ACOUSTIC COMMUNICATION SYSTEM WITH BROADBAND PHASEMANIPULATED SIGNALS.

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1. INTRODUCTION

Autonomous bottom station and free floating buoys are the important devices for the effective ocean monitoring. The oceanological stations can provide independent measurements from the ocean bottom, submerge to the given depth, change the horizons and scan across. The control of the stations is performed either automatically or by means of a acoustic remote systems. The efficiency of the stations is greatly increased by means of telemetry and remote control systems. To improve noise stability of acoustic communication systems in the case of multichannel signal propagation broadband phasemanipulated signals are used. Adaptive systems with broadband signals operate with minimum signal intensity to spectral noise density ratio. The low data rate and complicated algorithms of signal processing are the limitations of these systems.

In this article the results of testing of the acoustic digital communication system with the broadband phasemanipulated signals and noise immunity code are considered. The communication system was successfully used for telemetry and remote control of autonomous oceanological stations and free floating buoys.

2. BROADBAND SIGNAL CODE CONSTRUCTION

The broadband signal code construction [1,2] is built on the base of the matrix 32x31 with the binary digital elements. The zero line of this matrix is a 31 symbol code of M-sequence. Each next line is the sum (on modulus two) of the zero line and the every possible cyclic displaces of the M-sequence code with the another raising polinom with the same length (31 symbols). Such matrix is one of the Gold code adjacent classes. The pseudorandom signal ensemble has been formed by phase manipulation of the CW signals on π in the discrete time moments at the intervals 2 msec in accordance with the elements of the matrix. The line with number i is corresponding to information code i $(0 < i < 32)$.

ACOUSTIC COMMUNICATION SYSTEM

The noise immunity code (the modification of Gilbert code [3]) protects the 5 binary symbols batch. The initial binary information has been written in form of the matrix 4x6 and supplemented with the checking symbols. The dimension of the resulting matrix is 5x7. This matrix is reading obliquely by slanting lines:

$a_{00}, a_{11}, \ldots, a_{44}$; $a_{01}, a_{12}, \ldots, a_{45}$; $a_{02}, a_{13}, \ldots, a_{46}$;

$a_{03}, a_{14}, \ldots, a_{40}$; $a_{04}, a_{15}, \ldots, a_{41}$; $a_{05}, a_{16}, \ldots, a_{42}$;

$a_{06}, a_{10}, \ldots, a_{43}$.

Each slanting line is corresponding to one of pseudonoise signals, transmitted through the channel. Each code combination carries 24 information and 11 checking symbols.

During of the decoding process the received information is written in the form of matrix 5x7 in the same order, using slanting lines. Then checking symbols on lines and columns are calculated. The decoder makes the cyclic displaces of the initial information in the groups of 5 symbols until the line checking symbols disagrees with the first five column checking symbols. At the case of coincidence the errors will have position in the first group of five symbols and can be corrected by summing with the checking symbols. Such code can correct five error symbols batch after putting out of one of the 32 pseudonoise signals.

The signal sequence with central frequency f_o = 8500 Hz contains three parts: CW signal, 294 msec; seven periods of M-sequence with period T = 32x2 msec, full duration 434 msec; seven pseudonoise signals with duration T, 434 msec.

The first part of this sequence is used for the signal detection and fast estimation of doppler frequency shift, the second part is used for the adaptation to the channel impulse response, and the third part is used for the data transmission.

3. SIGNAL PROCESSING ALGORITHM

The signal processing algorithm contains three stages: detection, adaptation, demodulation and decoding. The received

ACOUSTIC COMMUNICATION SYSTEM

signal with the bandwidth $2\Delta f$ = 1 kHz is transformed into the sequence of the pairs simultaneous samples from the quadrature components, taking at intervals $\tau_d=(2\Delta f)^{-1}$.

The detection of the signal sequence beginning is done with the help of fast Fourier transform (FFT). The samples x_i from the detected signals sequence are written into the memory of the microprocessor system. Doppler frequency shift and corresponding time scale variations, caused by the displacements of source and receiver of sound, are compensated by the frequency removing on the value of Doppler shift with the opposite sign and by the sampling interval changing by means of linear interpolation.

The digital matched filtration of periodical M-sequence is realized with the help of samples transposition and fast Walsh transform (FWT), using combinatorial equivalence of M-sequence and Hadamard code. The accumulation of impulse channel response estimation is realized taking into account the variations of the doppler shift due to acceleration of source and receiver of sound. One of the acceleration source is the jerks of the hydrophone owing to the ship tossing. This algorithm can be presented in the form:

$$\theta(kT,\ l\tau_d,\ m\Delta f_d) =$$

$$\left|\sum_{i=0}^{62}(x(i\tau_d-l\tau_d-kT)+x(i\tau_d-l\tau_d-kT-\tau_d))s(i\tau_d)\exp(-j2\pi m\Delta f_d i\tau_d)\right|^2,$$

$$Z(l\tau_d) = \sum_{k=0}^{7}\theta(kT,l\tau_d,m(k)\Delta f_d),$$

where $\theta(kT,l\tau_d,m\Delta f_d)$ - the scattering function estimation on the interval $[kT,(k+1)T]$ with frequency doppler shift $m\Delta f_d$ and delay $l\tau_d$; Δf_d - discrete of doppler frequency shift equal 0.25/T.

Sequence m(k) is Markov chain with five states in accordance to doppler frequency shift m(k) = {-2, -1, 0, 1, 2} (fig.1). The transition to each next state from the last one is possible only if the last number differ from the next number no more then one unit. The time sequence of this transitions is described by lattice (fig.1). When the nocoherent accumulating of scattering function $\theta(\Delta\tau,\Delta f)$, the moving through the lattice occurs, and when the different paths coincide, the path with maximum value $M = \max_{\langle m(k)\rangle} Z(l\tau_d)$ is selected (algorithm Viterbi [4]).

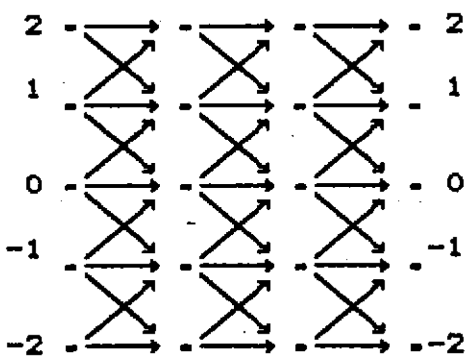

Fig. 1 The Markov chain lattice.

This algorithm takes into account the variations of doppler shift of frequency owing to acceleration of sound source or receiver $a \le \Delta f_d c/(f_o T) = 11$ m/sec^2, c - sound velocity.

Three estimations of delays $l\tau_d$ with maximum values $Z(l\tau_d)$ are

used for taking down manipulation by M-sequence code from zero line of matrix A. Then intercorrelations of the pseudonoise signals with the signals, arriving with each of these delays, are defined with the help of transposition of samples, fast Walsh transform, and transposition of results. The amplitude squares of the correlations are summing with the coefficients, which are proportional to the ray signal intensities. So the paths metrics are calculated for each of 5 frequencies, corresponding to the different states of Markov chain. The decision is corresponding to maximum metric R of every possible lattice paths, when different paths coincide, the path with maximum R(j) was selected.

ACOUSTIC COMMUNICATION SYSTEM

$$R(j) = \max_{e(j)} \sum_{k=0}^{j} r(m(k), e(k)),$$

$$r(m,e) =$$

$$\sum_{l} a_l^2 \left| \sum_{i=0}^{62} (x(\tau_d(i-1)-kT)+x(\tau_d(i-l-1)-kT))s(e,i\tau_d)\exp(-j2\pi m\Delta f_d i\tau_d) \right|^2,$$

$$R(n) = \sum_{k=0}^{n} r(m(k), e(k)).$$

$$\max(m(0),\ldots,m(n));\ e(0),\ldots,e(n).$$

The multiciphered symbols from the alphabet 2^5 are decoded by described method.

4. EXPERIMENTAL TECHNIQUES

The four identical autonomous bottom stations with possibility of information exchanging by means of acoustic communication system between them and the ship took part in the experiment. The five transceivers had identical construction. One of them was placed on the scientific ship, others on the autonomous stations. Only one of the experiments is described here.

Each transceiver consisted of microprocessor , power key amplifier, acoustic source, hydrophone and receiving amplifier with clipping. The receiving signals after selective amplifier and clipping came into microprocessor system. The quadrature components were selected with the help of coincidence schemes with sinusoidal and cosinusoidal oscillations and counters. The adaptive signal processing was realized in the form of microprocessing program. The microprocessor of low consumption was realized on the chip V40 (NEC) with the command system i8086.

5. TESTING CONDITIONS

The communication system was tested in the Atlantic ocean in august - september 1991 year during 8-th tour of scientific ship "Akademik Sergey Vavilov" in the region of underwater mountain Meteor.

The diving depth of autonomous station was 895 m. The ship drifted from the distance 10 km in the direction on the station.

ACOUSTIC COMMUNICATION SYSTEM

The submerging depth of the receiving hydrophone was 100 m; the wind velocity was 12 knots; the ship drift was 1.1 knots; the sea state was 5 balls on the Beafast scale.

The high negative gradient of the sound velocity near the sea surface created a shadow for water rays on the depth 100 m at the distance 10 km. The acoustic system of the drifting ship crossed the caustic from shadow zone to the region with water and surface ray. During the error probability analysis the inclined distance from the ship to the station changed from 8967 m to 8580 m.

The acoustic communication system was used for the control of all station working regimes. Once a minute the station transmitted a digital information. The checking of the receiving information was done simultaneously with writing of quadrature components of signal together with support signal on the tape recorder. In the laboratory these records had been recalled and introduced into computer, where independent noise had been summed to the signal.

The signals had been processed in the computer with the different noise dispersion by the same program as the real time processing program.

6. RESULTS OF EXPERIMENT

The dependence $P_e(S/N)$ of error probability from signal to noise power ratio (in the bandwidth of signal) is shown on fig. 2. The curve has been obtained for 2×10^{4} information bits. On this fig. one can see also the dependence of error probability from the signal to noise ratio for the ideal system in the computer model of one-ray channel without fluctuations of signal amplitude. The same program has been used for the signal processing in this model. The energy losses nearly 3 dB ($P_e(S/N) = 10^{-2}$) are caused by the amplitude fluctuations and the instrument errors. The system operate with ratio of total received signal energy to noise spectral density per bit of transmitting data - 12 dB and 55 bit/sec data rate. The analogous system with phasemanipulated recurrent complex signals operated with signal to noise ratio per one infomation bit 9 dB at 12 bit/sec data rate in the underwater sound channel [5].

ACOUSTIC COMMUNICATION SYSTEM

Simultaneously with the digital information transmission the system allows to define the signal propagation delay on every ray with the precision 1 ms, and then calculates the distance to the station taking into account the measured hydrology. It is also possible to measure the difference in the timers values of the autonomous bottom stations by means of transmitting this values from the stations after calling from the ship. It has been defined by the experience that the precision of such measurement is about 1 ms (the distance was 10 km, the drift was 1 knot). The communication system was successfully used as a part of the acoustic long base navigation system with autonomous bottom station. The navigation system have measured the signals delays under the water and then have transmitted them from the bottom station to the ship. Time by time the bottom stations timers were corrected by means of the communication system.

This autonomous bottom stations with pseudonoise phasemanipulated signals were used also for the tomography of the internal waves near the underwater mountain, and for testing communication system between the bottom vehicles using sound refraction on the sea surface.

7. REFERENCES

[1] A K MOROZOV & V A STEPIN, 'Characteristics of broadband cascade construction in the multipath channel with Rice fading', Radiotekhnika i electronika., Moscow, 1992, t.37, v.8, pp. 1450-1456.

[2] A K MOROZOV, 'The upper border for reception error probability of signals on the base of blocks and recurrent codes', Izvestiya vysshikh uchebnykh zavedenii seriya radioelectronika., Kiev, 1990, t.33, n.1, pp. 48-52.

[3] A K MOROZOV, 'Signals reception in multipath channel with the help of Viterbi algorithm', Radiotekhnika i electronika., Moscow, 1989, t.34, v.4, pp. 767-773.

[4] M BLAUM, P G FARREL & H C A VAN TILBORG, 'A class of burst error-correcting array codes', IEEE Trans. on Inform. Theory, vol. IT-32, No 6, 1986, pp. 836-839.

[5] A K MOROZOV, B B SEMENOVITCH & A N KOLMOGOROV,'A system for transfer of oceanological information from research buoy station', Okeanologiya., Moscow, 1987, t.27, v.3, pp. 525-531.

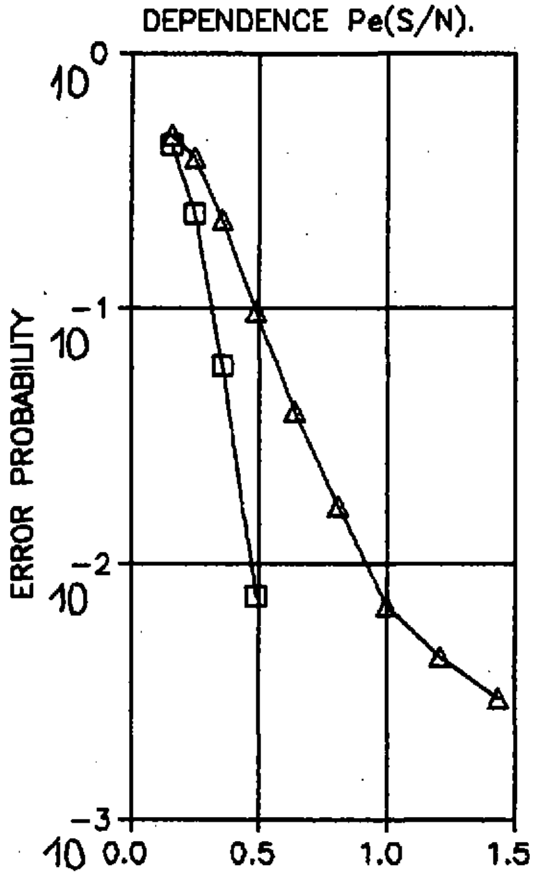

FIG.2 S/N RATIO

□ model   Δ experimet