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# AN AFFORDABLE SOFTWARE DEFINED RADIO

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# ABSTRACT

This paper discusses the utilization of a relatively inexpensive wideband radio receiver in combination with a digital downconverter (DDC) based data recorder to capture and record real world radio signals. The resulting in-phase (I) and quadrature (Q) data files are then imported into MATLAB for processing. This batch processing of real world radio signals allows for a tremendous amount of classroom flexibility in the discussion of software defined radio topics.

*Index Terms*— Communication, digital signal processing, real time systems, software defined radio, SDR

# 1. INTRODUCTION

There is a great deal of interest in the DSP algorithms necessary to demodulate communications signals. While a number of existing courses cover these topics, the use of real world communications signals to develop and test these algorithms can be problematic. For many universities, the largest challenge in working with real world signals is the cost of the equipment necessary to detect, track, and capture the signals of interest. Two instrument grade, but costly, solutions to this signal capture problem can be found in references [1] and [2].

An alternative to the instrument grade test and measurement equipment solution is the use of a commercial-off-theshelf system that was originally designed to support the amateur radio community. A photograph of the high speed streaming digitizer, SDR-14 [3], is shown in Figure 1. In this capacity the system provides filtering, amplification, and samples for signals from 0.1 MHz to 30 MHz. The resulting information is then streamed as decimated in-phase (I) and quadrature (Q) data to a host computer using a USB connection. Figure 2 shows a typical display for a system setup to capture a weak commercial AM radio station's signal.

Unlike a number of available signal capture devices, this system is reasonably priced (approximately 1,000 USD) and is only limited in its recording capability by the available storage of the host computer's hard drive. For example, a one minute recording of an AM radio station created a 10 MB file.



Fig. 1. The SDR-14 is a high speed streaming digitizer.

#### 2. COMMERCIAL AM

Using only a simple loop antenna connected directly to the SDR-14, the signal is captured and the resulting file is imported into MATLAB for processing and algorithm development. For AM demodulation this only requires a few lines of MATLAB code. Specifically,

envelope = abs(I + j\*Q);

which extracts the signal's envelope from the I and Q data.

which removes the DC bias from the envelope. The message is now available for playback using the computer's soundcard. If multirate signal processing is a topic of concern, as shown in Figure 3, full control of the SDR-14's digital down converter's decimation and filtering processes is possible, in order to create the required I and Q data.

#### 3. COMMERCIAL FM

Another common signal is the commercial frequency modulation (FM) radio station signal. An FM signal (88–108 MHz,



**Fig. 2**. Screen capture of the SpectraVue software application capturing a weak AM radio station centered at 1140 kHz using a span of 50 kHz.

Undata Eirauara	A/D Sample Freq(Hz)	Bef 1000000	RF Input Source
Opuale Filliwale	66666687 Me	as 10000000	C Direct Input(ch 0
External HW Sync		Cala	© 0.30 MHz (ch 1
External Radio, I	F / Panadapter Mode Setup	Laic	
None Select Ext Radio SDR14 Select Serial Port 3600 Select Bit Rate IF Center Freq(Hz) Ext 0	ADR IAA Inverted II HF Tracks IF BW(Hz) Ext Center	ne Center Freq F Ing Mode Frequency(Hz)	Ch 1 RF Gain Ch 1 RF Gain Ch 1 RF Gain Ch 20 dB Ch 2
Filter Bandwidth	6620 Di	igital Downconver	ter Settings
C 5 KHz C 10 KHz C 25 KHz C 25 KHz Demod 0k C 1000 KHz	CIC2 Rate 5 CIU CIC5 Rate 30 CIU RCF Rate 4 RU	C2 Scale 3 C5 Scale 20 CF Scale 0	6620 IF Gain • +24 dB +18 dB +18 dB +12 dB +12 dB
50 KH2	RCF TAPS 256 Total Decimation = 600 Final Sample Rate = 111111		
• 100 KHz			

Fig. 3. SDR-14 setup/controls (to include digital downconverter settings).



Fig. 4. AR 5000A communications receiver.



**Fig. 5**. A strong FM signal (94.3 MHz) captured using the SDR-14 connected to the AR5000A's IF output.

in the United States) would be a challenge for the SDR-14 to capture without additional analog RF signal conditioning circuitry. An alternative to designing and implementing this analog RF signal conditioning circuitry is the use of a radio receiver that has its intermediate frequency (IF) signal available for processing by the high speed digitizer (the SDR-14). The radio system we selected is shown in Figure 4.

With only minor configuration changes to the SDR-14's software controls, the system can capture the 10.7 MHz IF signal. An example of such a signal is shown in Figure 5. The MATLAB processing of this captured signal involves numerous steps. Specifically,

- Import the wavfile into the MATLAB workspace
- Convert the wavfile's data to I and Q format
- Recover the FM signal's message using the MATLAB command,



**Fig. 6**. In-phase and quadrature components of a commercial FM signal.

message=diff(unwrap(angle(I + j \* Q))); A plot of a typical commercial FM signal in I and Q format is shown in Figure 6. A *perfect* FM signal would plot as a circle instead of the wide ring shown. Spectral analysis of the recovered message results in Figure 7.

At this point in the message recovery process, the FM mono message signal can be listened to by playing the message through the host computer's soundcard. This process uses the analog audio circuitry as the lowpass filter to remove the undesired portions of the FM composite baseband signal. Basically, the soundcard and its attached speakers will filter out any signal above approximately 20 kHz. Any remaining signal above this frequency would not be heard by *normal* human hearing.

#### 4. RBDS

Most commercial FM radio stations in the United States transmit a radio broadcast data system (RBDS) signal [4]. The RBDS (or RDS) signal is a significant next step in radio sophistication in that this signal has a 57 kHz carrier (3 times the 19 kHz pilot shown in Figure 7) and uses biphase digital communication techniques to represent the bits that eventually result in an ASCII-based character display on a fairly new radio receiver's display. To recover these bits several steps are required. Specifically,

• The RDS signal centered on 57 kHz must be isolated using a bandpass filter. The results, in the sample do-



**Fig. 7**. The composite baseband spectrum of the FM signal's message.

main, of such a filtering operation are shown in Figure 8.

- The filtered signal must be resampled to ensure that there are an integer number of samples in a symbol period (1/1187.5 seconds). This seemingly odd bit rate (1187.5 bps) is due to the integer relationship (48) between 1187.5 and 57,000. The details of this relationship are available in reference [4]. If the resampling operations are accomplished properly, this will only result in a new sample frequency. In this example, the initial sample frequency was 158,730 Hz. Using P and Q values of 5700 and 5291, respectively, results in a new sample frequency of 171 kHz, which is related to 1187.5 by the integer 144.
- Mix the signal to baseband using a local oscillator or a phase locked loop (PLL).
- Lowpass filter this signal to recover the desired biphase signal.
- Plot the signal's eye pattern. The result of timing recovery is shown in Figure 9.

From the perspective of a communications course, our work is now complete, since we have achieved an open eye pattern. However, most students prefer to return the signal to a character-based display for a more intuitive result.

# 5. CONCLUSIONS

We have offered a relatively inexpensive alternative to the commercially available vector signal analyzer hardware and



Fig. 8. The results of filtering (isolating) the RBDS signal.



Fig. 9. The RBDS signal's biphase eye pattern.

software. While this approach is much more labor intensive to use, it results in considerably more student understanding of the underlying algorithm associated with analog and digital communications systems. This approach has also resulted in new interest in both our communication and DSP course offerings.

In a perfect world, all students would be exposed to both the low cost and the instrument grade approaches to vector signal analysis. However, budget realities of individual institutions may not make this possible. The monetary investment required to implement the low cost approach described in this paper should be within reach of nearly any university.

# 6. REFERENCES

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