Digital Microwave Receiver Technology

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Invited Paper

***Abstract*—This paper reports the impact of digital signal processing on microwave receiver technology. The majority of modern receiver designs are based on digital technology. Wide- and narrow-band receivers will be presented. The wide-band receivers cover approximately 1-GHz instantaneous bandwidth and are used to intercept radar pulses. Current narrow-band receivers cover up to 50-MHz instantaneous bandwidth and are primarily used for receiving communication signals. Two approaches for wide-band receiver design will be discussed. One is the conventional digital receiver. The other one is called the monobit receiver, which has slightly inferior performance in some respects, but can be built on a single chip. Narrow-band receivers are best implemented in software because they can more adapt to changes. Two types of receivers will be discussed. One is the software global positioning system receiver. The other one is called a transform-domain communication system. The object of this system is to avoid interference in a hostile communication environment.**

***Index Terms*—Digital receiver, digital signal processing, digital technology, GPS receiver, microwave receiver, monobit receiver, transform-domain communication.**

I. Introduction

MICROWAVE receivers can be divided into two categories: wide- and narrow-band. Wide-band receivers are usually used to intercept radar signals and often referred to as electronic warfare (EW) receivers. Narrow-band receivers are primarily used for communication or intercepting communication signals. Communication receivers usually only receive one signal at a time and are designed for a known signal. The military is sometimes interested in searching for unknown signals. An intercepting receiver usually is required to receive simultaneous signals this includes the communication intercept receiver. This is a difficult goal to accomplish because the receiver must be able to detect a weak signal in the presence of a strong one, which requires high instantaneous dynamic range. To fulfill this requirement, the receiver must detect a genuine weak signal and avoid the detection of spurs generated in the receiver. Often a receiver is required to receive up to four simultaneous signals.

In the past, all receivers were built using analog technology. The types of receivers that can process simultaneous signals are: 1) channelized; 2) Bragg cell; and 3) compressive. Channelized and Bragg cell receivers are similar and both have parallel outputs. Channelized receivers use filters to separate the signals and Bragg cell receivers use optical techniques to separate signals. Compressive receivers have series outputs and use dispersive delay lines to separate signals. After the frequency separation, the signals are typically converted into video signals by using crystal video detectors. The video signals are digitized and further signal processing is applied.

The channelization function can be accomplished more easily with the advent of digital circuitry. The main advantage in using digital channelization is the better control of filter shape. Therefore, one can see why digital techniques rather than analog are being used in the development of wide receivers.

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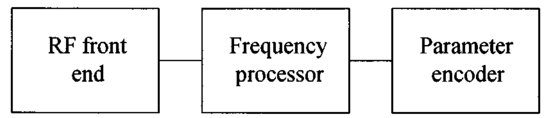

Fig. 1. Main blocks of a wide-band digital receiver.

II. Wide-Band Digital Receivers [1]

The main difference between an analog and a digital receiver is that, in an analog receiver, the signal is converted into a video signal. The conversion from RF to video signals will lose information. For example, the carrier frequency will not appear in the video signals, while in a digital receiver, the signal is usually down converted to a lower frequency and digitized. This process retains all the information. In this sense, the digital receiver can produce better results than an analog one.

A wide-band digital receiver usually contains three major functions: the RF front end including the digitizers, the frequency sorting function, and the parameter encoding circuits, as shown in Fig. 1. The typical bandwidth of the receiver can be anywhere from 1 to 4 GHz. In general, 1 GHz is used in a digital receiver because of the limitation in sampling rate of the analog-to-digital converter (ADC). The RF front-end is usually selected in the second aliasing zone of the ADC to avoid the second harmonic, which is generated in the first aliasing zone, as shown in Fig. 2. In this arrangement, the ADC must be able to digitize a signal close to the sampling frequency. For example, If the ADC samples at 3 GHz, it should digitize an input of 3 GHz. The RF design must compromise between the sensitivity and dynamic range. Usually higher sensitivity leads to lower dynamic range and vise versa. There is little difference between an analog and a digital RF front-end in the design of a receiver.

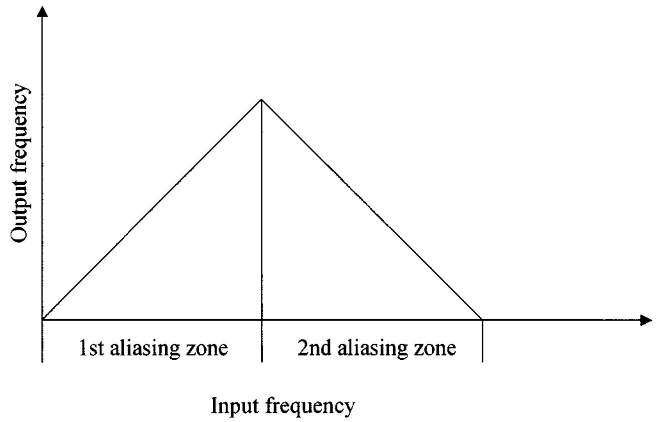

Fig. 2. First two alias zones of an ADC.

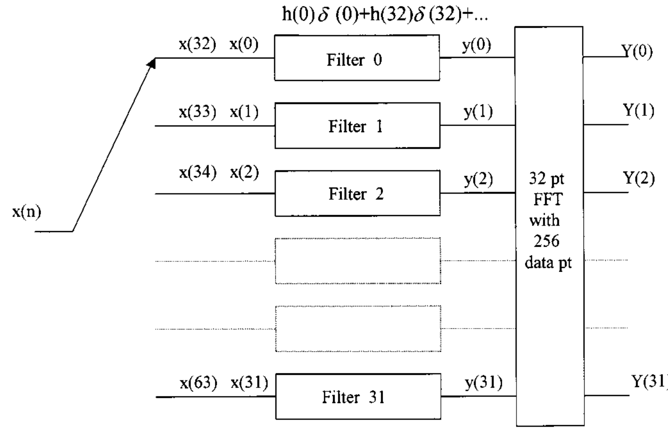

Fig. 3. 32-channel filter bank.

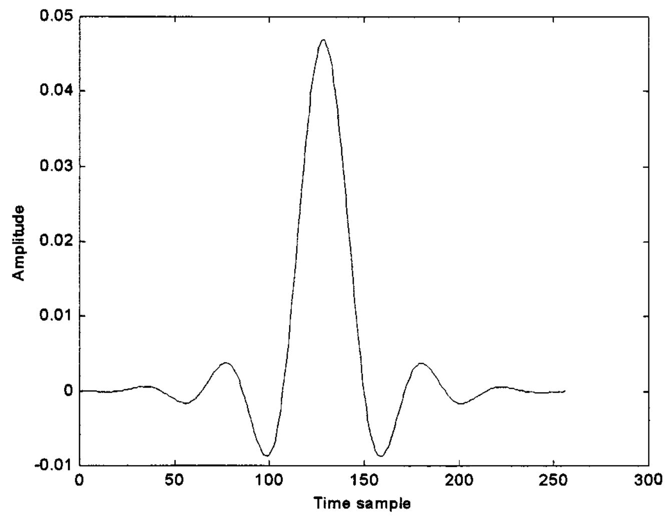

Fig. 4. Time- and frequency-domain responses of filter.

The frequency processor is a bank of digital filters, which separates the input signals according to their frequency. This discussion on wide-band receivers is concentrated on this subject.

The parameter encoding circuit converts the input signals (most are pulses) into a pulse descriptor word (PDW). Usually, a PDW includes frequency, pulse amplitude, pulsewidth, and time and angle of arrival. Depending on the requirements of the receiver, the PDW can only include such parameters as frequency and pulsewidth. From past experience, the encoder is the most difficult function to build. Most of the receiver problems can be improved by modifying the encoder design. However, improving one parameter may influence another one. Therefore, the design must consider the overall capability of the receiver and make design tradeoffs among the parameters. Since specific encoder design can be complex, it will be discussed very briefly.

III. Conventional Digital Filter Bank [1]–[6]

The digital filter bank is designed based on the fast Fourier transform (FFT). The actual design depends on the ADC and FFT technology. In order to assist in illustrating the filter bank, a numerical example will be used. At the present stage, the state-of-the-art ADC can operate at about 3 GHz with 8 bits and the corresponding sampling time is about 0.33 ns. It takes about 85 (256×0.33) ns to collect 256 points, which determines the minimum pulsewidth of the receiver. The FFT must be performed at least every 85 ns. However, existing hardware cannot perform the FFT at this speed.

To solve this problem, a 32-point FFT with 256 points input data is used. A filter of 256 points is used to shape the output of the FFT. This arrangement is shown in Fig. 3. In this figure, the input is decimated by 32 and the filter is also decimated by 32. The 32-point FFT operation is performed at a speed of is 93.75 (3000/32) MHz. The outputs of the filters at steady state are

$$\begin{aligned} y(0) &= x(0)h(0) + x(32)h(32) + \cdots + x(224)h(224)\\ y(1) &= x(1)h(1) + x(33)h(33) + \cdots + x(225)h(225)\\ &\cdots\\ y(31) &= x(31)h(31) + x(63)h(63) + \cdots + x(255)h(255). \end{aligned} \tag{1}$$

The outputs from the FFT operation are

$$Y(k) = \sum_{n=0}^{31} y(n)e^{-j2\pi kn/32}. \tag{2}$$

This operation continues and the input to the FFT shifts 32 points for every operation. The time and frequency domains of the filter used in this example are shown in Figs. 4 and 5, respectively. The filter has more than 70-dB dynamic range. The outputs of the 16 filters are shown in Fig. 6. If 12 out of the 16 filters are used in the receiver design, the overall bandwidth is approximately 1125 MHz (12×93.75). The time resolution is about 10.7 ns (32×0.33). With this filter, if two signals are within 93.75 MHz, it is difficult to separate them. The resolution of the frequency is also about 93.75 MHz, which is too coarse for signal sorting. The desirable frequency resolution is approximately 1 MHz. A parameter encoder can be designed to provide the desired results. A receiver built based on this principle is shown in Fig. 7.

Test results show that the basic approach is very sound. However, the parameter encoder needs further improvement to achieve the desired design goal.

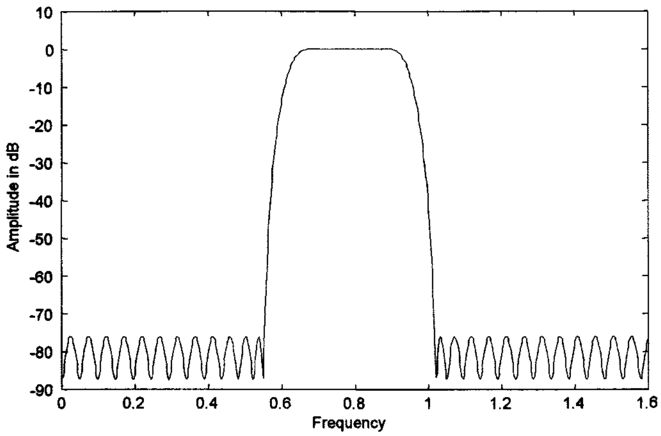

Fig. 5. Outputs of filter bank.

IV. MONOBIT RECEIVER [1], [7]–[9]

The basic idea of a monobit receiver is to simplify the FFT operation. One obvious way to simplify it is to eliminate the multiplication. The discrete Fourier transform (DFT) can be written as

$$X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi nk/N} \tag{3}$$

where $x(n)$ is the input data, k is the frequency component, and N is the length of the DFT. In order to avoid multiplication, either the input $x(n)$ is 1 bit or the Kernel function is 1 bit. That is why the name "monobit receiver" is used. The Kernel function is complex, which means the Kernel function has four values (± 1 and $\pm j$).

Both arrangements were evaluated using simulated data. It appears that with a 1-bit Kernel function better results are obtained. Once the Kernel function is determined, the number of bits of the input signal is determined. If the number of bits is increased from 1 to 2 bits, a significant improvement can be observed. Increasing the number of bits to three gives very little improvement. Thus, two input bits were chosen for the design.

An ADC with 2 bits operating at 2.5 GHz was used to build the receiver. The input is from 1.375 to 2.375 GHz with a bandwidth of 1 GHz in the second alias region. The receiver processes 256 points in 102.4 ns and there is no data overlap in consecutive FFTs. The FFT is built using the conventional butterfly structure. There are eight layers for the 256-point FFT and 128 outputs. The receiver covers a bandwidth of 1250 MHz and has a channel width of approximately 9.77 MHz (1250/128).

Since the input has only 2 bits, the system is nonlinear. The advantage of this approach is that the analog input can be nonlinear and the design can be simplified. The input consists of two filters and a limiting amplifier, as shown in Fig. 8. The first filter limits out-of-band signals. The input signal is amplified to a desired level. The second filter limits the noise. The disadvantage is that the instantaneous dynamic range (the capability to measure a weak signal in the presence of a strong one) is low.

A chip was built with 812 931 transistors. This chip includes thresholds to detect the presence of signals. For simplicity, the receiver only processes two simultaneous signals. The receiver can produce three possible outputs, i.e., 0, 1, and 2. "0' indicates no signal. "1" and "2" indicate one and two signals with the corresponding frequency reading. This function can be considered as a simple parameter encoder. Preliminary tests show that they can process two simultaneous signals. The only shortcoming is the low instantaneous dynamic range. When two signals are separated by over 5 dB, the receiver can only detect the strong one. Even when two signals are of the same amplitude, the receiver can only detect one most of the time. However, the receiver does not produce erroneous information under simultaneous signals like an instantaneous frequency measurement (IFM) receiver. Another unexpected result is that the receiver can receive signals directly at 10 GHz. The ADC operates as the down converter and encodes the frequency correctly.

Since the receiver and front-end are very simple, it is anticipated that the entire receiver including the ADC can be built on one chip. This is attractive for many applications, especially for space applications.

V. SOFTWARE GLOBAL POSITIONING SYSTEM (GPS) RECEIVER [10]–[12]

This discussion is limited to the coarse/acquisition (C/A) coded signal, which is used by civilian users. The signal is centered at 1575.42 MHz and has a bandwidth of 2.046 MHz from null to null. Since this signal is narrow-band, the receiver can be built in software. The term "software" used here is that the digitized signal is processed through software, as shown in Fig. 9. In conventional receivers, the acquisition and the tracking are accomplished in hardware, but the following steps are performed in software. The hardware used in the receiver is to digitize the input signal, which includes amplifiers, a down-converter, and an ADC. Actually, the signal can be either digitized at 1575.42 MHz or down converted to a lower frequency.

The main advantage of a software approach is flexibility. In a hardware receiver, any change in design requires the modification of hardware, while in a software receiver, the modification does not require hardware changes. This is especially convenient for studying antijamming problems. One can collect signals with various jamming signals and process them offline to find an optimum approach.

Acquisition and tracking programs are available in software. There are several different acquisition programs. Some are faster and others are more suitable for weak signals. It has been demonstrated that it is possible to transition from acquisition to tracking using software techniques. The tracking program can be performed on a personal computer (PC) or with a digital signal processing (DSP) board. The PC version can track four satellites in real time. The DSP version has 12 channels. It is anticipated that the sensitivity of the software GPS receiver can be improved by better processing methods.

It is predicated that, in the future, most of the narrow-band receivers will be built using software approaches. This might be particularly valuable for military applications. One set of hardware can be used to build many different types of receivers by selecting the proper software associated with them.

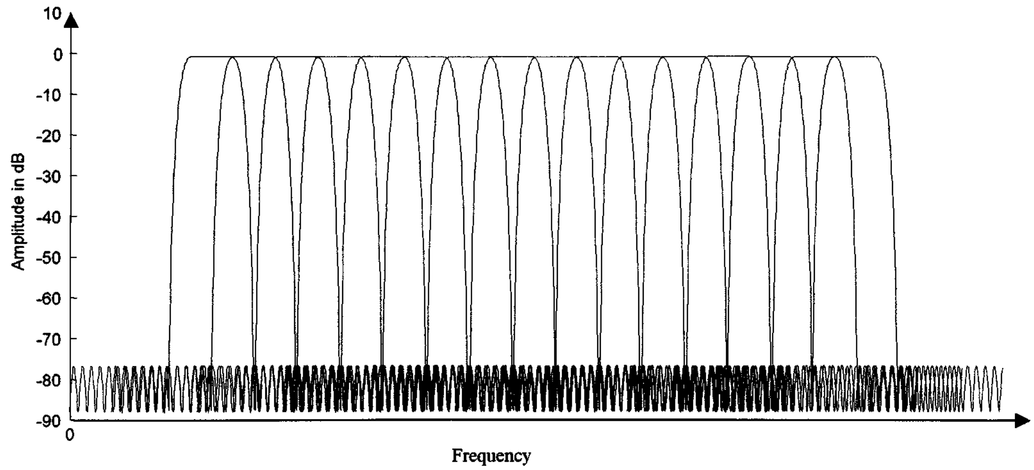

Fig. 6. Channelized receiver.

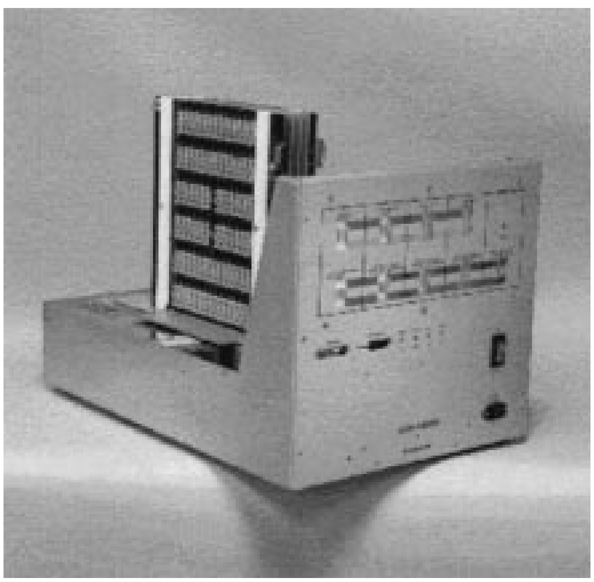

Fig. 7. Front-end for monobit receiver.

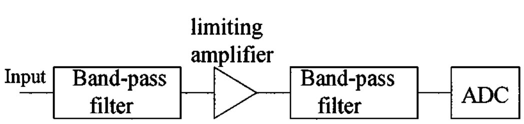

Fig. 8. Block diagram of a software GPS receiver.

VI. Transform-Domain Communication System (TDCS) [13]–[17]

The ability to communicate reliably in the presence of interference, both intentional and unintentional, is an important issue for both military and commercial applications. Efforts to mitigate the effects of interference usually involve filtering or techniques such as spread-spectrum modulation. Traditionally, communicators design waveforms in the time domain and accept the frequency-domain characteristics as a consequence. The spectral characteristics of the waveform may be tailored with the selection of operating parameters and filtering. Future waveforms will be designed in the transform domain (e.g., frequency domain, time-scale domain, etc.) to satisfy the requirements of spectral effectiveness, interference avoidance, and information throughput efficiency. This section discusses a new method for avoiding interference in a direct manner using transform-domain techniques.

A TDCS provides two major enhancements to traditional spread-spectrum signaling, which increase signal robustness against interference. First, interference-avoiding waveforms are generated simultaneously at the transmitter and receiver location to avoid spectral interference; via adaptive notching, spectrally crowded regions are avoided altogether. Second, no carrier modulation is employed, rather a "noise-like" basis function (BF) is data modulated. The basic theory for a TDCS is very simple. The transmitter and receiver sample the environment and produce an estimate of the local interference in the transform domain. The transmitter and receiver are assumed to operate in the same environment, thus, the estimates produced by the transmitter and receiver will be the same. This assumption is valid for short-range communications such as aircraft flying in close formation. For more flexible applications, this assumption can be relaxed and protocols can be established for exchanging spectral estimates between the receiver and transmitter, but this is not addressed in this paper. From this estimate, a BF established in the transform domain, which contains no (or very little) energy in the areas, occupied by the interference. The time-domain version of the BF is then obtained through an inverse transform operation. At the transmitter, the BF is then modulated with data and transmitted. A block diagram for the general process is shown in Fig. 10. Since the BF generated at the receiver is used for correlating the received signal, it must match the waveform generated by the transmitter. A key element of the system is the identical application of a pseudorandom code in the generation of the waveform by the transmitter and receiver. The following examines the specific details for systems according to the transform-domain technique.

A Fourier-based system like the one described in Fig. 10 attempts to identify what spectral components of the environment contain interference and nulls those frequency in the process of generating the waveform. The key parameters of Fourier analysis are the magnitude and phase of each frequency. When sam-

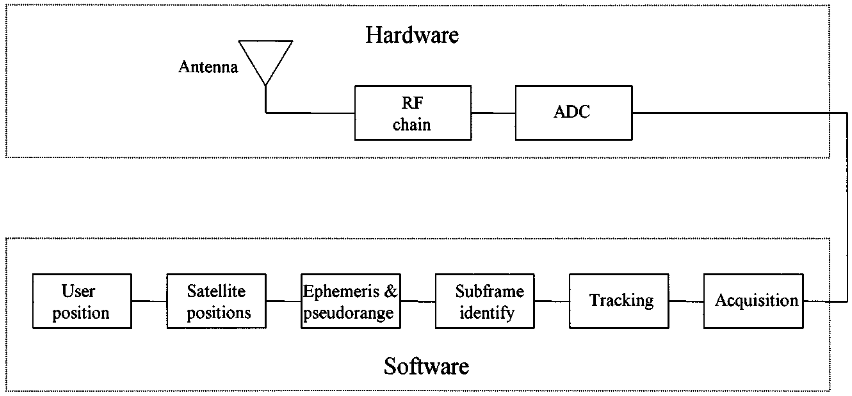

Fig. 9. Front-end of a GPS software receiver.

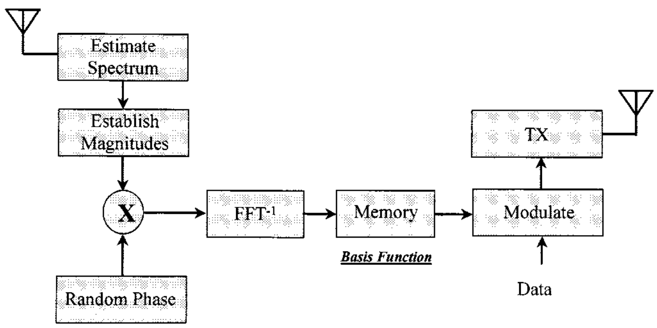

Fig. 10. Fourier-based system transmitter.

pling the environment, the system is only concerned with the magnitude of the spectrum. The magnitude of the Fourier transform is used to determine what frequencies contain interference and what frequencies are clear for communication. To generate a BF, the clear frequencies are given a magnitude of one and the frequencies containing interference are set to zero. Next, a pseudorandom code is used to generate phase information for each frequency. The resulting spectrum of magnitudes and pseudorandom phases is then inverse transformed to produce the time-domain version of the BF. Fig. 11 shows the spectrum of a sampled environment containing a single tone, a binary phase-shift keyed (BPSK) signal, and additive Gaussian white noise (AGWN). Fig. 12 shows the estimated spectrum using a tenth-order autoregressive process and a threshold simply set at the mean of the smoothed spectrum's magnitude. This method of spectral estimation or threshold is by no means considered optimal.

The spectral estimate undergoes a thresholding process, resulting in a magnitude vector containing zeros and ones. Spectral regions exceeding the threshold are assigned a value of zero and other regions a value of one. Effectively, this process yields a vector representing an ideal notched rectangular spectrum, as shown in Fig. 13. The application of the pseudorandom phase is important because it ensures the time-domain version of the BF is noise-like. Without the phase information, the time-domain waveform for a spectrum of uniform magnitude would be the discrete time equivalent of an impulse function. The pseudorandom phase given to the BF's spectrum is as much responsible for the time-domain signal's shape as is the magnitude of its spectrum. The magnitude vector is multiplied by a

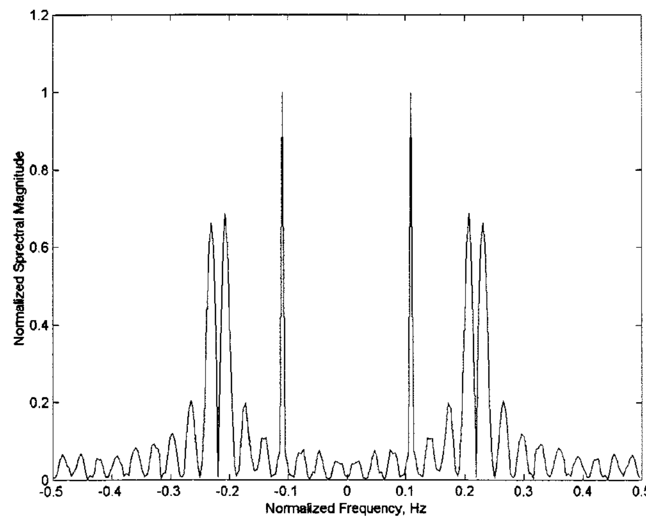

Fig. 11. Spectrum of sampled environment.

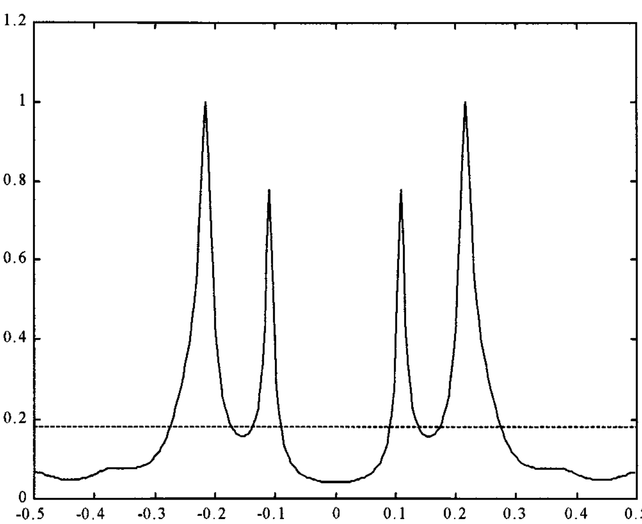

Fig. 12. Estimated spectrum and threshold.

random phase vector $e^{j\phi(\omega)}$, where $\phi_i \in (0, 2\pi/2^r, 4\pi/2^r, \ldots, 2\pi/(2^r-1)/2^r)$, and $i = 1, 2, \ldots, N$. Here, the outputs of a linear feedback shift register are used with a phase mapper to randomize 2^r possible phase positions for each of the sampled frequency bins. This complex pseudorandom phase vector

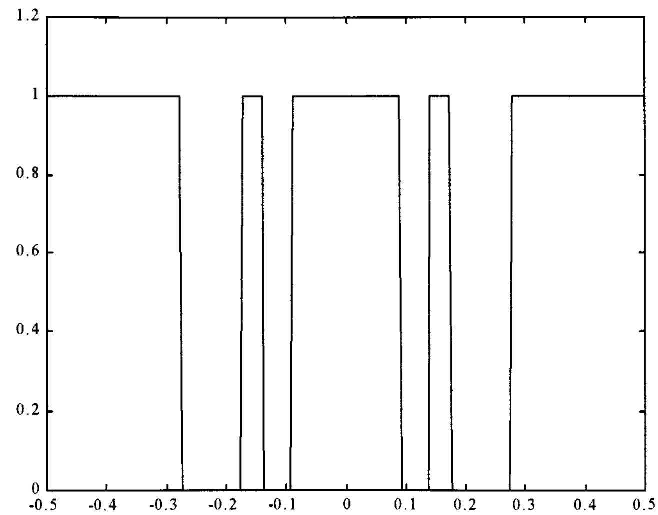

Fig. 13. Theoretical waveform spectrum.

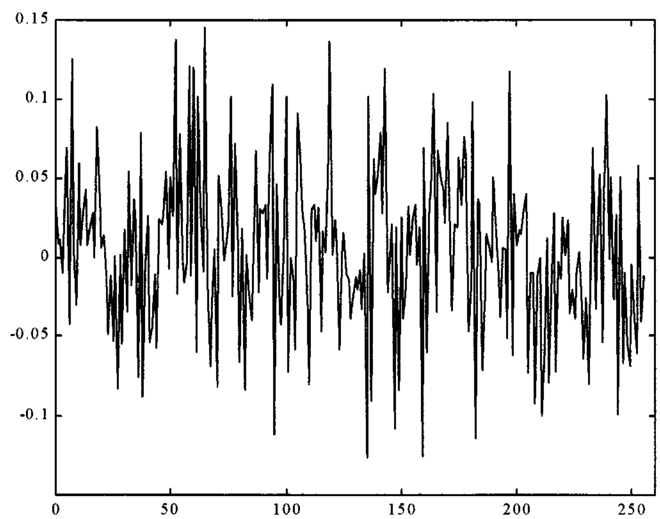

Fig. 14. Time-domain signal (BF).

is multiplied element by element with the previous theoretical, thresholded, and scaled sampled spectrum. Values of $r = 2, 3$, and 4 have been simulated. The time domain, BF of 256 samples, is shown in Fig. 14.

Data may be modulated on to the BF in a number of ways. One of the most basic modulations is antipodal signaling, where the BF represents one binary value and the negative of the BF represents the other binary value as follows:

$$\begin{aligned} S_1(t) &= \mathrm{BF} \\ S_2(t) &= -\mathrm{BF}. \end{aligned} \tag{4}$$

All of the transform systems may be used with antipodal signaling, and their error performance in the presence of AWGN is identical to other antipodal signaling schemes in AWGN. As has been demonstrated, the benefits of the TDCS become evident when there are interferers present in addition to the AWGN. Cyclic shift keying (CSK) is another means of data modulating the BF. A binary CSK system would use the BF for one symbol

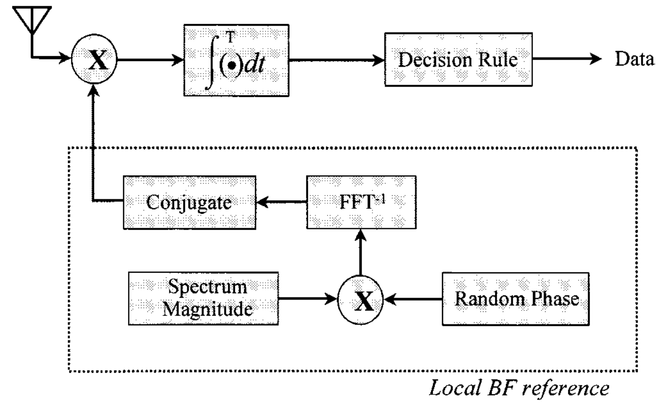

Fig. 15. Fourier-based TDCS receiver.

and a half-period cyclically delayed version of the BF for the other symbol as follows:

$$\begin{aligned} S_1(t) &= \mathrm{BF} \\ S_2(t) &= S_1\big((t - T/2)\big)_T. \end{aligned} \tag{5}$$

Binary CSK is an orthogonal modulation and as such has a 3-dB performance loss compared to antipodal signaling; however, improvements are realized when $m$ary CSK is employed. CSK modulation has no effect on the BF spectrum magnitude, as is consistent with the time shift property of the Fourier transform; thus, interfering frequencies are avoided regardless of which symbol is sent.

The basic block diagram of a Fourier-based TDCS receiver is shown in Fig. 15. The incoming signal is correlated with one of M locally generated reference signals. The reference signals are locally generated versions of the normalized and conjugated signal set and are generated from the same BF used in the transmitter. The correlators each provide a test statistic as output. A decision rule is applied to the test statistics and a decision is made as to what data symbol was transmitted.

The performance of a TDCS is dependent on a number of factors. Chief among these is the accuracy of the estimate of the environment. The Fourier transform-domain technique illustrated here is probably the most familiar and is capable of characterizing most man-made interferers; however, wavelets have been shown to offer benefits for dealing with certain types of intentional interference and may have additional benefits for synchronization, combating multipath, and faster processing. Since the BF is created from samples of the environment, the samples used to generate the BF must present a true characterization of the environment. For example, if the environment contains a slow-moving swept tone interferer and the TDCS samples the environment for only a short amount of time, then the system may judge that the interference is a constant tone. The result would be that the BF generated does not avoid the interferer once the interferer moves sufficiently far enough away from the frequency it was at when the samples were collected. As with other estimation problems, the longer the system spends sampling the environment, the better it can estimate it. However, this impacts the amount of time that the system can be used to transmit data. The estimate of the environment is also affected by any smoothing of the transform coefficients and the selection of the threshold. CDMA has also been demonstrated through the random phase code by using PN codes with good auto- and cross-correlation properties.

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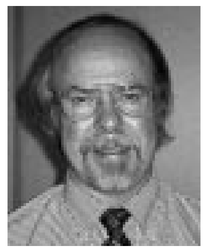

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