

# spread spectrum and digital communication techniques: a primer

Not quite sure  
how speech can be  
transformed into a  
digital signal?  
*Read on.*

Ever since the inception of Amateur Radio, hams have kept abreast of the latest and most innovative methods of communication. From the advent of spark-gap radiotelegraphy, to the early FM transmitters, through the 2nd World War — when hams played an important part in concocting the first reliable pulse radar systems — to RTTY, SSTV, SSB, satellite communications, and packet radio, Amateurs have been ardent users of new and fascinating modulation methods.

In recent years there has been interest in a relatively new type of communication technique. While the foundation for this communication method was laid with the advent of ranging radar, it has only been in the past 10 to 15 years that it has received so much attention from both the military and the private sector. This technique, known as *spread spectrum* (SS) is unlike any communication method previously tried by Amateurs. However, judging by our track record, it would seem that it's only a matter of time before we familiarize ourselves with it.

The purpose of this article is to provide an overview of spread spectrum communications for those not familiar with it. While the topic is much too broad to be fully discussed here, the major concepts will be highlighted in a manner that can, I hope, be understood by those with little math background. As well

as the concepts governing spread spectrum and digital communications, typical station hardware requirements will also be addressed.

## why spread spectrum?

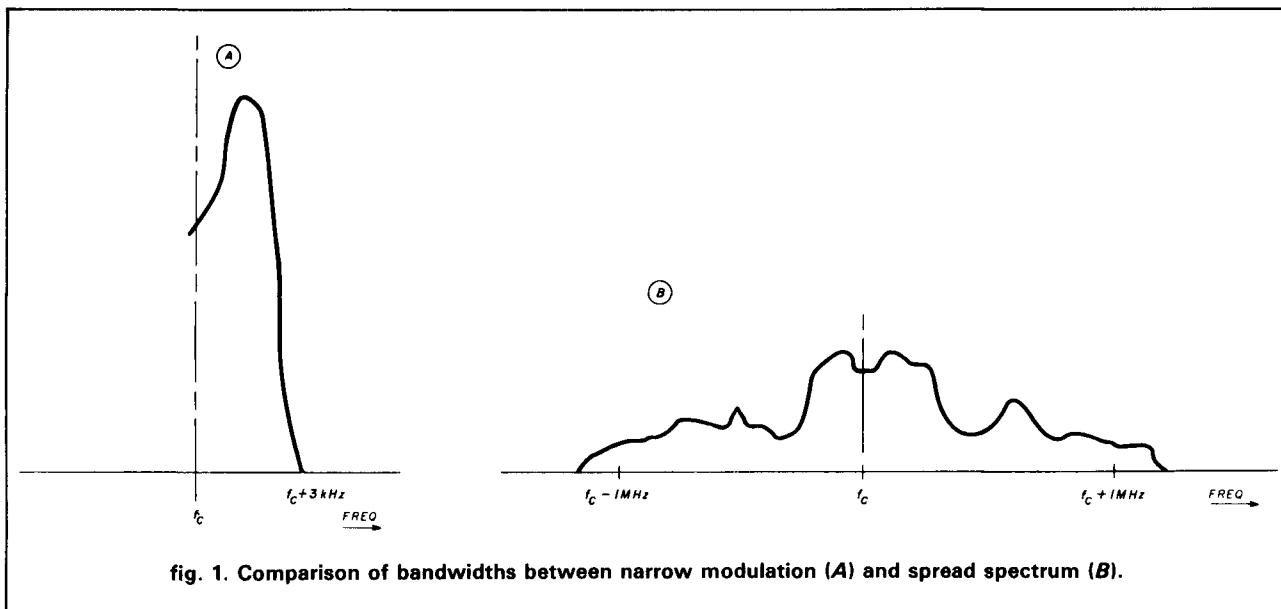
One might wonder how and why spread spectrum evolved, and how it could be applicable to the Amateur Radio bands. To address the first query, it is necessary to consider the problems associated with military communications during World War II. At the time, jamming and antijamming techniques were the order of the day. By 1945 every Allied Bomber plane was equipped with two jamming transmitters, while it is estimated that as many as 90 percent of all electrical engineers in Germany were involved in an antijamming program of gargantuan proportion.<sup>1</sup>

To combat the effects of jamming, spread spectrum was used to spread the signal out, thereby rendering narrow band jammers virtually ineffective. In addition, the fact that spread spectrum could be used with a *low probability of intercept* (LPI) made this an ideal method of communicating while appearing "radio silent" to conventional receivers.

Today the quest for a signal that cannot be jammed continues in military circles; commercial applications, such as banking and private mail systems, require security. As jamming and intercept capabilities become more sophisticated, methods of communicating become increasingly complex. Spread spectrum continues to evolve into a highly complicated mode of communication. Those fascinated with the history of radio would find the accounts of the development of spread spectrum to be very exciting reading. Several excellent accounts are listed in the references.<sup>1,2,3</sup>

The above-mentioned attributes hardly seem appropriate for Amateur Radio! The FCC rules prohibit any kind of coded or secure communications, and intentional jamming is a problem we would ideally never

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have to deal with. There are several other benefits, however, that might be of use to us in the future as the Amateur Radio spectrum becomes saturated with users. In fact, a look at why mobile telephone companies are considering SS sheds some light on some of the possible rewards.

In metropolitan areas, where there are many mobile telephone users in a small area, cellular radio has been introduced to alleviate the congestion in the mobile telephone spectrum. In a cellular radio system, as the term suggests, the city is broken into "cells," with each cell having its own multichannel repeater capable of handling a limited number of users within the cell.<sup>4</sup> As the user travels into an adjacent cell, the adjacent repeater takes over the communication. The cellular technique has been used to increase the maximum number of mobile telephone and commercial radio users from several hundred to several thousand in many cities across the country. Of special interest to the industry is the fact that compared to conventional narrow band modulation techniques, SS has the capability of supporting a larger number of users for a given cell size!<sup>5</sup>

Other advantages that SS offers to both the military and the mobile communication industry include selective addressing capability, code division multiplexing, and interference and multi-path rejection. With SS, it is possible for a transmitter to selectively communicate to one or several receivers while remaining oblivious to other users. Also, several stations may use the same band of frequencies simultaneously without interfering with one another. Since SS signals have very wide bandwidths, conventional narrow band users may also use the same spectrum without adversely

affecting the SS communication. Conversely, the average power of a SS signal in any narrow band region is small, so the narrow band modulation is not severely QRM'ed, either.

As will be demonstrated shortly, SS requires more complex hardware than does conventional narrow band equipment; as the spectrum stands today, its use in Amateur Radio is probably not currently warranted. However, with increased HF and VHF/UHF activity, it is conceivable that we may eventually need a drastically different approach to communications. Progress has recently been made by hams in such areas as coherent CW and packet communications.<sup>6,7,8,9</sup> The inevitable thrust toward digital communications makes SS appear to be a likely modulation method in the future.

## overview of spread spectrum

Figure 1 illustrates the bandwidth of an SSB speech signal compared to a typical spread spectrum signal. As its name implies, spread spectrum is a modulation method whereby the energy of the transmitted signal is spread out over a very wide bandwidth. This is quite unlike SSB or narrow band FM (NBFM), where the transmitter output has a bandwidth on the order of that of the modulating signal (the usable audio bandwidth for speech is about 3 kHz). However, wide band FM (WBFM) transmitters have bandwidths that are many times greater than that of the modulation. Clearly, though, WBFM is not spread spectrum! This is where the second important distinction between spread spectrum and conventional modulation methods must be made.

A spread spectrum communication system uses a

special generated wide band signal that is independent of the message modulation. At the transmitter end, the message modulation (a voice signal) may be multiplied by this independently generated signal, and the resulting mix then transmitted on a carrier. This is known as *Direct Sequence Spread Spectrum* (DS). Another type of SS, known as *Frequency Hopping Spread Spectrum* (FH), can be generated by using the independently generated signal to cause the carrier signal to frequency hop in a prescribed manner.

It is the independent signal that determines the amount of bandwidth spreading at the transmitter output. It also determines the immunity the SS signal has to narrow band interference. This independent signal is always digital, and is generated by digital logic devices (such as TTL). The term *pseudo noise code* (PN) is used to describe this independent signal since to an uninformed observer the PN code looks like a random jumble of 1s and 0s. Actually, though, the PN code is a periodic sequence that can be easily gener-

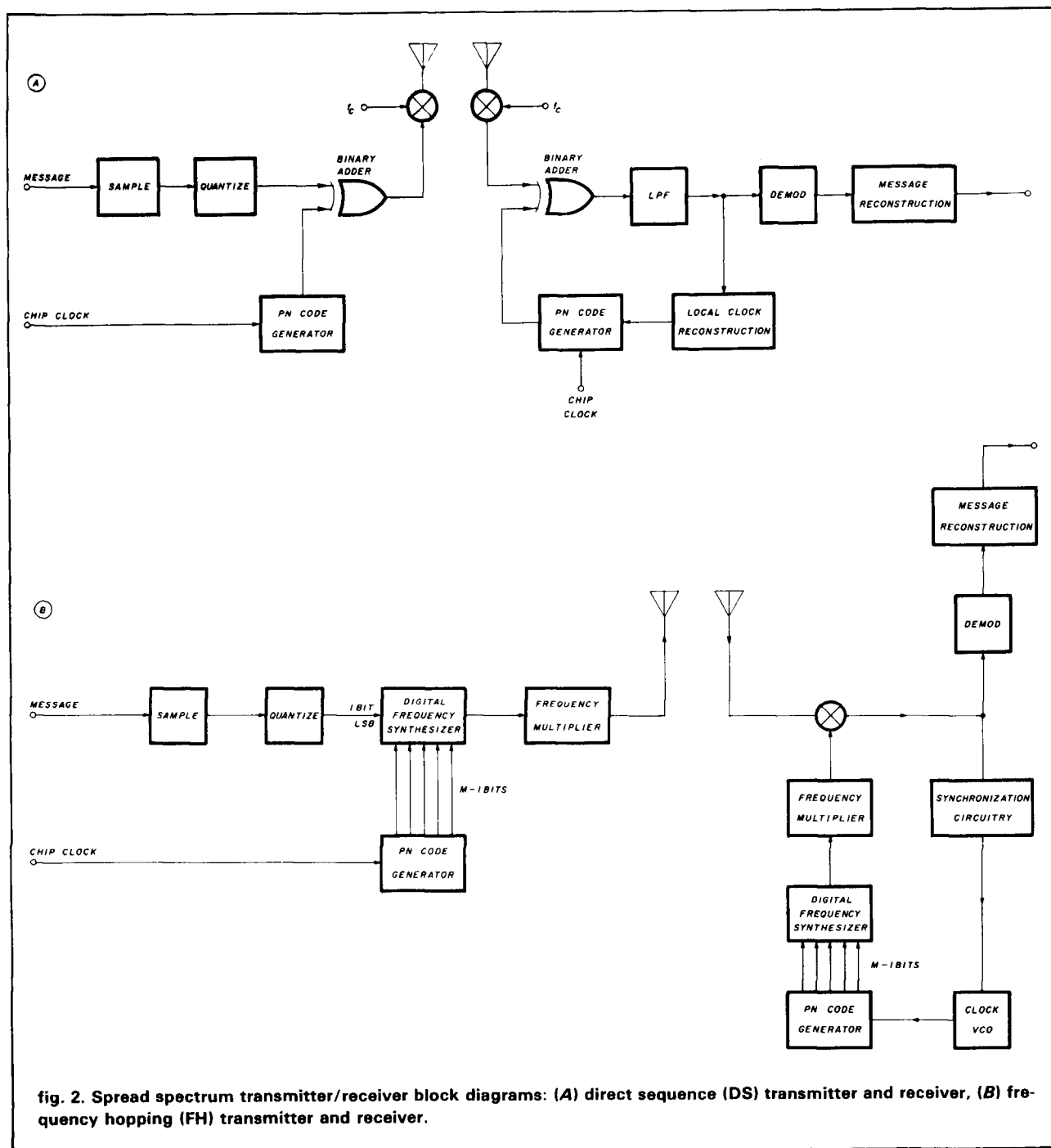
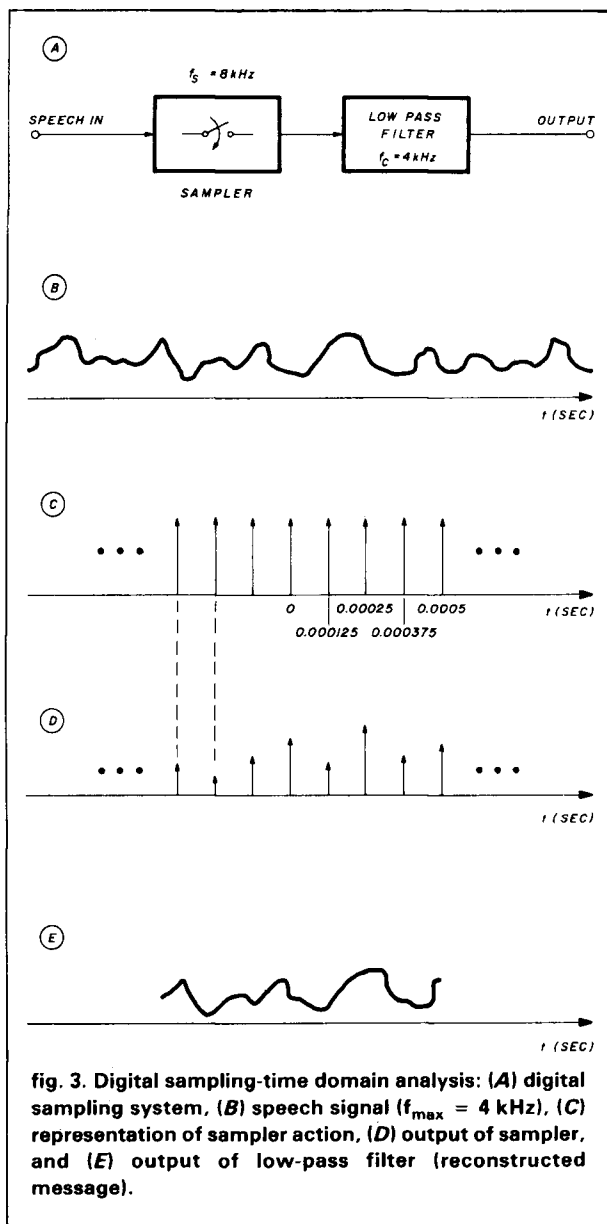


fig. 2. Spread spectrum transmitter/receiver block diagrams: (A) direct sequence (DS) transmitter and receiver, (B) frequency hopping (FH) transmitter and receiver.



ated by a sequence of shift registers. In order to recover the original message, the receiver must be able to reconstruct the same PN code used by the transmitter. When the transmitter and receiving encoding signals are identical, and when they are synchronized in time, then the message is detected. Figure 2 illustrates block diagrams of spread spectrum transmitter and receiver pairs for both the DS and FH case. A more detailed look at the generation of the PN code is considered subsequently. However, before delving into the details of SS, it is first necessary to become familiar with some basic concepts of digital communications.

### digital communication concepts

Because the encoding signal is digital, spread spec-

trum can be considered to be a special form of digital communications. Unlike SSB, AM and FM, which are analog, continuous time communication methods, digital communication systems work on the principle of the sampling theorem.

The sampling theorem, developed by Nyquist in 1924,<sup>10</sup> states that a continuous time signal may be represented by a sequence of discrete time snapshots, or samples, without any loss of information in the signal, provided that the samples are taken at a rate which is at least twice as great as the highest frequency component of the original continuous time signal. A basic relationship which relates the sampling frequency ( $f_s$ ) to the time duration between successive samples ( $T_s$ ) is

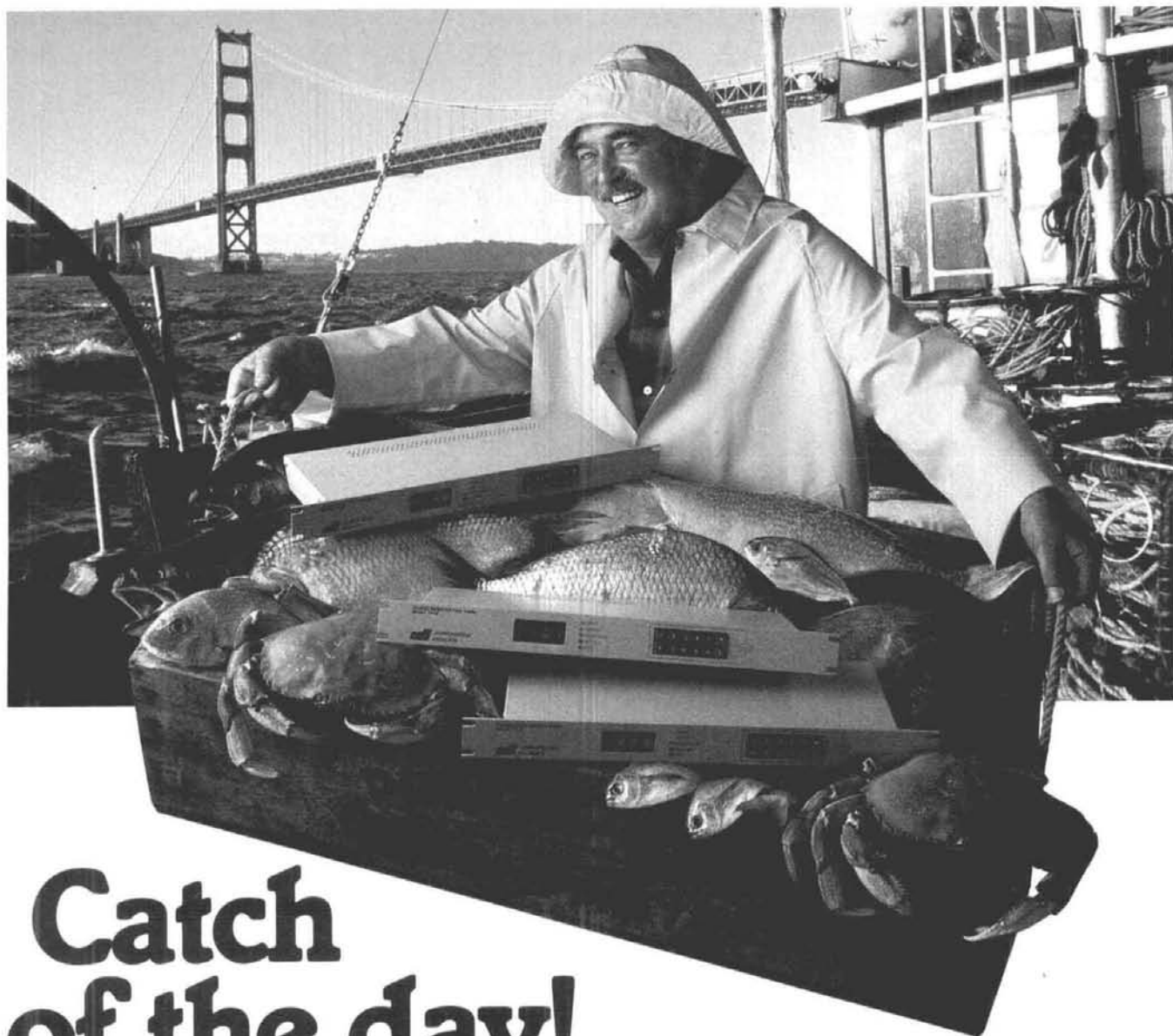
$$f_s = \frac{1}{T_s} \quad (1)$$

Figure 3A shows the components of a typical sampling system. Figure 3B illustrates a typical speech signal that has been band limited to have a peak frequency component of 4 kHz. The action of the sampler is shown in fig. 3C. Figure 3D illustrates the output of a sampler that is taking samples at a rate of 8000 samples per second (twice the rate of the highest message component). Figure 3E shows the recreated message waveform after the samples are placed through a low-pass filter having a cutoff of 4 kHz.

In order to lay a foundation for the analysis and understanding of SS, it is instructive to look at the sampling theorem from a different point of view. In the early 1800's, Fourier, a famous mathematician, observed that most functions could be represented by a summation of sinusoids having different amplitudes and periods. In short, he laid the groundwork for the development of the celebrated Fourier transform. This transformation allows one to analyze a signal in the frequency domain rather than in the time domain.

Frequency domain analysis can directly give information pertaining to the bandwidth of a signal. Tables such as table 1 have been compiled which lists the Fourier transforms of many common signal shapes.<sup>11,12,13</sup> By analyzing the sampling circuit of fig. 3A in the frequency domain, we can better explain how and why the sampling theorem holds.

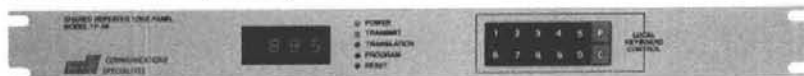
From table 1, the Fourier transform of an impulse sampler is an impulse train in the frequency domain. Note that in the time domain (fig. 3C), the sampling action is effectively multiplying the input signal by a "1" at each sampling instant, and multiplying by "0" in the interval between samples. Just as time signals have Fourier transforms, so do time operations such as addition and multiplication. The Fourier transform of a time multiplication is known as *frequency convolution*. Convolution is a fundamental concept in control



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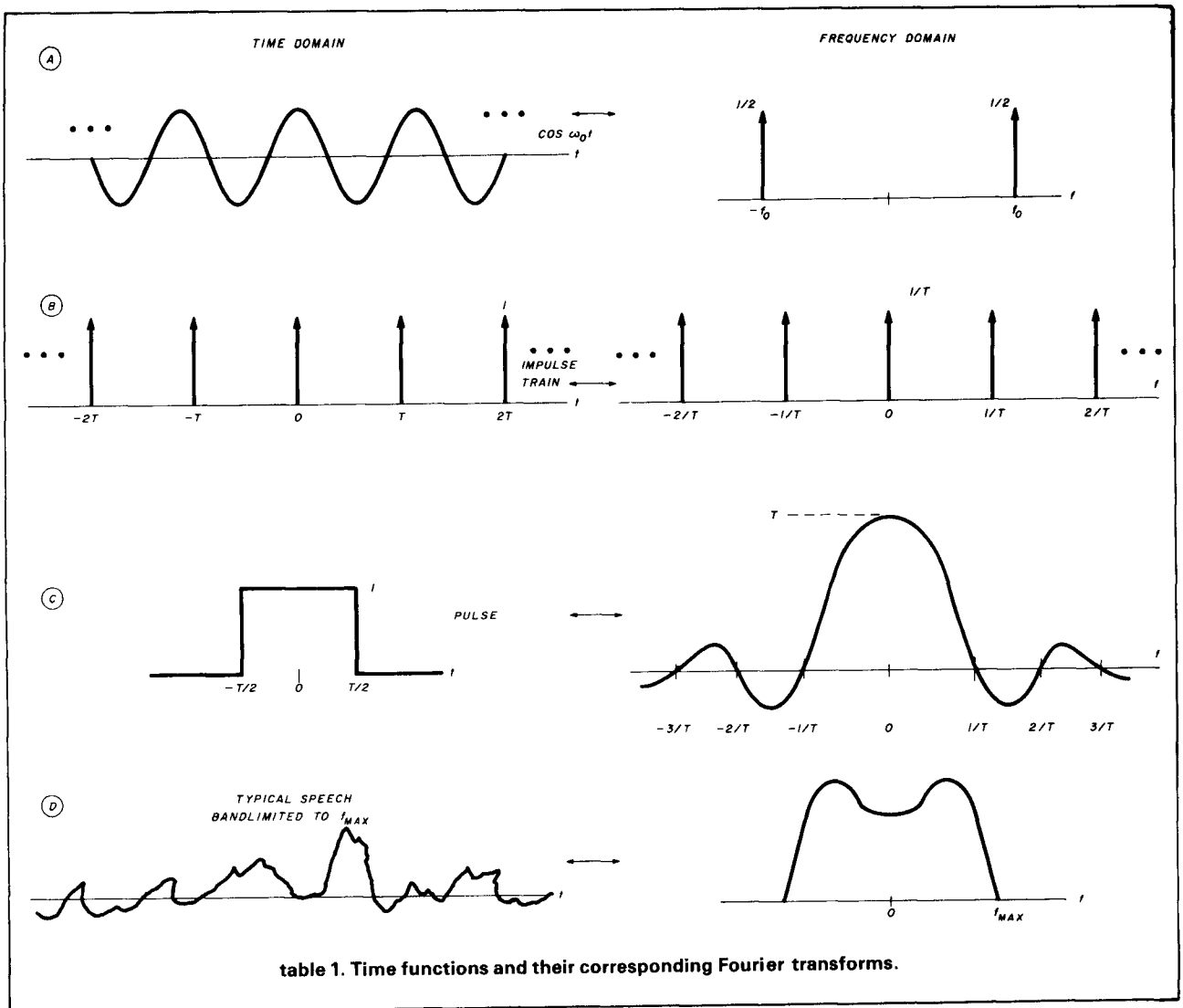


table 1. Time functions and their corresponding Fourier transforms.

and communication theory, and is used to express the output of a system or filter in terms of the input signal and the system impulse response, (i.e., the response of the system to a sudden input signal). For this discussion, it is necessary to know only that the convolution of a band-limited spectrum (fig. 4A) with a frequency pulse train (fig. 4B) yields the original message spectrum replicated at each of the pulse train harmonics. Hence, by frequency domain techniques, we find that the output of an ideal sampler is the original message spectrum replicated throughout the entire frequency domain and separated by integer multiples of the sampling frequency (fig. 4C). By low-pass filtering the sampler output, we can recreate the original message exactly (fig. 4D). By the same token, we could bandpass filter the output of the sampler and also recreate the message, although this is not usually done.

If the highest frequency of the input message exceeds one half of the the sampling rate, then an

undesirable effect known as *aliasing* occurs. As can be seen in fig. 4E, each adjacent message spectrum overlaps so that the LPF output is not the original message, but rather a distorted signal. With frequency domain analysis, it becomes clear why the sampling frequency must be at least twice that of the peak frequency component of the input.

Before moving on, it should be noted that this quick look at the sampling theorem assumes an "ideal" sampler — one for which the sample durations are infinitely small. In reality, the sample durations are small, but finite. Taking this into account yields similar, but slightly more complicated, results. Figure 5 illustrates the spectrum of the output of a typical "real world" sampler. As the sample widths become wider, there is less energy at the higher frequencies. This is why the sampler is followed by a low-pass filter rather than a band pass filter. Also neglected here are some amplitude scaling factors that are involved in transforming between the time domain and frequency domain.

These subtleties are required in exact problem solving, but are not important in gaining a good understanding of the sampling theorem. Those interested in the finer details of the sampling theorem and Fourier transform techniques might find the references helpful.<sup>11,12,13,14,15</sup>

## data communication

Certain digital communication systems such as RTTY and packet radio, where the message text is originated by a keyboard rather than continuous-time speech, are known as data communication. In this case the sampling theorem does not apply, since there is no continuous time signal to sample. However, the data rate (the rate at which information can be sent) is a function of the number of bits used to represent each character, and is also a function of the time duration of each bit.

For example, the American Standard Code for Information Interchange (ASCII) prescribes that each keyboard character be represented by a unique 7-bit data word. The letter i, for example, is represented by the binary word 1101001. If each bit has a time duration of 1 millisecond, then any character may be sent down a channel in a time of 7ms. If a start bit, a stop bit, and a parity bit are sent along with the data, then one character can be sent every 10 ms. The data rate for this set-up would be 1000 bits per second (bps), or 100 characters per second.\* Data communications of this type are termed "asynchronous" since the receiver never knows when the sender will depress the keyboard. The start bit and stop bit are the necessary overhead to identify each of the keyboard entries. The parity bit is used to validate the received data. **Figure 6** illustrates a complete asynchronous character word for the letter i.

There is a trade-off between bit rate and occupied bandwidth of a digital signal. A good rough estimate is that the required bandwidth of a digital signal is equal to the reciprocal of the bit duration. For the previous example, the required bandwidth would be on the order of

$$BW = \frac{1}{t_{bit}} = \frac{1}{0.001 \text{ s}} = 1000 \text{ Hz}$$

For a faster data rate, more bandwidth is required. In a band-limited channel, such as a commercial telephone line, there is an upper limit on the bit rate. This is why home computer modems seldom exceed a data rate of 1200 bps.<sup>16</sup> While spread spectrum is well suited for either voice or data messages, the remainder of this article will consider only a voice message. Once the voice is "digitized," it is sent through the channel in the same manner as data.

\*In digital communications the bit rate is the same as the baud rate, so for this example the data rate is equivalently 1000 baud.

## quantization

From the sampling theorem, we know that it is necessary to send only the voltage values of the samples rather than the continuous time signal. If the sample values could be represented in a digital fashion, then we could take advantage of schemes that have been developed expressly for digital communication. In short, digital communications systems are able to outperform analog methods because signal reception is based on distinguishing whether a "0" or a "1" was sent, rather than trying to recreate a random continuous time waveform directly. Schemes such as *error correction coding* and *minimum probability of error receivers* can be used to provide far superior performance when compared to analog communication techniques.

To represent the height of a sample value digitally, it is necessary to quantize the sample. For binary data (standard digital logic), the quantizing action truncates the actual sample value so that each sample is represented by a fixed number of bits (i.e., every sample is expressed by N bits) in the time between successive samples. Since it is conceivable that the samples can take on a continuum of values, there is some error introduced by the quantizer. However, if a limiter is used (to contain the voice signal voltage within the limits of the quantizer) and if there is a sufficient number of levels in the quantizer, then this error, known as *quantization noise*, is quite small.

The quantizer is an important concept in digital communications. The resolution, or the accuracy in which a sample can be represented, is directly related to the N, the number of bits used to represent each sample. For N bits, there are  $2^N$  quantizer levels (sometimes called *bins*).

An example is useful to clarify how speech can be quantized and sent down a channel as a digital bit stream. **Figure 7** demonstrates how the sample values are assigned data words in a three-bit quantizer (N=3). As indicated in **fig. 7**, there is some error introduced because of the fact that each sample is represented by only a three-bit word. By using more bits, each sample can be more accurately represented. Surprisingly, though, even with only three bits of quantization, intelligible speech can be transmitted.<sup>17</sup> Once assigned a quantization data word, the truncated sample is converted into 1s and 0s and sent down the channel. The output of the quantizer is known as *pulse code modulation*, since the message has been coded into a train of digital pulses. Since N bits must be sent in the time between adjacent samples, the bit duration of the quantized data is

$$T_{bit} = \frac{T_s}{N} \quad (2A)$$

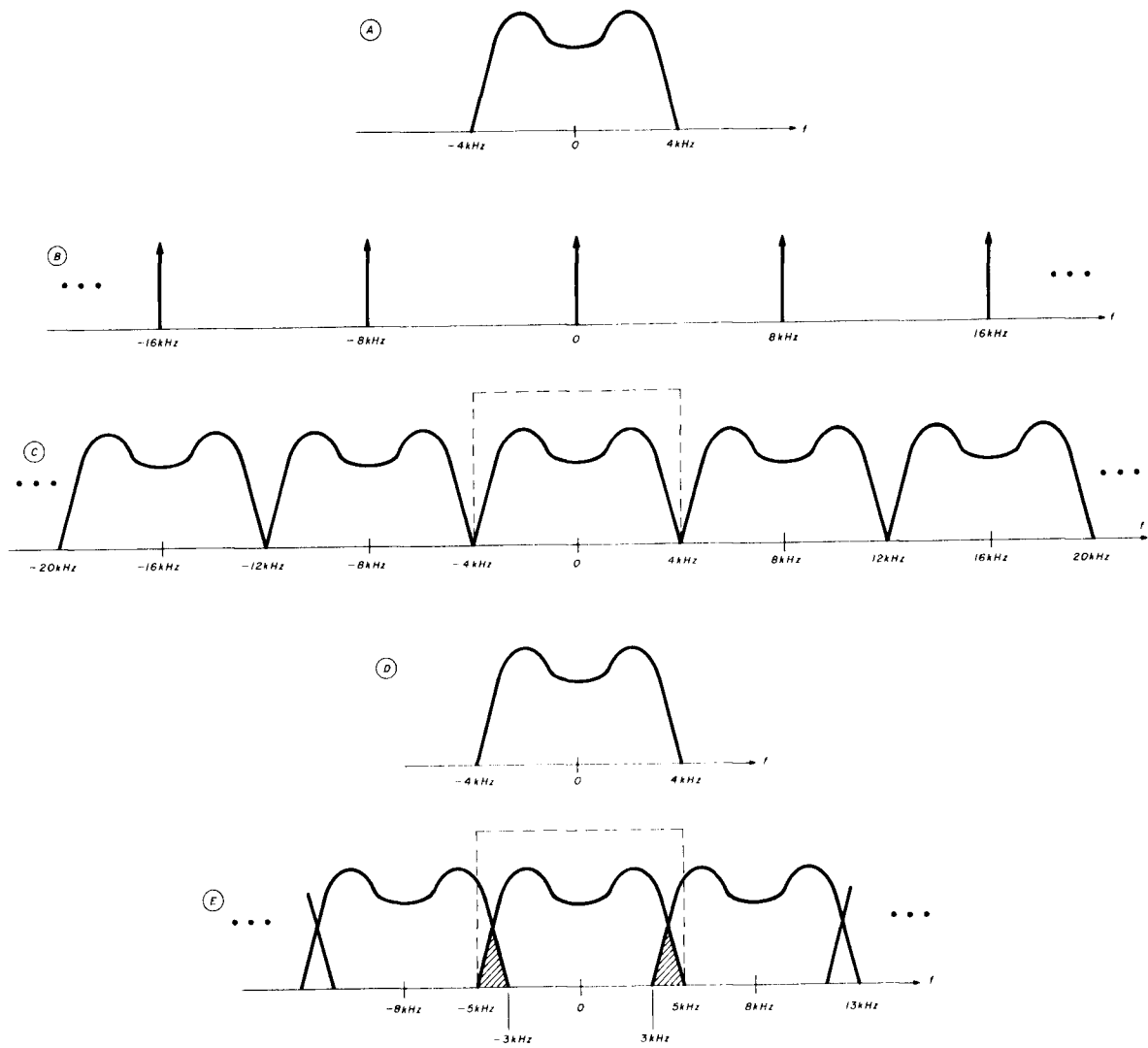


fig. 4. Digital sampling — frequency domain analysis: (A) spectrum of speech signal ( $f_{\max} = 4$  kHz), (B) spectrum of sampler action, (C) output of sampler, (D) output of recreated message spectrum, and (E) aliasing example,  $f_s = 8$  kHz and  $f_{\max} = 5$  kHz.

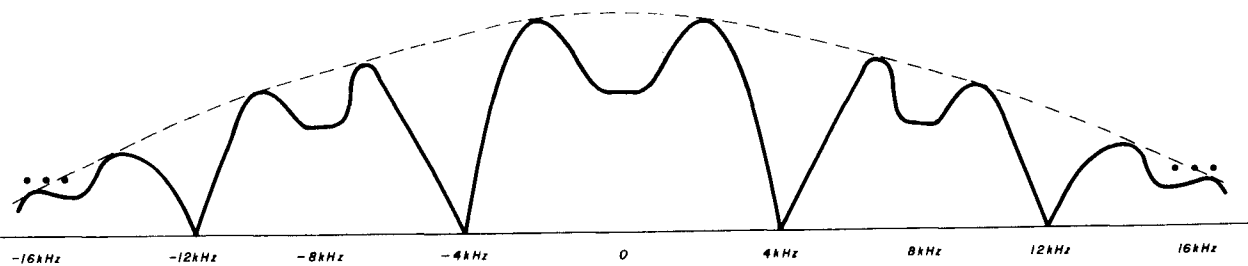


fig. 5. Spectrum of output for non-ideal sampling.

where  $T_s$  is the time between adjacent samples. Since

$$T_s = \frac{1}{f_s} \quad (2B)$$

each quantizer bit duration is given by

$$T_{bit} = \frac{1}{Nf_s} \quad (2C)$$

Hence, the bandwidth is given by

$$BW = Nf_s \quad (2D)$$

A practical method of sampling and quantizing is to use a *sample and hold* circuit followed by an *analog-to-digital* (A/D) converter. The sample and hold is similar to the ideal sampler, except the sample height is held for the entire time duration between samples, and is updated at each new sampling instant. While the sample value is held at a constant level, it is converted into a digital signal by the A/D converter. The end result is identical to that of **fig. 7**.

To recreate the message, the digital bit stream is clocked into a *digital-to-analog* (D/A) converter. This undoes the effect of quantizing which the A/D had

upon the original message samples. The D/A output is then low-pass filtered to transform the reconstructed samples into the original message.

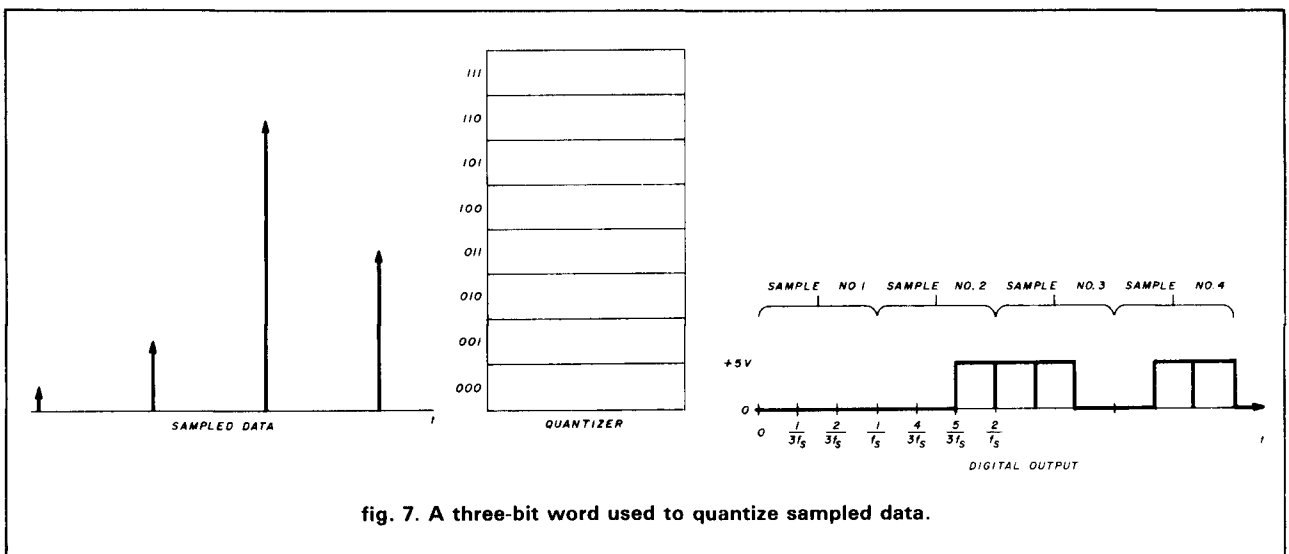
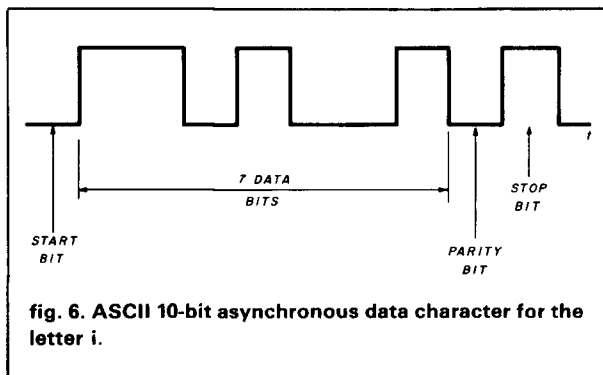
## spread spectrum systems

There are many types of spread spectrum systems. These include direct sequence (DS), frequency hopping (FH), time hopping (TH), chirp, and hybrid systems which combine several techniques at once. Only DS and FH are considered here, since these seem to be most easily implemented. Common to both of these types of SS systems is the need to generate and reconstruct a PN (pseudonoise) code.

To produce the DS spread spectrum signal, a PN code signal must be produced that has a bit rate (bandwidth) much greater than that of the quantized message. For FH spread spectrum, it is not so much the bit rate that matters as does the number of bits used in a complete cycle of the PN code.

The PN code can be generated by a feedback arrangement of flip-flop stages. A flip-flop is a digital device which can store a binary value (either a 0 or a 1). Flip-flops can be connected in series to form shift registers. As the term implies, shift registers store several binary digits and shift them to the left or right each time an external clock pulse is received.

The simplest PN codes (there are several types) are known as *maximal linear codes*, or *m-sequences*. These are produced by m-stage shift registers which use feedback to produce periodic codes that have N bits before recycling. For an m-stage shift register, there are  $N = 2^m - 1$  bits in each period. **Figure 8** illustrates a three-stage shift register. All three flip-flops of the shift register are clocked to the right simultaneously, and each time a clock pulse appears, a new binary digit appears at the output. **Table 2** indicates the value held by each flip-flop for a given time inter-



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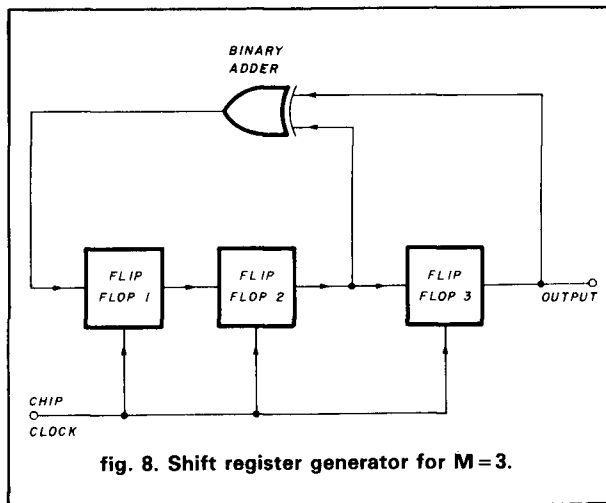
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val. **Figure 9** shows the digital signal that would be produced by a standard TTL circuit.

To avoid confusion between the PN code and the actual message bit stream, the term chip is used to describe each bit of a PN code. For a chip duration of  $t_1$  seconds, the periodic PN sequence repeats itself every  $Nt_1$  seconds. For the example shown in **fig. 9**, the same chip value would be seen every  $7t_1$  seconds apart.

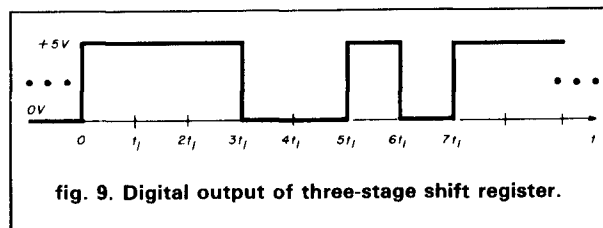
By changing shift register feedback paths, it is possible to generate many unique m-sequence codes, each having a period of  $N$  chips. The number of possible codes is important since it defines the maximum number of users that can be uniquely addressed, assuming that each user has the same length shift register. The exact number of unique codes is dependent upon the number of shift register stages and the possible feedback paths. In general, the longer the shift register, the more unique codes exist. However, as can be noted in **table 3**, when the shift registers consist of a prime number of stages, there is a maximum of codes for a minimum of hardware.

Modular Shift Register Generators (MSRGs) such as the MC8504 are available for easy PN code generation. The MC8504 is a 16-pin chip that features four stages of an expandable shift register. They may be cascaded, and additional flip-flops (such as a 7474) may be added to implement an arbitrarily long m-sequence. A nine-stage MSRG capable of producing 48 selectable 511 chip codes is shown in **fig. 10**.

**Direct sequence (DS) spread spectrum.** As shown in **fig. 2A**, the PN code is added with the digitized message (usually PCM) to produce a digital signal that can be readily modulated. If one of the adding signals is wide band, then the resulting adder output signal is also wide band. Since the PCM has a bandwidth on the order of the original message, it is necessary to use a PN code which has a bandwidth several orders of magnitude larger than the message band-

table 2. Contents of Shift Register Stages for  $m = 3$ .

	stage 1	stage 2	stage 3
initial contents	1	1	1
clock pulse 1	0	1	1
clock pulse 2	0	0	1
clock pulse 3	1	0	0
clock pulse 4	0	1	0
clock pulse 5	1	0	1
clock pulse 6	1	1	0
clock pulse 7	1	1	1



width in order to obtain large bandwidth spreading. For this reason, the chip rate is typically run at speeds of several Megachips per second (Mcps).

The output of the binary adder is fed into a balanced modulator. This modulator produces a particular carrier phase for a logic "1" and a 180-degree shifted carrier for a logic "0." Hot-carrier diodes are used in conjunction with wide band transformers to produce the final RF signal. **Figure 11** illustrates a double-balanced mixer, which is the most commonly used type of balanced modulator. The balanced modulator input and output signals are shown in **fig. 12**. While the output waveform appears simple in the time domain, frequency domain analysis reveals that there is a wide band of frequency components centered around the nominal carrier frequency.

In a DS-SS receiver, just as with any RF receiver, it is first necessary to bring the modulation down to baseband. This is accomplished by mixing the received signal with a local oscillator that is adjusted to the transmitter's frequency. Then, the coded signal must be matched with an internally generated PN code. This is accomplished by binary addition. If the internal PN code generator is not correlated with the incoming signal, the resulting adder output is called *code noise*. However, when the internal code is synchronized with the incoming signal, the output of the binary adder collapses to the original digital message bit stream. A manual tuning dial can be used to adjust the phase of the code generator until synchronization is obtained. Better yet, a microprocessor can be used to automatically adjust the receiver PN code phase. Once synchronized, the demodulation may be accomplished by using a D/A converter followed by a low pass filter.

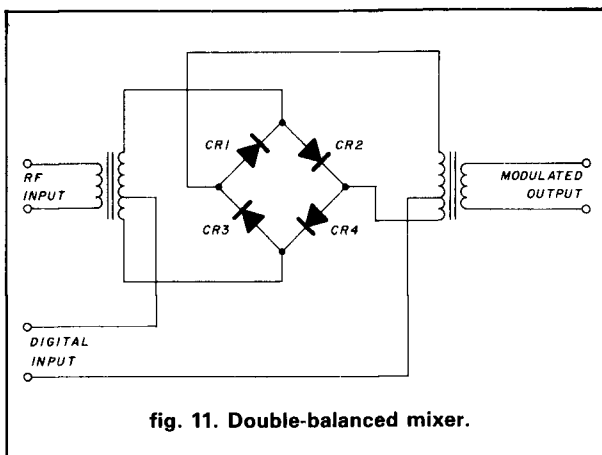
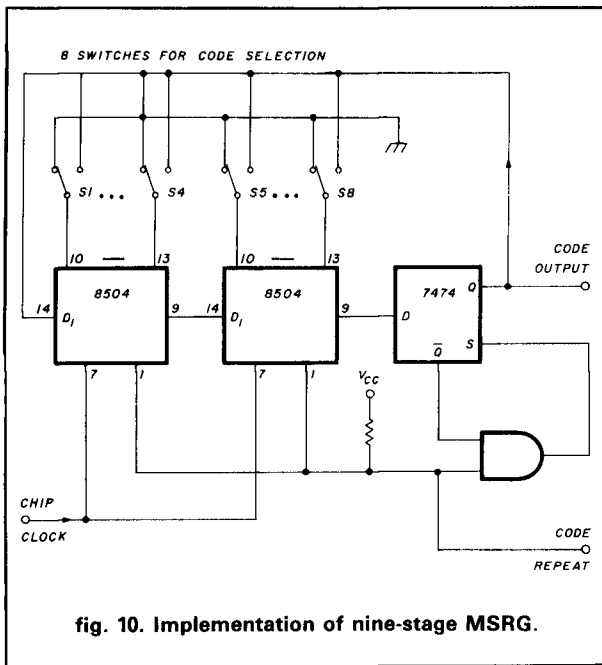


table 3. Characteristics of M-sequences of various length.

M stages	maximum number of unique codes	number of chips per period
3	2	7
4	4	15
5	6	31
6	4	63
7	18	127
8	16	255
9	48	511
10	60	1023
11	176	2047
13	630	8191
20	19,200	1,048,575
30	11,880,000	1,073,741,823
61	$3.1 \times 10^{16}$	$2.3 \times 10^{18}$

Since the receiver uses an internal clock signal for code reconstruction, the low pass filter could be of the switched capacitor variety.

It is important to note several points of practical interest. Obviously, the receiver must tune to the sender's transmitting frequency in order to establish the possibility of communication. This suggests that calling frequencies would be advisable for the first Amateur attempts at DS-SS communication. Furthermore, a standard shift register length (or a few agreed upon lengths) seems mandatory, since the senders and receivers must be able to match each other's PN code. A fixed chip duration is necessary, too, so that all stations would be ensured that they could synchronize with the other users.

Before leaving DS-SS, a word should be said about PN code length. In military applications, where security is an important consideration, it is not uncommon to find PN coding schemes which use shift register stages of length 40 or greater. If each chip duration is 1 microsecond, then one cycle of an  $m = 40$  PN code is completed in a time of

$$(2^{40} - 1) \text{ chips} \times \frac{10^{-6} \text{ sec}}{\text{chip}} = 1.1 \times 10^6 \text{ sec} = 12.7 \text{ days}$$

Even if an interceptor listened in on this signal for several hours, it would appear to be a random jumble of binary digits. For long codes such as this, it takes a very long time to synchronize the receiver. On the other hand, if a shorter PN code is used (say,  $m = 13$ ), then the entire code sequence is repeated every 8.3 milliseconds! A code having such a short period can be synchronized quite quickly at the receiver end.

**Frequency hopping (FH) spread spectrum.** In a FH-SS system (fig. 2B) there is a narrow band transmission occurring at any given time instant. However, there is a wide range of frequencies from which the transmitter may select. The particular frequency selected for use at any given moment is determined jointly by the digitized message bit stream and the PN code generator.

The message bit stream is used as the least significant bit (LSB) of an M bit data word. The PN code generator supplies the other M-1 bits. The data word is then used to determine the additive offset required to generate the proper frequency. Frequency hopping takes place over  $N = 2^M$  frequencies separated by integer multiples of  $f_1$ , where  $f_1$  is the gap between adjacent hop frequencies. The repetition rate of the frequency hopping sequence is determined by the number of stages in the PN code generator and by the speed of the chip clock. Since the generator supplies M-1 chips for each hop, the frequency hop sequence repeats every

$$\frac{(2^m \text{ stages} - 1)}{(M-1)} \text{ hops}$$

table 4. Indirect frequency synthesizer ICs and their characteristics.

part number	number of frequencies	reference frequency	maximum divider input frequency	control
Hughes HCTR0347	45	50 Hz-500 kHz	10 MHz	8 bits parallel
Nitron 6410	100	4.00 MHz	1.6 MHz	8 bits parallel
Motorola 145104	256	10.24 MHz	4.0 MHz	8 bits parallel
National DS8906	16384	10.24 MHz	120 MHz	20 bits parallel
National MM55110	1024	10.24 MHz	3.0 MHz	10 bits parallel
Fairchild 11C84	128	10.24 MHz	20 MHz	7 bits parallel
AD-TECH FS-2574	1000	10.00 MHz	258 MHz	10 bits parallel

Figure 13 illustrates a typical FH transmission for the case of  $M=3$  and a nine-stage m-code generator using a 1-kHz chip clock.

Typical values for a suitable FH system might be  $f_1=500$  kHz and  $M=3$  bits. For this example, the total RF bandwidth of the system would be

$$2^3 \text{ frequencies} \times \frac{500 \text{ kHz separation}}{\text{frequency}} = 4.0 \text{ MHz}$$

If the lowest frequency of the transmitter were 420 MHz, the highest frequency used by this system would be 424.0 MHz.

The frequency synthesizer is the key to an FH-SS system. Its operating characteristics (such as frequency range, switching speed, and hop duration) determine a system's capability. There are two major classes of synthesizers, the direct type and the indirect type. The direct frequency synthesizer uses filters and mixers and is seldom found in current Amateur gear. The indirect type uses phase-lock-loops (PLLs) to generate the desired frequency set. As a rule, indirect synthesizers are not as quick to switch frequencies, but are easier to implement.

Figure 14 shows a block diagram of an indirect frequency synthesizer. Those familiar with PLLs will immediately recognize the structure. The reference frequency,  $f_1$ , is related to the output frequency,  $f_j$ , by

$$f_j = n_j \times f_1$$

since the VCO and the feedback loop forces  $f_j/n_j$  to equal  $f_1$ . As can be seen in table 4, indirect frequency synthesizers, manufactured by several IC companies, can produce output frequencies above 100 MHz. To achieve greater frequencies, multiplier stages must be added.

The duration of a single frequency hop ( $t_h$ ) may be longer or shorter than the duration of a message bit. If the message bit duration is longer than  $t_h$ , then the system is called a *fast hop FH* system since the hop rate is greater than the message bit rate. Otherwise, the system is termed *slow hop FH*. The advantage of a fast hop system is that if there is interference on one of the hop frequencies, the garbled message bit may

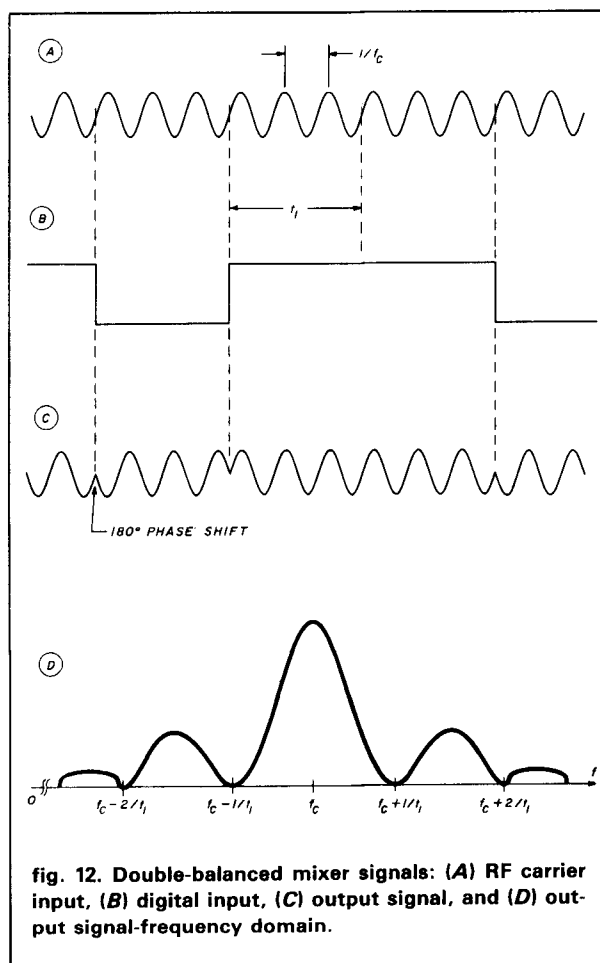


fig. 12. Double-balanced mixer signals: (A) RF carrier input, (B) digital input, (C) output signal, and (D) output signal-frequency domain.

still be present at the next hop frequency. For slow hop systems, error correction coding is needed since QRM on a given hop frequency could obliterate several message bits.

Reception of an FH-SS signal is achieved by synchronizing the receiver frequency with that of the transmitter's. Once synchronization occurs, the receiver output is identical to FSK, with the mark and space separated by  $f_1$  Hz. The demodulator can consist of band pass filters which are compared to determine the value of the message bit. The actual message

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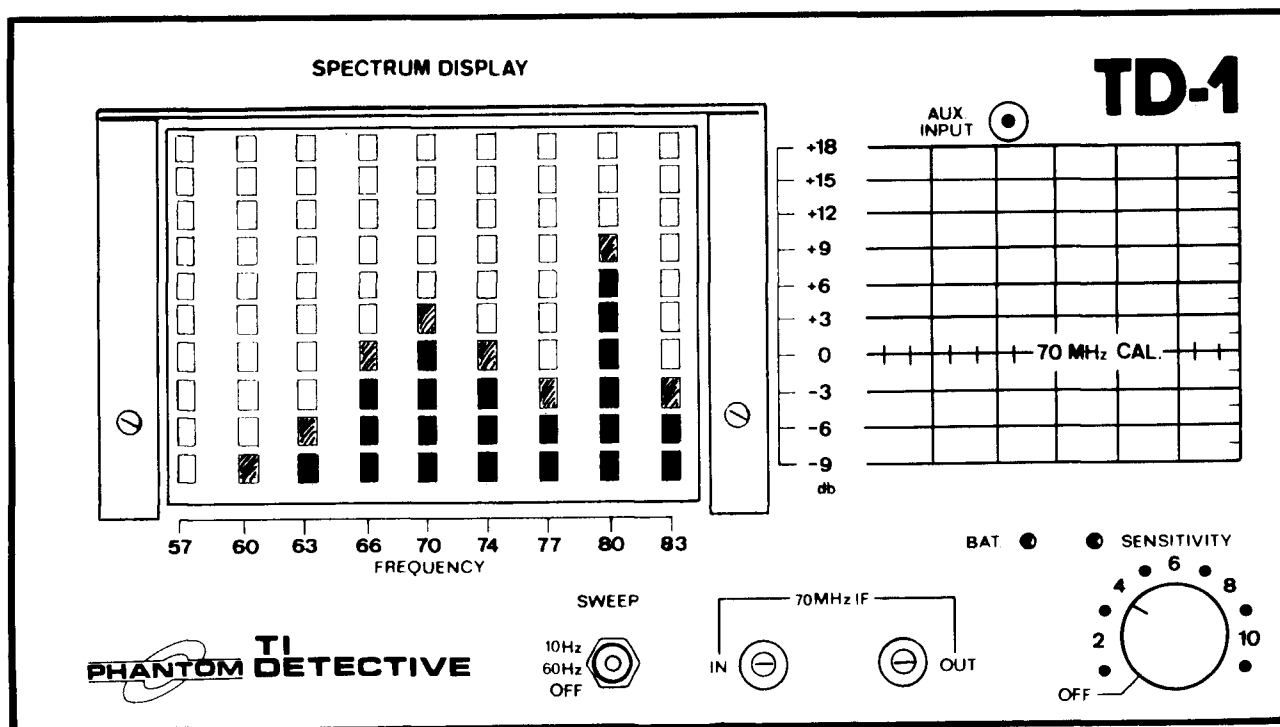
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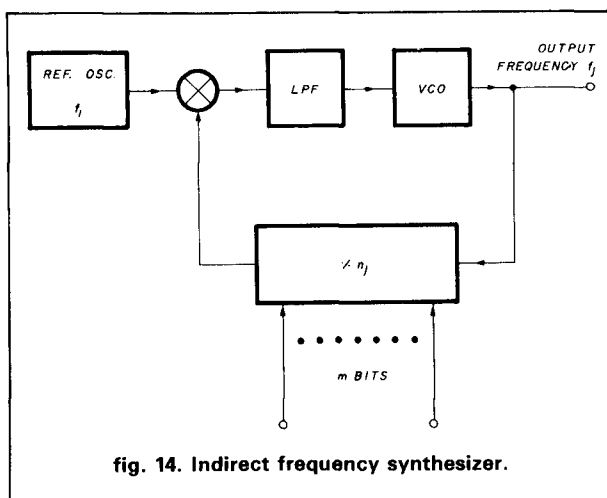
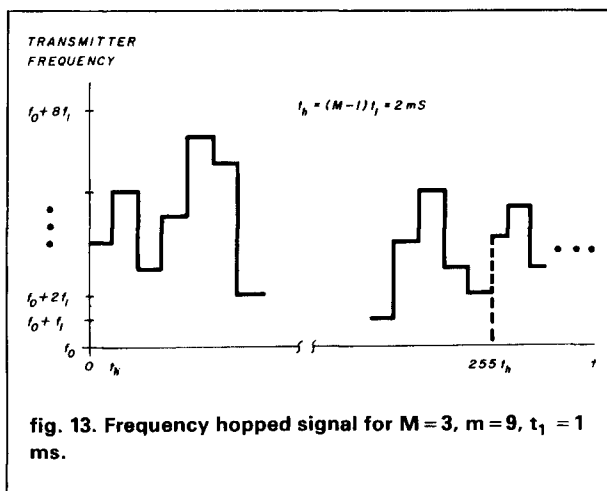
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is then reconstructed by using a D/A and a low-pass filter.

Synchronization must be acquired in roughly the same manner as in the DS case. A computer shifts the  $M-1$  bit PN code until the receiver tracks the transmitter through each hop. As cited in the DS case, a shorter PN code period ensures quicker acquisition of the transmitted signal.

With indirect synthesizer chips and MSRG chips readily available, it appears that FH systems could be constructed with slight modification to existing VHF/UHF gear. Certain standards such as hop duration, frequency band allocation, and the number of hop frequencies need to be developed, however.

### future of spread spectrum

In late May, 1985, the FCC approved docket 81414, which allows the use of spread spectrum on all ham frequencies above 420 MHz as of May, 1986. Temporary authorizations are now being given to those Amateurs interested in experimentation.

In less than six months from now, we'll have a new

mode of communication unlike any other we've ever tried. With this new mode, our hobby may take a big step toward reaching state-of-the-art digital communication techniques. There's a lot of work to be done, though; defining protocols for Amateur SS will not be simple.

### conclusion

The world of digital communications is a new, exciting technological field, and the future is being shaped daily by advances in this area. As Amateur operators, part of our charter is to increase the reservoir of electronics experts. While we don't have to be experts on digital communications, it's probably good for us to know the how's and why's of what is going on around us. Perhaps this article has shed some light on a subject that, as timely as it is, has not been widely discussed in the Amateur Radio literature.

For those interested in learning more about spread spectrum systems, the definitive reference is *Spread Spectrum Systems*, by Robert C. Dixon.<sup>17</sup> Dixon's book was the first on the topic, and has recently been revised to include discussions about practical hardware considerations. Also good is a book just released: *Modern Communications and Spread Spectrum* by C.D. McGillem and G.R. Cooper.<sup>13</sup> This book treats practically every type of modulation method and highlights some of the more important concepts of SS communication.

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