Increasing Spectrum Utilization for Voice Systems

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ABSTRACT

The United States Department of Defense and researchers throughout the world have been addressing the overcrowding of the radio frequency (RF) spectrum. This has led to numerous studies concerning cognitive radios, networks, and radar systems. In this paper we demonstrate the possibility of sending both analog voice and digital signals simultaneously using legacy analog radios without appreciably degrading the intelligibility of the audio signals and thereby increasing communication without requiring additional bandwidth.

Index Terms: multiple access interference, digital modulation, speech processing, electromagnetic compatibility.

1.0 INTRODUCTION

The radio frequency (RF) spectrum is crowded and more space is needed for wireless internet access, cell phone communications, and for military and civilian usage. The US Congress passed a bill to open up more spectra [1] to auction off RF frequencies belonging to the television broadcast industries. To slow down the need for more frequencies and to reduce the number of cell towers to accommodate the growing number of mobile phones, the industry is deploying microcell, picocell and femtocell technologies. However, these initiatives alone will not solve spectrum crowding.

When the frequency spectrum is measured over time, technologists have shown that the spectrum is underutilized. Recognizing this, there have been numerous research projects funded by the US Department of Defense (DOD). The Defense Agency Research Project Agency (DARPA) has probably funded the most projects in this area. Through this research, we now have two distinct users defined as the primary user (PU) (i.e. those who own the license for the frequency range) and the cognitive user (CU) (i.e. those users trying to share the spectra either by using broadband signals or sampling the spectra in time and transmitting when the PU is not transmitting). Most significant projects in this area include the DARPA XG program and the Wireless Network after Next (WNaN) program. In addition to these efforts, there has been a move to apply Cognitive Radio (CR) technologies to the radar domain (Cognitive Radar efforts) and radio networks [2-7]. Some of these systems sample the spectrum and transmit if no one else is transmitting at any given frequency. This approach can cause electromagnetic interference (EMI) in nearby receivers. Many people have recognized this problem and have addressed it in many different ways [8-14].

The DOD has many legacy radios that can only transmit and receive voice signals over very low bandwidths using amplitude modulation (AM) or frequency modulation (FM). Our challenge was to investigate these legacy radios with limited bandwidths and attempt to send both analog voice and digital signals simultaneously without appreciably degrading the intelligibility of the audio signals as opposed to CUs transmitting in the same frequency bands while not knowing they are interfering with the nearby PUs. Achieving this goal will allow the DOD to extend legacy analog radios effective digital bandwidth and provide digital information to PU radios without requiring any additional bandwidth.

There have been very few attempts in the literature to address the problem of simultaneous transmission of analog voice and digitally modulated data over analog channels. In order to search for ideas and suitable approaches one has to resort to literature that deal with somewhat broader problems such as information embedding and watermarking [15-17]. While some general ideas presented in these and other related papers can be applied to our problem, it is important to note that they usually do not have very strong restrictions on the available bandwidth (as we are faced with), and often, especially in the case of watermarking, the digital data represents a specific signature and not information bearing data streams.

2.0 TECHNICAL APPROACH

Our goal is to simultaneously transmit analog voice signals and digitally modulated data over a very narrowband communications link using an analog radio. Thus, our digitally modulated data have to be considerably lower in amplitude than the analog signal and yet robust enough to sustain transmission over a narrowband analog RF link so that the digital signal can be successfully reconstructed at the receiver. These conflicting requirements make the selection of the appropriate modulation format challenging. In this paper we describe our investigation in using a Direct Sequence Spread Spectrum (DSSS) approach for modulating the digital data. The DSSS modulation technique has been very popular in the communication world for many years but for very different types of applications. It is primarily used in wideband multiuser systems. However, some of its features make DSSS suitable for meeting our goals as well. In this Section we will briefly outline the basic principles of DSSS.

The block diagram of a spread spectrum digital communications system is shown in Figure 1.



Figure 1. Spread spectrum digital communications system (from [18])

Most of the blocks in Figure 1 can be found in almost any digital communication system. Channel encoder is used to introduce redundancy in the information sequence so that the receiver can overcome the effects of noise and interference encountered in the transmission. In this paper we will not model a channel encoder and decoder. Modulation format can, in general, be any digital modulation format such as Phase Shift Keying (PSK), Frequency Shift Keying (FSK), etc.

What is characteristic for a spread spectrum system is a presence of two *identical* pseudorandom pattern generators at both the transmitter and receiver. These two generators typically produce a pseudorandom or pseudonoise (PN) binary valued sequence which is used to spread the transmitted signal at the modulator and to despread the received signal at the demodulator.

Let us for simplicity assume that the information bearing signal is modulated using the BPSK modulation. Then the modulated baseband signal can be expressed as:

$$v(t) = \sum_{n=-\infty}^{\infty} a_n g(t - nT_b)$$

where $a_n \in \{-1,1\}$ and g(t) is a rectangular pulse of duration T_b . T_b is referred to as a symbol interval (same as bit interval in the case of BPSK) and is inversely proportional to the data rate (throughput of the system). The generation of a DSSS signal is shown in Figure 2. The BPSK modulated baseband signal is multiplied with the signal from the PN sequence generator c(t) defined as:

$$c(t) = \sum_{n = -\infty}^{\infty} c_n p(t - nT_c)$$

where $c_n \in \{-1,1\}$ and p(t) is a pulse (or chip) assumed in Figure 2 to be rectangular with duration T_c . T_c is called chip interval and is inversely proportional to the bandwidth of the system. Intervals T_b and T_c are selected so that their ratio is an integer commonly referred to as spreading factor or spreading gain $L = \frac{T_b}{T}$.



Figure 2. Generation of a DSSS signal (from [18])

In other words, in DSSS systems each information symbol is multiplied (modulated) with a spreading sequence and, as a consequence, the resulting signal is spread across a frequency band wider than the original signal band. As a consequence, a DSSS signal has noise-like spectral characteristics, it is robust to jamming, noise and interference, and many users can share the same bandwidth with little interference. Some of these properties are exactly what we are looking for in our application. Robustness to noise and interference will allow us to lower the digitally modulated signal in amplitude as compared to the analog voice and still be able to reconstruct it at the receiver. In addition, its noise-like spectral characteristics and low amplitude will result in minimal degradation of the audio reception. The biggest challenge of this approach is how much data throughput one can achieve since we are spreading the original signal bandwidth over an already very narrowband channel.

The demodulation of the DSSS signal at the receiver is multiplied (despread) by exactly the same signal c(t) used

for spreading the signal at the transmitter. As expected, the synchronization of the PN sequence used at the transmitter and the PN sequence used at the receiver is required in order to properly despread the received signal. In this paper we will assume a very simple synchronization mechanism and model it as a sliding window correlator where needed. In practical systems the synchronization is usually established prior to the transmission of information by transmitting a specially designed fixed PN bit pattern.

Since $c^2(t)=1$ for all *t*, the received signal (in the absence of noise) after despreading becomes:

$$r(t)c(t) = Av(t)c^{2}(t)\cos 2\pi f_{c}t = Av(t)\cos 2\pi f_{c}t$$

where A denotes the signal amplitude and f_c is a carrier frequency. Thus, we are able to completely recover the information bearing signal v(t).

At the transmitter, analog voice and digitally modulated data are multiplexed at the baseband. The resulting signal is then used to modulate the transmitting RF signal using AM (or FM) modulation. The received RF signal is mixed with the Local Oscillator (LO) signal to produce an Intermediate Frequency (IF) signal. The IF signal is then applied to the AM/FM demodulator which provides the baseband signal. This signal is then demultiplexed – one output is sent to the analog circuitry that provides appropriate filtering and amplification, the other output is sent to the digital demodulator.

3.0 MODELING AND SIMULATION

In our application, because of the restrictions imposed on the bandwidth, we are forced to use short codes of length 4, 8, etc. What we do have to take into consideration are the spectral characteristics of the spreading code and its autocorrelation properties. We decided to perform all our simulations with Walsh-Hadamard sequences. However, in the future it might be worth trying different sequences, or even multiplexing multiple digital users for the purpose of increasing the overall digital throughput.

In order to shape the spectrum of the digital signal a root raised cosine filter was applied at the transmitter. This filter does not introduce the Inter Symbol Interference (ISI) which allows for a better symbol reconstruction at the receiver. Unless stated otherwise it is assumed that the effective bandwidth of the digital signal is approximately 3 KHz.

Our goal is to establish and maintain a digital data flow while making virtually no modifications to the analog part of the system. That is why our modeling of the analog system is very simple. It does not include any type of signal processing algorithm and only accomplishes the necessary discretization of the analog signal so that it can be mixed with the digital signal in the MATLAB domain that is inherently discrete. In our simulations we used two speakers, a male and a female, each represented with a speech segment approximately 20 seconds long that were sampled at 44.1 KHz with 16 bits per sample used for quantization. It was assumed that amplitude modulation was used to modulate the analog signal.

Finally, a channel noise was modeled as a zero mean white Gaussian process and added to the signal of interest obtained by multiplexing the speech signal and the digital signal.

There are several parameters that define a single simulation experiment: 1. L - spreading factor (number of chips per information symbol), 2. Signal to Noise Ratio (SNR) strength of the analog signal versus the strength of the noise (modeled as additive white Gaussian process) at the receiver, 3. Signal to Interference Ratio (SIR) – strength of the analog signal versus the strength of the digital signal and 4.T - throughput of the digital link.

Our goal is to select these parameters while accomplishing several goals: 1. Maintain the intelligibility of the speech, 2. achieve reasonable Bit Error Rates (BERs) that, with or without additional error correction coding, will guarantee a reliable digital connection and 3. Achieve significant throughputs worthy of implementation.

Meeting these multiple goals is a challenge since some of them are in direct conflict. For example, increasing the spreading factor will improve the BER but will reduce the throughput, while increasing the SIR will help the speech intelligibility but hurt the quality of the digital reception.

The results of our MATLAB simulations (for the male speaker and SNR = 15 dB) are presented in Figure 3. Table 1 shows the results for the male and female speakers, with SNR = 15 dB. We provide a name and a link [19] to all wav files so that a reader can assess how well the speech intelligibility was preserved.



Figure 3. BER results for male speaker

Speaker	L	T [bps]	SIR [dB]	BER	Wav file name
М	-	-	-	-	maleNN
М	1	4400	-5	0.0458	male-5L1n
М	2	2200	10	0.0730	male10L2n
М	4	1100	15	0.0780	male15L4n
F	-	-	-	-	femaleNN
F	2	2200	10	0.0653	female10L2n
F	4	1100	15	0.0609	female15L4n

Table 1. Sample simulation results for digitally mixed signals

After analyzing the above presented BER results and the corresponding wav files we can make several rather loose observations that will help us narrow down the simulation space for future measurements. First, in order to maintain the intelligibility of the speech it is necessary to keep SIR above

10 dB (preferably in the 15 dB range). Second, if SIR is selected to be in the 15 dB range, the spreading factor has to be at least 4 to keep the BER under control. Finally, it can be seen that the BER results are rather poor throughout the parameter space. However, due to the nature of the analog signal, errors mostly occur in bursts and it is reasonable to expect that with some standard error correction coding and interleaving they can be brought down to meet basic Quality of Service (QoS) requirements based on the application.

4.0 MEASUREMENT AND SIMULATION RESULTS

While the MATLAB simulations presented in the previous Section provide a good initial assessment of the feasibility of the proposed approach, they do not fully represent what might happen in the actual system once the concept of mixing the speech and digitally modulated data is implemented. One reason is the fact that all signals in MATLAB simulations are discretized versions of the analog signals that will be transmitted in the actual system. The other reason is that in MATLAB simulations signals are not transmitted over an RF link. Instead we modeled the RF channel as additive noise.

We addressed the first shortcoming by converting the digital signal into a wav format and then mixed it with the speech signal using an analog mixer. To perform the analog mixing we used the Behringer XENYX 802 Mixer where we had the flexibility to arbitrarily select signal strengths and control SIR parameter. We addressed the second shortcoming by employing Insignia FM radios, half duplex, with operating powers less than 4 watts. Channel 3 at 462.6125 MHz was used in our experiments. In order to meet very limited bandwidth requirements, a 3.3 KHz low pass filter was placed in the line before transmission. The measurement results are shown in Table 2 for SNR = 15 dB.

Speaker	L	Т	SIR	BER	Wav file name
_		[bps]	[dB]		
М	2	2200	10	0.0404	male10L2TAnoR
М	4	1100	15	0.0503	male15L4TAnoR
F	2	2200	10	0.0236	female10L2TAnoR
F	4	1100	15	0.0273	female15L4TAnoR

 Table 2. Measurement results for acoustically mixed signals
 with wireless transmission

Let us now investigate what kind of performance improvement we can expect if we assume that it is feasible to expand the digital data bandwidth to 10 KHz. Our primary goal is to increase the data throughput so we will select our SIR and L parameters accordingly. Our measurement setup is the same as above, with the only exception being that the 10 KHz filter instead of 3.3 KHz filter is applied. Digital noise was not added to the signals. The results are shown in Table 3. All audio files are created after the 10 KHz filter was applied.

Speaker	L	T [bps]	SIR	BER	Wav file name
			[dB]		
М	4	3300	15	0.0516	male15L4TARex
F	4	3300	15	0.0318	female15L4TARex

Table 3. Measurement results for acoustically mixed signals with wireless transmission (10 KHz digital data bandwidth)

Another approach for a potential throughput increase is to change the modulation format. Tables 4 & 5 show simulation results for QAM modulations where SNR = SIR = 15 dB and for 3 KHz and 10 KHz bandwidths, respectively.

Speaker	L	T [bps]	BER	Wav file name
М	2	4400	0.0830	male15L2nq
М	4	2200	0.0522	male15L4nq
F	2	4400	0.0765	female15L2nq
F	4	2200	0.0446	female15L4nq

 Table 4. Sample simulation results for digitally mixed signals with digital bandwidth of 3 KHz

Speaker	L	T [bps]	BER	Wav file name
М	2	13200	0.0496	male15L2nqx
М	4	6600	0.0231	male15L4nqx
F	2	13200	0.0565	female15L2nqx
F	4	6600	0.0258	female15L4nqx

 Table 5. Sample simulation results for digitally mixed signals with digital bandwidth of 10 KHz

Le us briefly comment on the speech quality. Readers are invited to listen to all wav files specified in our tables (see [19]) and personally assess the intelligibility. In Table 2 the subjective voice quality is best for both the male and female speakers with L = 4 and SIR = 15 dB. As expected, the voice quality is better when the bandwidth is increased to 10 KHz. Comparing the results of Table 4 and 5 show that the subjective voice quality is approximately the same for L = 2 and L = 4 for the male and female speakers respectively, e.g. wav file male15L2nq has approximately the same voice quality as wav file male15L4nq.

5.0 CONCLUSIONS

In this paper we have demonstrated that one can send digital data streams simultaneously with analog voice thereby increasing spectrum utilization without a cognitive RF system. We have shown that it is possible to achieve throughputs of over 13 Kbps with a 10 KHz bandwidth channel and still preserve the intelligibility of the speech while maintaining bit error rates that are acceptable for certain applications.

FOOTNOTE

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