A Technical report on the Design and Simulink Implementation of

Anti Jam Communication System

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Introduction to Spread Spectrum Modulation:

Spread spectrum signals used for the transmission of information are distinguished by the characteristic that their bandwidth W is much greater than the information rate R in bits/s. That is, the bandwidth expansion factor B = W/R for a spread spectrum signal is much greater than unity. The large redundancy inherent in spread spectrum signals is required to overcome the severe levels of interference that are encountered in the transmission of information over wireless channels. Since coded waveforms are also characterized by a bandwidth expansion factor greater than unity and since coding is an efficient method for introducing redundancy, it follows that coding is an important element in the design of spread spectrum signals.

A second important element employed in the design of spread spectrum signals is pseudo randomness, which makes the signals appear similar to random noise and difficult to demodulate by receivers other than the intended ones. This element is intimately related with the application.

To be specific, spread spectrum signals are used for

1. Combating the detrimental effects of jamming, interference arising from other users in the channel and self interference due to multipath propagation.
2. Hiding a signal by transmitting it at low power and thus making it difficult for an unintended listener to detect in the presence of background noise.
3. achieving message privacy in the presence of other listeners.

In combating jamming, it is important to the communicators that the jammer who is trying to disrupt the communication does not have prior knowledge of the signal characteristics except for the overall bandwidth and the type of modulation being used. If the information is just encoded, a sophisticated jammer can easily mimic the signal emitted by the transmitter and thus confuse the receiver. To circumvent this possibility, the transmitter introduces an element of randomness in each of the transmitted coded signal waveforms that is known to the intended receiver but not to the jammer. As a consequence, the jammer must synthesize and transmit an interfering signal without knowledge of the pseudo random pattern.

Interference from the other users arises in multiple access communication systems in which a number of users share a common channel bandwidth. At any given time, a subset of these users may transmit information simultaneously over the common channel to corresponding receivers. Assuming that all the users employ the same code for the encoding and decoding of their respective information, the transmitted signals in this common spectrum may be distinguished from one another by superimposing a different pseudo random pattern also called a code, in each transmitted signal. Thus a particular receiver can recover the transmitted information intended for it by knowing the pseudo random pattern, i.e the key, used by the corresponding transmitter. This type of communication technique, which allows multiple users to simultaneously use a common channel for transmission of information is called Code Division Multiple Access.

Resolvable multipath components resulting from time dispersive propagation through a channel may be viewed as a form of self interference. This type of interference may also be suppressed by the introduction of a pseudo random pattern in the transmitted signal. A message may be hidden in the background noise by spreading its bandwidth with coding and transmitting the resultant signal at a low average power. Because of its low power level, the transmitted signal is said to be covert. It has a low probability of being intercepted by a casual listener and hence is also called a low probability of intercept signal.

Finally message privacy may be obtained by superimposing a pseudo random pattern on a transmitted message. The message can be demodulated by the intended receivers, who know the pseudo random pattern or key used at the transmitter but not by any other receivers.

Model of Spread Spectrum Communication System:

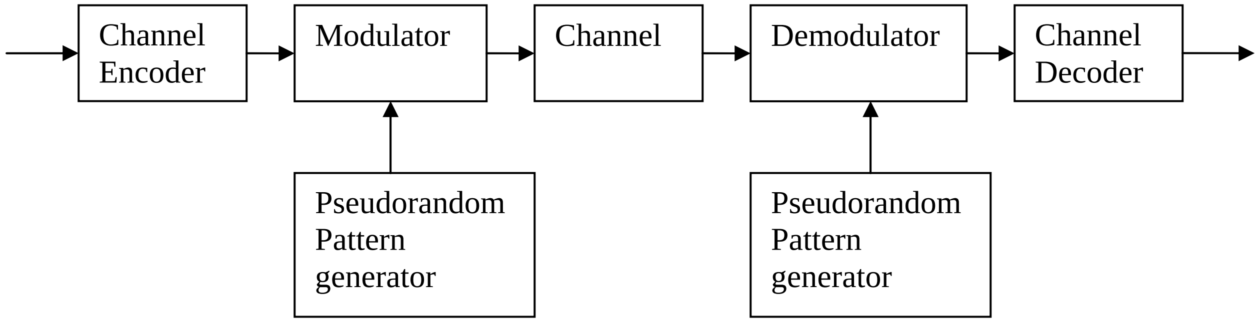


Diagram 1: Model of a spread spectrum digital communication system.

The block diagram shown in figure 1 illustrates the basic elements of a spread spectrum communication system with a binary information sequence at its input at the transmitting end and at its output at the receiving end. In addition to elements like encoder and decoder, modulator and demodulator, there are two identical pseudo random pattern generators, one that interfaces with the modulator at the transmitting end and a second that interfaces with the demodulator at the receiving end. The generators generate a pseudo noise sequence which is impressed on the transmitted signal at the modulator and removed from the received signal at the demodulator.

Synchronization of the PN sequence generated at the receiver with the PN sequence contained in the incoming received signal is required in order to demodulate the received signal. Initially prior to the transmission of information, synchronization may be achieved by transmitting a fixed pseudo random bit pattern that the receiver will recognize in the presence of interference with a high probability. After time synchronization of the generators is established, the transmission of information may commence. Interference is introduced in the transmission of the information bearing signal through the channel. The characteristics of the interference depend to a large extent on its origin. It may be categorized as being either broadband or narrowband relative to the bandwidth of the information bearing signal and either continuous or pulsed in time. For example, a jamming signal may consist of one or more sinusoids in the bandwidth used to transmit the information. The frequencies of the sinusoids may remain fixed or they may change with time according to some rule. As a second example, the interference generated in CDMA by other users of the channel may be either broadband or narrowband depending on the type of spread spectrum signal that is employed to achieve multiple access. If it is broadband, it may be characterized as an equivalent additive white Gaussian noise.

The PN sequence generated at the modulator is used in conjunction with the PSK modulation to shift the phase of the PSK signal pseudo randomly. The resulting modulated signal is called a Direct sequence spread spectrum signal. When used in conjunction with binary or M-ary FSK, the pseudo random sequence selects the frequency of the transmitted signal pseudo randomly. The resulting signal is called frequency hopped spread spectrum signal.

Direct Sequence Spread Spectrum Modulation: (DSSS)

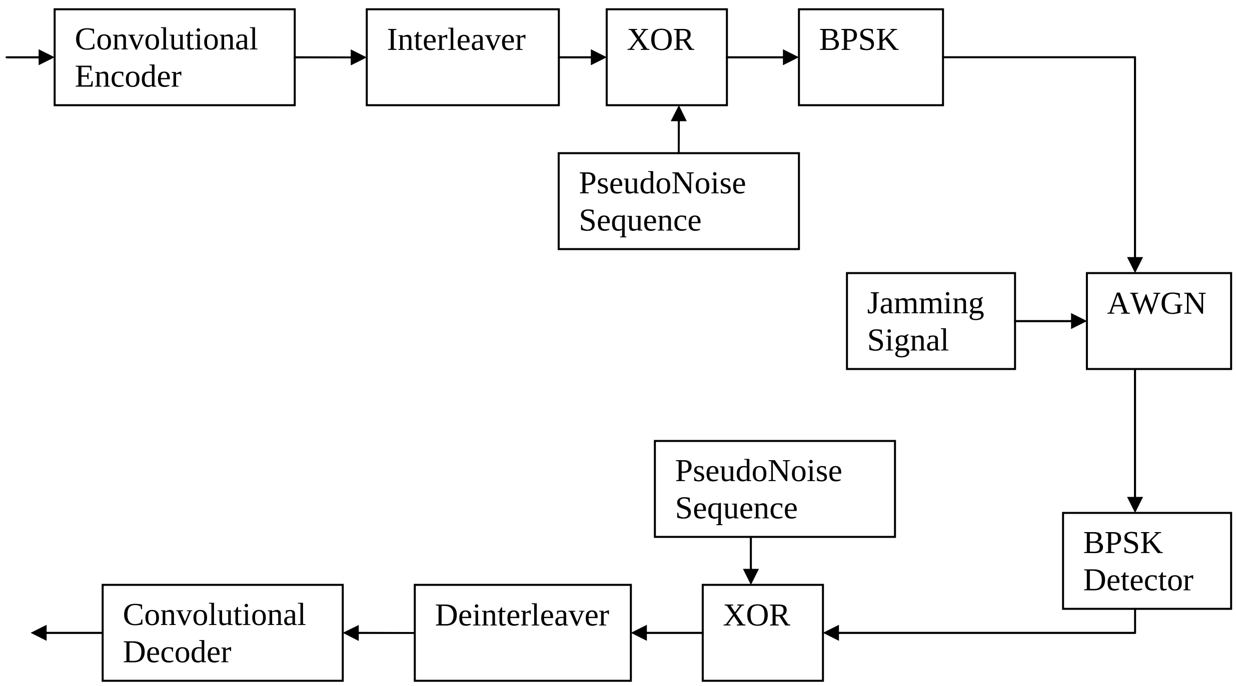


Diagram 2: Direct Sequence Spread Spectrum (DSSS) Modem.

A direct sequence spread spectrum modem is shown in the figure 2. The information rate at the input to the encoder is R bits/s and the available bandwidth is W Hz. The modulation is assumed to be binary PSK. In order to utilize the entire bandwidth, the phase of the carrier is shifted pseudo randomly according to the pattern from the PN generator at the rate W times/s. The reciprocal of W, denoted by T_c, defines the duration of a rectangular pulse, which is called a chip while T_c is called the chip interval. The pulse is the basic element in a DS spread spectrum signal.

Define $T_b = 1/R$ to be the duration of a rectangular pulse corresponding to the transmission time of an information time of an information bit, the bandwidth expansion factor W/R may be expressed as $B_e = W/R = T_b / T_c$.
In practical systems, the ratio T_b / T_c is an integer,
$L_c = T_b / T_c$ is the number of chips per information bit. That is, L_c is the number of phase shifts that occur in the transmitted signal during the bit duration $T_b = 1/R$.

Suppose that the encoder takes k information bits at a time and generates a binary linear (n, k) block code. The time duration available for transmitting the n code elements is k T_b sec. The number of chips that occur in this time interval is kL_c. Hence the block length of the code is $n = kL_c$. If the encoder generates a binary convolutional code of rate k/n, the number of chips in the time interval kT_b is also $n = kL_c$. The following theory applies to both block codes and convolutional codes.

One method for impressing the PN sequence on the transmitted signal is to alter directly the coded bits by modulo-2 addition with the PN sequence. Thus, each coded bit is altered by its addition with a bit from the PN sequence. If b_i represents the i^{th} bit of the PN sequence and c_i is the corresponding bit from the encoder, the modulo-2 sum is
$a_i = b_i + c_i$.

Hence, $a_i = 1$ if either $b_i = 1$ and $c_i = 0$ or $b_i = 0$ and $c_i = 1$; also $a_i = 0$ if either $b_i = 1$ and $c_i = 1$ or $b_i = 0$ and $c_i = 0$. The sequence { a_i } is mapped into a binary PSK signal of the form $s(t) = Re[\, g(t) \exp(j2\pi f_c t)]$ according to the convention

$g_i(t) = g(t-iT_c)$ if $a_i = 0$ & $-g(t-iT_c)$ if $a_i = 1$.
where g(t) represents a pulse of duration T_c s and arbitrary shape.

The modulo-2 addition of the coded sequence $\{c_i\}$ and the sequence $\{b_i\}$ from the PN generator may also be represented as a multiplication of two waveforms. To demonstrate this point, suppose that the elements of the coded sequence are mapped into a binary PSK signal according to the relation

$c_i(t) = (2c_i - 1)g(t - iT_c)$

Similarly define a waveform $p_i(t)$ as

$p_i(t) = (2b_i - 1)p(t - iT_c)$
where p(t) is a rectangular pulse of duration T_c.

Then the equivalent lowpass transmitted signal corresponding to the i^{th} coded bit is
$g_i(t) = p_i(t)\, c_i(t)$.

Consequently, modulo-2 addition of the coded bits with the PN sequence followed by a mapping that yields a binary PSK signal is equivalent to multiplying a binary PSK signal generated from the coded bits with a sequence of unit amplitude rectangular pulses, each of duration T_c and with a polarity which is determined from the PN sequence.

The received equivalent lowpass signal for the i^{th} code element is
$r_i(t) = p_i(t)\, c_i(t) + z(t)$, $iT_c \le t \le (i+1)\, T_c$
where z(t) represents the interference or jamming signal corrupting the information bearing signal. The interference is assumed to be a stationary random process with zero mean.

If z(t) is a sample function from a complex valued Gaussian process, the optimum demodulator may be implemented either as a filter matched to the waveform g(t) or as a correlator. In the matched filter realization, the sampled output from the matched filter is multiplied by (2bi – 1), which is obtained from the PN generator at the demodulator when the PN generator is properly synchronized. Since $(2bi – 1)^2 = 1$ when $b_i = 0$ and $b_i = 1$, the effect of the PN sequence on the received coded bits is thus removed.
The cross-correlation can be accomplished in either one of two ways. The first involves pre multiplying ri(t) with the waveform $p_i(t)$ generated from the output of the PN generator and then cross correlation with g*(t) first, sampling the output of the correlator and then multiplying this output with (2bi – 1), which is obtained from the PN generator.

If z(t) is not a Gaussian random process, the demodulation methods stated are no longer optimum.

Frequency Hopped Spread Spectrum Modulation: (FHSS)

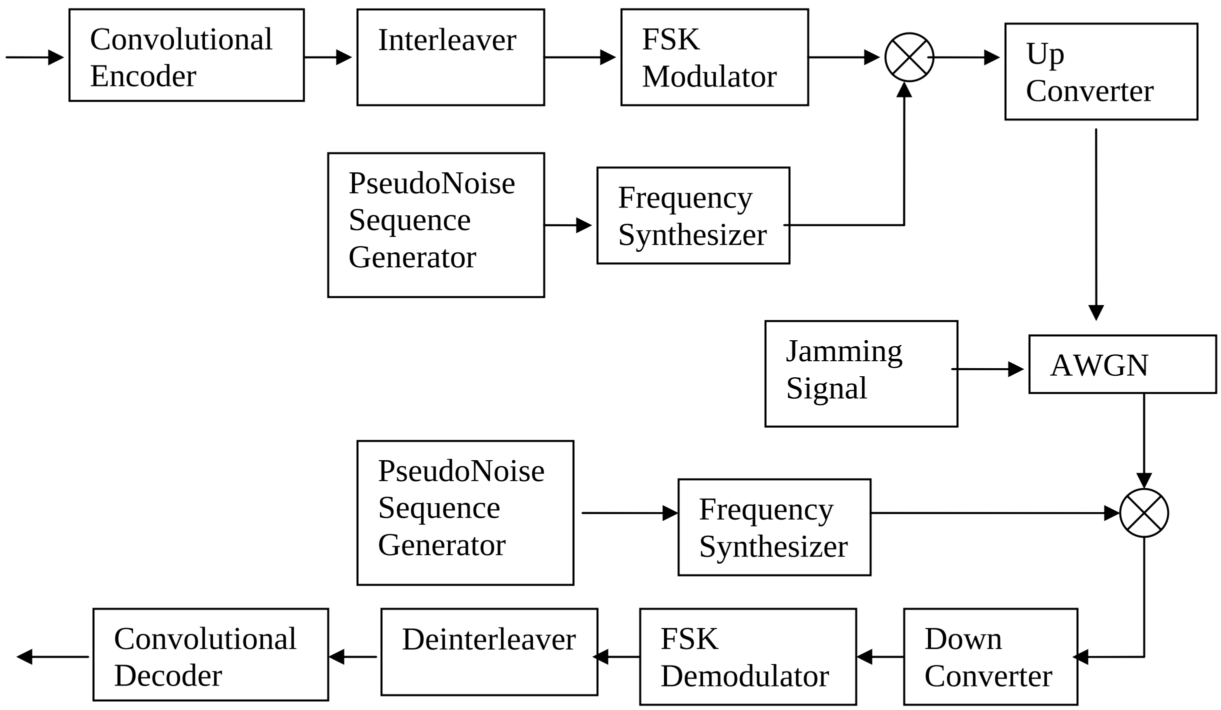


Diagram 3: Frequency Hopping Spread Spectrum (FHSS) Modem.

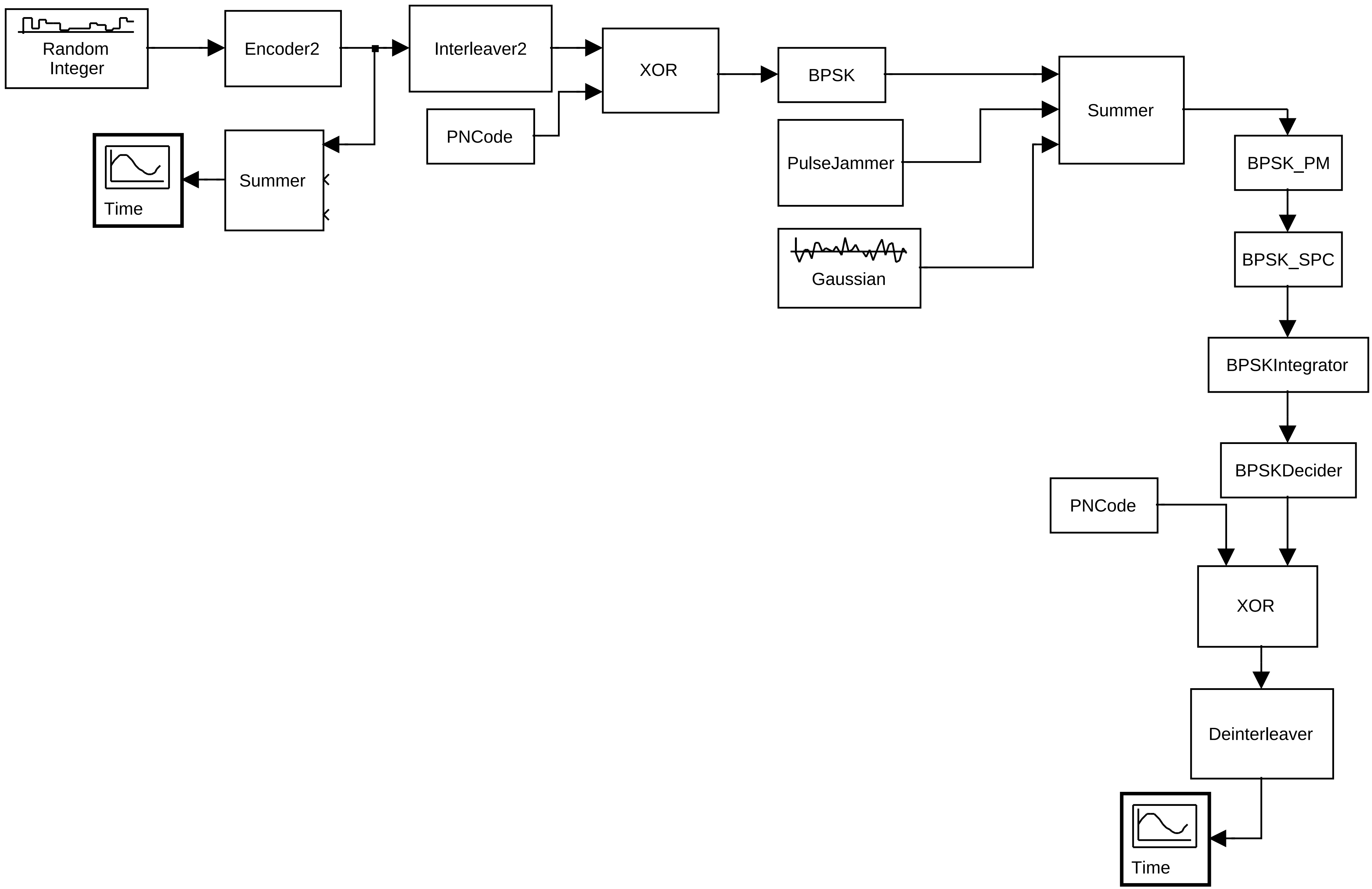
In a frequency hopped spread spectrum communications system the available bandwidth is subdivided into a large number of contiguous frequency slots. In any signaling interval, the transmitted signal occupies one or more of the available frequency slots. The selection of the frequency slots in each signaling interval is made pseudo randomly according to the output from a PN generator.
A block diagram of the Frequency hopped spread spectrum modem is shown in figure 3.

The modulation is usually either binary or M-ary FSK. For example, if binary FSK if employed, the modulator selects one of two frequencies corresponding to the transmission of either a 1 or 0. The resulting FSK signal is translated in frequency by an amount that is determined by the output sequence from the PN generator, which in turn, is used to select a frequency that is synthesized by the frequency synthesizer. This frequency is mixed with the output of modulator and the resultant frequency translated signal is transmitted over the channel. For example, m bits from the PN generator may be used to specify 2^m -1 possible frequency translations.

At the receiver, we have an identical PN generator, synchronized with the received signal, which is used to control the output of the frequency synthesizer. Thus, the pseudo random frequency translation introduced at the transmitter is removed at the receiver by mixing the synthesizer output with the received signal. The resultant signal is demodulated by means of an FSK demodulator. A signal for maintaining synchronism of the PN generator with the frequency translated received signal is usually extracted from the received signal.

The frequency hopping rate is usually selected to be either equal to the symbol rate or faster than that rate. If there are multiple hops per symbol, it is fast frequency hopping. If the hopping is performed at the symbol rate, it is slow frequency hopping.

DSSS Modem implemented in Simulink:



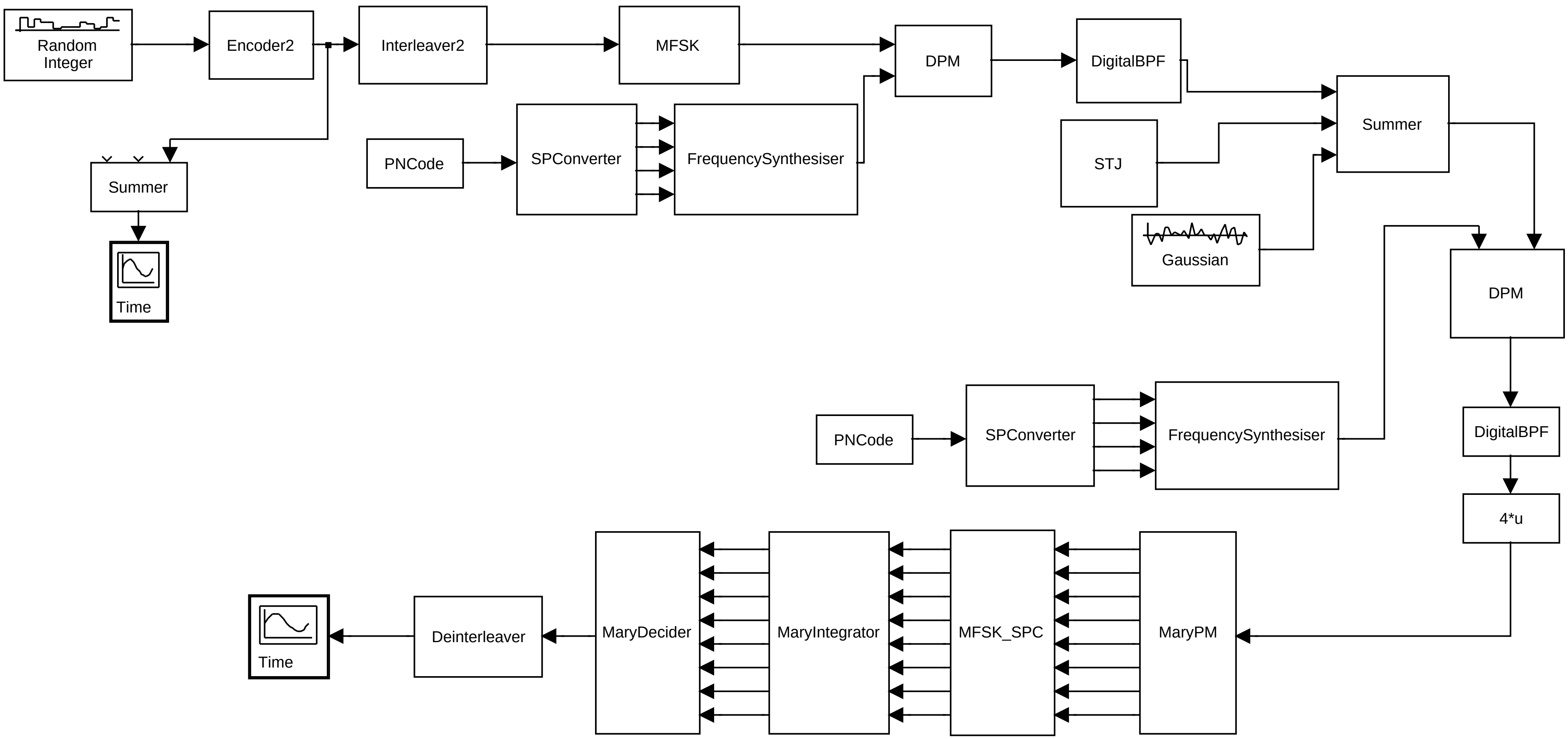
Description of Blocks in the DSSS Modem implemented in Simulink:

Sl	Block Name	Description
1	Random Integer	Random Bit Generator - Simulink Built In Block
2	Encoder 2	Convolutional Encoder Rate 1/3 - Coded Block
3	Interleaver 2	Convolutional Interleaver - Coded Block
4	PNCode	Generates the Maximum Length spreading sequence using a Linear Feedback Shift Register - Coded Block
5	XOR	This block spreads the coded & interleaved data sequence by a Pseudo Noise sequence - Coded Block
6	BPSK	Generates the BPSK signal for every chip duration - Coded Block
7	STJ	Single Tone Jammer - Coded Block
8	MTJAM	Multi Tone Jammer - Coded Block
9	Pulse Jammer	Pulse Jamming signal - Coded Block
10	Gaussian	Gaussian noise generator - Simulink Built In Block
11	BPSK_PM, BPSK_SPC, BPSKIntegrator, BPSKDecider	The mentioned blocks in cascaded connection simulate a standard Matched Filter followed by a Maximum Likelihood detector for BPSK - Coded Block
12	Deinterleaver	Convolutional Deinterleaver - Coded Block
13	Summer	Block that adds the input signals - Coded Block
14	Time	Block used for data display - Simulink Built In Block

Input Parameters to the Simulink Blocks of DSSS Modem:

Sl	Block Name	Input Parameters	Values
1	Encoder2	1. Input Symbol Duration 2. Symbol Size	1/50 2
2	Interleaver2	Input Symbol Duration	1/150
3	XOR	Sample Time	1/150
4	PNCode	Chip Duration	1/150
5	BPSK	1. Bit Energy 2. Bit Duration 3. Frequency 4. Sample Time	1/100 1/150 3000 1/15000
6	Pulse Jammer	1. Amplitude 2. High Duration 3. Pulse Duration 4. Sample Time	1 1/900 1/300 1/15000
7	MTJAM	1,3,5 Amplitude(s) 2,4,6 Frequencies 7. Sample Time	
8	STJ	1. Amplitude 2. Frequency 3. Sample Time	1 3900 1/15000
9	BPSK_PM	1. Bit Duration 2. Frequency 3. Sample Time	1/150 3000 1/15000
10	BPSK_SPC	Parallel Width	100
11	BPSKIntegrator	1. Bit Duration 2. Parallel Width	1/150 100
12	BPSKDecider		
13	Deinterleaver		
14	Summer	Sample Time	variable

FHSS Modem implemented in Simulink:



Description of Blocks in the FHSS Modem implemented in Simulink:

SI	Block Name	Description
1	Random Integer	Random Symbol Generator - Simulink Built In Block
2	Encoder 2	Convolutional Encoder Rate 1/3 - Coded Block
3	Interleaver 2	Convolutional Interleaver - Coded Block
4	MFSK	Generates the M-ary Frequency Shift Keying signal for every symbol duration and tone size M - Coded Block
5	PNCode	Generates the Maximum Length spreading sequence using a Linear Feedback Shift Register - Coded Block
6	SPConverter	Serial to Parallel: Using the serial bits of the spreading sequence, generates a parallel bit vector which is used by the Frequency Synthesizer - Coded Block
7	Frequency Synthesizer	The block generates a single tone signal whose tone is dependent on the input parallel bit vector - Coded Block
8	DPM	Discrete Product Modulator: Frequency hopping/dehopping of the M-ary FSK signals is done by the block - Coded Block.
9	Digital BPF	Band Pass Filter: The Upconverted Frequency hopped M-ary FSK signals are generated by the Band Pass Filter by passing the higher of the two tones at the output of DPM. The BPF implemented is an IIR filter. The IIR filter coefficients are fed as inputs to the block. The passband frequency range & thus IIR filter coefficients are determined by the symbol alphabet size/ FSK tone size M & Frequency Synthesizer output. Currently M = 8. At the receiver, the Upconverted Frequency hopped M-ary FSK signals are converted back to plain M-ary FSK signals by Down conversion - Coded Block
10	STJ	Single Tone Jammer - Coded Block
11	MTJAM	Multi Tone Jammer - Coded Block
12	Pulse Jammer	Pulse Jamming signal - Coded Block
13	MaryPM, MFSK_SPC, MaryIntegrator, MaryDecider	The mentioned blocks in cascaded connection simulate a standard Matched Filter followed by a Maximum Likelihood detector for M-ary FSK - Coded Block
14	Deinterleaver	Convolutional Deinterleaver - Coded Block
15	Gaussian	Gaussian noise generator - Simulink Built In Block
16	Summer	Block that adds the input signals - Coded Block
17	Time	Block used for data display - Simulink Built In Block

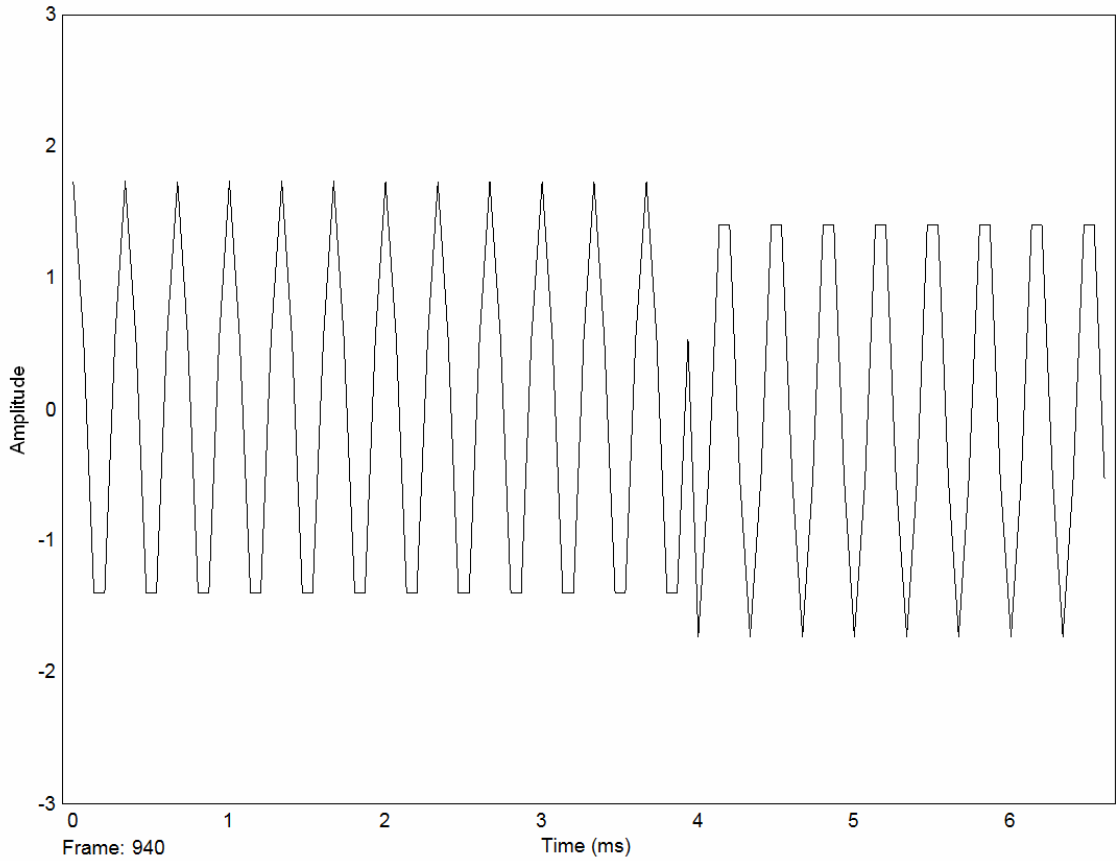
Input Parameters to the Simulink Blocks of FHSS Modem:

SI	Block Name	Input Parameters	Values
1	Encoder2	1. Input Symbol Duration 2. Symbol Size	1/50 8
2	Interleaver2	Input Symbol Duration	1/150
3	PNCode	Chip Duration	1/150
4	SPConverter		
5	Frequency Synthesiser	1. Sample Time 2. Chip Duration	1/25000 1/150
6	MFSK	1. Symbol Energy 2. Symbol Duration 3. Frequency 4. Sample Time	1/100 1/150 3000 1/25000
7	DPM	Sample Time	1/25000
8	Digital BPF	1. Filter Coefficients **a** 2. Filter Coefficients **b** 3. Length of **a** 4. Length of **b**	$\mathbf{a}_t/\mathbf{a}_r$ $\mathbf{b}_t/\mathbf{b}_r$ 59/23 59/23
9	MaryPM	1. M-ary Number 2. Symbol Duration 3. Frequency 4. Sample Time	8 1/150 3000 1/25000
10	MFSK_SPC	1. M-ary Number 2. Parallel Width	8 166
11	MaryIntegrator	1. M-ary Number 2. Symbol Duration 3. Parallel Width	8 1/150 166
12	MaryDecider	M-ary Number	8
13	Deinterleaver		
14	Pulse Jammer	1. Amplitude 2. High Duration 3. Pulse Duration 4. Sample Time	1 1/900 1/300 1/25000
15	MTJAM	1,3,5 Amplitude(s) 2,4,6 Frequencies 7. Sample Time	
16	STJ	1. Amplitude 2. Frequency 3. Sample Time	0.01 3900 1/25000
17	Deinterleaver		
18	Summer	Sample Time	1/25000

Note:

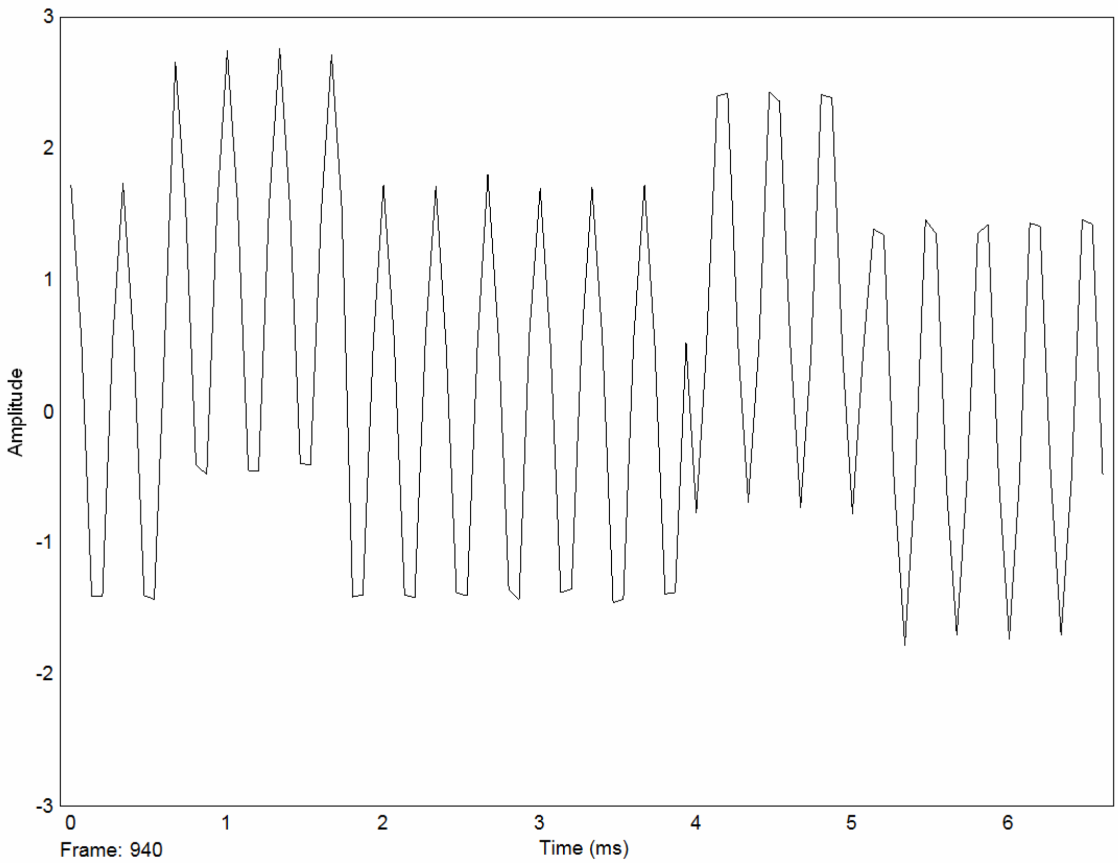
1.) $\mathbf{a}_t$ & $\mathbf{b}_t$ Transmitter IIR Digital Bandpass filter coefficients.
2.) $\mathbf{a}_r$ & $\mathbf{b}_r$ Receiver IIR Digital Bandpass filter coefficients.

AWGN Input for the DSSS Modem:

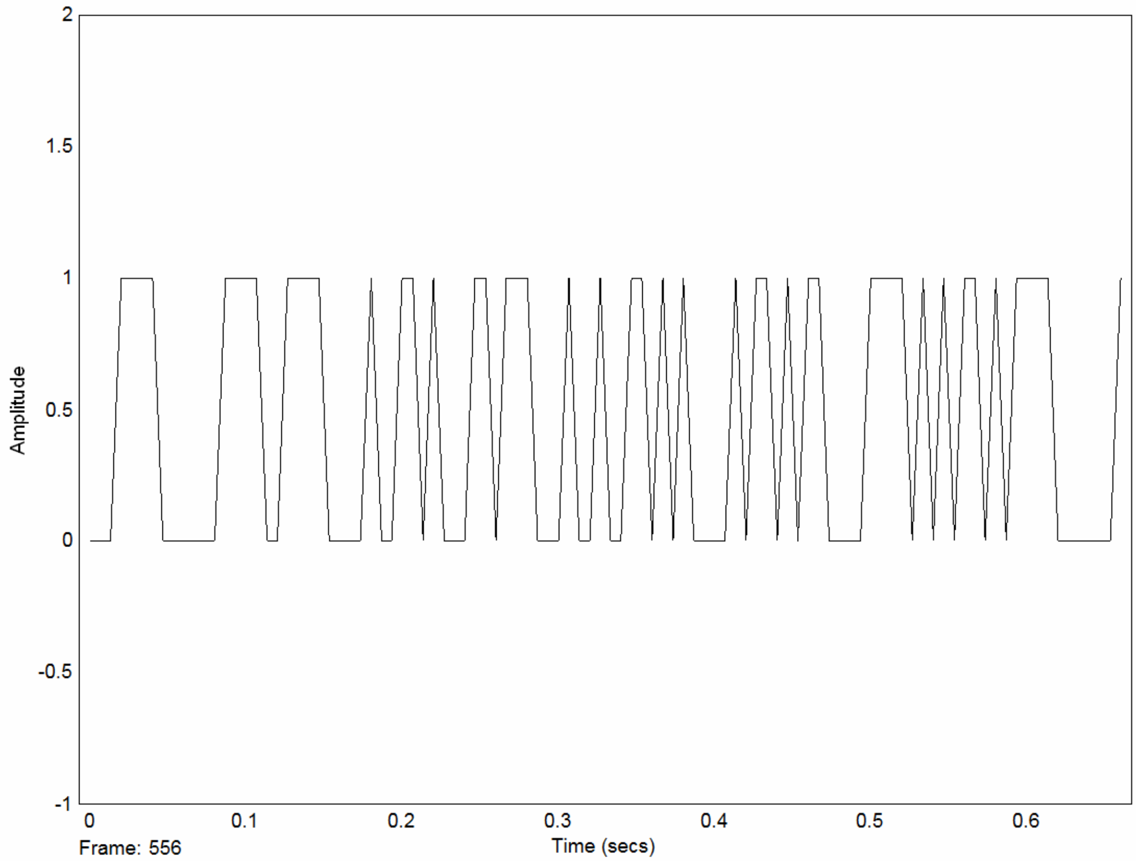


AWGN Output for the DSSS Modem:

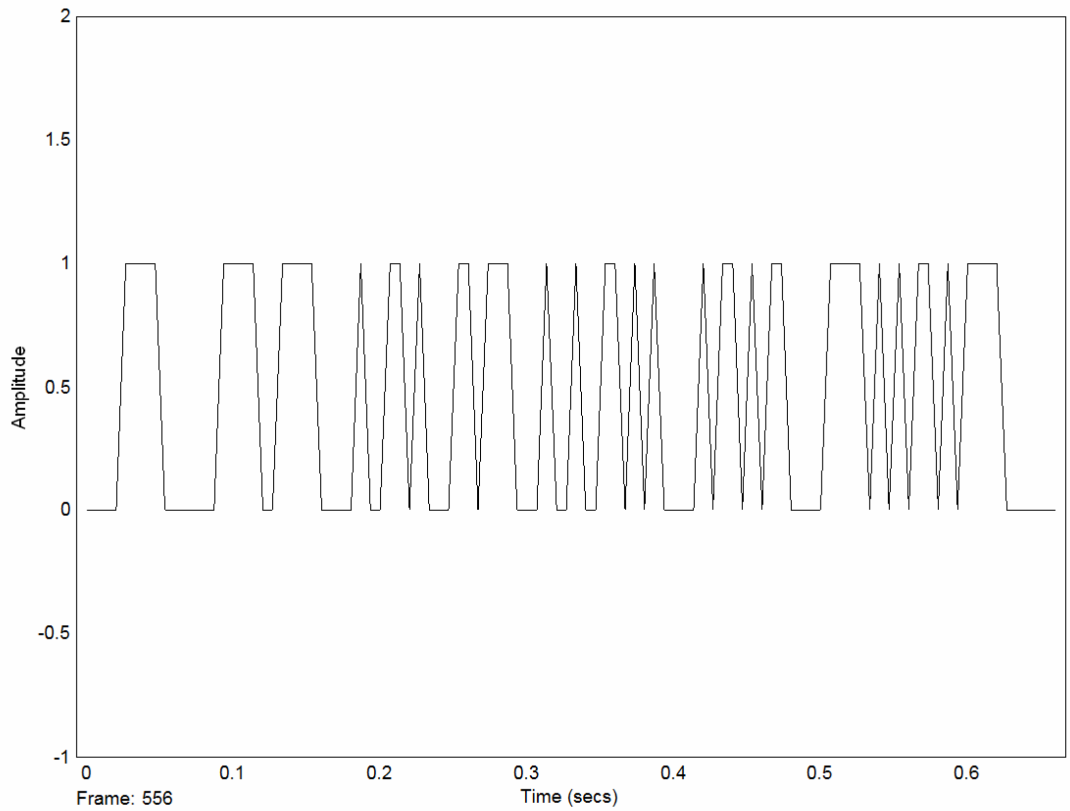
Periodically some of the AWGN output samples are shifted in amplitude. This is because of the pulse jamming in the channel, which is a periodic jamming signal.



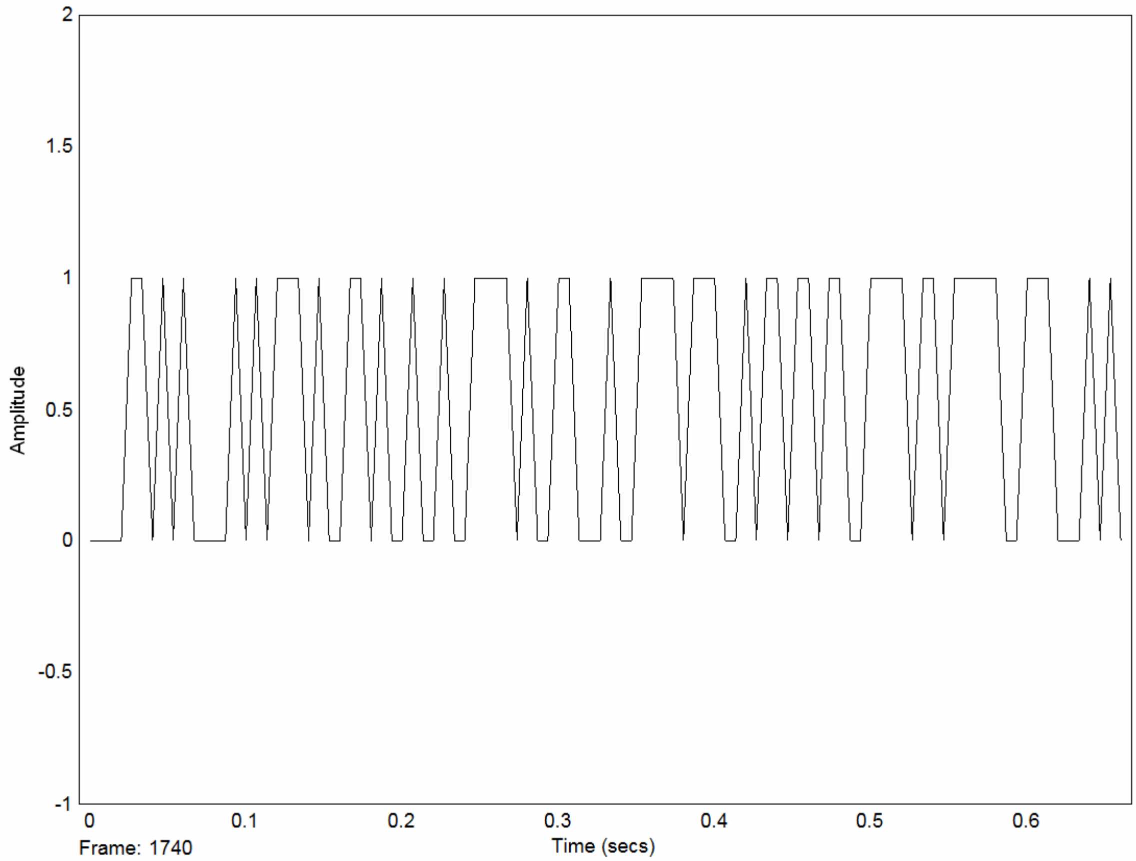
BPSK Modulator Input Symbols for the DSSS Modem:



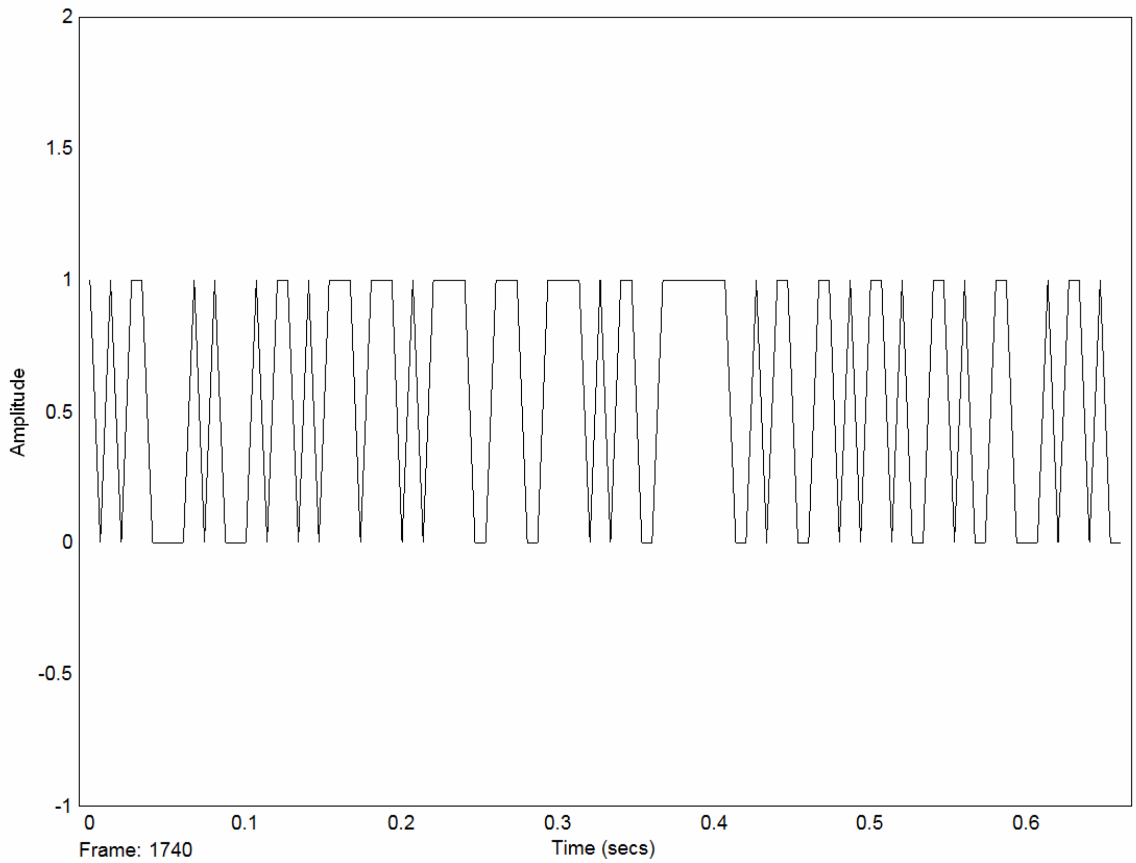
BPSK Detector Output for the DSSS Modem:
BPSK Detector output symbols are delayed from the BPSK Modulator input symbols by a small delay.



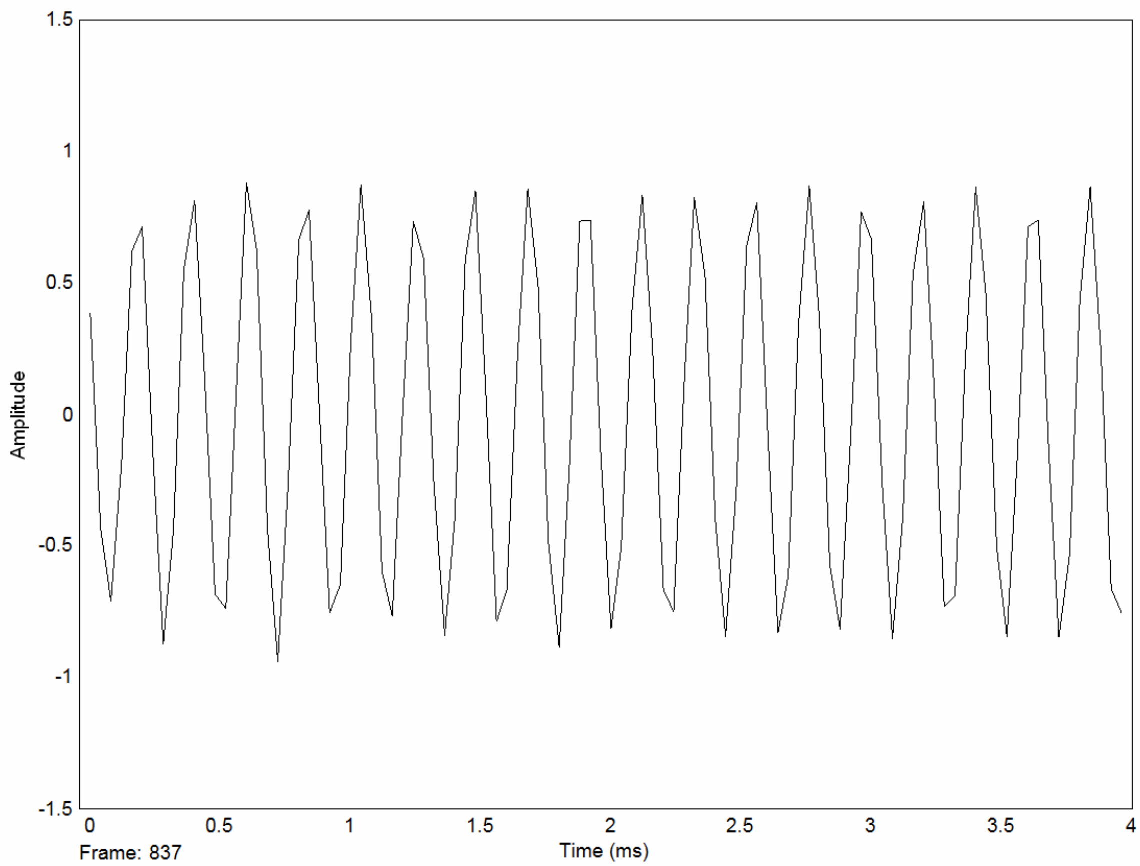
Interleaver Input for the DSSS Modem:



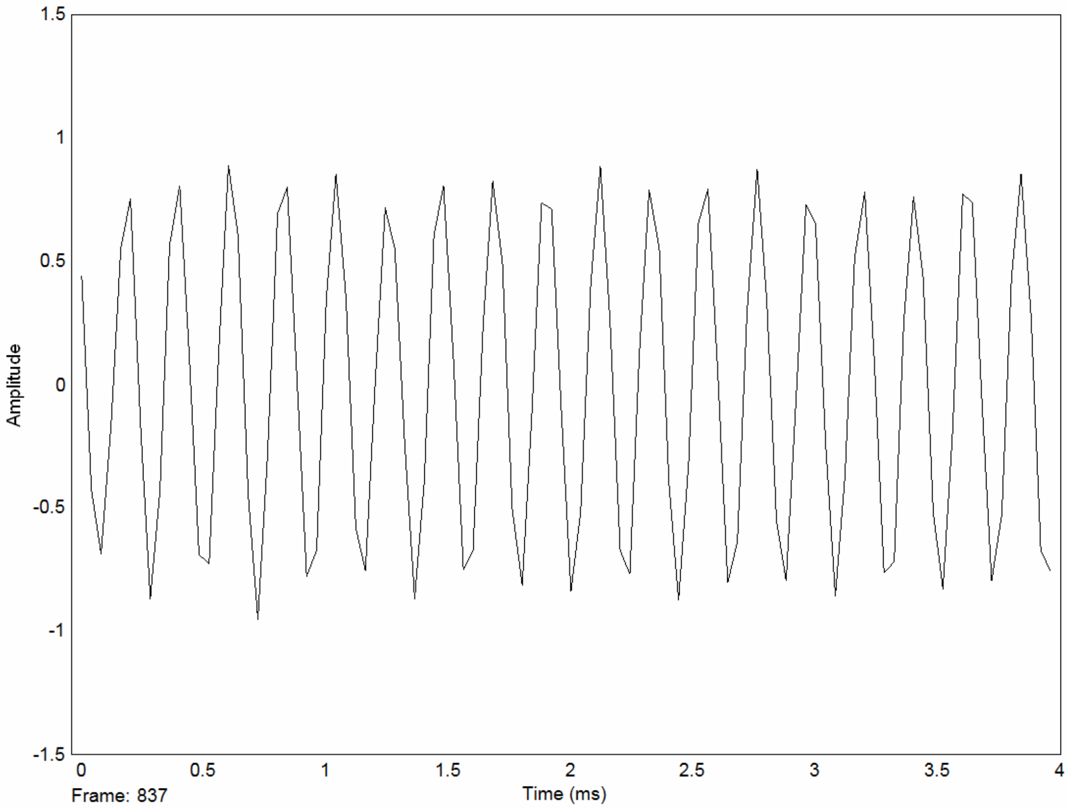
Deinterleaver Output for the DSSS Modem:
Due to the lack of time synchronization of the PN code generator at the receiver, errors are introduced into the deinterleaved symbols.



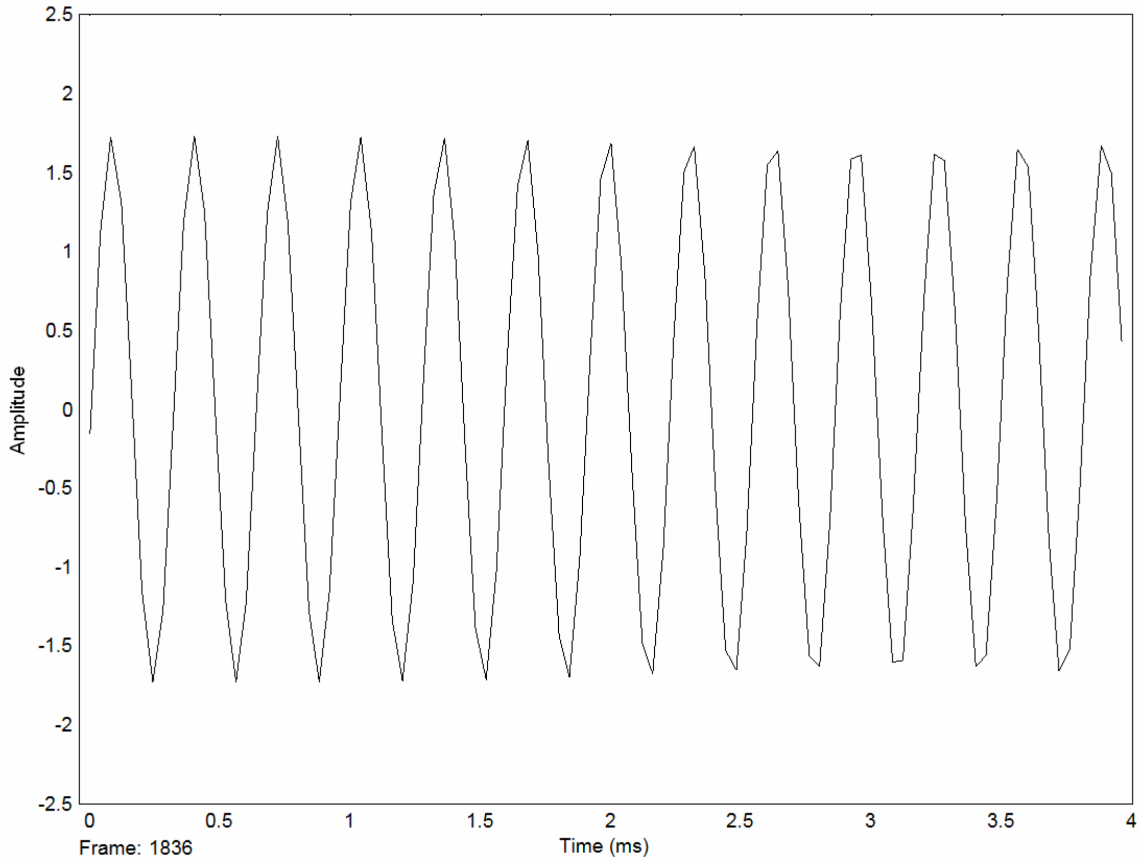
AWGN Input for the FHSS Modem:



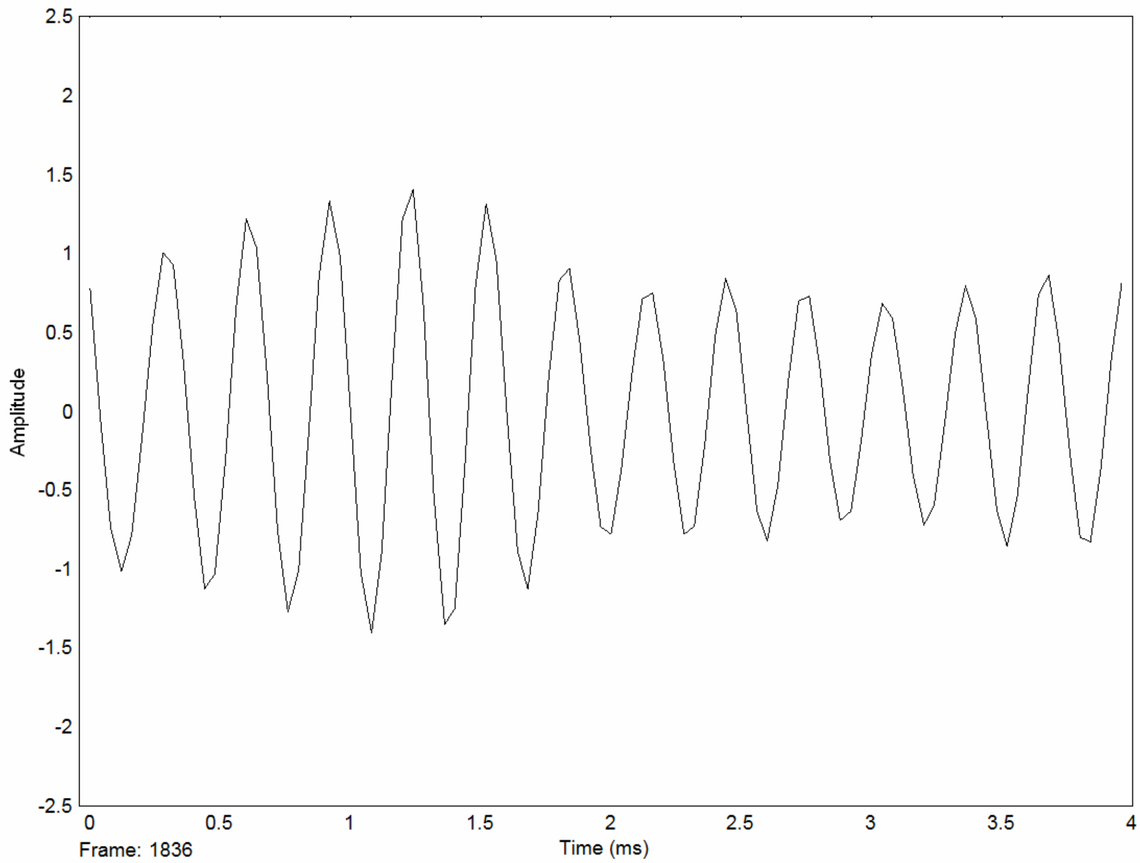
AWGN Output for the FHSS Modem:



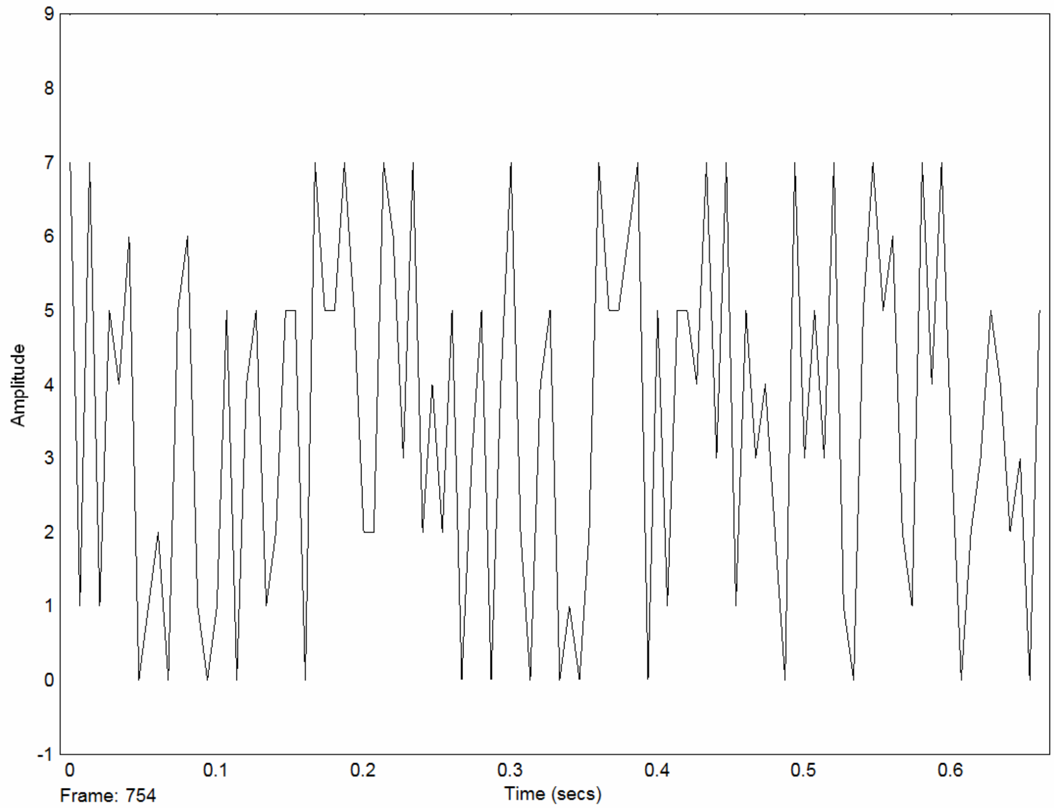
MFSK Modulator Output for the FHSS Modem:



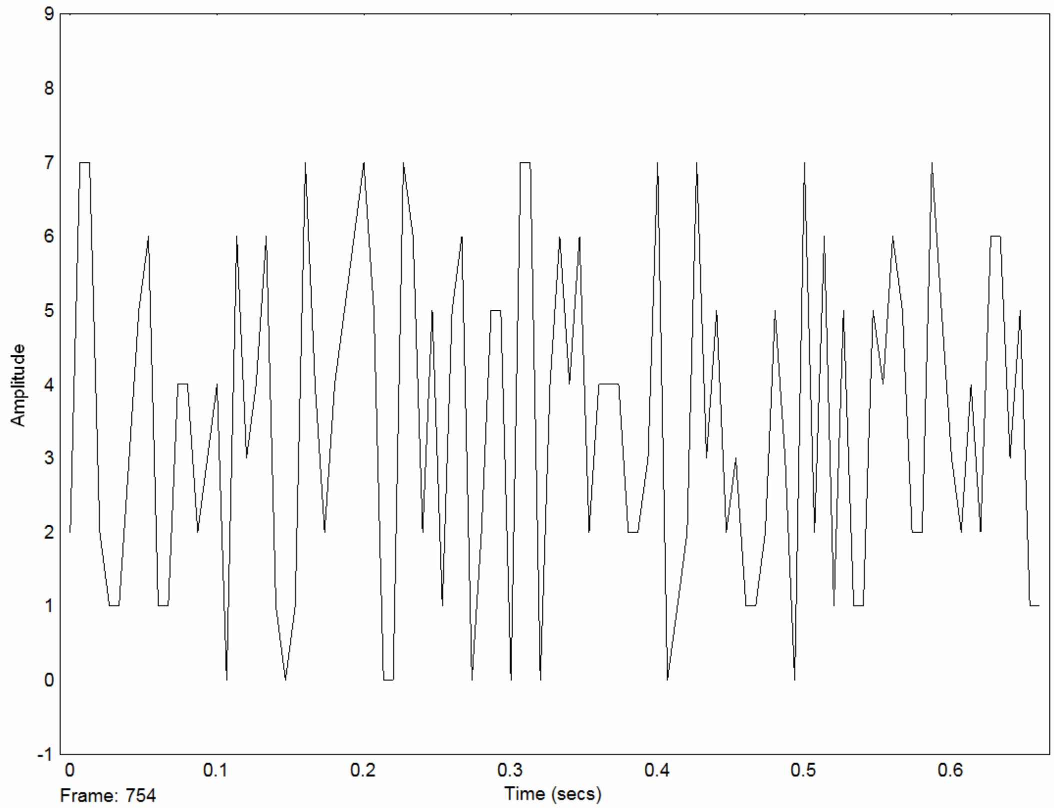
MFSK Detector Input for the FHSS Modem:
Due to the imperfect down conversion by the digital Bandpass filter at the receiver, errors are introduced into the signal input to the MFSK detector.



MFSK Modulator Input for the FHSS Modem:



MFSK Detector Output for the FHSS Modem:



Conclusions:
Future work can include a RAKE detector for operating in multipath channels. Use of a more sophisticated digital bandpass filter replacing the current IIR filter can improve the system performance. If the pseudo noise sequence generator at the receiver is synchronized to the received signal, error rate can be reduced. Use of LDPC and/or Turbo codes in the system will increase the performance against jamming signals.

References:
[1] "Digital Communications" John G Proakis, McGraw Hill, 4 edition, August 15, 2000.
[2] "Communication Systems" Simon Haykin, Wiley, 4 edition, May 15, 2000.