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DETECTION AND SIGNAL PROCESSING IN THE ACOUSTIC ECHOLOCATION SYSTEM FOR THE BLIND

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The paper presents a method of design, the general features and parameters of the acoustic, multibeam echolocation system for the blind which uses an electret microphone array and a digital beamformer with second order sampling. The system in question detects various obstacles that a blind person encounters on his or her way. It also determines the location of obstacles and the presumable shape and gives verbal messages about the situation observed. The paper includes a detailed description of signal processing in the beamformer and the conditions of detecting the observed obstacles. Theoretical and measured beam patterns of the system have been presented as well as the results of detecting and locating obstacles.

1. Introduction

In recent years, thanks to technological achievements, there has been a great deal of interest in designing aids for visually handicapped people. However, up till now only few of them have become commonly available as commercial products. The most significant progress has been made in the accessibility of printed materials, computers and telecommunications systems for the blind. There are several speech output systems for blind computer users which are available on the market in a great number of languages of the world. Current research on screen and print access technology – speech synthesis and text recognition is focused on naturalness, prosody, flexibility and improvement of optical character recognition techniques. Relatively more difficulties have been encountered in the area of travel and navigation aids for the blind. Visually disabled people who travel on their own have two kinds of problems: first "micro" navigation, e.g. walking along clear paths and avoiding obstacles around a known environment, and second – planning a journey to new places and keeping to the planned route. Although a number of navigation aids using optical, infrared and ultrasound sensors have been presented as prototypes, none of them have been commonly accepted, [2, 4, 6]. The main problem lies in the interaction between the system and the blind person, [3, 5]. Those devices use complicated audio signals which send out too much information to the user and obscure the external sounds from the environment. Technologically it is easy to obtain a digital representation of the surrounding with a miniature camera, however, the extraction of useful information on a safe path or nearby obstacles and then transmission to the user needs implementation of complicated signal processing procedures. That process takes too much time and in spite of the recent progress in signal processing technology and artificial intelligence methods, it cannot be used in real time systems. Infrared information systems have been presented in several places such as the London Underground, Paris and Hamburg Underground. These methods, however, have not become popular as they require the installation of a set of transmitters which is a very costly undertaking [7]. In radar and acoustic systems the image of the scrutinized space is relatively simpler and that is why the obstacles can be detected and signalized in a satisfactory time by a laptop computer. However, radar systems use a harmful microwave radiation which additionally is difficult to process due to its high frequency. Taking into account the restrictions mentioned above, an acoustic echolocation method seems to be a more attractive way of solving the problem of navigation aids for the blind because of its:

- simplicity of emission and detection of acoustic waves,
- small size of the acoustic array,
- low operating frequency,
- harmlessness to the user and environment.

Basing on the data pointed out, a multibeam echolocation system for the blind has been designed [8]. The results of pilot experiments confirmed the correctness of the assumptions and the effectiveness of the design and signal processing methods.

2. General operating features of the device

In line with its purpose, the system should perform the following functions:

- detection of obstacles in front of the blind person, in the road surrounding that he is using,
- determination of the position of the obstacles detected,
- classification of the obstacles,
- analysis of the spatial situation and providing short, verbal messages to the blind user about the recommended direction to follow or to stop.

In echolocation systems the first three of the above functions are called detection, parameter estimation and target classification. The fourth function is not related to acoustics and constitutes a separate problem in the area of image recognition and analysis. That is why it will not be discussed in this paper.

The above functions are realized in a system whose block diagram is shown in Fig. 1. The computer generates periodically every 100 ms discrete sinusoidal pulses at a frequency of 18 kHz and of a rectangular envelope. These pulses are processed in a digital-analog converter into the analog form, amplified in the power amplifier and radiated through a high frequency loudspeaker in a beam of approximately 60° width. Pulse duration takes 1.1 ms which shows that the system's range resolution is about

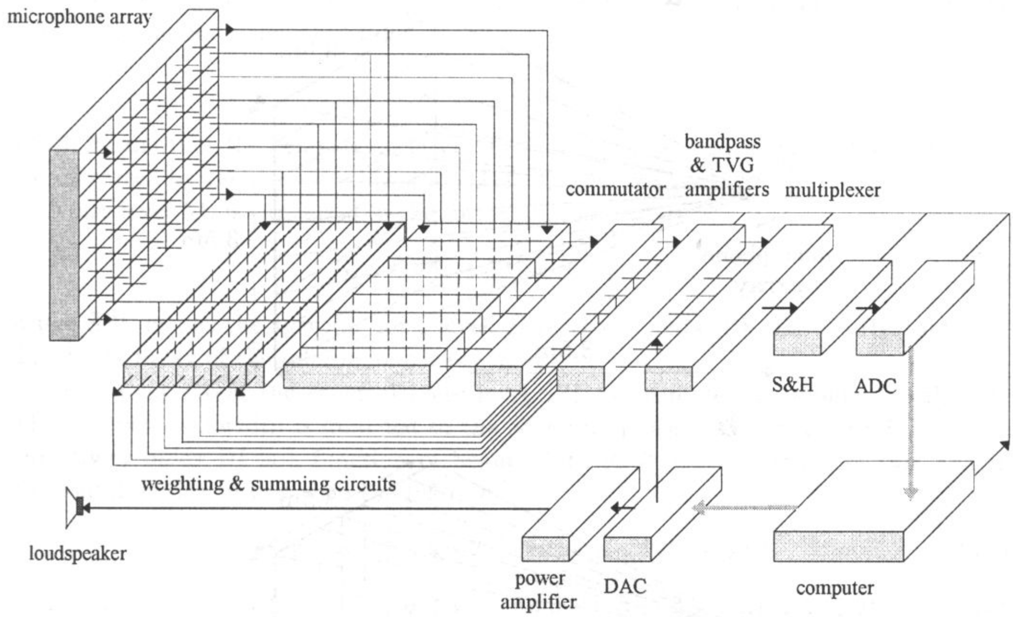
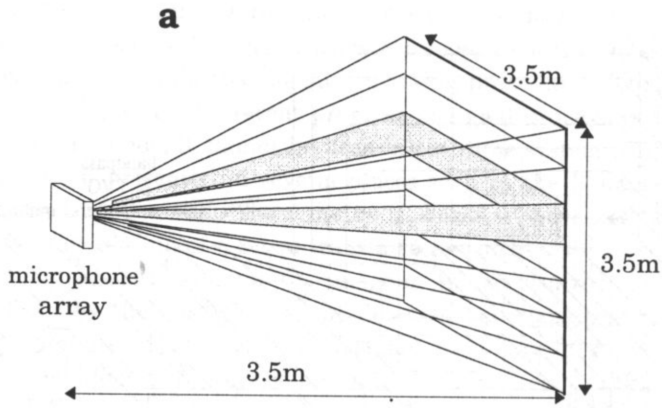


Fig. 1. Block diagram of the echolocation system for the blind.

20 cm(1). After they are reflected from an obstacle, echo pulses are received by an array of 8×8 electret microphones, then weighted and summed up along eight rows and eight columns. The commutator in one transmission feeds to the eight channel receiver the signals from the eight rows and in the next transmission – the signals from the eight columns. They are amplified, filtered and normalized in the eight channel receiver. Analog echo signals are then successively sampled and fed into the memory of the signal processor. The processor realizes numerically the function of the beamformer which simultaneously produces 7 receiving beams at about 12° width mutually deflected by 9°. They cover the observation area at angular dimensions of about $60° \times 60°$. As a result of the beamformer and commutation set performance in one transmission we get 7 vertical beams and in the next transmission we get 7 horizontal beams located as shown in Fig. 2. The method of beams determining is described comprehensively in Sec. 4.

The digital signals from the 7 beams of the beamformer are transferred to a laptop computer which is used to determine the location and to classify the detected obstacles. The results of the calculations are then analyzed for their possible threat to the blind person. It is only in the case of a coming collision that the synthetizer produces verbal messages which warn the blind individual against a collision or indicate a safe direction of a further route. This method of giving information is preferred by the blind because they have to continuously use their hearing to communicate with the surrounding.

(1) The range resolution ΔR is theoretically equal $\Delta R = c\tau/2$, where c is the velocity of acoustic wave in air and τ is the pulse duration. For $c = 340$ m/s and $\tau = 1.1$ ms value of the range resolution is 18.7 cm.



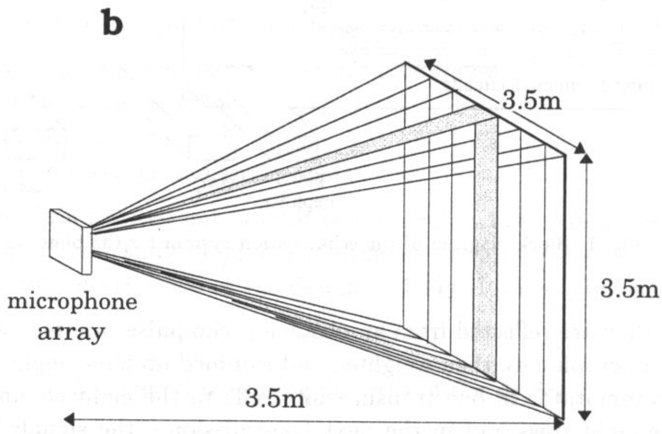


Fig. 2. Receiver beams configuration, a) horizontal configuration, generated by 8 columns of the microphone array, b) vertical configuration, generated by 8 rows of the microphone array.

3. Obstacle detection

The primary condition of an effective performance of the device is to detect the obstacle from a sufficient distance. For a normal speed of walking of a blind person, the distance must be long enough to accommodate the measurements, analyze the situation, give the message and stop or change the direction of movement. Based on the experience collected by the blind and having estimated the pace of the device's operations, it was assumed that the total time necessary to perform these activities is 3.5 s. Given the walking speed of 1 m/s, the necessary range is equal to 3.5 m. Calculations and experimental examinations showed that the target strength of the majority of dangerous obstacles (poles, trees, curbs, chains, people) is not less than TS = -30 dB(2). That is why the system's parameters must be selected in a way that will enable detection of obstacles at target strength TS = -30 dB at a distance of 3.5 m. To determine these parameters, a design procedure was adapted, developed by R.J. Urick for underwater acoustics [11].

(2) Target strength TS = -30 dB is exhibited by a perfectly reflected sphere at a diameter of 12 cm.

According to this procedure it is assumed that the following inequality must be fulfilled as the detection condition:

$$\mathrm{EL} \geq \mathrm{NL} + \mathrm{DT}, \tag{3.1}$$

which means that the echo signal level EL([3]) must be at least equal to the noise level NL increased by the detection threshold DT.

Noise level NL is expressed in the formula:

$$\mathrm{NL} = \mathrm{SPL} + 10\log B/B_0 - \mathrm{DI}, \tag{3.2}$$

where SPL means the spectral noise level, B receiver bandwidth, $B_0 = 1\,\mathrm{Hz}$, and DI – directivity index of the microphone array used.

On a busy street the spectral noise level SPL is estimated at about – 85 dB [12]. The receiver bandwidth is matched to the sounding pulse and amounts to $B = 1\,\mathrm{kHz}$. Directivity index DI is a function of beam width θ, Ψ in the vertical section and in horizontal section in line with the relation:

$$DI \cong -10\log\sin(\theta/2) - 10\log\sin(\Psi/2) + 4\mathrm{dB}. \tag{3.3}$$

Substituting $\theta = 12°$ and $\Psi = 60°$ we obtain DI = 17 dB. Formula (3.2) shows that the expected level of acoustic noise in the system is $NL = -72\,\mathrm{dB}$.

Detection threshold DT expresses in decibels the signal to noise ratio at the system's input, necessary to perform detection of the useful signal. Due to the stochastic character of noise, detection is taking place at a certain PD probability. Sporadically we can observe so called false alarms where with no useful signal and a temporary high amplitude of noise it is assumed that a target is there. For the Gaussian noise, probability of detection and probability of a false alarm PFA explicitly define the necessary value of detection threshold DT. With respect to the blind person's safety, the designed system should assume a high value of the detection threshold. On the other hand, a low value of probability of a false alarm will ensure that a blind person will not be unnecessarily bothered by apparent obstacles. In line with this reasoning it was assumed that PD = 98% and PFA = 10^{-5}([4]). For known signal and the Gaussian noise, the assumed values PD and PFA are matched by the detection threshold DT = 15 dB, [9].

The calculations made so far enable the determination of the minimum echo signal level in the system. In line with formula (3.1) the level amounts to EL = −57 dB. Given the microphone sensitivity VR used in the acoustic antenna we can calculate the voltage level UL at the input of the receiver. It is:

$$\mathrm{UL} = \mathrm{EL} + \mathrm{VR}. \tag{3.4}$$

([3]) All dimensions that denote the level are defined as $20\log(p/p_0)$, where p is the acoustic pressure of the respective signal and $p_0 = 1\,\mathrm{Pa}$.

([4]) Further increase in detection probability is not justified because it increases on its own when the blind person is nearing an obstacle. This is caused by an increase in the amplitudes of the echo and consequently by an increase in the signal to noise ratio. This effect is not observed in respect to false alarm probability. It is constant for a detection threshold set at a specific value. False alarm probability pertains to a time range that is equal to the sounding pulse duration. The value PFA = 10^{-5} in the system in question, means that on the average one false alarm comes up approximately every 8 minutes.

The sensitivity of the electret microphones is equal to 10 mV/Pa, that is VR = −40 dB([5]). The above formula shows that UL = −97 dB which means that the minimum voltage of a useful echo signal is equal to $u = 14\,\mu\text{V}$([6]).

The last step in designing the system from the standpoint of detection conditions is to determine the source level SL. This level can be determined by transforming the basic equation of echolocation:

$$\mathrm{EL} = \mathrm{SL} - 2\mathrm{TL} + \mathrm{TS} \tag{3.5}$$

to the following form:

$$\mathrm{SL} = \mathrm{EL} + 2\mathrm{TL} - \mathrm{TS}, \tag{3.6}$$

where the symbol 2TL means transmission loss.

Assuming a spherical model of propagation, transmission loss is calculated from the formula:

$$2\mathrm{TL} = 40 \log R/R_0 + 2\alpha R, \tag{3.7}$$

where range $R = 3.5$ m, $R_0 = 1$ m and α is the absorption coefficient. Given a frequency of 18 kHz, its value is $\alpha \cong 1$ dB/m, [10], and so 2TL $\cong$ 29 dB.

After substituting the parameters determined earlier of the range equation to formula (3.6) we obtain SL = 2 dB. This source level ensures a loudspeaker radiating an acoustic wave whose acoustic pressure at a 1 m distance from the membrane on the acoustic axis is equal to 1.2 Pa. Electric power of the power amplifier is experimentally set at a value where the acoustic pressure of the sounding signal is equal to the above conditions.

Formulas (3.5) and (3.7) show that when a blind person is nearing an obstacle at the same time echo signal level increases. The amplitude of the echo signal is not just a function of target strength (the size of the obstacle) which makes analysis of the situation difficult. In order to reduce this negative effect, the receiver uses time varied gain (TVG). The echo level at the output of the TVG is expressed in the general formula:

$$\mathrm{EL_{TVG}} = \mathrm{SL} + \mathrm{TS} - 2\mathrm{TL} + K, \tag{3.8}$$

where K is the variable in time function of amplification, selected for the purpose of a complete or partial compensation of transmission loss and a possible change of target strength together with the distance. The system uses time varied gain which realizes the function:

$$\begin{aligned} K &= -\infty && \text{for } R \le R_1 \text{ and } R > R_M, \\ K &= 20 \log R/R_1 + 2\alpha R + K_0 && \text{for } R_1 < R \le R_M, \end{aligned} \tag{3.9}$$

where $R_1 = 0.5$ m is the minimal distance of observation, $R_M = 3.5$ m is the system range, and K_0 is the constant receiver gain([7]). Given this gain function the echo level

([5]) Voltage response VR $= 20 \log[(u/u_0)(p_0/p)]$, where $u_0 = 1$ V, $p_0 = 1$ Pa, u = voltage at the output of the microphone, p – acoustic wave pressure falling on the microphone. Voltage level UL $= 20 \log u/u_0$.

([6]) It should be noted that rms voltage of acoustic noise at the output of the microphone is greater by DI − DT = 2 dB (amounts to about 18 μV) because the reduction of noise happens only at the output of the beamformer. This does not pertain to the electric noise because for this noise, an improvement of the noise ratio at the output of the beamformer amounts only to about $\sqrt{n}$-th times (n – effective number of microphones considering weighting factors). For $n = 16$ rms voltage of electric noise at the output of the microphone should not exceed 10 μV at detection threshold DT = 15 dB and absence of acoustic noise. To ensure acoustic noise dominance in the system, rms voltage of electric noise should be 2 to 3 times lower.

([7]) The TVG gain changes in the function of time according to the formula: $K(t) = k_0(t/t_1)10^{\alpha ct/20}$, where $k_0 = 10^K$ and $t_1 = 2R_1/c$.

in the range of distances $R_1 < R \le R_M$ is described with the following formula:

$$\mathrm{EL_{TVG}} = \mathrm{SL} + \mathrm{TS} - 20\log R/R_1 + K_0 - 12\,\mathrm{dB}. \tag{3.10}$$

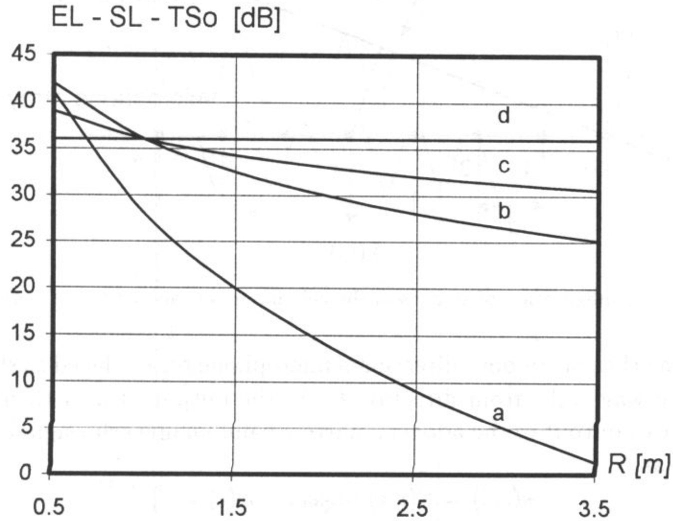


Fig. 3. Changes of the echo level EL in the function of the distance normalized in relation to the source level SL and the target strength coefficient $\mathrm{TS_0}$. a) $\mathrm{TS_0} = \mathrm{const}$, no TVG, b) $\mathrm{TS_0} = \mathrm{const}$, TVG on c) $\mathrm{TS} = \mathrm{TS_0} + 10\log R/R_0$, TVG on, d) $\mathrm{TS} = \mathrm{TS_0} + 20\log R/R_0$, TVG on.

Figure 3 shows the changes of echo level for a case of a constant target strength and target strengths described using the relations $\mathrm{TS} = \mathrm{TS_0} + 10\log R/R_0$ (poles, tress) and $\mathrm{TS} = \mathrm{TS_0} + 20\log R/R_0$ (large surfaces, walls). As the presented charts show for the most negative and practically rare case of an object of a constant target strength, the dynamics of echo signal decreased from 40 dB for the system without TVG (formula (3.7)) to 17 dB with the TVG system. The system in question used linear gain control in the receiver and absorption loss is compensated numerically in the computer.

4. The beamformer principle

The above principle of the device shows that in order to identify an obstacle it is necessary to generate seven mutually deflected receiver beams. These beams are generated in the beamformer that cooperates with the microphone array. The method of beam generation is identical for vertical deflection and horizontal deflection. For vertical deflection the signals are summed and weighted in each of the eight rows of the microphone array. As a result we obtain 8 independent signals from eight sources that form a linear column array. For vertical deflection as a result of the commutator's operations the situation is reverse: signals from each of the eight columns of the array are summed up and in the end effect we have at our disposal eight independent signals from a linear row array. The position of the sources in both linear arrays is identical and the placement of the sources is shown in Fig. 4.

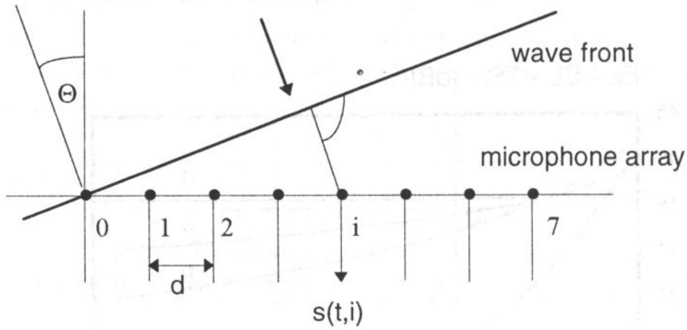


Fig. 4. Configuration of sources in the column or row array ($d = 11.5$ mm).

Let us assume that on an omnidirectional microphone array placed as shown in Fig. 4, a plane acoustic wave falls from direction θ. At the output of an i-th microphone (in practice at the i-th output of the adder) a narrow band analog echo signal appears which can be written as:

$$s(t,i) = S(t,i)\sin[\omega_0 t - \alpha(i) + \varphi], \tag{4.1}$$

where $S(t,i)$ is the slow changing envelope of echo signal, $\omega_0 = 2\pi f_0$ is the angular frequency of sounding signal ($f_0 = 18$ kHz), φ is the phase of carrier signal, and $\alpha(i)$ is phase shift resulting from the direction of the acoustic wave incidence. For an incidence angle of θ the phase shift is equal to:

$$\alpha(i) = \omega_0(id/c)\sin(\theta), \qquad i = 0, 1, ..., I-1, \tag{4.2}$$

where c is the velocity of the acoustic wave in the air, d is the distance between the neighboring microphones, and I – the number of microphones in a row or a column ($I = 8$).

In the device in question a delay-sum beamformer was used with second order sampling. The first stage of the signal processing in this beamformer was a special sampling which for discrete signals – as will be shown later – is the equivalent of quadrature detection. The idea of second order sampling is to cyclically collect two samples of each signal at a time interval equal to 1/4 of the period of carrier signal. The principle of sampling is illustrated in Fig. 5.

As the result of sampling we obtain two sequences of samples that consist of N segments with I samples in each segment. As is shown in Fig. 5, the values of the samples in both sequences can be written as:

$$\begin{aligned} s_s(i,n) &= S\left[\frac{T}{2}(nI+i)\right]\sin\left[\omega_0\frac{T}{2}(nI+i) + \alpha(i) + \varphi\right], \\ s_c(i,n) &= S\left[\frac{T}{2}(nI+i) + \frac{T}{4}\right]\sin\left\{\omega_0\left[\frac{T}{2}(nI+i) + \frac{T}{4}\right] + \alpha(i) + \varphi\right\}, \end{aligned} \tag{4.3}$$

where $T = 1/f_0$ is the period of the sounding signal.

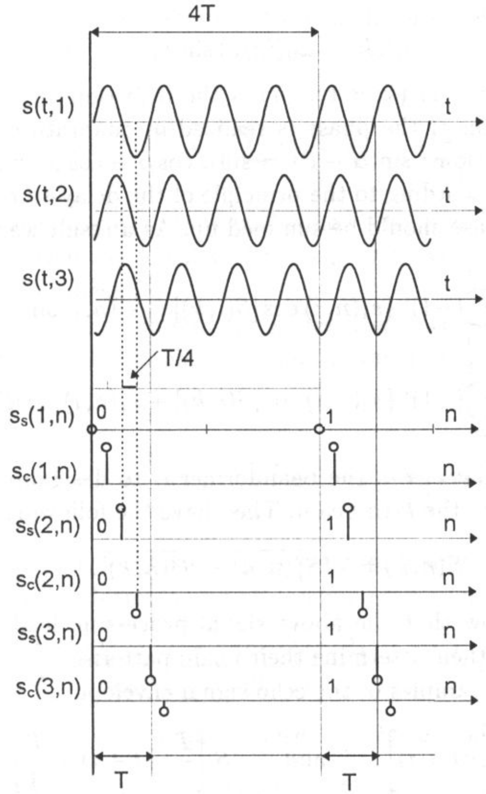


Fig. 5. Second order sampling of echo signals realizing quadrature detection ($T = 1/f_0 = 55.5\,\mu s$).

After applying simple trigonometric identities, the expressions (4.3) are transformed to the following form:

$$\begin{aligned} s_s(i,n) &= (-1)^i S\left[\frac{T}{2}(nI+i)\right]\sin[\alpha(i)+\varphi], \\ s_c(i,n) &= (-1)^i S\left[\frac{T}{2}(nI+i)+\frac{T}{4}\right]\cos[\alpha(i)+\varphi]. \end{aligned} \tag{4.4}$$

The above formulas show that as a result of second order sampling we obtain from each i-th microphone two sequences of samples of echo signal envelopes multiplied by sinus and cosine of the phase shift. That is why they are in fact samples of quadrature components of echo signal. The samples of the i-th sequence appear in time intervals $IT/2$ which must be smaller than the sampling period that results from Nyquist's theorem: $IT/2 < 1/2B$ so $I < f_0/B$. In the system in question this condition is met with a surplus because $I = 8$, and $f_0/B = 18$.

In order to generate a deflected beam, the phases of the received signals should be changed so that in each i-th channel the phase of the signal received from the direction of the beam axis is identical. This can be achieved by subtracting from phase $\alpha(i) + \varphi$

phase$\beta(i)$ that is equal to:

$$\beta(i,k) = \omega_0(id/c)\sin(\theta_k), \tag{4.5}$$

where $\theta_k = -27° + \Delta\theta(k-1)$ $(k = 1, ..., 7)$ is the defection angle of the k-th axis of the beam. The subtraction of the phase is realized in quadrature detection using the known trigonometric relations: $\sin(\alpha - \beta) = \sin\alpha\cos\beta - \cos\alpha\sin\beta$ and $\cos(\alpha-\beta) = \cos\alpha\cos\beta + \sin\alpha\sin\beta$. According to the principle of the delay-sum beamformer signals that are shifted in the phase should be summed up. As a result we obtain:

$$\begin{aligned} S_s(n,k) &= \sum_{i=0}^{I-1}(-1)^i\{s_s(n,i)\cos[\beta(i,k)] - s_c(n,i)\sin[\beta(i,k)]\}, \\ S_c(n,k) &= \sum_{i=0}^{I-1}(-1)^i\{s_c(n,i)\cos[\beta(i,k)] + s_s(n,i)\sin[\beta(i,k)]\}. \end{aligned} \tag{4.6}$$

The last step of calculations in the beamformer is to designate the samples of the echo envelope coming from the k-th beam. They have the following form:

$$S(n,k) = \sqrt{S_s^2(n,k) + S_c^2(n,k)}\,. \tag{4.7}$$

Now we are going to show that the above signal processing leads to a generation of deflected beams. We will then determine their beam patterns.

Let us notice that the samples of the echo signal envelope

$$S\left[\frac{T}{2}(nI+i)\right] \quad \text{and} \quad S\left[\frac{T}{2}(nI+i) + \frac{T}{4}\right]$$

are practically equal because given the narrow envelope spectrum the changes of its temporary values cannot happen all too quickly. In line with this reasoning a significant error is avoided by assuming that

$$S\left[\frac{T}{2}(nI+i)\right] \cong S\left(\frac{T}{2}nI\right),$$

which means that within one cycle of sampling, the envelope assumes an almost constant value. Given these assumptions, the relations (4.6) are reduced to the form:

$$\begin{aligned} S_s(n,k) = S\left(\frac{T}{2}nI\right)&\sum_{i=0}^{I-1}\sin[\alpha(i,k) - \beta(i,k) + \varphi] \\ &= S\left(\frac{T}{2}nI\right)\sum_{i=0}^{I-1}\sin[\omega_0(id/c)(\sin\theta - \sin\theta_k) + \varphi], \\ S_c(n,k) = S\left(\frac{T}{2}nI\right)&\sum_{i=0}^{I-1}\cos[\alpha(i,k) - \beta(i,k) + \varphi] \\ &= S\left(\frac{T}{2}nI\right)\sum_{i=0}^{I-1}\cos[\omega_0(id/c)(\sin\theta - \sin\theta_k) + \varphi]. \end{aligned} \tag{4.8}$$

After substituting the above relations to formula (4.7) and after some elementary transformations, we obtain:

$$S(n,k) = S\left(\frac{T}{2}nI\right)\sqrt{\left[\sum_{i=0}^{I-1}\sin(ix)\right]^2 + \left[\sum_{i=0}^{I-1}\cos(ix)\right]^2}\,, \tag{4.9}$$

where $x = \omega_0(d/c)(\sin\theta - \sin\theta_k)$. As is visible now, the effect of the operations of data using formula (4.7) is that the results of the calculations become independent of the unknown and accidental phase φ.

In order to determine the sums of the series from formula (4.9) we substitute the trigonometric functions with complex exponential functions and after algebraic operations, we obtain:

$$S(n,k) = S\left(\frac{T}{2}nI\right)\sqrt{\sum_{i=0}^{I-1}\exp(jix)\sum_{i=0}^{I-1}\exp(-jix)}\,. \tag{4.10}$$

The sums in the above formula are the sums of geometric series. After they are computed using the known formula and after some reductions, we obtain:

$$S(n,k) = S\left(\frac{T}{2}nI\right)\frac{\exp(jIx/2) - \exp(-jIx/2)}{\exp(jx) - \exp(-jx)} = S\left(\frac{T}{2}nI\right)\frac{\sin(Ix/2)}{\sin(x/2)}\,. \tag{4.11}$$

After substituting $x = \omega_0(d/c)(\sin\theta - \sin\theta_k)$ we have finally:

$$S(n,k) = S\left(\frac{T}{2}nI\right)\frac{\sin\left[(\pi I d/\lambda)(\sin\theta - \sin\theta_k)\right]}{\sin\left[(\pi d/\lambda)(\sin\theta - \sin\theta_k)\right]} = S\left(\frac{T}{2}nI\right)b_k(\theta), \tag{4.12}$$

where $\lambda = c/f_0$ is the length of the acoustic wave in the air and $b_k(\theta)$ is the beam pattern.

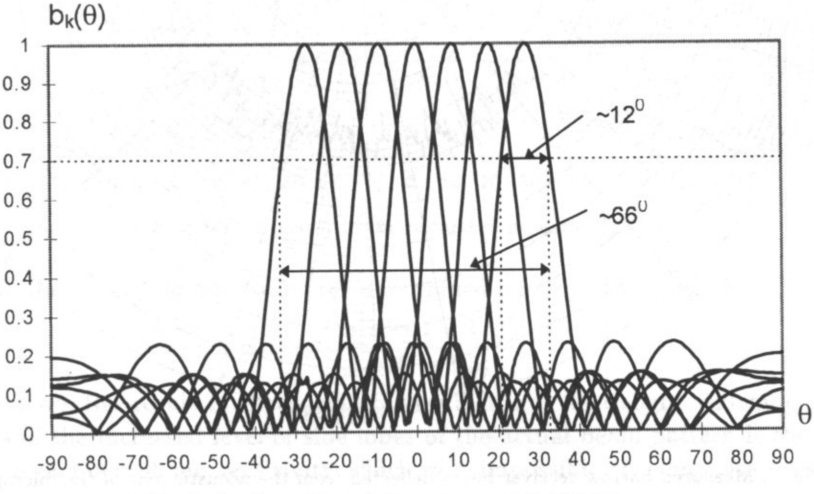


Fig. 6. Configuration of 7 narrow receiver beams.

As is known, the fraction expression seen in formula (4.12) describes beam pattern $b_k(\theta)$ of the linear array consisting of I omnidirectional microphones placed at a distance d from each other. The beam axis is deflected by the assumed angle θ_k which shows that the above described beamformer with second order sampling correctly performs its function. Figure 6 shows the calculated patterns of the microphone array used in the device. As a comparison, Fig. 7 illustrates a measured beam of the device deflected to the right by $\theta_6 = -27° + 5 \cdot 9° = 18°$. As the chart shows, the beam width is slightly greater than the width determined analytically which is due to the measurement errors and the technical imperfection of the beamformer's performance.

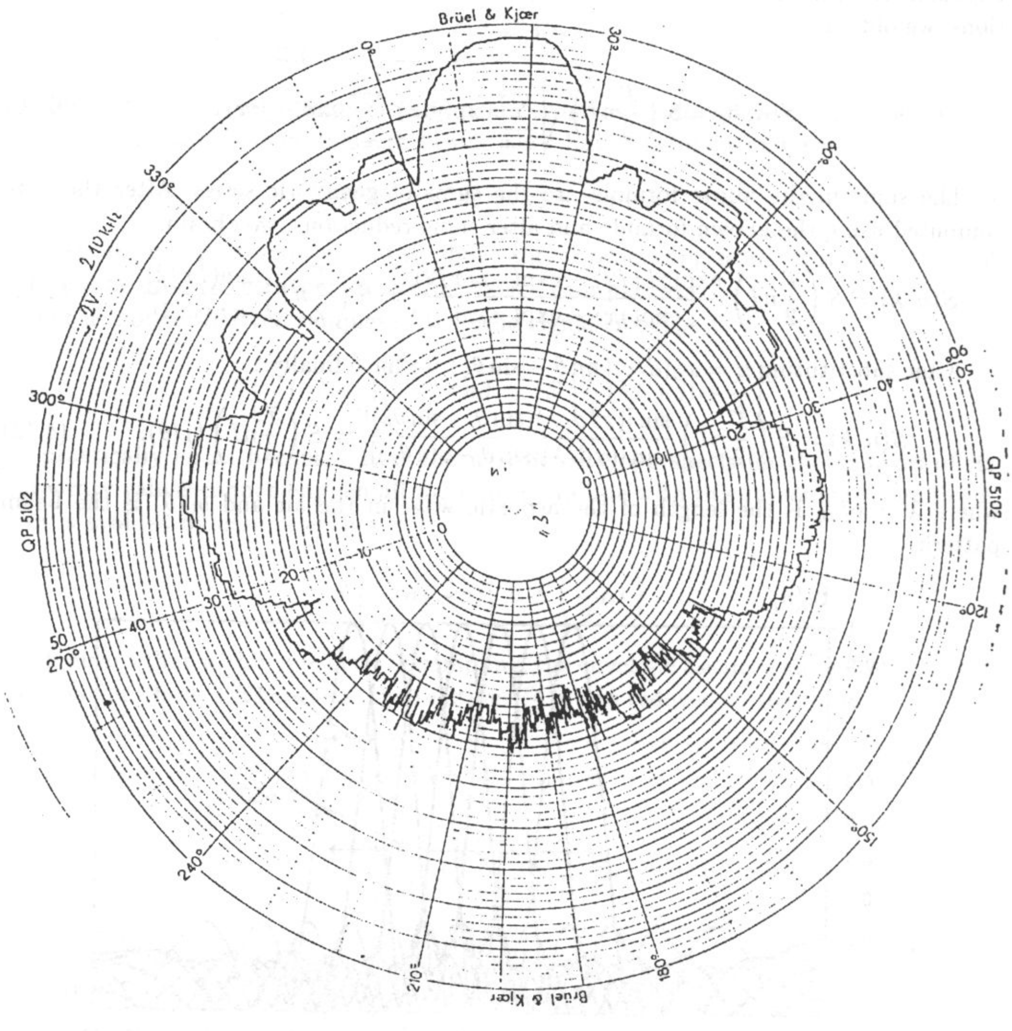


Fig. 7. Measured narrow receiver beam deflected from the acoustic axis of the microphone array by 18°.

5. Method of increasing the beam width

The majority of echolocation systems strive to reduce the level of side lobes the side effect of which is an increased beam width. In the device in question, the technique used to reduce the level of side lobes was applied in order to increase the beam width and to cause it to adopt the demanded shape. According to the adopted method of field searching, beam width in both sections must be significantly differentiated and amount to about $12° \times 60°$. Width of 12° is provided by a linear set of microphones cooperating with the beamformer. In order to obtain a width of 60° in the second section a strong amplitude of echo signals was used with reversing of the phase. A function of the type $\sin x/x$ was selected as the weighting distribution. This function ensures a significant increase in the beam width and caused it to assume a shape close to a rectangle. Beam patterns of a linear series of 8 microphones with amplitude weighting is described by the formula:

$$b(\theta) = \left[\sum_{i=0}^{7} w(i) e^{j\frac{\omega_0}{c} d(i-3.5)\sin\theta}\right] \Big/ \left[\sum_{i=0}^{7} w(i)\right], \tag{5.1}$$

where $w(i)$ is a weighting function.

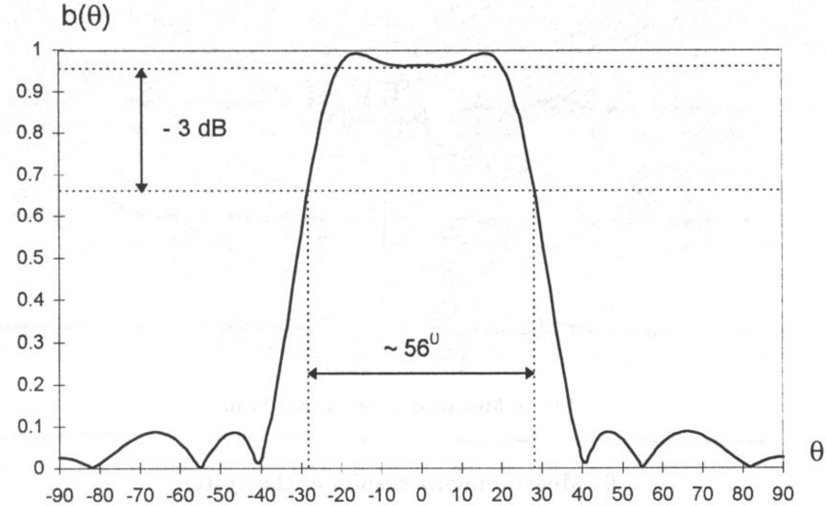


Fig. 8. Wide receiver beam calculated numerically.

Figure 8 illustrates beam pattern $b(q)$ determined numerically in which

$$w(i) = \frac{\sin[1.94(i-3.5)]}{1.94(i-3.5)}$$

and Fig. 9 shows the result of measuring of this pattern. The slight widening of the main lobe and the increased level of side lobes of the actual beam pattern is the effect of dispersion of microphone sensitivity which was impossible to compensate using a correction of the weighting factors.

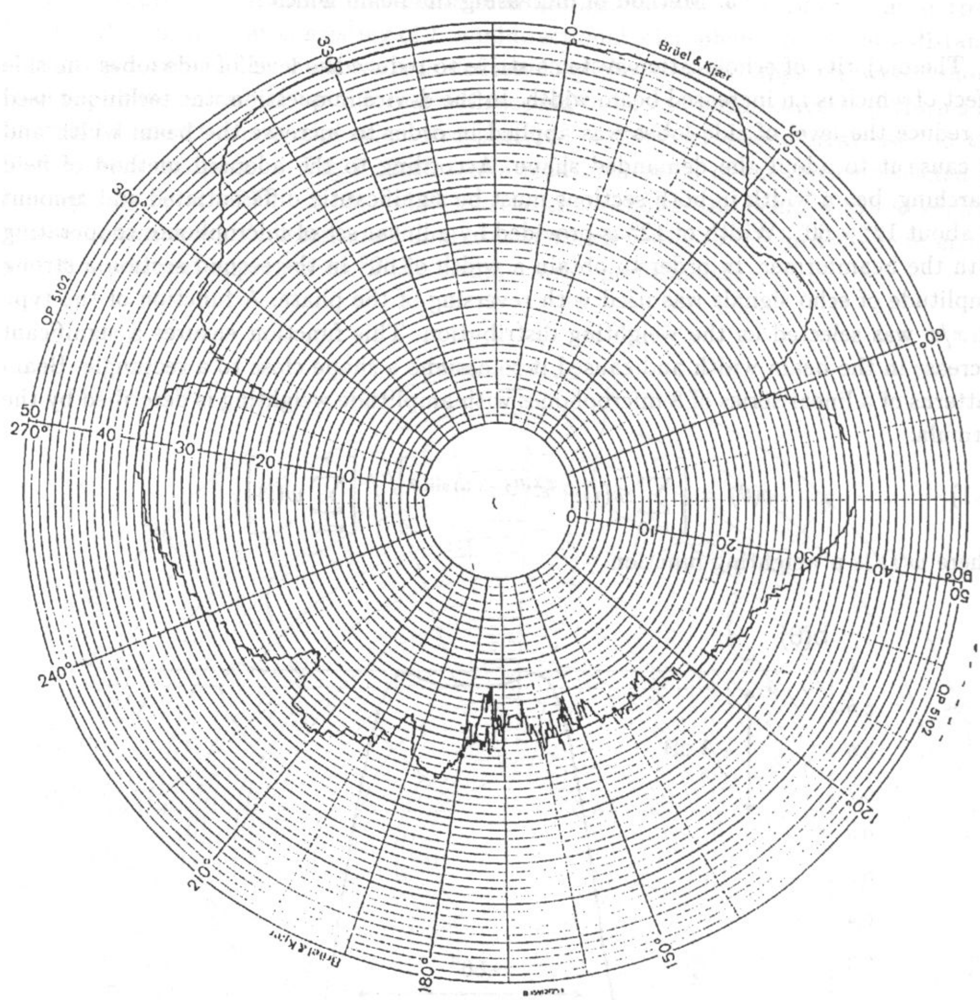


Fig. 9. Measured wide receiver beam.

6. Measurements results of the system

The constructed and launched system was then examined. The purpose of the examination was to verify the applied method of signal processing and the adopted design assumptions. All beam patterns of deflected narrow beams were measured as well as the patterns of non-deflected wide beams. It was found that the deflection angles are in line with the design assumptions, and the beam widths – despite a slight widening – fit into the permissible tolerances. Beam patterns of the loudspeaker covers with just a slight surplus the angular observation sector at a width of 60°. The electric power of the signal was selected in a way that caused the acoustic pressure at a distance of 1 m from the loudspeaker on its acoustic axis to be 1.2 Pa which matches the assumed value of the source level SL = 2 dB. In order to determine the range of the system, an observation was made of the echo signals from a sphere at a diameter of 12 cm (TS = −30 dB). In a quiet, closed laboratory room, the echo from the sphere was clearly visible in the background of noises when the sphere was on the axis of each of the beams 3.5 m away from the microphone array. The assumed range of the system was accomplished even though the electric noise level adopted in design calculations was exceeded. This voltage, related to the output of the microphone, amounted to 30 μV and dominated the voltage of acoustic noises. The echo signal from the sphere was also masked with a noise, however, at the input to the system its amplitude exceeded at least twice the noise level. The result of the increase in noise level is therefore a reduction of the detection threshold DT from 15 dB to 6 dB. This defect will be removed in the system's prototype by a more careful separation of analog and digital systems (use of transoptors).

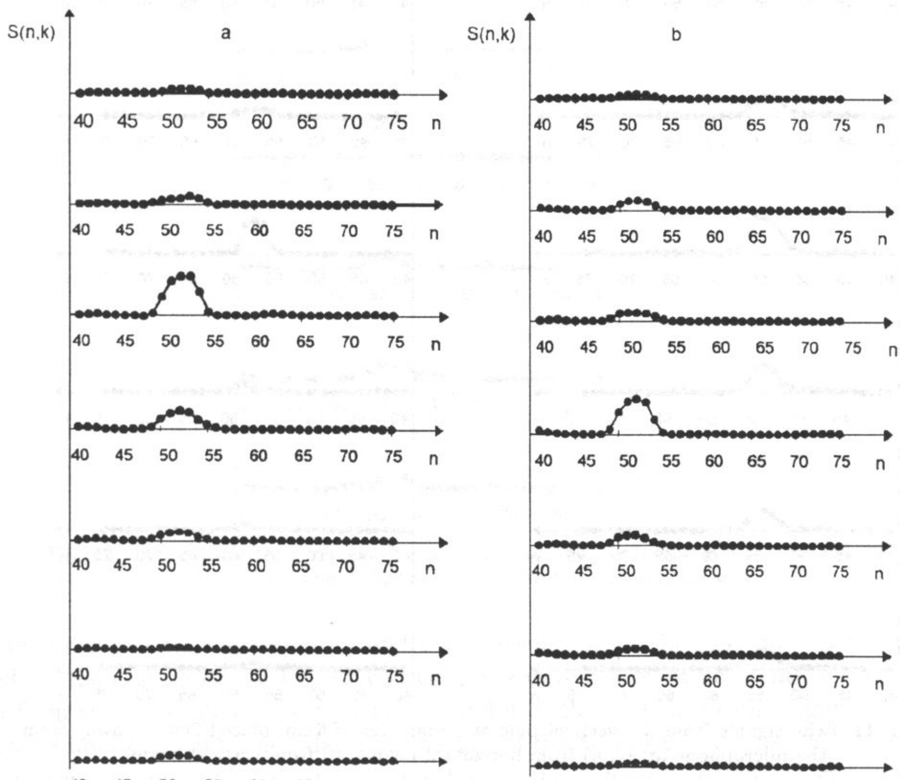


Fig. 10. Echo signals from a sphere at a diameter of 12 cm placed 200 cm away from the microphone array: a) from horizontal beams, b) from vertical beams.

Further measurements pertained to the system's ability to define the location and identify obstacles. Examples of measurement results are shown in the consecutive figures. Figure 10 shows the echo from a sphere at a diameter of 12 cm placed 2 m away from the microphone array. The sphere was hanged at the point where the vertical beam 3 and horizontal beam 4 intersect. As was expected in the respective beams clear echo signals were observed, much bigger than the background interference and the small echoes in the other beams caused by the side lobes of beam patterns.

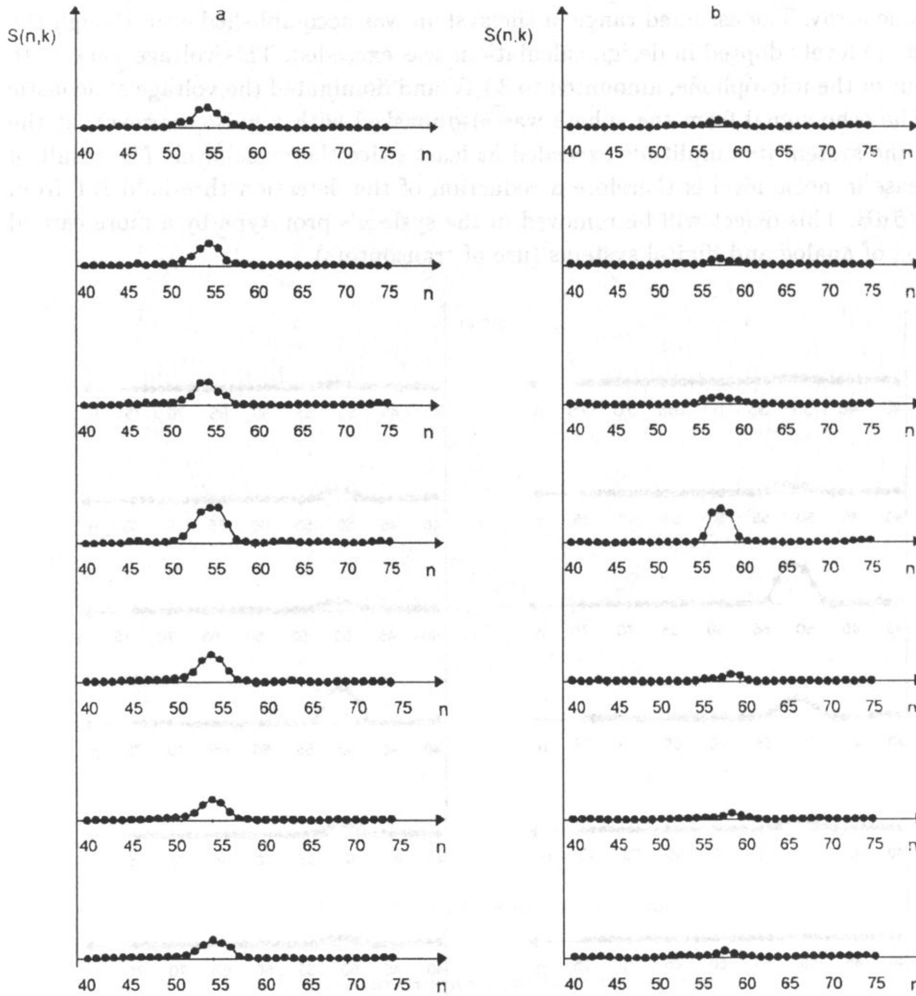


Fig. 11. Echo signals from the vertical pole at a diameter of 6 cm, placed 200 cm, away from the microphone array: a) from horizontal beams, b) from vertical beams.

The idea of the next experiment was to place a vertical or a horizontal pole of a 6 cm diameter in the observation field of the system. Figure 11 shows visible echo signals from the pole placed vertically in the axis 4 of the vertical beam. In line with the principle of the system's performance, a strong echo appeared in 4-th vertical beam and in all horizontal beams, [8]. The highest amplitude was exhibited by the echo in 4-th central horizontal beam because the direction of incidence of the acoustic wave on the pole is at that time perpendicular. The smallest echo amplitudes are observed in extreme end beams (the first and seventh) because the wave in these beams falls on the pole at an angle of about 30°. Analysis of the distribution of the echoes in the particular beams allows both a determination of the object's location ($R \approx 2\,\mathrm{m}$, $\theta \approx -18°$), as well as its identification (vertical pole).

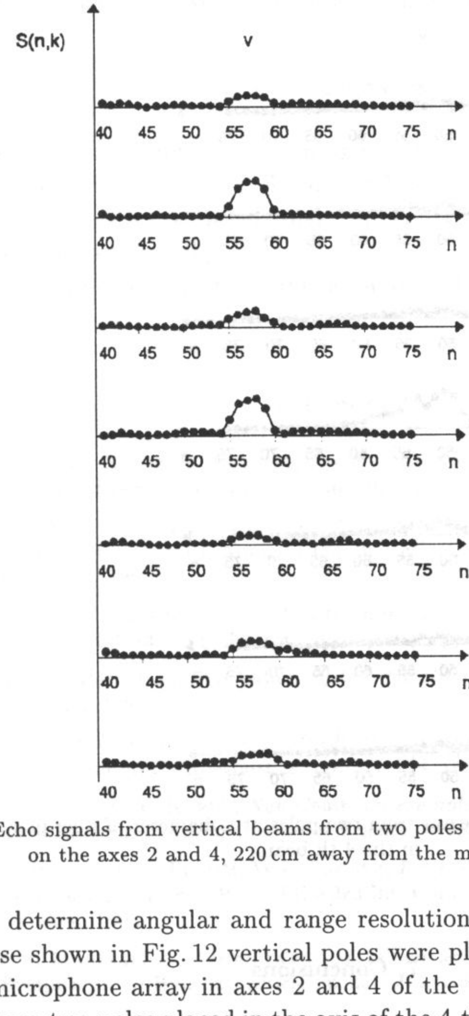


Fig. 12. Echo signals from vertical beams from two poles at a diameter of 6 cm placed on the axes 2 and 4, 220 cm away from the microphone array.

In order to determine angular and range resolution of the system, two poles were used. In the case shown in Fig. 12 vertical poles were placed at an identical distance of 2 m from the microphone array in axes 2 and 4 of the vertical beam. Figure 13 shows visible echoes from two poles placed in the axis of the 4-th beam at a distance of 2 m and 2.2 m from the microphone array. The echoes in both figures match the expectations. Therefore the system has an assumed angular resolution of about 9° and range resolution of about 20 cm([8]). Despite the fact that the design assumptions were met, the system is not fully explicit upon determining the number of observed objects and their location. As an example, when a single pole is placed at a point of the beams' intersection, the echo signals are equal in both beams. The same picture is formed when in the axes of the

([8]) The range resolution ΔR on the display is equal $\Delta R = c\Delta\tau/2$ (see footnote 1). Here $\Delta\tau$ signifies the time between well distinguishable maxima of adjacent received signals. Figure 13 shows that $\Delta\tau \cong 5T_s$, where $T_s = 4T$ is the sampling time in the system. For $T = 55.5\,\mu\mathrm{s}$ we obtain $\Delta R \cong 19\,\mathrm{cm}$.

neighboring beams two identical poles are placed. This lack of identical identifications is eliminated when the microphone array is moving (the blind person) because as we near the objects they appear in other beams. At the same time, for angular resolution, the spatial resolution of the system improves.

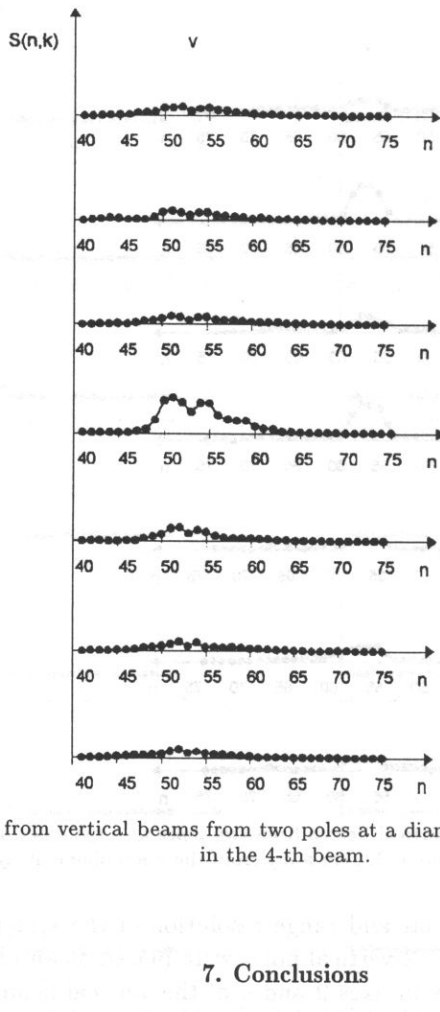


Fig. 13. Echo signals from vertical beams from two poles at a diameter of 6 cm placed at 20 cm gaps in the 4-th beam.

7. Conclusions

The acoustic echolocation system was built following the principles presented in the paper and has technical and operating parameters that enable detection, positioning and identification of obstacles that are a threat to the blind person. Currently it is being used for measurements in the field. The collected measurement data will be used to verify the algorithms that analyze the pictures observed and generate adequate verbal messages. The algorithms developed so far define the position of simple obstacles (poles, walls) that are placed in uncomplicated configurations. Verbal messages only tell the direction of the obstacles discovered. If the results of the studies are positive, further research should concentrate on reducing the size of the device and its full automation.

What is especially necessary is replacing the laptop with a specialized processor. It is estimated that given the current state of technology it is possible to fit the entire device (including the battery) to the size of a medium size book. Parallel research is being carried out on the use of a satellite navigation system which – by cooperating with an acoustic echolocation system – would facilitate walking in a vast, unknown area.

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