Development and Study of Demodulation Techniques for Frequency-Modulated Signals

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***Abstract*— We suggest several techniques for demodulation of frequency-modulated signals. The techniques are compared using the criterion of error rate of a demodulated signal relative to the signal before modulation. The experimental research is conducted for different signal-to-noise ratios.**

Keywords— demodulation; frequency-modulated signal; error rate; signal-to-noise ratio

I. INTRODUCTION

A communication channel is aimed at transmitting some information. The communications theory tells us that there are two main reasons for low transmission variability [1]. One of the reasons is low signal-to-noise ratio (S/N — Signal to Noise, or SNR — Signal-to-Noise Ratio). The other one is signal distortion. The signal under study can be an information signal, video impulse or modulated harmonic signal. When we deal with analogue signals we can mention intermodulation distortions (e.g., CSO and channel distortions) [2]. Digital communication systems are mostly based on intersymbol interference. The present paper discusses only error rate computation depending on SNR and the modulation technique.

II. INPUT SIGNAL

The input signal is a frequency modulated signal, which is the type of analogue modulation when the information signal determines the carrier frequency [3]. Logical "1" and "0" are calculated using the formula $f \pm f_0$, where f is the carrier frequency, f_0 is the deviation of frequency values of logical "1" and "0". In our research f_0 has the values of 100, 250, and 500 Hz. The channel bandwidth is determined by f_0 and is twice greater than the value of this carrier frequency.

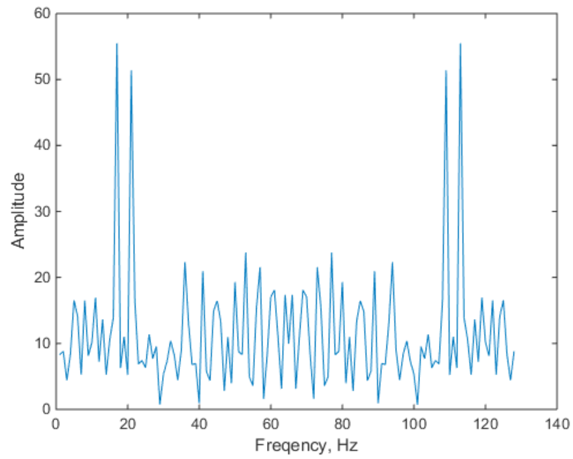

Fig. 1. Spectrum of input signal

III. MODEL FOR DEMODULATORS

Consider a model simulating communication channels and computing the error rate. The communication channels have ideal synchronization. The sampling rate of a modulated signal is 12800 Hz and each symbol has 128 samples. The model has a synchronizer which produces a synchroimpulse each 128 samples and this impulse indicates the start of operation. Such a construction provides "pure" reception of one symbol.

The input signal is passed from the output of a random number generator. This generator produces "1" and "0" or a constant value.

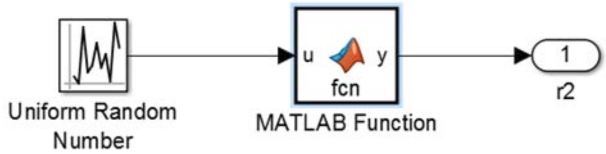

Fig. 2. Input signal generator

After the generator the signal is sent to a modulator for forming a frequency-modulated signal. The frequency of logical "1" is 2000 Hz. The frequency of logical "0" is 1600 Hz.

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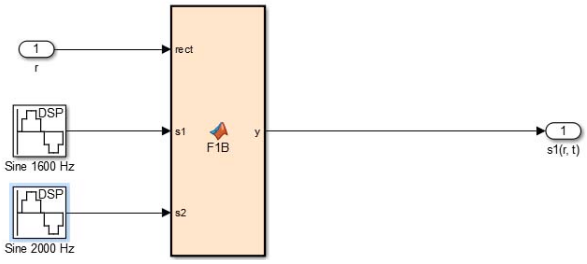

Fig. 3. Signal modulator

Next, the signal is sent to a block imitating Rayleigh fading, and another block adding white Gaussian noise. The noise is generated by the block AWGN (Additive White Gaussian Noise) in Matlab/Simulink.

There are different ways of setting the required noise level depending on an input message: signal power divided by noise power (SNR), relation of information bits per symbol to noise power spectral density (Eb/N0), ratio of information symbol energy to noise power spectral density (Es/N0). The model uses Es/N0.

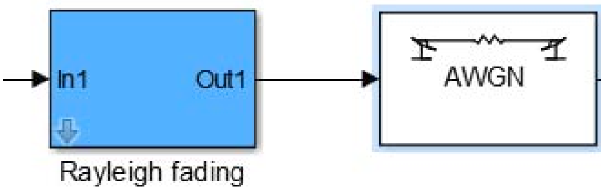

Fig. 4. Rayleigh fading and Additive White Gaussian Noise

Next, the signal is sent to decoders and then is compared to the original signal. Finally, demodulation errors are found.

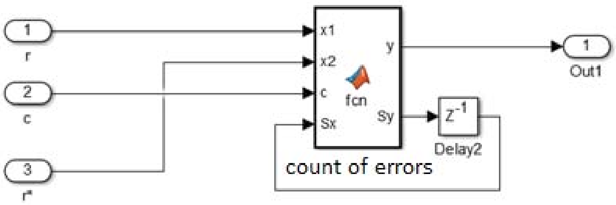

Fig. 5. Block for computing demodulation errors

This paper covers several techniques of decoder design for demodulating frequency-modulated signals.

IV. Decoder Based on Fast Fourier Transform

This technique is based on the fast Fourier transform (FFT). FFT is performed using the signal set of 128 samples. The 17th channel is the one corresponding to logical "0", while the 21st corresponds to logical "1". Signals in the channels are compared and then we carry out decision-making regarding the demodulated symbol. Decision-making is carried out in a separate block by comparing two values and finally we can conclude about the signal itself.

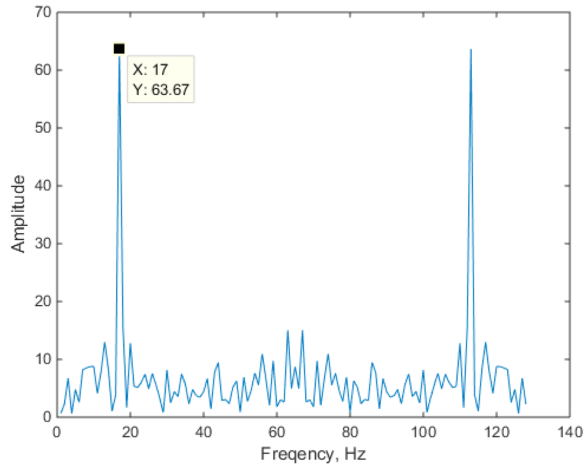

Fig. 6. 17th channel

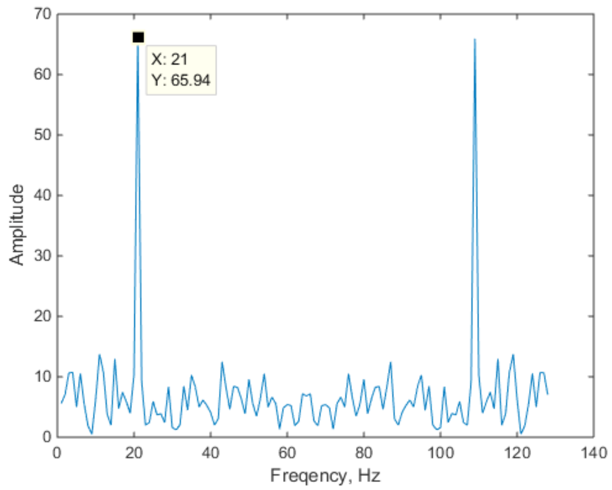

Fig. 7. 21st channel

V. Decoder Based on Correlation Function

Correlation function is used for optimal signal detection for any signal configuration. As we know, the correlation function of a sinusoid is also a sinusoid of the same frequency and this property can be used for harmonic signal filtering in a particular channel [4].

The cross-correlation function is calculated by the following formula:

$$R(\tau) = \sum_{i=1}^{N} x(i)y(i+\tau)$$

The autocorrelation function is calculated as

$$\Psi(\tau) = \sum_{i=1}^{N} x(i)x(i+\tau)$$

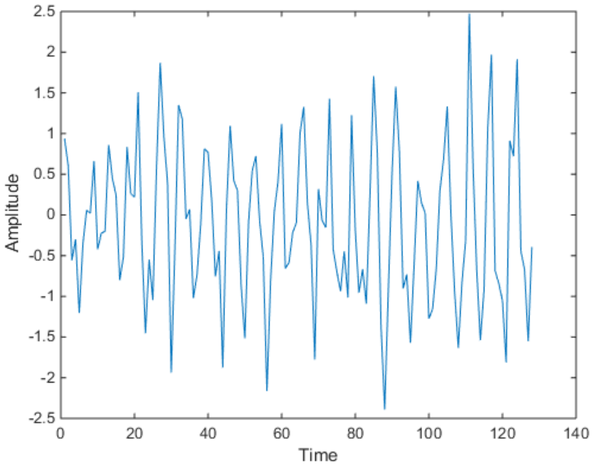

Fig. 8. Harmonic signal contaminated with noise

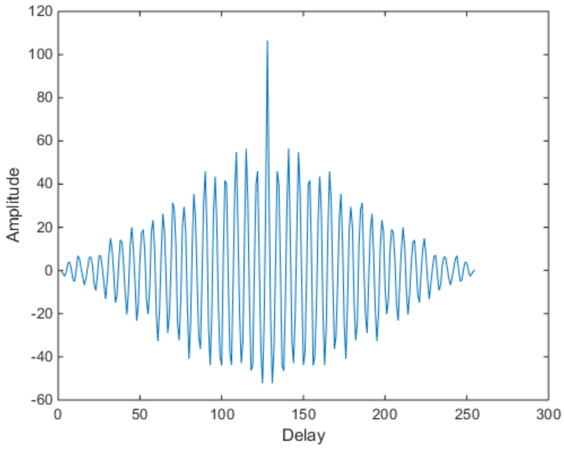

Fig. 9. Autocorrelation of harmonic signal with noise

Synchroimpulse chosen from the set of 128 symbols is used for computing an autocorrelation function. Then we compute the autocorrelation function with the frequencies corresponding to logical "0" and "1". The correlation levels of logical "0" and "1" are compared and a decision is made about a new symbol.

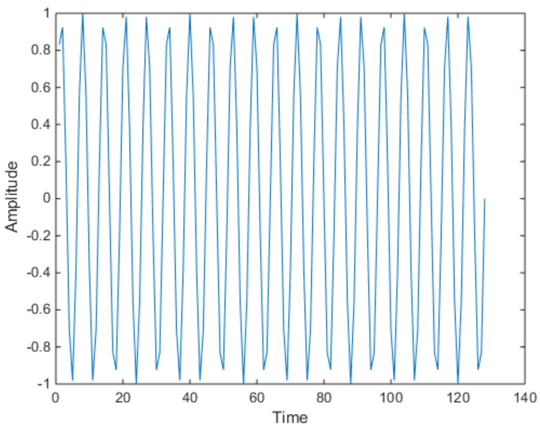

Fig. 10. Input harmonic signal (carrier frequency 2 kHz)

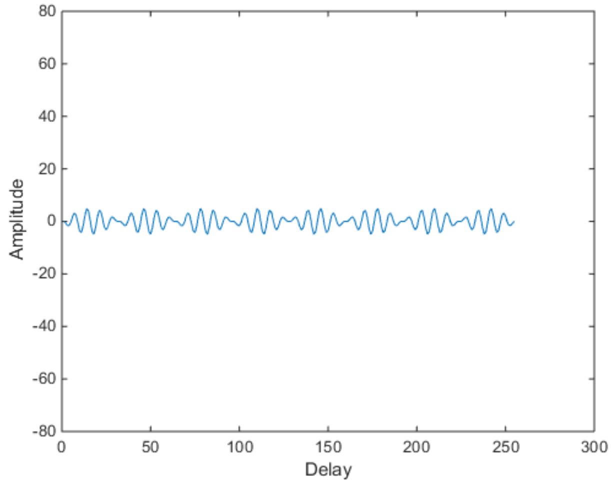

Fig. 11. Correlation of input signal (reference frequency 1.6 kHz)

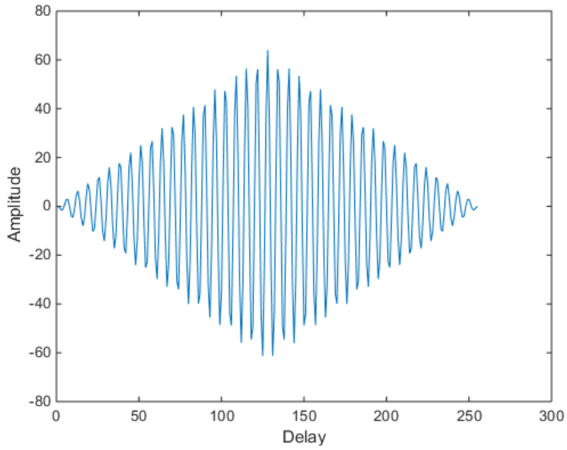

Fig. 12. Correlation of input signal (reference frequency 2 kHz)

VI. Decoder Based on Double Correlation

Calculation of the autocorrelation function is carried out twice and then we repeat all the operations from the previous section. The 2nd autocorrelation of the input signal helps extract the harmonic signal more distinctly than the 1st autocorrelation [5], [6]. Fig. 14-16 illustrate single, double, and triple autocorrelation of the input signal with SNR 0 dB (Fig. 13).

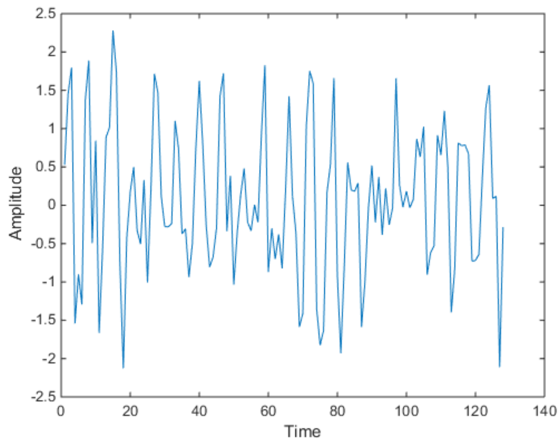

Fig. 13. Input harmonic signal with SNR 0 dB

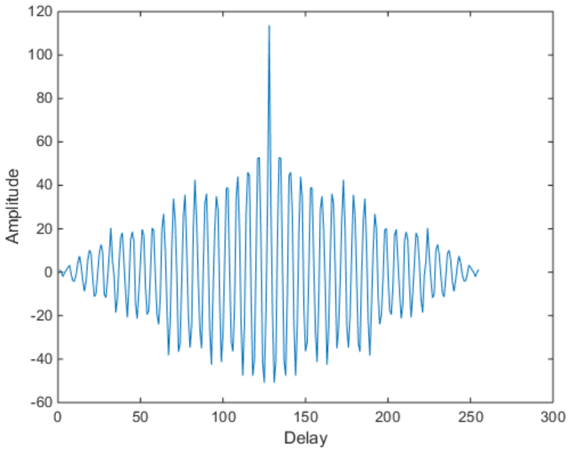

Fig. 14. Autocorrelation of harmonic signal with noise

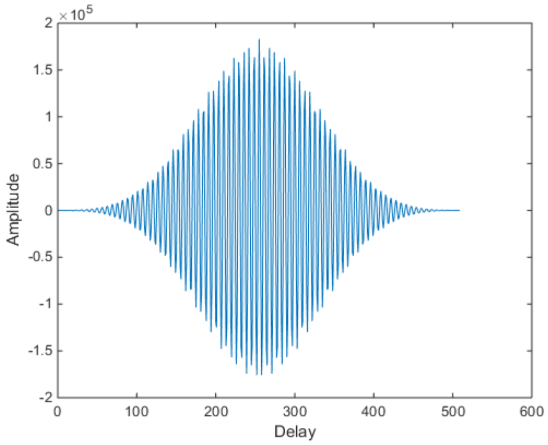

Fig. 15. Double autocorrelation of harmonic signal with noise

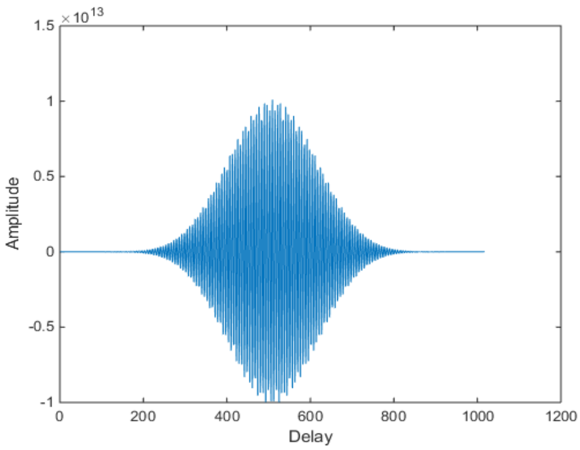

Fig. 15. Triple autocorrelation of harmonic signal with noise

VII. RESULTS

This section contains error rates for different demodulation techniques. The table was obtained for different signal-to-noise ratios. Overall, we simulated 100 sec. (9980 samples) of continuous pulses.

DEMODULATION ERRORS

E_s/N_0	**Demodulation technique**		
	FFT	***Correlation***	***Double correlation***
15	0	0	0
9	0.0091	0.0094	0.0085
8	0.0219	0.0212	0.0217
7	0.0428	0.0445	0.0427
6	0.0727	0.0732	0.0719
5	0.1067	0.1095	0.1072
4	0.1439	0.1470	0.1441
3	0.1817	0.1842	0.1817
2	0.2238	0.2262	0.2249
1	0.2634	0.2642	0.2620
0	0.2971	0.3022	0.2983

IV. CONCLUSIONS

The results provided in Table I show that the best results are provided by the techniques with double modulation (for SNR exceeding 5 dB) and FFT-based technique (for SNR lower than 5 dB)

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