DIRECTIONAL SONOBUOY SYSTEM FOR DETECTION OF SUBMARINES

ROMAN SALAMON

Gdansk University of Technology,
Faculty of Electronics, Telecommunications and Informatics,
Department of Marine Electronics Systems
G. Narutowicza 11/12, 80-952 Gdańsk
roman.salamon@eti.pg.gda.pl

This article outlines the methods of detecting and locating submarines. Special attention is paid to the airborne system utilizing directional sonobuoys. The article discusses operating principles of such systems, a method of detecting and estimating the approach direction of the acoustic wave emitted by the vessel, and practical visualization methods. The article is related to the successful development of a Polish prototype of a directional sonobuoy acoustic system processor.

INTRODUCTION

Anti-submarine warfare is among the most important tasks of the navy. The first and indispensable stage is to detect a submarine, the second stage is to determine its position, and the third stage – to classify or identify it. Submarines are detected, located and classified mainly by hydroacoustic methods which are presently regarded as the most effective. Hydroacoustic submarine detection methods are usually divided into two groups: active and passive. Active methods use the acoustic signal echo reflected on the submarine, while passive methods use acoustic signals emitted by the submarine. The advantage of active methods rests in the possibility of detecting submarines which do not emit any acoustic signals (e.g. when the submarine is not moving) or emit very weak signals (such as the so-called quiet submarines). The main disadvantage is the need to emit sounding signals which reveal the presence of the enemy echo ranging system. Passive methods use acoustic signals emitted by the submarine, which is an obvious drawback; however, they do not reveal the presence of the system. With regard to the complementary advantages of both methods, they are usually combined in submarine detection.

1. GENERAL CHARACTERISTICS OF PASSIVE SYSTEMS FOR SUBMARINE DETECTION

Hydroacoustic submarine detection systems are installed on board of ships, helicopters and airplanes, or used as stationary versions, which is especially true for passive systems. The methods and the technical solutions to be adopted are largely determined by the location of devices. Generally, the larger the distance between the system carrier and the acoustic array, the easier the detection of a submarine owing to the lower level of noise emitted by the carrier. Therefore, airborne systems are preferred, where the very large acoustic wave reflection coefficient of the air/water interface serves as an additional barrier for introducing noise to water. Another important advantage of airborne systems is their greater mobility compared to vessel mounted systems.

In most systems the hydroacoustic arrays are connected to the remaining part of the system with electric wires implying permanent, mechanical connections with the system carrier. In such solutions, transducers are required to move at the same velocity as the submarine, airplane or helicopter. This is virtually impossible for airplanes because transducers cannot be kept submerged at velocities reaching hundreds of kilometers per hour. Moreover, acoustic noises caused by turbulences around the rapidly moving array would completely disturb the reception of signals emitted by submarines. Therefore, the only practical solution is to abandon the permanent connection of transducers with the remaining part of the hydroacoustic system and replace it with a radio connection. Airborne submarine detection systems are thus necessarily radiohydroacoustic systems.

The problem of a mechanical connection between the array and the on-deck part of the system is not that big for helicopters. A hydroacoustic array can be lowered from a hovering helicopter, but such an operation is quite risky, and the array must be small. Therefore, systems with lowered arrays are usually active high-frequency systems which ensure satisfactory directivity despite their small size. Helicopters, as well as airplanes, are also used as radiohydroacoustic system carriers.

Radiohydroacoustic systems for detecting, locating and classifying submarines utilize acoustic waves which propagate well under water. Acoustic signals received by submerged hydrophones are transmitted by a radio link to the airborne part of the system. There are two types of radiohydroacoustic systems: active or passive. Mobile passive systems are most popular, in which hydrophones and related electronic circuits are installed on floating buoys transported by airplanes or helicopters and dropped onto the water in regions where submarines are suspected to be operating. Sometimes the buoys are equipped with sounding signal transmitters – such systems are active systems. Usually, active searching merely supplements the basic passive searching function. The mobility of the system is achieved by the possibility of throwing the buoy to any sea region. Since the buoy is used as a carrier for ultrasonic transducers (hydrophones), such systems are called radiohydrobuoy systems.

The system consists of two basic parts: the radiohydrobuoy unit and the airborne part. Radiohydrobuoys are manufactured in three basic versions:

- Omnidirectional Passive Sonobuoy -LOFAR
- Directional Passive Sonobuoy –DIFAR
- Command Active Multibeam Sonobuoy – CAMBS

LOFAR sonobuoys are equipped with a single omnidirectional hydrophone or a vertical array of such hydrophones. This layout makes it virtually impossible to determine the approach direction of the wave emitted by the submarine. Therefore, the functionality of systems utilizing such sonobuoys is limited to detecting submarines and, conditions allowing, to determine their region of presence. This region is determined based on the level of the signal emitted by the submarine: if a signal received by a sonobuoy is stronger than the signals received by the other sonobuoys, then the submarine is probably near the first one. Due to the low precision of this method, LOFAR sonobuoys are rarely used by the navies of technologically developed countries.

The basic submarine detection devices used by passive airborne systems are DIFAR sonobuoys. These sonobuoys incorporate directional arrays with crossed pairs of piezoceramic transducers and an omnidirectional reference transducer. Despite their very small size relative to the length of the received acoustic waves, they allow to determine the wave approach direction with sufficient accuracy. In the remaining part of the article, we will present the construction of a DIFAR sonobuoy system, methods of processing and displaying the received signals, and tactical applications in submarine detection.

CAMBS type sonobuoys are generally used as complementary to passive directional sonobuoy systems. Their operating principle is similar to that of active multibeam sonars. They are single-use devices, thrown into water from airplanes or helicopters. To provide good directivity, they incorporate rather large unfolding multielement arrays. With regard to the much higher cost compared to DIFAR sonobuoys, they are mainly used after detecting a submarine by a passive method to determine its position more precisely.

2. GENERAL DESCRIPTION OF A SYSTEM WITH PASSIVE DIRECTIONAL SONOBUOYS

A passive sonobuoy system consists of two basic parts: an airborne part and a set of buoys. The airborne part incorporates a radio signal receiver with an antenna, an acoustic processor, and a visualization unit. The set of sonobuoys consists of several or several dozen identical DIFAR sonobuoys placed in special launchers installed in the airplane or helicopter. Upon arrival at the submarine search area, the airplane or the helicopter drops the sonobuoys onto predetermined locations of the sea region so that they form a certain geometric figure, such as a circle or a line. The sonobuoys fall on parachutes, and automatically unfold after hitting the water surface. A radio antenna remains above the surface supported by a floating container which is filled with air when the sonobuoy hits the water. The body housing electronic circuits and power supply is submerged. A hydroacoustic array descends from the body to a predefined depth with a built-in acoustic signal receiver. When the sonobuoy unfolds, electronic circuits switch on automatically and the sonobuoy begins to monitor acoustic signals. These signals modulate the radio signal which is transmitted by the antenna and is then received by the airborne part. The receiver demodulates the signal and transfers the acoustic signals to the acoustic processor which scans them for any signals emitted by a submarine in an attempt to estimate the arrival direction of the waves.

After dropping on the water, the sonobuoys can operate up to 6 hours; this time can be preset in 1 hour steps using a dial mounted on the body. When the preset time elapses, the sonobuoy sinks and electronic circuits are automatically destroyed.

The construction of a radiohydrobuoy system with DIFAR sonobuoys is shown in Figure1.

The method of determining the position of the detected submarine is shown in Figure 2. The presumed position of the submarine is shown as the circle around the intersections of bearing lines set out by respective sonobuoys.

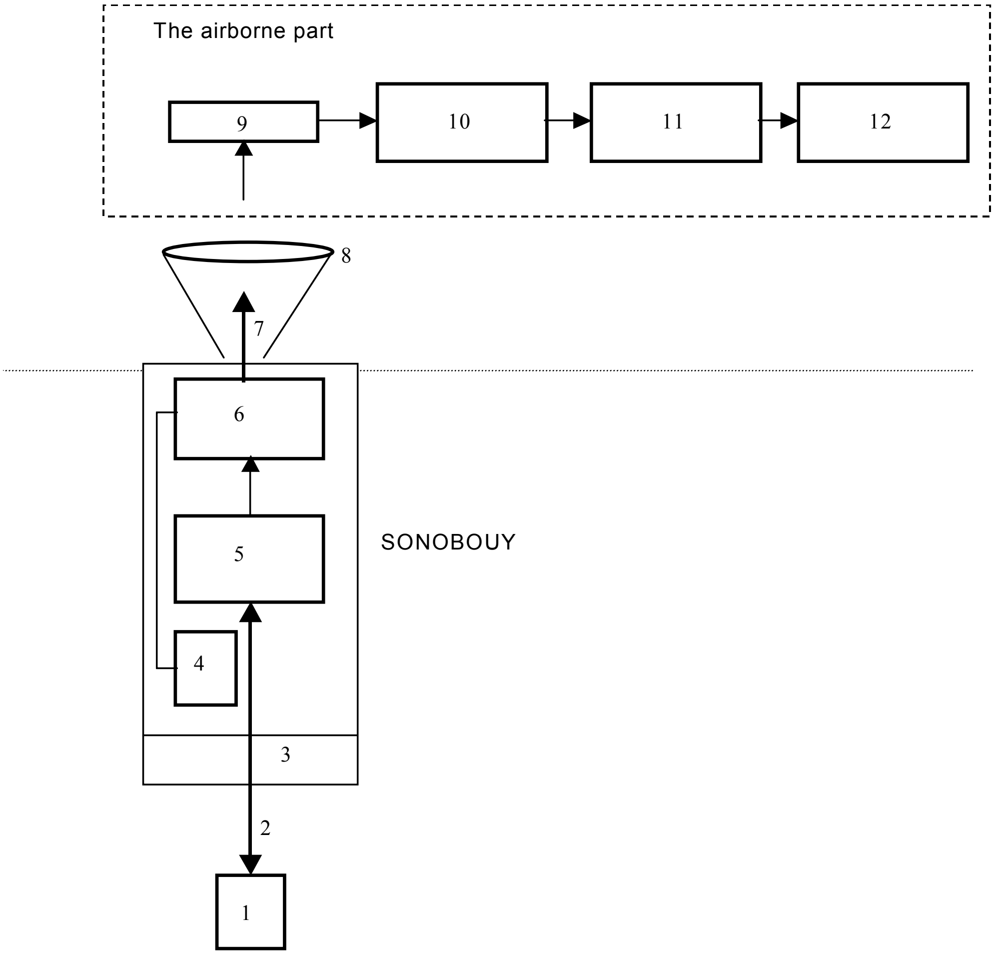

Fig. 1. Block diagram of a passive radiohydrobuoy system
(1- acoustic array unit, 2- umbilical, 3- umbilical container, 4- PSU, 5 – modulator, 6 - radio transmitter, 7- radio transmitter antenna, 8- parachute, 9- radio receiver antenna, 10- radio receiver, 11- acoustic processor, 12- display unit).

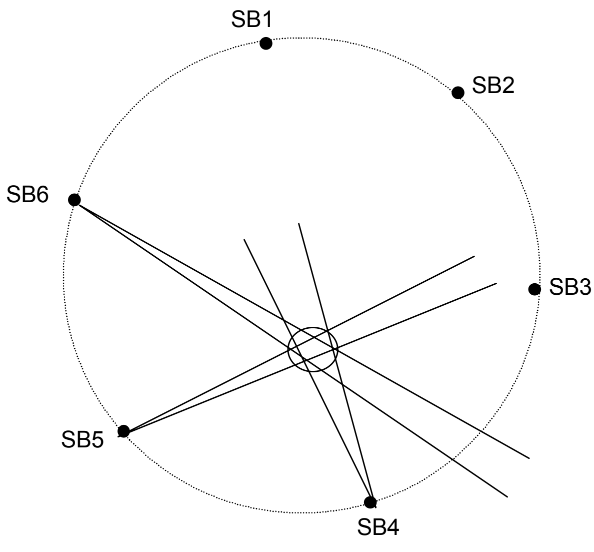

Fig. 2. The method of determining the position of the detected submarine

3. THE METHOD OF DETECTING AND ESTIMATING WAVE APPROACH DIRECTION

The radiohydrobuoy is equipped with a directional set of gradient type hydrophones. This set consists of five ultrasonic transducers positioned as shown in Figure 3.

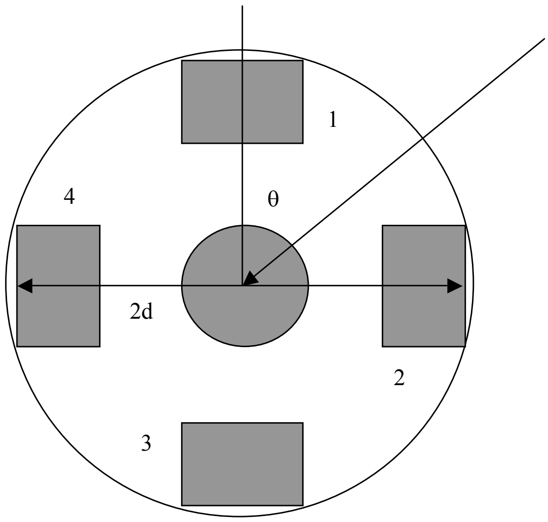

Fig. 3. The diagram of the directional set of hydrophones

If an acoustic wave reaches the array at the angle θ, then signals received by respective hydrophones can be expressed as:

$$\begin{aligned} s(t) &= u(t) + n(t) \\ s_1(t) &= u(t+\tau_c) + n_1(t) \\ s_2(t) &= u(t+\tau_s) + n_2(t) \\ s_3(t) &= u(t-\tau_c) + n_3(t) \\ s_4(t) &= u(t-\tau_s) + n_4(t) \end{aligned} \qquad (1)$$

where *u(t)* is the signal on the output of the central hydrophone, $n(t),\ldots,n_4(t)$ are noises on the outputs of respective hydrophones, $\tau_c = (d/c)cos\theta$ and $\tau_s = (d/c)sin\theta$ are delays, and *c* is the acoustic wave velocity in water.

The signals from hydrophones located on the perimeter of the array are subtracted in pairs which leads to the following results:

$$\begin{aligned} s_c(t) &= s_1(t) - s_3(t) \\ s_s(t) &= s_2(t) - s_4(t) \end{aligned} \qquad (2)$$

The differential signals and the *s(t)* signal from the central hydrophone modulate carrier signals thus creating a composite signal described later in the article. The composite signal modulates the radio signal transmitted to the airborne receiver. The signal is demodulated in the receiver, and the retrieved composite signal is input to the acoustic processor. The processor reproduces the reference signal *s(t)* and two differential signals, $s_c(t)$ and $s_s(t)$. Then, the processor calculates the spectra of these signals, which have the following form:

$$\begin{aligned} S(\omega) &= \Im\{s(t)\} = U(\omega) + N(\omega) \\ &, \\ S_c(\omega) &= \Im\{s_c(t)\} = U(\omega)(e^{j\omega\tau_c} - e^{-j\omega\tau_c}) + N_c(\omega), \end{aligned} \qquad (3)$$

$$S_s(\omega) = \Im\{s_s(t)\} \cong U(\omega)(e^{j\omega\tau_s} - e^{-j\omega\tau_s}) + N_s(\omega),$$

where $\omega = 2\pi f$ is the pulsation of a signal with frequency f, and $U(\omega) = \Im\{u(t)\}$, $N_c(\omega) = \Im\{n_1(t) - n_3(t)\}$, $N_s(\omega) = \Im\{n_2(t) - n_4(t)\}$, $N(\omega) = \Im\{n_1(t) + ... + n_4(t)\}$ are Fourier transforms of respective signals and noises.

Since the length of received acoustic waves λ is always much larger than distance d between hydrophone surfaces, we have $\omega\tau_c = 2\pi(d/\lambda)cos\theta << 1$. The same inequality is valid for delay τ_s. Using theses inequalities, formulas (3) can be simplified to the following forms:

- $$S(\omega) = U(\omega) + N(\omega),$$

$$S_c(\omega) \cong 2jU(\omega)sin(\omega\tau_c) + N_c(\omega) \cong 4j\pi(d/\lambda)U(\omega)cos\theta + N_c(\omega), \quad (4)$$

$$S_s(\omega) \cong 2jU(\omega)sin(\omega\tau_s) + N_s(\omega) \cong 4j\pi(d/\lambda)U(\omega)sin\theta + N_s(\omega).$$

The useful signal is detected by computing and displaying the periodogram which is an estimate of the power density spectrum, [2]:

$$P(\omega) = |S(\omega)|^2 \quad (5)$$

In numerical calculations, the periodogram $P(k)$ is expressed by the relationship, [3]:

$$P(k) = \frac{1}{N^2}\left|\sum_{n=0}^{N-1} s(n)e^{-j2\pi nk/N}\right|^2 \quad (6)$$

where N is the number of signal samples, and k is the number of periodogram line. The frequency of k line equals $f_k = k/T$, where T is the signal observation period containing N samples.

The main source of acoustic waves emitted by a submarine is the propeller. The acoustic signal generated by the propeller is a periodic signal; the harmonic whose frequency is proportional to the rate of rotation and the number of blades dominates in its spectrum. Figure 4 shows the signal $s(t)$ and its periodogram assuming that the useful signal $u(t)$ is a square wave, and the noise $n(t)$ is a white Gaussian noise.

The signal to noise ratio is much better compared to the received signal $s(t)$. The signal to noise ratio increases with signal observation time, because the height of the useful signal spectrum is constant, and the noise variation is inversely proportional to the square of the observation time (i.e. it is proportional to $1/N^2$), [1]. Therefore, long observation times are used in passive systems, from 1s to 8s.

The wave approach direction is estimated using the following operations:

- $$Y(\omega) = Im[S_c(\omega) \cdot S^*(\omega)] \cong 4\pi\,|U(\omega)|^2\, cos\theta + Im[N_c(\omega)S^*(\omega)],$$

$$X(\omega) = Im[S_s(\omega) \cdot S^*(\omega)] \cong 4\pi\,|U(\omega)|^2\, sin\theta + Im[N_s(\omega)S^*(\omega)]. \quad (7)$$

By multiplying differential signal spectra by conjugate spectrum of the central hydrophone signal, two spectra are obtained whose lines are nearly proportional to the cosine and sine of the wave incidence angle. This proportionality is disturbed by noises whose spectrum lines of random value add to the lines of the useful signal spectrum.

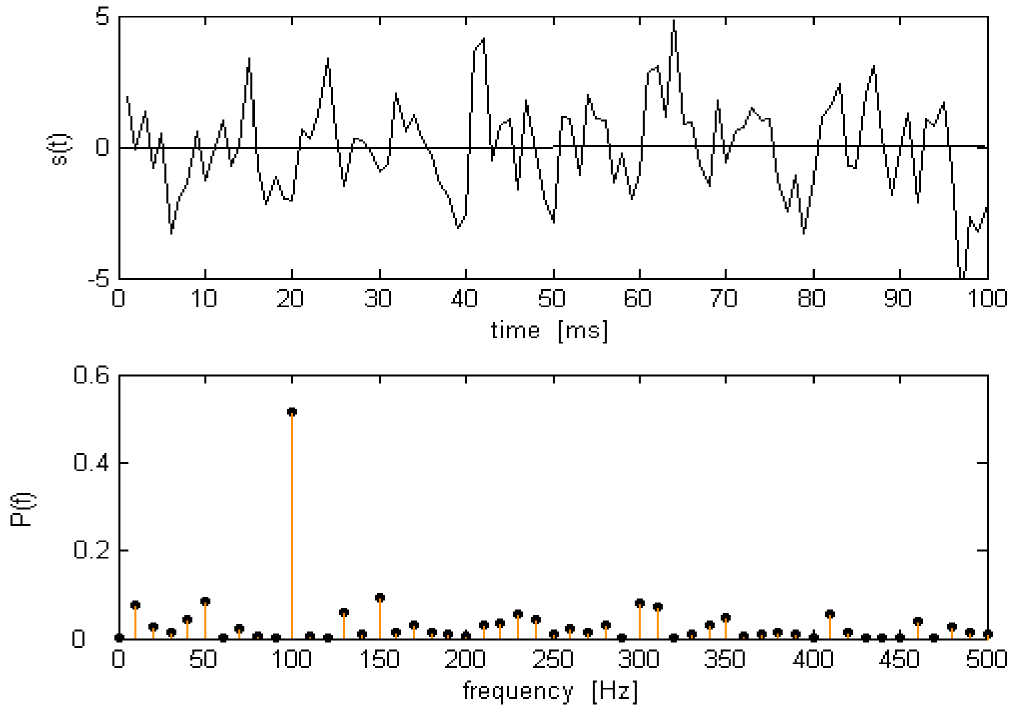

Fig. 4. The signal *s(t)* and its periodogram *P(f)* (signal amplitude = 1, useful signal period = 10 ms, standard deviation of noise = 1.5)

In numerical calculations, terms of sequences *X(k)* and *Y(k)* are treated as rectangular coordinates of points. For each frequency of the spectrum one point is determined, whose distance Z*(k)* to the origin of coordinates is $Z(k) = \sqrt{X^2(k) + Y^2(k)}$, and angle θ*(k)* satisfies the equation $sin[\theta(k)] = X(k)/Z(k)$. For large signal to noise ratios, the distance *Z(k)* is proportional to the power of the sine signal emitted by the vessel. The line linking such a point with the origin of coordinates shows the wave approach direction relative to the current position of the array (with error related to noises).

Figure 5 shows the results of computer simulation of a case where a sine signal is received arriving to the set of hydrophones at the angle $\theta = 45^0$. The received signal is biased with an uncorelated white noise, and the signal to noise ratio is 20 dB. 10 sequences were created from signal samples, each containing *M*=2048 samples.

Points which determine the wave approach direction gather around the point determined without noise. The dispersion of points, and thus the measurement error, is therefore caused by noises present in the system. Lines of noise spectrum are represented by points located around the origin of coordinates. At constant amplitude of the sine signal, the size of the area determined by these points decreases with increasing signal to noise ratio. Simultaneously, the scattering of points determining the wave approach direction is lower for larger signal to noise ratios.

The system allows to determine approach directions of sine waves of different frequencies. At the same signal amplitudes, points representing lower frequencies are placed closer to the origin of the coordinates. The system also works well when periodical signals of different periods are received. Calculations are performed numerically by the acoustic processor on digital signal samples, following the procedure described by the formulas presented above.

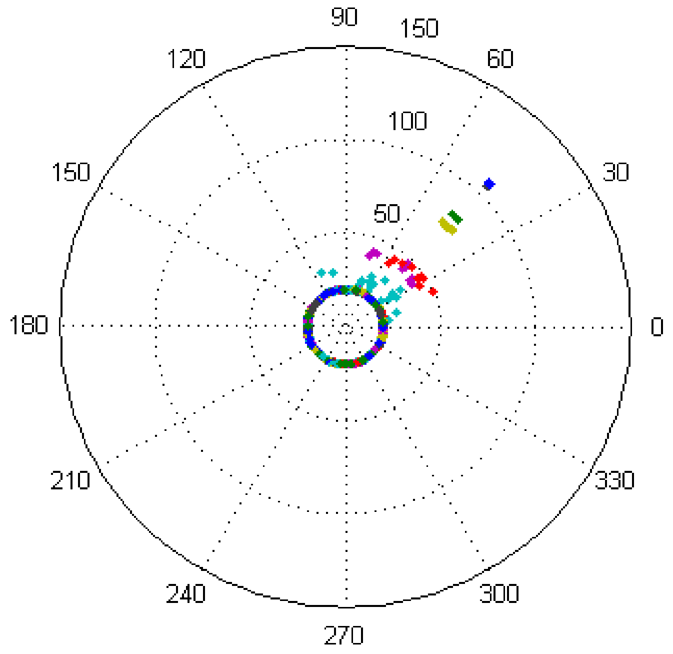

Fig. 5. Determining submarine bearing in a system using directional sonobuoys

4. CREATING THE COMPOSITE SIGNAL

Three signals are needed for the detection and estimation of direction: the central hydrophone signal and differential signals are transmitted to the receiver a single, modulated radio signal. This requires the three signals to be converted to a special signal called the composite signal. Another information included in the composite signal is the angle θ_0 of array relative to north. This angle is measured by a compass installed in the array unit which descends to the preset depth.

The method of creating the composite signal is interesting enough to be discussed in detail. The array unit generates three signals whose waveforms are shown in Figure 6.

The period of the two first rectangular waves is $T_0=1/f_0=1/15$kHz. The period of the third signal is twice as long; therefore, its first harmonic frequency is 7.5 kHz. This signal is shifted relative to keying signals by a time proportional to the angle between the array axis and north. For the full turn of the compass needle, the signal delay is equal to period T_0. The composite signal is created according to the following algorithm:

$$y(t) = s(t) + s_c(t) \cdot s_k(t) + s_s(t) \cdot c_k(t) + s_k'[t - \tau(\theta_0)] \tag{8}$$

The basic components of the spectrum of this signal are obtained by replacing square waves with a sine or cosine signal. For positive frequencies, the significant part of the composite signal spectrum can be expressed as follows:

$$\begin{aligned} Y(f) = S(f) + A[S_s(f-f_0) - jS_c(f-f_0) + S_s(f+f_0) + jS_c(f+f_0)] + \\ + S_k'(f)e^{-j2\pi f\tau(\theta_0)} \end{aligned} \tag{9}$$

where $S(f) = \Im\{s(t)\}$ and A is constant.

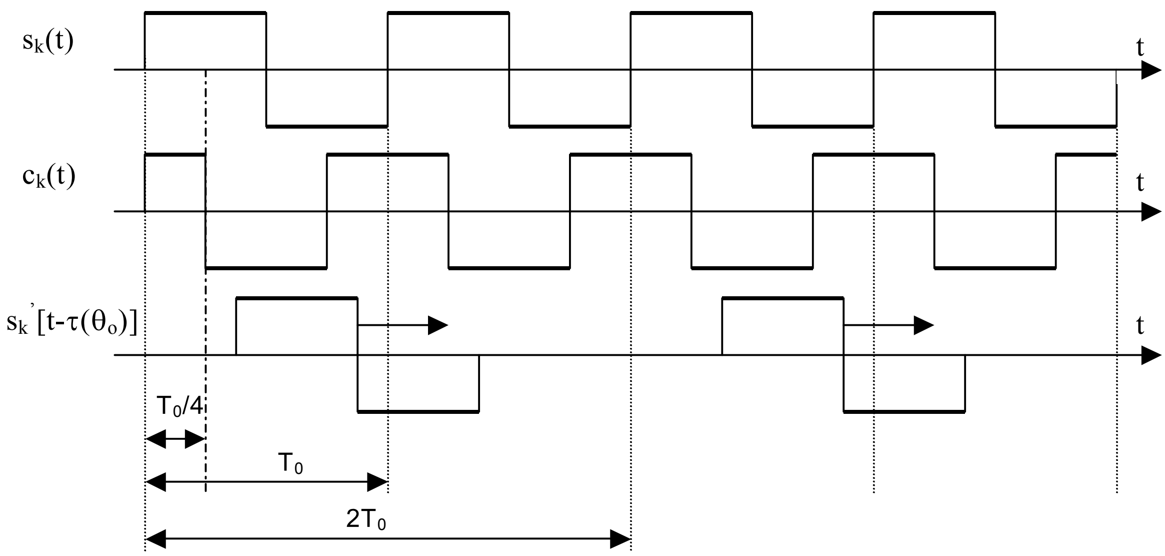

Fig. 6. The keying signals generated in the array unit

Neglecting the noise spectra for the sake of clarity, and using formulas (4), we get for sine signals:

$$Y(f) \cong S(f) + B\{\delta[f-(f_o - f_s)]e^{-j(\theta+\varphi)} + \delta[f-(f_0+f_s)]e^{-j(\theta-\varphi)}\} + \\ + C\delta(f-f_0)e^{-j\theta_0} \tag{8}$$

where B, C are constants and f_s is frequency of received signal.

The composite signal spectrum for a sine signal received with Gaussian noise is shown in Figure 7.

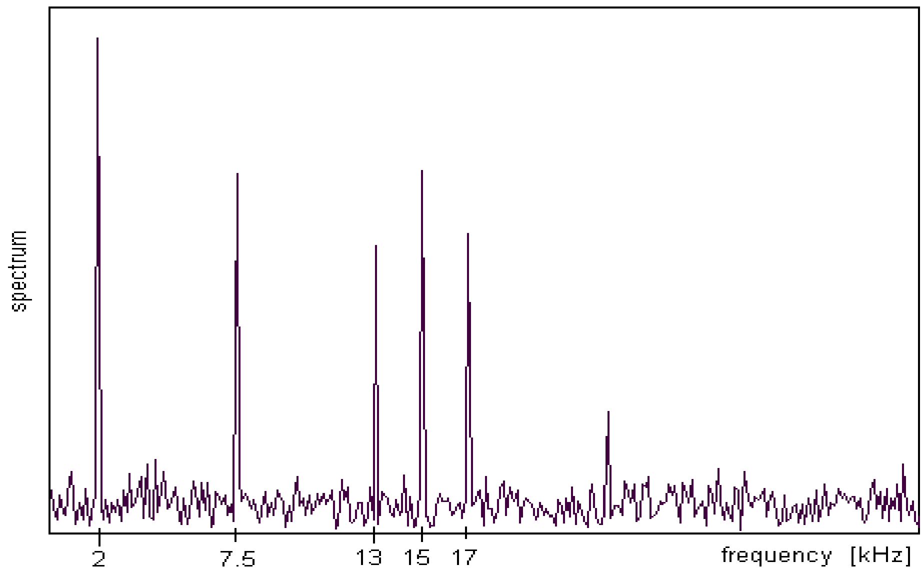

Fig. 7. The composite signal spectrum

The composite signal modulates the radio signal of one of 99 frequencies of the band ranging from 136 MHz to 173.5 MHz. The radio frequency raster is 375 kHz. The frequency of the modulating signal is preset for each sonobuoy prior to throwing from the aircraft. In DIFAR sonobuoys, frequency modulation (FM) is used with +/- 105 kHz maximum deviation. The minimum frequency of the acoustic signal is 5 Hz; the maximum frequency is 2.4 kHz.

5. RETRIEVING THE COMPOSITE SIGNAL IN THE ACOUSTIC PROCESSOR

Radio signal receivers are modular, which allows to simultaneously receive signals from several to several dozen sonobuoys (usually 8, 16 or 32). The receiver demodulates radio signals and outputs composite signals of the respective sonobuoys. These signals are sampled in the respective channels of the acoustic processor, the number of which corresponds to the number of receiver channels. Processing of composite signals by the processor is the same for each channel. Modern acoustic processors are digital devices, so the processing is preceded by A/D conversion. The sampling frequency should satisfy the Nyquist stability criterion, so it should be higher than about 35 kHz in the discussed system. It is convenient to choose a frequency for which the number of sequence 1 samples is a power of 2, which allows to use the FFT algorithm without the need to complement the sequence with zeros.

Processing the digital composite signal has the following purposes:

- retrieving the acoustic signal from the central hydrophone,
- retrieving differential signals,
- retrieving the keying signal containing the information about the angle of array relative to north.

This tasks can be achieved by processing the signal in the time domain or in the frequency domain. Digital processing of the signal in the domain provides all functions of an analog processor whose functional diagram is shown in Figure 8.

The composite signal $y(t)$ is filtered by digital filters; the first one (7.5 kHz midband frequency) separates the signal $s_k'(t)$ containing information about the angle of the array relative to north. The second filter (15 kHz midband frequency, about 6 kHz bandwidth) separates a narrow band signal modulated by the differential signals. The third (low-pass) filter separates the acoustic signal of the central hydrophone. The 7.5 kHz signal is squared and filtered by a 15 kHz midband frequency narrow band filter. The phase of this filter's output signal is shifted from the original sine keying signal by θ_0 being the angle of the array relative to north. A cosine signal of the same phase shift is available on the output of the Hilbert transformer shown marked HT. Multiplying the narrow-band, 15 kHz midband frequency by the sine and cosine signals, and applying low-pass filtration gives separated differential signals. When noise is absent, the amplitudes of these signals are proportional to the sine and cosine of the incidence angle of the wave relative to north. For each of the three signals, a Fourier transform is computed. Detection is performed on the basis of the central hydrophone signal transform; all transforms are used to determine the approach directions of waves in the way described in chapter 3.

Signal processing in the time domain is an inefficient method requiring numerous digital operations. Nearly all digital filters should have linear phase characteristics and steep slopes. FIR or IIR filters with data series inversion are preferred.

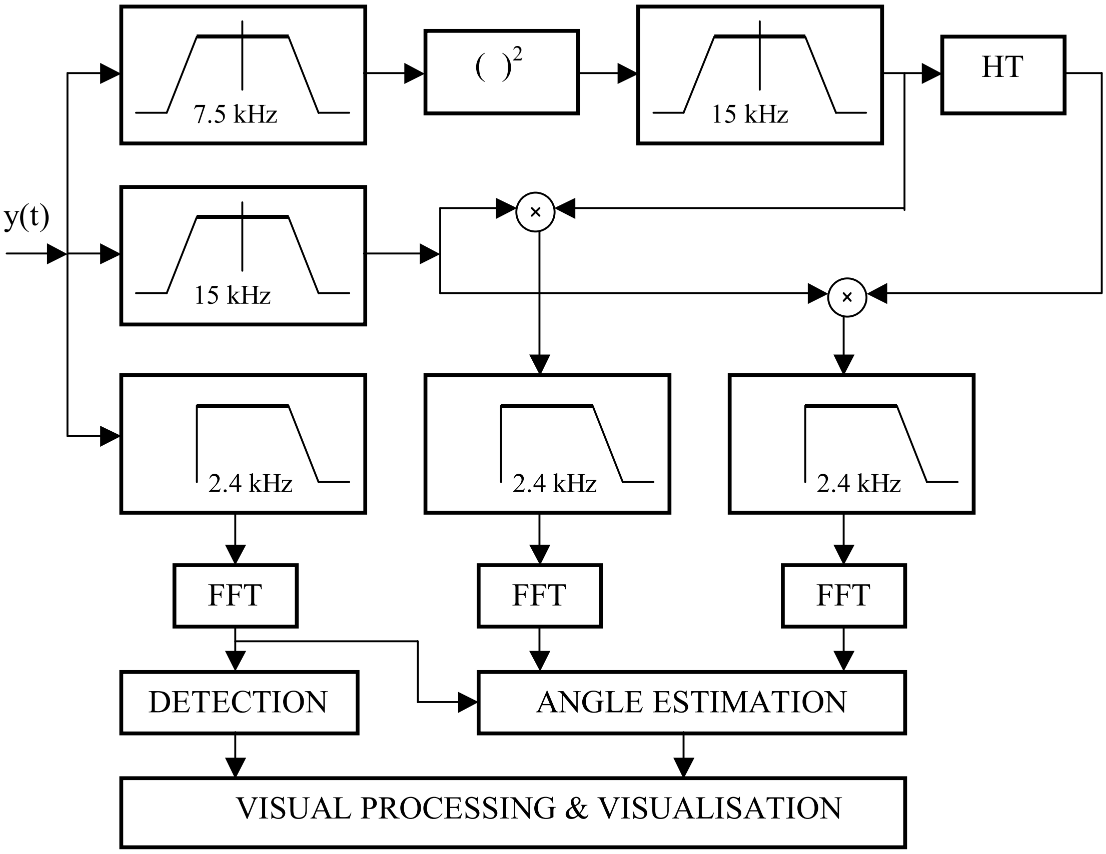

Fig. 8. Functional diagram of an acoustic processor

6. SIGNAL PROCESSING IN THE FREQUENCY DOMAIN

Signal processing in the frequency domain is much simpler and more efficient compared to the method described above. This type of processing does not require retrieving the composite signal and consists of simple operations on the spectrum of this signal. This procedure follows directly from (8) and includes the following operations:

1. computing Fourier transform of the signal *y(n),*
2. isolating from the Fourier transform the lines of the central hydrophone signal (equivalent of the low-pass filtration in the time domain),
3. determining the periodogram of the acoustic signal,
4. isolating spectral lines of the narrow-band 15 kHz midband frequency signal (equivalent of the narrow-band filtration in the time domain),
5. computing phase angles of spectral lines,
6. finding the 15 kHz spectral line,
7. computing the phase angle for this line,
8. computing the incidence angle of the wave relative to north.

The number of operations required to compute the Fourier transform using the FFT method is not significantly greater than for computing the three transforms of acoustic signals using the method described above. Operations 2, 4 and 6 are basically address manipulations. The only complex operation is computing the phase angles of spectral lines; it requires numerical calculations of arcsine and arccosine functions.

7. VISUALIZATION METHODS

In sonobuoy-based detection systems, determining the direction of sound sources and the position of detected submarines is done by an operator based on watching monitors showing the results of system operation in a user-friendly way. To make the detection possible, the monitor shows periodograms of signals received by all deployed sonobuoys. The periodograms are presented as spectral lines and in time-frequency form, where periodogram line heights are represented by spot color. The examples of this visualization method are shown in Figure 9. The latter method proves especially efficient with low signal to noise ratios, because it allows an experienced operator to detect a submarine based in characteristic features of the image recorded during long observation. In addition to detecting a submarine, the periodogram allows to determine the frequency of the sine signal emitted by the submarine which may help to identify it. The time-frequency visualizing can also show frequency changes of the received signal, which may result from Doppler effect or variations of propeller rotation rate, [4].

Periodogram spectral lines can be drawn in several colors indicating the angle sector which encompasses the acoustic wave approach direction. This additional information can help to make proper decisions related to target detection. Colored periodogram is shown in bottom chart of the Figure 9.

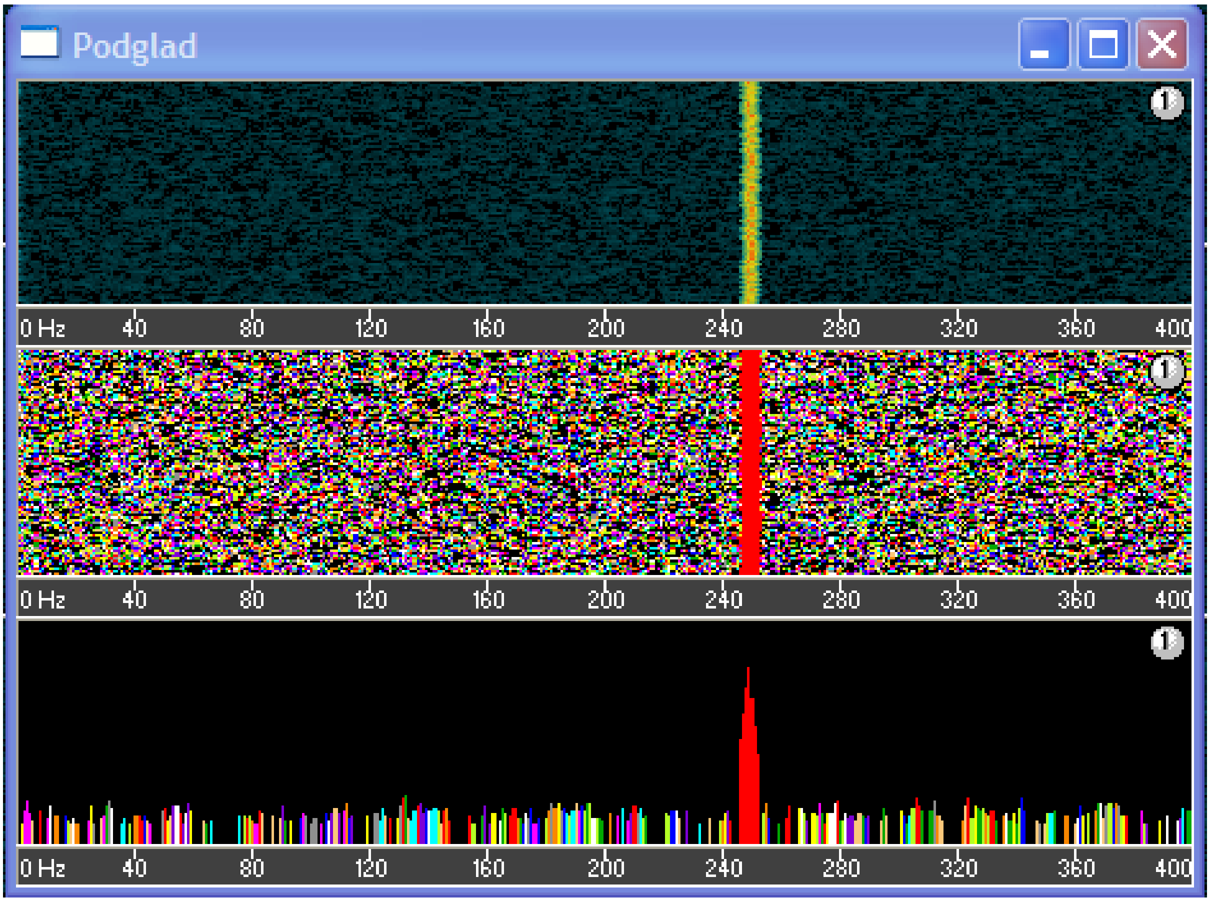

Fig. 9. Visualization of periodogram in time-frequency domain

Wave approach directions for each sonobuoy can be determined based on the visualization shown in Figure 5. However, this type of visualization is not very useful for observing signals from larger number of sonobuoys. The visualization shown in Figure 10 is much better - determining the wave approach direction is combined with determining the position of detected submarines. The directions of lines leading from the locations of sonobuoys show the wave approach direction, and brightness of the lines is proportional to the heights of periodogram lines, i.e. to the intensity of the detected wave. The brightness increases in intersection points, and very bright areas show the probable positions of detected submarines.

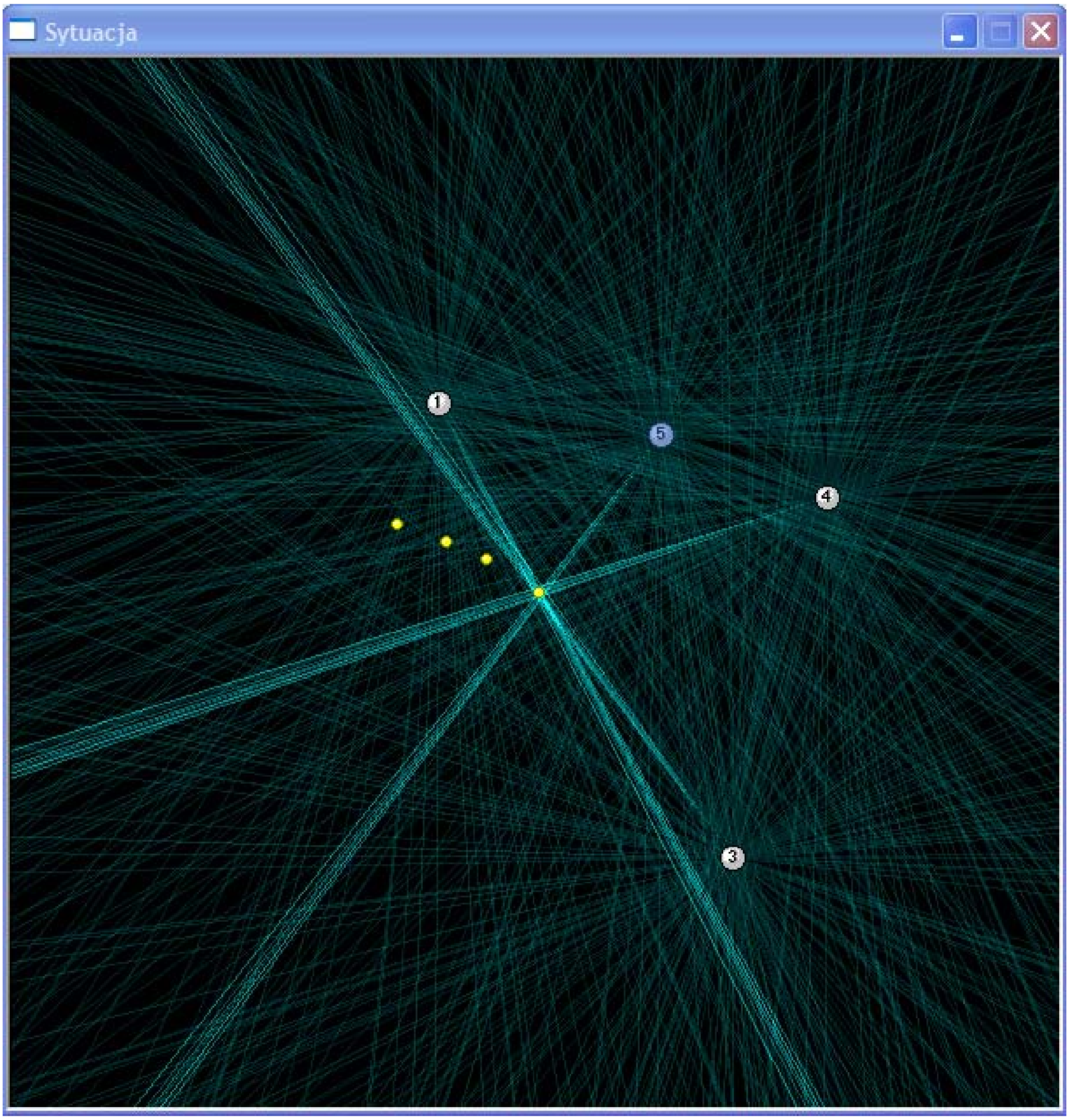

Fig. 10. Graphical method of target positioning in sonobuoy system

8. SONOBUOY POSITIONING SYSTEMS

The proper operation of the system requires the knowledge of the current position of all sonobuoys. The simplest but most inaccurate method of positioning the sonobuoys is by using the aircraft navigation system when deploying the sonobuoys. The positions are then assumed to be the same as drop positions. This assumption is often incorrect because wind and sea currents may significantly change the initial position of sonobuoys. To avoid these errors, radio systems are used for positioning sonobuoys.

A radiohydrobuoy positioning system is based on the well-known navigation principle of determining at least two bearings from at least two points of known positions. The position of these points is obtained from the aircraft navigation system. Bearings of sonobuoys are determined by a radio receiver tuned to the carrier frequency of the sonobuoy transmitter. The receiver incorporates a radio antenna with at least 3 independent elements. Bearings are determined by comparing phases of radio signals received by these 3 elements of the antenna. Electronic circuits used for measuring the phase may be embedded in the radio receiver or operate separately.

A promising solution of the sonobuoy positioning issue, providing better accuracy, is installing GPS receivers in sonobuoys. Information on the current position must then be periodically transmitted to the aircraft by radio. This requires modifying the standards of transmitted signals as well as the designs of sonobuoy transmitters and the receiver of the airborne system.

REFERENCES

[1] Burdick W. S. *Underwater Acoustic System Analysis.* Prentice-Hall, Englewood Cliffs, NJ, 1984

[2] Nielsen R. O. *Sonar Signal Processing.* Artech House, Boston 1991

[3] Oppenheim A. V. and Schafer R. W. *Digital Signal Processing.* Prentice-Hall, Englewood Cliffs, NJ, 1975

[4] Urban H. G. *Handbook of Underwater Acoustic Engineering.* STN ATLAS Elektronik GmH, Bremen 2002.

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