HYDROACOUSTICS
4, 261-264 (2001)

CODE SIGNALS TRANSMISSION USING MFSK MODULATION IN SHALLOW WATERS

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The paper presents an acoustic system for the transmission of digital data in shallow coastal waters of which multi-path propagation and intense reverberations are characteristic. To ensure a small error probability of moderate speed transmission, the system uses corrective codes and a technique of incoherent modulation MFSK (Multi Frequency Shift Keying). To synthesise the transmission signal, estimate the spectrum of the signal received and code and decode the channels, a DSP processor was used. The paper includes the results of a computer simulation in which the system was put to work in the presence of Gaussian noise. The system was tested in shallow coastal waters and the results are discussed in the paper.

INTRODUCTION

Similarly to the majority of other telecommunications systems, the task of the underwater communications system presented in the paper is to send a lot of data error-free in the shortest possible time. The actual system, however, usually has some limitations to deal with such as the transmission channel and advancement of technology. In the case of a shallow underwater channel these limitations mostly come from multi-channel transmission (interference), acoustic wave dispersion at the limits of the medium and acoustic noise. Limitations of the equipment mainly come from the small bandwidth of ultrasonic transducers. To resolve these problems, analogue signals transmission usually uses SSB modulation while digital signals transmission uses FSK modulations and versions of it. In an effort to maximally shorten the transmission time, the system in question uses MFSK modulation [1], [2]. To reduce the number of errors it uses Hamming code with one error correction [3].

1. SIGNAL STRUCTURE AND SYSTEM DESCRIPTION

The transmitted signal consists of a sinusoidal pilot signal of constant frequency f_p and a sequence of signals containing a combination of 7 sinusoidal signals of frequencies $f_1, f_2, ..., f_7$ (4 information bits, 3 corrective bits). In between the pilot signal and the successive signals are intervals of a fixed duration t_i, when no signals are emitted (Fig.1).

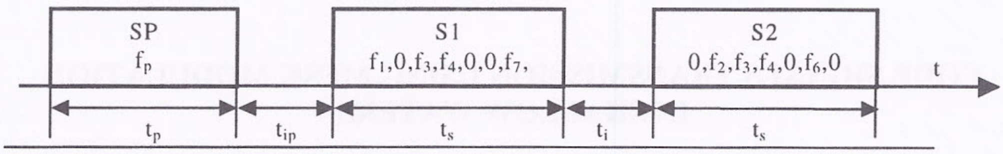

Fig.1. Signal structure (SP – pilot signal, S1, S2,...- information signals).

The signals being transmitted undergo digital synthesis in the transmitter's DSP processor to be later transmitted to a DAC digital-to-analogue converter, PA line power amplifier and UT ultrasonic omni-directional transducer (Fig. 2). In the receiver the pilot signal's listening channel is in constant operation. It consists of an AGC automatic gain control amplifier, BF narrow band matched filter, Q square system, I integration system of integration duration t_p and a T threshold system. AGC and the threshold system's parameters are such that the signal from the integration system will exceed the threshold when the level of the pilot signal received is greater by a specific value from the spectral level of the acoustic noise in the system's operating band. The listening channel of the pilot signal is made using energy-saving analogue circuits to minimise power consumption from the power source. The other digital systems are powered only within a short time after the pilot signal detection.

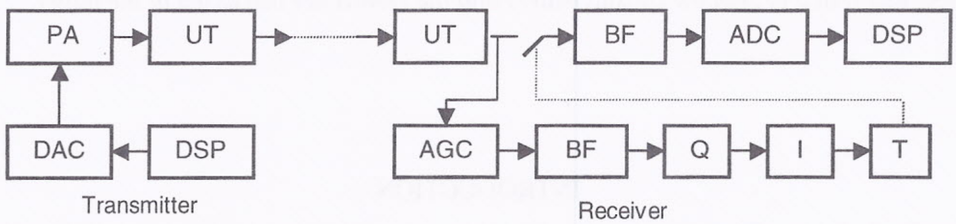

Fig. 2. Block diagram of the underwater communications system.

When the threshold is exceeded after time t_i the information signals reception channel is switched on. Following amplification and filtration in a filter whose bandwidth is $B \cong (f_1+f_2+...+f_7)/7$, the information signal is sampled in time t_s and converted into a digital signal at the ADC. The DSP processor computes the signal's discrete Fourier transform and the estimate of its PSD. The computer scans the frequency bands to see where the PSD lines exceed the given threshold. It assigns a logical "1" to those bands and a logical "0" to the others. The resulting binary sequence undergoes the error correction procedure based on Hamming code principle. The other elements of the received signal are processed in the same way.

The output signal to noise ratio in the pilot signal's channel is proportional to integration time t_p and inversely proportional to the transmission bandwidth of the BF filter. The band has to be wide enough to allow for the expected frequency changes of the pilot signal caused by Doppler effect. To obtain a high signal to noise ratio long pulses and long integration times have to be applied.

The application of Fourier transformation for the detection of the information signal is equal to the use of a matched receiver. The output signal to noise ratio is equal to the quotient of the received signal's energy and power spectrum density of the noise. With equal times t_p and t_s it is greater than the pilot signal to noise ratio. As a result, it is possible to shorten the information signals which speeds up the transmission of information in the system. In practice the time t_s and t_i are selected to match the level of reverberation and the degree to which pulses caused by multi-channel transmission are extended.

2. RESULTS OF THE SIMULATIONS AND MEASUREMENTS

Fig.3 shows the result of a digital simulation of how the system operates. The information signal is transmitted periodically and consists of segments of varying envelope whose shape depends on the code being transmitted (Fig. 3a). The signal's spectrum contains lines that match the code's "1" only (Fig. 3b). With a low signal to noise ratio at the receiver input in the BF filter band, the information signal is completely masked with noise (Fig. 3c). The code, however, can be read from the spectrum (Fig. 3d).

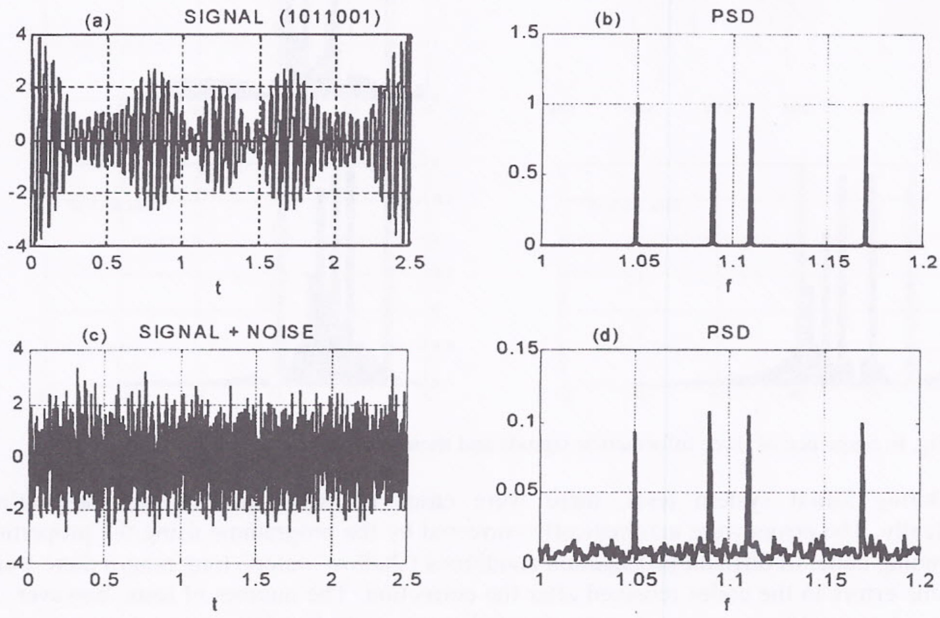

Fig.3. Results of system simulation.

The actual transmitted and received signals are shown in Fig. 4 and Fig. 5. Fig. 4 shows the final fragment of the pilot signal and one information pulse, Fig. 5 shows two information pulses with different codes.

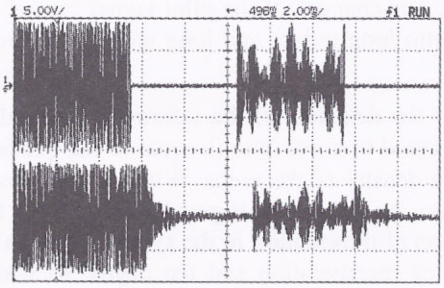

Fig.4 The pilot and information signals.

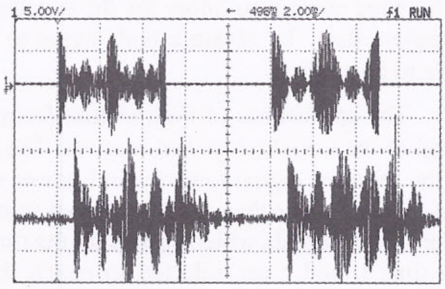

Fig.5. Sequence of two information signals.

Fig. 6 shows a received information signal containing three different codes. The pulses have a minimal duration and minimal spacing. When the detection threshold is set at 1, the spectral lines are a faithful representation of the codes transmitted.

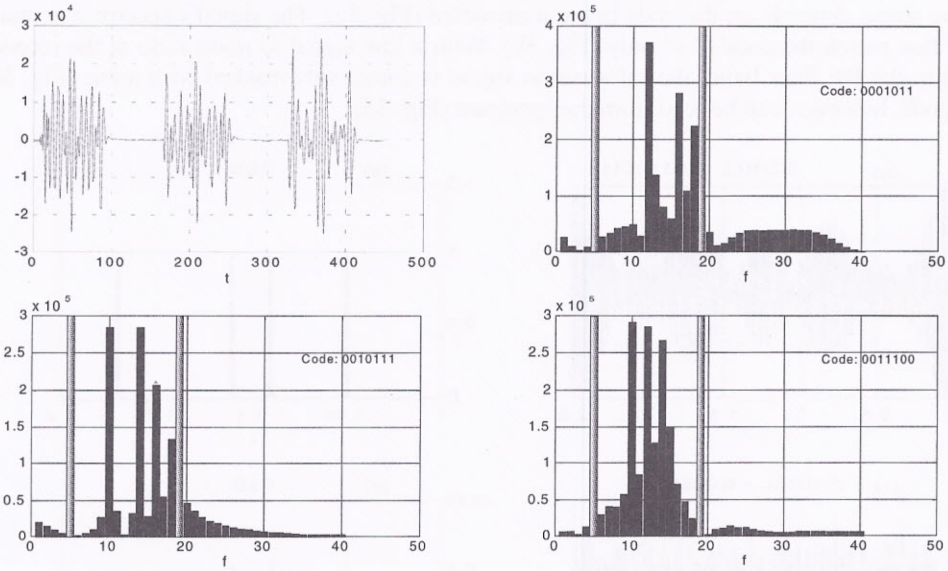

Fig. 6. Sequence of three information signals and their spectrums.

During initial system tests, there were cases of false spectral lines appearing sporadically. The errors were automatically corrected by the programme using the properties of Hamming code. In difficult propagation conditions (shallow waters, long range) there were also some errors in the codes received after the correction. The number of tests, however, is insufficient to enable a statistical estimate of the error rate for different system operating conditions.

REFERENCES

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