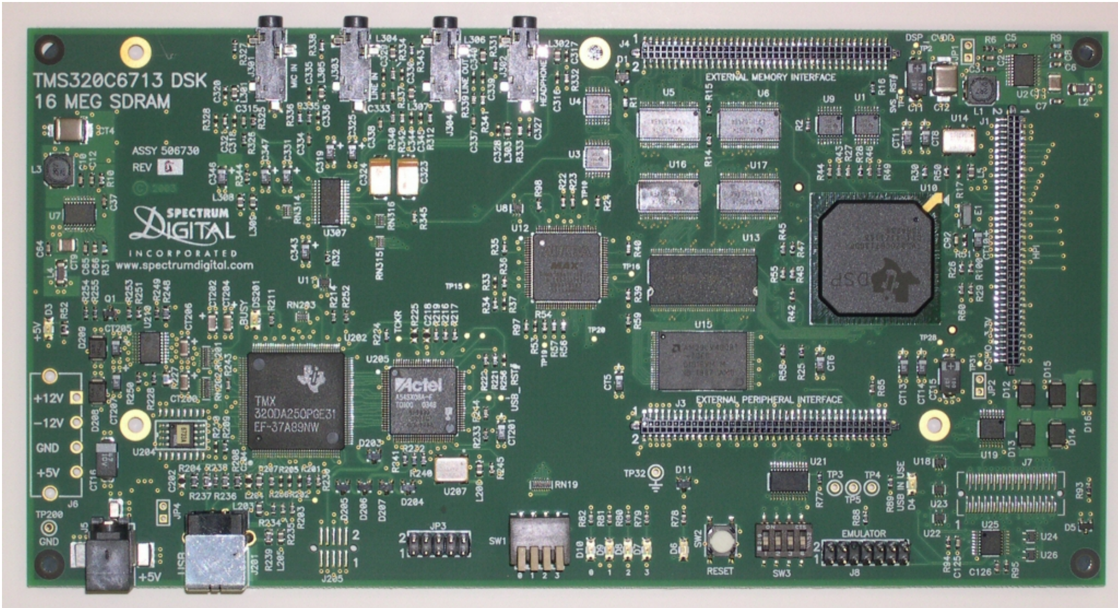

Figure 7: A Texas Instruments TMS320C6713 DSP Starter Kit (DSK).

We call “sampling,” “FIR filtering,” “IIR filters,” “signal generation,” and “frame-based signal processing” the *enduring fundamentals* of hardware-based DSP and spend a great deal of time on each of these topics.

After a review of the DSP theory associated with one of the enduring fundamentals; the remainder of that week and all of the following week is spent implementing student projects. This “2 weeks per topic” approach to teaching allows the students to become much more familiar with the prerequisite DSP theory, the available implementation strategies, and the procedures associated with testing their DSP hardware projects. Students, or groups of students, work largely at their own pace and given the self-selected nature of their projects, usually tend to fill the in-class time without difficulty. Finally, this approach ensures that the enduring fundamentals are very well covered prior to moving on to the final project. This approach has dramatically reduced the level of student frustration and increased student enthusiasm and learning.

Given that we are looking for real world audio frequency communication signals to demodulate for these DSP hardware courses, the telephone, and its associated digital signaling, is a natural choice. If you look closely at the block diagram of the caller ID demodulation system in Figure 3 you will see that almost *every* “enduring fundamental” of a DSP course is needed to successfully implement the system. Specifically,

- Sampling: Rate selection of the codec.
- FIR filtering: Hilbert transforming filter design and implementation at the input to the FSK demodulator and the matched filter at the output of the FSK demodulator.
- IIR filtering: Parallel filter design and implementation of the FSK demodulator’s loop filter.
- Signal generation: Complex local oscillator design and implementation associated with the FSK demodulator’s voltage controlled oscillator (VCO).

Additionally, the implementation and integration of the entire FSK demodulator, timing recovery loop, bit decision process, the framing of multiple bits into ASCII characters, verification of the check sum, and display of the resulting caller ID information are all required tasks of the caller ID DSP system. During this project, our students must also decide if sample-by-sample processing or frame-based processing will be used, and if any output other than just ASCII text will be provided. As discussed in,[7] the opportunity to use the caller ID information to control (turn "ON" or "OFF") various electrical loads can be a novel addition to many DSP-based projects.

6 Conclusions

We have described the use of a modified telephone coupler to allow student access to caller ID signals and have shown a number of the DSP techniques that are required to successfully demodulate, decode, and display the caller ID information. This project has been exceedingly well received by both our undergraduate and graduate students at our respective institutions. It represents a DSP-based solution to a real-world communications problem associated with a ubiquitous consumer device which every student has seen and used. Just as importantly, this project involves a very broad array of real-time signal processing concepts for proper implementation, yet the project is not so overwhelming in scope that it cannot be completed by a motivated student in a reasonable amount of time.

We highly recommend educators consider a project similar to this for any DSP class that includes exposure to real-time DSP hardware.

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