AC 2008-120: USING REAL RF SIGNALS SUCH AS FM RADIO TO TEACH CONCEPTS IN COMMUNICATION SYSTEMS

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Abstract

The theory of communication systems can be made more concrete and interesting by using real-world signals to illustrate the various theoretical concepts. The FM radio signal is, in many ways, an ideal signal to illustrate many of the concepts that are taught in communication system courses. This radio frequency (RF) signal is readily available in most populated areas, the students are familiar with the signal, and most FM stations broadcast both analog audio signals and digital Radio Data System (RDS) signals which can be used to illustrate digital receivers. The RDS signal contains auxiliary information such as the radio station identification, type of program, name of the current song, artist name, current time, traffic reports, etc. The RF signal that is broadcast from local FM radio stations, as well as most other RF signals, can be captured with hardware such as a real-time spectrum analyzer, a vector signal analyzer, or the Universal Software Radio Peripheral (USRP). The analog audio signals can be demodulated in software and used to demonstrate FM demodulation, frequency division multiplexing, double sideband suppressed carrier demodulation, mixing, and filtering. The digital RDS signal can be demodulated using a relatively simple software defined receiver which consists of a Costas loop, matched filter, timing recovery algorithm, and slicer. The FM radio signal can be used in classroom demonstrations or in student projects or homework assignments to enhance a communication systems course, motivate the students with a real-world system, and to provide students with the practical experience of creating and testing a software defined radio receiver.

Introduction

Some students are more motivated to learn material when it is clear that the material is useful in real systems. One way to show how concepts are used in practice is to examine existing commercial systems, but gaining access to commercial systems is impractical in many cases. However, capturing the wireless signals generated by real systems is often possible with commercially available test equipment, and these captured signals can be used to teach many of the concepts commonly found in communication systems courses. These signals can also be decoded in software to demonstrate concepts in software defined radio.

The FM radio signal is an excellent candidate for use in communication system courses because it (usually) contains both analog and digital signals, is widely available, and the students are familiar with the FM radio system. The digital signal is fairly easy to demodulate because it is a low bit rate system that does not necessarily require an adaptive equalizer, the bits are not interlaced, and it uses a fairly simple error control coding scheme. This paper explains how the FM radio signal can be used to teach many of the concepts in a communication system course.

Using Real RF Signals

The author's communication systems course uses real AM and FM radio signals to demonstrate various concepts in communication systems. In addition to the homework assignments which deal with the theory of communication systems, the students perform a series of five projects using real signals as shown below.

1. Spectrum Analyzer Laboratory Experiment
2. AM Demodulation and Frequency Division Multiplexing
3. FM Demodulation
4. RDS Demodulation
5. RDS Decoding

In the first project, the students use a Tektronix RSA3408A Real Time Spectrum Analyzer to examine and capture the RF signal from an AM radio station, an FM radio station, and the entire AM band. In the second project, the students load the signal from the AM station into MATLAB, demodulate it, and listen to the result. They also design a bandpass filter to isolate a single AM radio station from the recording of the entire AM band, filter the recorded signal, demodulate it, and listen to the result.

In the third project, the students load the signal from the FM radio station into MATLAB, demodulate it and recover the mono and stereo audio signals. This project requires the students to demodulate a double-sideband suppressed carrier signal to recover the stereo information. In the fourth project, the students isolate the digital RDS signal from the FM radio station and use MATLAB to demodulate it, which requires a digital receiver consisting of a Costas Loop, a matched filter, a timing recovery algorithm, and a slicer. Finally in the last project, the students decode the RDS bits to recover the type of program (sports/news/rock/etc.), the name of the program, the name of the song, and the name of the artist. This project requires the student to perform a CRC check to find the correct word alignment, and to decode the digital messages.

Capturing RF Signals

One way to capture RF signals is to simply connect an antenna to a Tektronix Real-Time Spectrum Analyzer. These spectrum analyzers have high speed analog-to-digital converters that capture the complex envelope of the RF signal in a data file that can be transferred to a computer by either a computer network or a USB drive. Some example recordings of the RF signal from FM radio stations can also be obtained for free from the author's website. The data file (filename.iqt) contains raw 16-bit integers that represent the in-phase and quadrature components along with scaling factors and frequency calibration data. A program available for free from the author can be used to load the data into the numerical package MATLAB and apply the scaling factors and frequency calibration. The output of this program is the complex envelope signal where the real part of the vector is the in-phase component and the imaginary part is the quadrature component.

Other methods of capturing RF signals include using an Agilent Vector Signal Analyzer[1] or the Universal Software Radio Peripheral[2] (USRP), which is designed to work with a Linux-based open source project called GNU Radio. In addition to being much less expensive than the real-time spectrum analyzer and the vector signal analyzer, the USRP also has the advantage that it can be used to build receivers (and transmitters) that operate in real-time.

Structure of the FM Radio Signal

Once the FM radio signal is captured and transferred to a computer, the next step is to demodulate the signal. Frequency modulation causes the angle of the complex envelope signal to change, and the rate of change is proportional to the amplitude of the original signal. Therefore the FM complex envelope signal can be demodulated using a phase-locked loop specifically designed for the complex envelope signal[3] as shown in Figure 1, or the angle of the signal can be calculated, unwrapped, and differentiated[4].

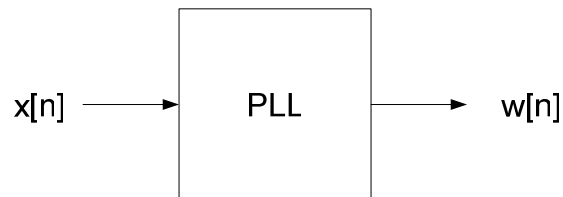$x[n] \longrightarrow$ PLL $\longrightarrow w[n]$

Figure 1: Demodulating the FM Signal

After the FM signal is demodulated, the spectrum of the signal can be examined to show that the demodulated signal contains several signals separated by frequency division multiplexing as shown in Figure 2. The L+R signal is an analog signal representing the mono audio signal. The L-R signal is an analog signal used to convey stereo audio information which has been double sideband suppressed carrier (DSB-SC) modulated using a carrier frequency of 38 kHz. The signal at 57 kHz is a digital signal called Radio Data System (RDS) which carries digital information such as the name of the current program, the title of the song, the artist's name, station ID, current time, traffic reports, etc. The demodulated FM signal could be used to show how frequency division multiplexing is used to allow several sources of information to be carried by a common channel. (Since the various radio stations are also separated by frequency division multiplexing, another way to show frequency division multiplexing is to record more than one radio station simultaneously and use filters to separate the stations.) The 19 kHz pilot is used to indicate the presence of the stereo signal and is sometimes used in demodulating the L-R and RDS signals.
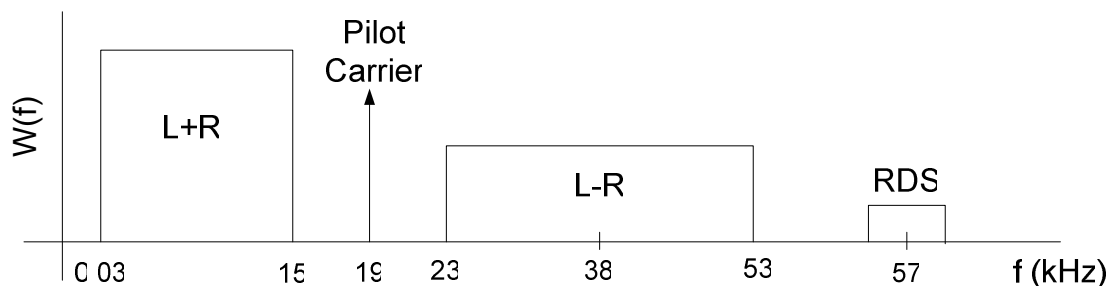
Figure 2: Spectrum of FM Radio Signal After Demodulation

Extracting the Analog Audio Signals

A bandpass filter can be used to isolate the mono audio signal as shown in Figure 3. In order to reduce the noise in the demodulated audio signals, FM radio systems use a pre-emphasis filter at the transmitter to amplify the high frequencies, and a corresponding de-emphasis filter in the receiver to attenuate the high frequencies back to their original strength.
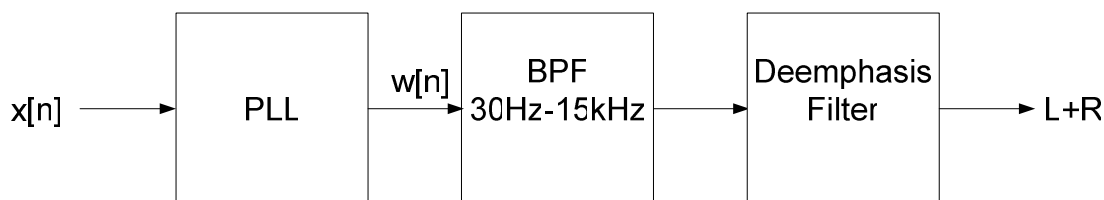

Figure 3: Mono Decoder

If implemented as an analog filter, the de-emphasis filter would be a single-pole lowpass filter with a frequency response given by $H(\omega) = \dfrac{1}{1 + j\omega/\omega_1}$ where $\omega_1 = \dfrac{1}{RC}$ and $RC = 75\mu S$ in North America[5]. In order to design a digital de-emphasis filter, we can pre-warp the frequency and use the bilinear z-transform as follows[6]: $\omega_{pp} = \tan\left(\dfrac{\omega_1}{2f_s}\right)$ where f_s is the sampling rate in

Hz, $a_1 = \dfrac{\omega_{pp} - 1}{\omega_{pp} + 1}$, and $b_0 = b_1 = \dfrac{\omega_{pp}}{\omega_{pp} + 1}$. The difference equation for the filter is then given by

$y[n] = b_0 x[n] + b_1 x[n-1] - a_1 y[n-1]$. The students can instantly verify that the signal was recorded and demodulated correctly by listening to the output of the deemphasis filter, especially if they listened to the radio station while recording the FM radio signal so they know what audio signal to expect.

A more complex system that can be used to obtain the left and right audio channels is shown in Figure 4. The L+R and L-R signals are separated by bandpass filters. The DSB-SC modulation of the L-R signal can be demodulated using a Costas loop[3], and then combined with the L+R signal to extract the left channel L and the right channel R. The scaling factor of 2 in the output signals can easily be removed if desired. The students can implement this processing in software and then listen to the output to see if they demodulated the signal correctly.
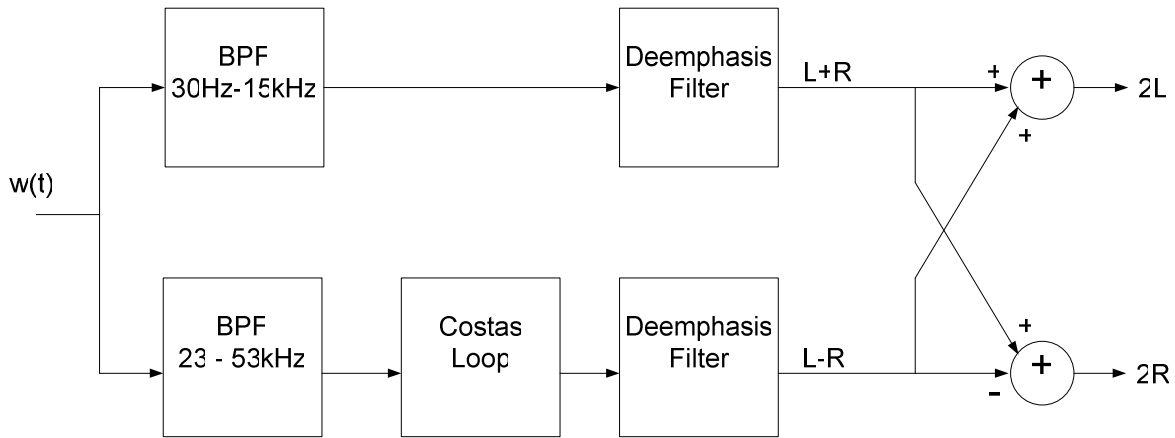
Figure 4: Stereo Decoder

A Radio Data System Receiver

The Radio Data System (RDS) system, which is also called Radio Broadcast Data System (RBDS) in the United States, transmits digital auxiliary information such as the program name, title of the song, artist's name, etc. This information enables a receiver to perform certain functions such as searching the available stations for a specific type of music, automatically switching to a station that is currently broadcasting a traffic report, and displaying the name and artist for the song that is currently playing. The RDS signal uses a carrier frequency of 57 kHz, bi-phase (or Manchester) encoding, a raised cosine pulse shape, and a bit rate of 1187.5 bits/sec[7,8].

A block diagram of one possible RDS receiver is shown in Figure 5. In order to demodulate the RDS signal, it is first necessary to isolate it from the audio signals using a bandpass filter whose center frequency is 57 kHz and whose bandwidth is about 5 kHz. Next the RDS signal is shifted back down to baseband by a Costas loop[3]. Then a matched filter whose impulse response is a raised cosine shape is used to remove as much noise as possible. The combined raised cosine pulse shape in the transmitter and the receiver has a rolloff factor of 100%.
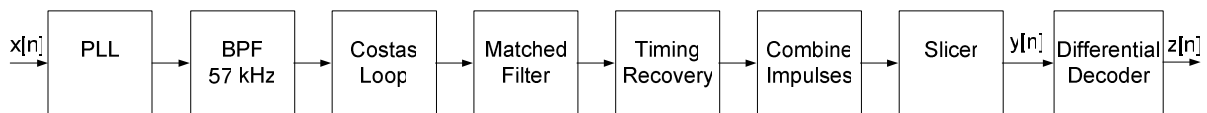

Figure 5: Block Diagram of one Possible RDS Receiver

A timing recovery algorithm is used to determine the best time to sample the signal by maximizing the power of the signal at the sample time[9]. Since two pulses are sent for each bit (for logic 1, +1 then -1 is sent, and for logic 0, -1 then +1 is sent), the sampling rate of the timing recovery algorithm is two times the bit rate, resulting in two values for each bit. The second of these two values is subtracted from the first value in the Combine Impulses block to form a single value which will be positive for a logic 1 bit and negative for a logic 0 bit. If the output of the Combine Impulses block is positive, the slicer decides logic 1 was sent, and if the value is

negative, the slicer decides logic 0 was sent. Finally the differential encoding is reversed by the Differential Decoder block whose operation is described in Table 1.

Table 1: Differential Decoder

Current Input y[n]	Previous Input y[n-1]	Current Output z[n]
0	0	0
0	1	1
1	0	1
1	1	0

It is also necessary with this receiver to have a mechanism to determine if a pulse is the first or second pulse in the pair of pulses that represents a bit. If the Combine Impulses block is properly aligned with the pulses, the two pulses in each pair should have opposite signs, so one way to pick the best alignment is to count how many pairs have opposite signs for each of the two possible alignments, and pick the best alignment. The output z[n] is a stream of bits that represent the RDS data, and the format of this data is discussed in the next section.

RDS Data Format

The RDS data is formatted in sets of 16 message bits and 10 checksum bits, and together these 26 bits are called a **block** (see Figure 6). Four blocks make up a **group**, so that each group has 4*26 = 104 bits.
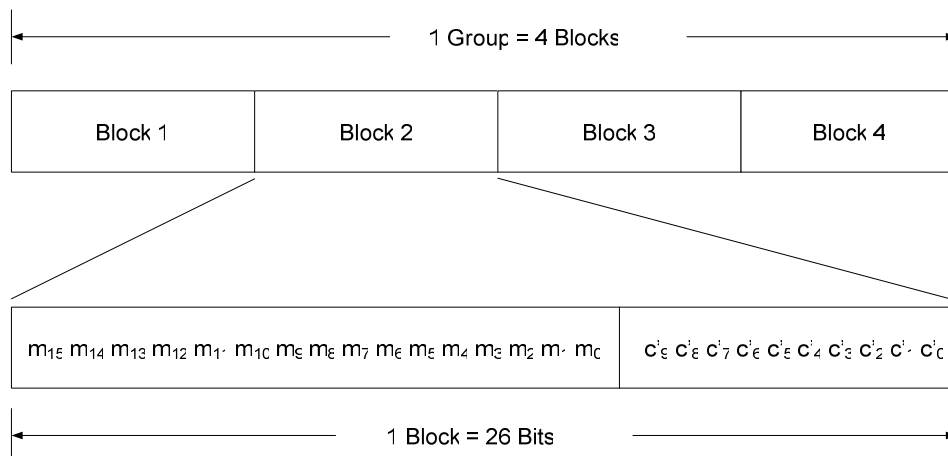

Figure 6: Blocks and Groups

The checksum bits are a cyclic redundancy check (CRC), and are computed by multiplying the message bits by x^{10} and then dividing (modulo 2) by the generator polynomial $g(x) = x^{10} + x^8 + x^7 + x^5 + x^4 + x^3 + 1$. By adding (modulo 2) the remainder to the message bits multiplied by x^{10}, a 26 bit block is formed that has the property that if it is divided by the generator polynomial, the remainder will be all zeros. This property can be used to check for bit errors.

The checksum is also used to determine the correct word alignment by performing the CRC for each possible word alignment to find the one that satisfies the CRC check. In order to prevent a bit shift from being mistaken for another valid code, an **offset word** is added (modulo 2) to the checksum bits before they are transmitted (see Table 2).

Table 2: Offset Words[7]

Offset Word	$d_9d_8d_7d_6d_5d_4d_3d_2d_1d_0$
A	0 0 1 1 1 1 1 1 0 0
B	0 1 1 0 0 1 1 0 0 0
C	0 1 0 1 1 0 1 0 0 0
C'	1 1 0 1 0 1 0 0 0 0
D	0 1 1 0 1 1 0 1 0 0
E	0 0 0 0 0 0 0 0 0 0

There are two types of groups that may be sent called Type A and Type B. The group type can be inferred from the offset words that have been added to the checksum bits. For Type A groups, the offset words A, B, C, and D are added (modulo 2) to the checksum bits of blocks 1, 2, 3, and 4 respectively. For Type B groups, the offset words A, B, C', and D are added to the checksum bits of blocks 1, 2, 3, and 4 respectively. If the offset word E is added to all four blocks, that indicates that the group is used by a paging system (MMBS), and so this group should be ignored by RDS receivers.

There are many different types of groups (32 Type A groups and 32 Type B groups) that may be used to transmit different types of information[7]. One of the groups that carries information that students may find interesting is Type 2A Radio Text, which contains text describing the current program such as the name of the song, the artist, and the name of the radio station. Other groups that may be interesting are Type 0B which carries basic tuning information including the program type name and the program service name, and Type 4A which carries the current time. Only the Type 2A Radio Text group is described here.

The format of a Type 2A Radio Text group is shown in Figure 7 and a list of the various items encoded in the Type 2A group is shown in Table 3. Each Type 2A group is called a segment and carries four characters (CH1 - CH4). Several Type 2A groups (up to a maximum of 16) are sent to convey a message that is up to 64 characters long. The TSA field is a 4-bit binary counter that starts at 0 with the first segment and counts up to the end of the message (with a maximum of 15). If the message requires fewer than 16 segments, the end of the message is marked with the hex character 0D. The value of the A/B flag changes at the beginning of each new message.
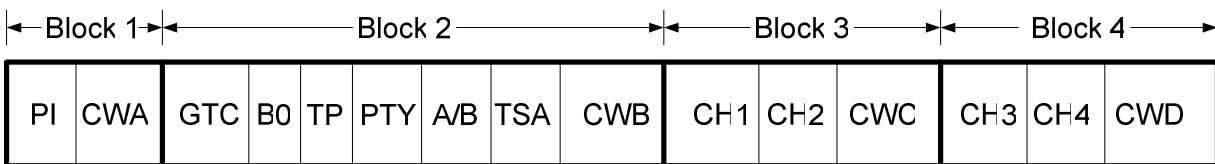

Figure 7: Format of Type 2A (Radio Text) Group

Table 3: Items Encoded in Type 2A (Radio Text) Group

Abbreviation	Number of Bits	Name	Description
PI	16	Program Identification	Unique number that Identifies Station and the program
CWA	10	Checkword + Offset Word A	CRC Checkword Plus Offset Word A
GTC	4	Group Type Code	Identifies Group Type (2 for Type 2A groups)
B0	1	Group Type	Identifies Group Type (0 - A, 1 - B) (0 for Type 2A groups)
TP	1	Traffic Program	Indicates if station broadcasts traffic information
PTY	5	Program Type	Type of program
A/B	1	Text A/B Flag	Changes when a new message begins
TSA	4	Text Segment Address	Counter for the groups in the Radio Text message
CWB	10	Checkword + Offset Word B	CRC Checkword Plus Offset Word B
CH1	8	Character 1	First Character in this group
CH2	8	Character 2	Second Character in this group
CWC	10	Checkword + Offset Word C	CRC Checkword Plus Offset Word C
CH3	8	Character 3	Third Character in this group
CH4	8	Character 4	Fourth Character in this group
CWD	10	Checkword + Offset Word D	CRC Checkword Plus Offset Word D

The PTY code indicates the type of program (sports/news/rock/pop/etc.), which can be used by a receiver to search for a radio station that carries a particular type of program. Since the PTY is a 5 bit code, there are 32 different possible values. Table 4 shows the 8 character string that the receiver displays in North America[7].

Table 4: PTY Codes for North America (RBDS PTY)

Value	8-Character String	Description
0	None	No Program Type
1	News	News
2	Inform	Information
3	Sports	Sports
4	Talk	Talk
5	Rock	Rock
6	Cls_Rock	Classic Rock
7	Adlt_Hit	Adult Hits
8	Soft_Rck	Soft Rock
9	Top_40	Top 40
10	Country	Country
11	Oldies	Oldies
12	Soft	Soft
13	Nostalga	Nostalgia
14	Jazz	Jazz
15	Classicl	Classical
16	R_&_B	Rhythm and Blues
17	Soft_R&B	Soft Rhythm and Blues
18	Language	Foreign Language
19	Rel_Musc	Religious Music
20	REL_TALK	Religious Talk
21	Persnity	Personality
22	Public	Public
23	College	College
24		Unassigned
25		Unassigned
26		Unassigned
27		Unassigned
28		Unassigned
29	Weather	Weather
30	Test	Emergency Test
31	ALERT_!	Emergency

Discussion

The FM radio signal can be used to demonstrate many of the concepts taught in communication system courses. The various parts of the signal (L+R, L-R, and RDS) are separated by frequency division multiplexing, and so separating the signals can be used to demonstrate filters and DSB-SC demodulation. The de-emphasis filter can be used to demonstrate a real-world application of filter design. The RDS signal can be used to demonstrate the Costas loop, matched filter, timing recovery, slicer, word synchronization, and error control coding.

Listening to the resulting audio signal gives the students immediate feedback on the quality of their receiver, which may be more interesting and satisfying than examining a graph or calculating a signal-to-noise ratio. The students can examine the RDS data, especially the radio text data, to see if it was decoded correctly and to see if it is consistent with the audio signal. Working with this real-world signal also teaches the students about a common commercial system with which they are likely to be familiar.

Conclusion

Using real signals to demonstrate concepts in communication systems can help show that the material that is being taught is useful in real systems. The FM radio signal is useful in teaching communication systems because it is widely available, contains both analog and digital signals, and results in output signals that are interesting to the students. Furthermore, the bit rate of the digital data is low enough that an adaptive equalizer is not necessary, and there is no bit interleaving, so the receiver is not as complicated as some practical digital receivers.

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