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Translation

OPERATION OF SHIPBOARD HYDROACOUSTIC STATIONS

By

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OPERATION OF SHIPBOARD HYDROACOUSTIC STATIONS

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[List of symbols, sections 2-5 of chapter 3, chapters 4-7, conclusion, references and table of contents from book "Operation of Shipboard Hydroacoustic Stations", by Vladlen Anatol'yevich Pokrovskiy and Gennadiy Aleksandrovich Shcheglov, Izdatel'stvo "Sudostroyeniye", 3500 copies, 192 pages]

CONTENTS

[Text] List of Symbols

α--1) probability of error of the first kind (false alarm); 2) probability of erroneous rejection of produced items (systems) due to unsatisfactory test results of some part of them (manufacturer's risk);

α_R, α_C, α_L--temperature coefficients of change in values of resistance R, capacitance C, inductance L;

β--1) coefficient of spatial attenuation; 2) probability of error of the second kind (missed signal); 3) probability of erroneous acceptance of production items (systems) due to satisfactory test results of some part of them (customer's risk);

γ--antenna concentration coefficient;

δ--discrimination coefficient in sonar equation;

$\Delta\alpha$--sensitivity angle of antenna directivity;

Δ_ϕ, Δ_r--accuracy of coordinate determination (for direction ϕ and distance r);

Δ--overall amplitude-phase nonidentity of electroacoustic antenna transducers;

$\zeta_{a.a}$--acoustic antenna sensitivity;

ζ_Σ--sensitivity of hydroacoustic facility as a whole;

η--efficiency;

ν_ϕ, ν_r--resolution of hydroacoustic facility with respect to direction and distance;

θ--width of principal maximum of directivity pattern;

κ--bulk modulus of elasticity;

λ--1) wavelength; 2) failure rate of elements and nonrestorable systems;

Λ--failure rate of restorable systems;

μ, ν--designation of limits for field of tolerance on parameter a (a_μ, a_ν);

ρ--material density;
σ--root-mean-square deviation;
τ--instantaneous time displacement, pulse duration;
ϕ--spatial angle;
ψ--phase angle;
ω--angular frequency;
a_i--1) i-th parameter of hydroacoustic facility; 2) linear dimension;
A_f--anomaly factor;
c--velocity of sound;
C--capacitance;
d--linear dimension, diameter;
D--1) range of action of hydroacoustic facility; 2) detonation coefficient;
E--Young's modulus; electromotive force (emf);
f--frequency;
$f(a)$--probability density function of instantaneous values of parameter a;
$F(a)$--distribution function;
g--acceleration of free fall;
$G(f)$--spectral density of power;
G--sensitivity of antenna to random errors;
h--depth;
H--amount of information;
J--acoustic signal intensity;
I--current;
k--coefficient;
l--linear dimension;
L--inductance;
q--signal-to-noise ratio;
Q--1) interference immunity; 2) storage factor of oscillatory tank circuit;
p--sound pressure
P--probability;
P_a--acoustic power;
r--distance, radius;
$R_{\text{э}}$--equivalent radius of target;
$R(\phi)$--directivity pattern;
S--area;
$S(j\omega)$--amplitude spectrum;
t--elapsed time
T--time interval, duration;
u--voltage;
v--velocity;
V--1) volume; 2) probability of recovery;
w--control confidence coefficient;
$W_{\text{эл}}$--electric power;
x, y, z--coordinates;
Э--efficiency.

§3.2. Instrument oscillators

In technical operation of hydroacoustic facilities, instrument oscillators are used to handle two major jobs: setting up a sound field in water jointly with the measurement emitter to determine electroacoustic parameters of the reception channel of the hydroacoustic station, and forming an electric signal calibrated with respect to amplitude, frequency, duration and so on (standard signal) to measure electric parameters, check operability of the electronic part of the equipment and locate malfunctions.

The principal technical characteristics of instrument oscillators are: the nature of the generated signal (shape, frequency spectrum and so on); dynamic and frequency ranges; graduation error (with respect to frequency, amplitude and the like) and stability of output signal parameters; output impedance; output power; coefficient of nonlinear distortions.

Each unit of the oscillator used in hydroacoustic measurements (Fig. 3.2) must meet specific requirements.

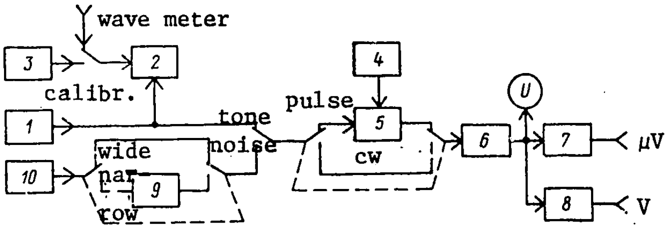


Fig. 3.2. Block diagram of instrument oscillator: 1--audio oscillator; 2--frequency meter; 3--crystal-controlled oscillator; 4--pulse modulator; 5--switching circuit; 6--line amplifier; 7--output attenuator; 8--power amplifier; 9--bandpass filter; 10--noise generator

The master oscillator determines the nature of the generated signal, the frequency band of the oscillator, the error of frequency calibration and signal frequency stability. In technical operation, hydroacoustic stations typically use sine-wave master oscillators with continuous frequency tuning, and frequency spectrum oscillators (noise generators).

Type RC oscillators are widely used as master sine-wave oscillators. The advantages of these oscillators are wide frequency band, uniform scale, high stability of signal parameters, simplicity of circuitry and construction. The accuracy of frequency setting and readings is improved by using special scales: either a mechanical scale of great length determined by the transfer number of the vernier ("GV-1M"), or an electronic scale with continuous monitoring of the signal frequency by a built-in digital frequency meter ("Generator").

The major requirement for master noise oscillators is continuity of the signal spectrum over the entire frequency range, and frequency independence of each frequency component ("white" noise). Noise generators based on thyratrons, and especially on semiconductor diodes have come to be most extensively used in special hydroacoustic equipment. The noise signals used in hydroacoustic facilities usually have a limited frequency spectrum, and therefore to increase the spectral density of the controlling noise signal, provisions may be made in the circuitry of instrument oscillators for inclusion of bandpass filters to isolate the necessary frequency band from the wide-band noise.

Pulse measurements play an important role in hydroacoustics, and therefore a pulse modulator is included in the oscillator circuit to shape square pulses via a switching circuit that are filled with sine-wave or noise signals. The pulses used in hydroacoustic measurements have durations from a few milliseconds to several seconds, and a recurrence rate of no more than a few hertz. There are no rigid requirements on the enumerated parameters, and pulsers are made in the form of simple switching devices in the output signal circuit of the master oscillator.

The signals generated by the master oscillator are sent to a preamplifier, for which the major requirement is uniformity of frequency response. This amplifier should have gain control (in the "one-volt" circuit) for setting voltage of a given value at the output. A built-in voltmeter is used to monitor this voltage.

The attenuator, or output divider, is a precision potentiometer with scale graduated in microvolts. The attenuator scale can be used after a voltage calibrated by voltmeter has been set at its input, usually equal to one volt.

From the standpoint of matching to the load, the attenuator must have minimum output resistance. The lower the internal impedance of the oscillator output as compared with the impedance of the external circuit, the closer will be the correspondence between the voltage drop across the external circuit and the reading of the attenuator scale. For practical purposes, the output resistance of the output divider should not exceed 10 ohms. To reduce the influence of electric pickups at low output voltages, the attenuator is reliably protected by a massive shield.

The power of the signal taken from the attenuator output (fraction of a volt) is totally inadequate for excitation of the emitters that produce the sound field, and therefore the signal from the preamplifier output is fed to a power amplifier. The following principal requirements are imposed on the power amplifier: uniformity of frequency response; minimum coefficient of nonlinear distortions (no more than 1-3%); capability of matching the oscillator output to different loads.

The last two requirements are interrelated since improper matching of the power amplifier output to the loads leads not only to a reduction of efficiency, but also to distortion of the sinusoidal waveshape, which inevitably produces harmonics in the signal spectrum. Matching involves ensuring equality of the normalized impedances of the power amplifier output and the emitter input. This is usually done by using a wide-band output transformer that has a sectionalized secondary winding. Depending on the impedance connected to the emitter output, more or fewer turns of the transformer winding are tapped. Matching is checked by an oscilloscope connected in parallel with the emitter, in accordance with the sinusoidal waveshape of the display.

Since emitter impedance depends on frequency, the matching of oscillator output to the load should be checked not only when changing emitters, but also when there is a change in output signal frequency. The power amplifiers of portable sets designed for signal power or more than 10 W are usually made as separate instruments.

The requirement for high accuracy of frequency readout (to 0.1%) leads to the necessity for periodic or continuous calibration of the master oscillator frequency scale. The various methods of calibration reduce essentially to measuring the frequency of the master oscillator by a crystal-controlled oscillator and frequency meter. The built-in frequency meter can also be used for measuring the frequency of external sources through a special input.

Specialized oscillators and power amplifiers are used in practical technical operation of hydroacoustic stations. Let us consider some of these.

The "GV-1M" wavemeter oscillator (Fig. 3.3) produces standard sinusoidal signals: cw, pulsed and AM. The built-in frequency meter (wave meter) of the device enables measurement of the frequency of cw signals and the fill factor of pulse signals of external sources.

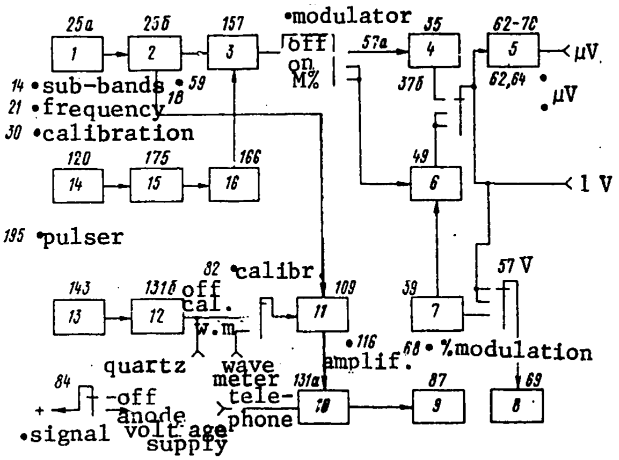


Fig. 3.3. Block diagram of "GV-1M" instrument: 1--master oscillator; 2, 12--buffer stages; 3--electronic switch; 4--linear amplifier; 5--attenuator; 6, 11, 16--mixers; 7--modulator; 8--monitoring voltmeter; 9--voltmeter display; 10--amplifier; 13--crystal-controlled oscillator; 14--clock multivibrator; 15--duration multivibrator

The oscillator of the instrument can operate in three modes: unmodulated sine waves, amplitude modulation, and pulsed.

In all modes, the master oscillator in RC circuit form generates sine-wave signals with frequency that can be continuously tuned from 0.3 to 100 kHz in five sub-bands. Over narrow limits, the frequency of the master oscillator can be tuned by the "calibration" regulator. High precision of frequency setting (0.12-0.25%) is achieved by using a long spiral scale (up to 1.2 m) calibrated by a crystal-controlled oscillator.

The one-volt potentiometer to which waveforms are sent from the master oscillator output is used to set a voltage of 1 V across the output divider. The voltage setting is monitored by a VTVM that should have the pointer set at the 1 V mark.

The oscillator has two outputs: "1 volt" from which the voltage set on the voltmeter scale is taken, and "μV" from which a voltage is taken that is continuously variable from 1 μV to 0.1 V. The impedance of the "μV" output is 10 Ω on all stages of the divider except the last (×10 000).

The pulse oscillator is connected by the pulser switch and shapes square pulses with durations of 2.5, 5, 10 and 20 ms with recurrence rate of 1 s.

The amplitude modulator is connected by the modulator switch, and provides continuously variable modulation up to 80% for a sine-wave signal on constant frequency of 400 Hz. In the M% position of the modulator selector, the VTVM is connected to the modulator output for checking percentage modulation. A high-frequency filter suppresses the audio frequency of the operating modulator.

The output divider and VTVM of the device can be used to get a calibrated voltage from an external source that does not have its own voltage divider. To do this, the signal toggle switch kills the power from all units of the device except the VTVM. Voltage from the external source is fed to the "one volt" jack, its magnitude is displayed on the 1 V scale of the voltmeter, and the required output voltage is taken from the μV plug.

The wave meter of the device is connected by the calibration selector, and operates in two modes "cal." and "w.m."

In the "w.m" mode, the wavemeter measures the frequency of the signal from an external source connected to the "w.m" input. Two signals--from the external source and from the master oscillator--are sent simultaneously to the wave meter mixer. A low-frequency filter isolates the difference frequency of the signals--the beat frequency--and the difference signal after amplification is sent to a telephone and through a detector to an electric eye cathode-ray tuning indicator tube (6Ye5S). The master oscillator is tuned until "zero" beats are achieved on both indicators, corresponding to the difference frequency. At this point, the earphones give minimum loudness, and the "pupil" of the electric eye is opened to the maximum. The value of the measured frequency is taken from the frequency scale of the master oscillator. Sensitivity with respect to the wave meter input is 0.5 V, and it can be reduced by the "amplif." regulator.

The scale of the master oscillator is calibrated in the "cal." mode. This involves establishing exact correspondence between the scale reading and the actual frequency of the master oscillator. In this mode, a crystal-controlled oscillator with resonant frequency of 5 kHz is connected to the mixer input instead of the external source. The voltage of the crystal-controlled oscillator is purposely limited, and the frequency spectrum contains a large number of harmonics that are multiples of 5 kHz; therefore zero beats are observed at corresponding points of the master oscillator scale. Less pronounced zero beats can be observed on frequencies of the master oscillator that are multiples of 2.5 kHz as well, which is attributed to the presence of second harmonics of the fundamental frequency in the voltage of the master oscillator. For example, for a scale point of 12.5 kHz, zero beats can be observed between the second harmonic of the 12.5 kHz frequency and the fourth harmonic of the quartz crystal.

For calibration, a scale value is taken that is a multiple of 5 kHz in the band in which the measurements will be made, and the "calibration" control is used to set the frequency of the master oscillator to give zero beats, which will indicate exact correspondence of frequency to the value of the scale.

The "GV" instrument differs from the "GV-1M" in scale design and calibration procedure. In calibrating this instrument, zero beats are first obtained near a scale point that is a multiple of 5 kHz, after which the cursor of the scale is brought to this point.

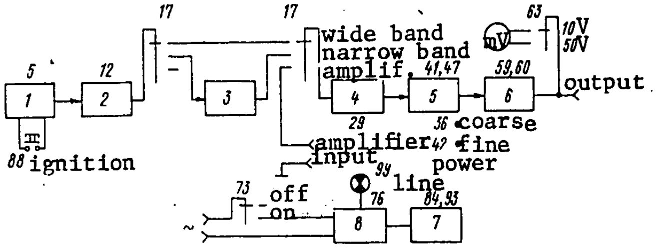


Fig. 3.4. Block diagram of "ShGU" instrument: 1--noise generator; 2, 4, 5, 6--amplifiers; 3--bandpass filter; 7, 8--power supplies

The "ShGU" noise-generator amplifier (Fig. 3.4) shapes a sound field in water jointly with an emitter incorporated into the device. Besides, the instrument can be used as a power amplifier for any electric signals in the frequency range from 0.5 to 100 kHz.

The instrument can be used in three modes: wide band, narrow band and amplifier.

In the first and second modes, the master noise generator based on a thyratron generates noise voltage in the band of frequencies from 0.5 to 100 kHz. The thyratron fires upon pressing the "ignition" switch. After preamplification, the noise signal is fed either directly to the next stage ("wide band" mode) or to the interchangeable bandpass filter ("narrow band" mode) that limits the frequency spectrum of the noise. An inverse-phase stage enables continuous control of signal power, a built-in VTVM with two measurement ranges being used to monitor the voltage sent to the emitter.

In the "amplifier" mode, the master oscillator is disconnected and an external signal source can be connected to the input jacks. After power amplification, this signal can be sent to the emitter and simultaneously taken from the output jacks. If the emitter is not connected, the power amplifier can be disconnected from work by a contact block in an output coaxial plug to protect the output tubes from the open-circuit state.

In addition to the emitter with stand, the instrument is equipped with a set of eight interchangeable bandpass filters and a boom for holding the emitter. The documentation furnished with the instrument includes curves for the mutual dependence of output voltage, sound pressure and distance.

The emitter of the instrument is a tubular magnetostriction transducer made of a tube of Nicosi alloy accommodating a permanent rectangular magnet with winding having 15 turns of wire with cross section of 0.55 mm. The resonant frequency of the transducer lies in a range of 25-35 kHz.

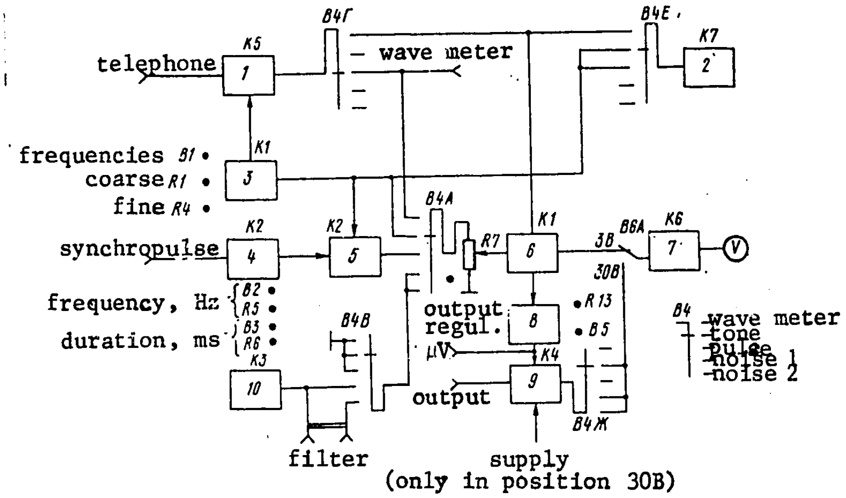


Fig. 3.5. Block diagram of "Generator" device: 1--mixer; 2--frequency meter; 3--audio-frequency oscillator; 4--pulser; 5--electronic switch; 6--buffer stage; 7--voltmeter circuit; 8--attenuator; 9--amplifier

The "Generator" oscillator-frequency meter (Fig. 3.5) produces standard cw sine-wave and noise signals as well as pulse signals with audio filler. The built-in frequency meter measures the instantaneous frequency of the master oscillator and the frequency of signals of external sources.

The technical data of the "Generator" are similar to those of the "GV-1M". The difference is the great expansion of capabilities of the pulse generator and increased precision of frequency readout.

The instrument can be used in five modes: "wave meter", "tone", "pulse", "noise 1" and "noise 2".

In the "tone" mode, sine-wave voltage from the master oscillator (audio-frequency oscillator K1) is fed from selector B4 through a plate to the frequency meter that acts as an electronic four-digit scale of the oscillator, and through a plate of the same selector to the "output regulator" (R7) that sets a voltage of 1 V on the output attenuator. The voltage setting is monitored by a voltmeter on the 3B [3-volt] scale. The attenuator design is analogous to the "GV-1M" instrument; a standard signal is taken off from the "μV" coaxial output.

The circuitry of the instrument includes a final amplifier (K4) for power amplification of the signal and transmission to an emitter connected to the output jack. Supply to the final stage is fed through switch B6A only in the 30B [30-volt] position. The voltage supplied to the emitter is simultaneously monitored in this case.

In the "pulse" mode, voltage from the output of the audio-frequency oscillator is fed to the "output regulator" through a switching circuit controlled by the pulse generator. The pulse generator includes an auxiliary "synchropulse" output for synchronizing instruments (oscilloscope and selector) connected in the channel of the measurement hydrophone.

In the "noise 1" and "noise 2" modes, a noise generator is connected to the output regulator instead of the audio-frequency oscillator, the frequency meter and pulse generator are not used, and the other circuits remain unchanged. In the "noise 1" mode, the signal from the output of the noise generator is fed directly to the output regulator, i. e. in a wide frequency band; in the "noise 2" mode it is fed via a bandpass filter formed in the "filter block" device (see §3.3). The filter block is connected to the "filter" plug on the rear panel of the instrument.

The "wave meter" mode is used for rough measurement of the frequency of cw and pulse signals from external sources by a zero-beam method, and for exact measurement of the frequency of cw signals by a digital frequency meter.

To realize the zero-beat method, mixer K5 is provided to which signals are sent from the output of the audio-frequency oscillator and from the "wave meter" plug -- from an external source (through the output regulator in the "wave meter" mode, and directly in the "pulse" mode). Headphones are used for detecting zero beats by ear.

The digital frequency meter (Fig. 3.6) of the instrument is used to measure the frequency of cw sinusoidal signals of an external source and of the master oscillator. The instrument uses the method of counting the number of voltage periods of the frequency being measured in a standard time interval.

The time-mark generator is a crystal-controlled oscillator with resonant frequency of 10 kHz. The frequency meter has three measurement ranges: ×0.1, ×1 and ×10, for which the measurement times are 10 s, 1 s and 0.1 s respectively.

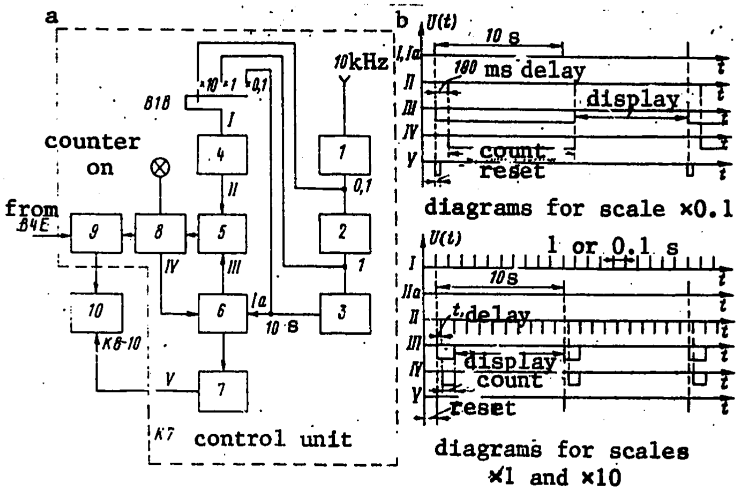


Fig. 3.6. Digital frequency meter: a--block diagram; b--voltage diagrams; 1--crystal-controlled oscillator; 2, 3--frequency dividers; 4--delay multivibrator; 5--coincidence circuit No 1; 6--preparatory flip-flop; 7--reset multivibrator; 8--controlling flip-flop; 9--coincidence circuit No 2; 10--decade counter

The integer part of the number is separated from the fractional part by a decimal point with position that depends on the selected measurement range.

On the ×0.1 measurement range, the frequency of the crystal-controlled oscillator is divided by 10^5, and pulses with interval of 10 s go to the preparatory flip-flop and delay multivibrator (diagram I and Ia). The preparatory flip-flop is flipped (diagram III) and a negative differential triggers the reset multivibrator that pulses the four-digit counter to zero (diagram V). At the same time, the preparatory flip-flop sends an enabling potential of negative polarity to one of the inputs of coincidence gate 5. The delay multivibrator delays the time-mark pulse by 100 μs, which is necessary for resetting the circuit to zero. The delayed negative pulse (diagram II) flips the controlling flip-flop for 10 s, i. e. until arrival of the next delayed pulse (diagram V). The negative enabling pulse of the controlling flip-flop is fed to coincidence gate 9, and the "counter on" light is simultaneously lit. The signal being measured goes continuously to the second input of coincidence gate 9, and for 10 s passes through the open coincidence gate to the counter. The second delayed pulse of the time-mark generator returns the controlling flip-flop to the initial state, coincidence gate 9 is blocked, counting stops and the light is extinguished. At the same time, a positive differential of the controlling flip-flop pulse through the second input returns the preparatory flip-flop to the initial state (the preparatory flip-flop does not react to the second positive pulse of the time-mark generator). The measurement result is fixed on the counter before arrival of the third pulse of the time-mark generator (display time). With arrival of the third pulse of the time-mark generator the process is repeated. Thus the total working cycle on the "×0.1" scale is 20 s.

On the "×1" and "×10" scales, the preparatory flip-flop also enables operation of coincidence gate 5 within 10 s, but since the controlling flip-flop is flipped only for the time between delayed pulses of the time-mark generator, the counter is on for 1 s or 0.1 s respectively. Thus in these ranges the total working cycle is 10 s.

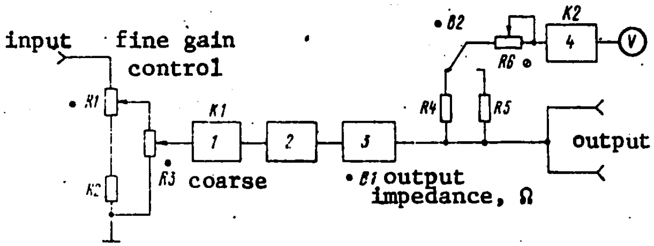


Fig. 3.7. Block diagram of "Power amplifier" set: 1--emitter follower; 2--preamplifier; 3--final amplifier; 4--voltmeter circuit

The "Power amplifier" set (Fig. 3.7) amplifies signals sent to its input and sets up the sound field in water jointly with the emitters that are part of the device.

Gain is controlled by the "coarse" and "fine" gain potentiometers. The load of the final amplifier is an output transformer with tapped secondary for matching the output of the amplifier to the load. The number of turns is determined by the setting of the "output impedance, Ω" selector. The voltage at the output of the set is monitored by a built-in voltmeter with two measurement scales.

§3.3. Sound pressure meters

Measurement of sound pressure is the most typical task in hydroacoustic measurements. The measurements are based on converting sound pressure to electric voltage by an instrument hydrophone (see §3.1). The sensitivity of instrument hydrophones is relatively low -- from units to tens of $\mu V/dyne/cm^2$ and therefore direct measurement of the emf at the hydrophone output is practically impossible. Instrument amplifiers (i. e. amplifiers with gain exactly determined throughout the frequency band) are used to amplify the emf developed by the hydrophone. An rms AC voltmeter is generally used for display.

The sound pressure p is defined as

$$p = \frac{u}{k\zeta_{1im}}, \tag{3.3}$$

where u is amplifier output voltage measured by the voltmeter, k is amplifier gain, and ζ_{1im} is the sensitivity of the hydrophone.

If k and ζ_{lim} are determined beforehand, the voltmeter scale can be graduated in units of sound pressure measurement.

The devices assembled into the block diagram corresponding to Fig. 3.7 have been given the name of "sound pressure meters."

The principal technical characteristics of sound pressure meters are: frequency response, amplifier input impedance, level of electrical set noises, gain, dynamic range, coefficient of nonlinear distortions.

The nonuniformity of the frequency response of sound pressure meters should not exceed 1-2 dB over the entier working frequency band. The shape of the frequency response curve is determined primarily by the properties of the hydrophone used in the sound pressure meter. The hydrophone can be treated as a voltage generator that has a certain internal impedance z_{int} and that develops an emf E. The input impedance of the amplifier z_{in} is the load impedance of the hydrophone. Impedances z_{int} and z_{in} must be matched on the basis of conditions of ensuring a sufficiently uniform frequency response and maximizing the ratio of signal voltage to the voltage of noises at the amplifier input.

The maximum power presented by the generator to the load is ensured by satisfying the equality $z_{int} = z_{in}$. It can be shown [Ref. 51] that for piezoelectric sensors working outside of resonance, $z_{int} \cong 1/\omega C_0$, i. e. the internal impedance of such hydrophones depends on frequency, and the condition of optimum energy matching cannot be met for a wide frequency band.

From Fig. 3.8 we can see that

$$u_{in} = \frac{Ez_{in}}{z_{in} + z_{int}}. \tag{3.4}$$

Expression (3.4) implies that to reduce the influence of z_{int} on the voltage across the load it is necessary to satisfy the condition

$$z_{in} \gg z_{int}. \tag{3.5}$$

In this case, the voltage across the load will be close to the emf developed by the hydrophone under no-load conditions.

It can be shown that satisfaction of condition (3.5) reduces thermal noises in the circuit including the hydrophone and the amplifier input. The level of electrical set noises of the amplifier normalized to the input should not exceed a few microvolts.

The internal impedance of spherical piezoceramic receivers usually does not exceed a few hundred ohms, and condition (3.5) is easily satisfied for these. It may be difficult to match high-frequency hydrophones because of the increased internal capacitive reactance for small-sized spheres.

Condition (3.5) may be violated when the connecting cable between hydrophone and amplifier is very long (more than 50 m) as a consequence of the shunting action of the distributed capacitance of the cable. Therefore in deep-water measurements preamplifiers are used that are combined with the hydrophone and power supply in a unified sealed enclosure. The preamplifier also compensates for signal attenuation in the cable and increases the signal level with respect to possible electrical pickups on the cable route. To reduce the level of acoustic interference of external sources during measurements, band-pass filters are used that are included in the instrument amplifier circuit. The passband of the filters is selected on the basis of the condition of undistorted transmission of a signal that has a finite spectrum. For example, for undistorted transmission of a pulse signal of duration τ the passband of the amplifier may be selected from the condition $\Delta f = 1.5\tau$.

The gain of the instrument amplifier must be exactly defined throughout the working frequency band since it is only in this case that the scale of the instrument can be calibrated in sound pressure units in accordance with formula (3.3). Therefore the gain is adjusted by calibrated attenuators and provision is made in the sound pressure meter circuit for calibrating the amplifier to compensate for variation of gain due to aging of components, influence of external conditions and the like. Calibration is done with the hydrophone connected to the amplifier input.

The dynamic range of the amplifier determines the range of input signals that can be transmitted without distortion to the display. The dynamic range of the amplifier that is part of the sound pressure meter must not be less than the dynamic range of the hydrophone, i. e. as wide as 120-140 dB. The dynamic bandwidth is determined mainly by the properties of the amplifier input stage in which sufficient gain reserve must be ensured at minimum interference level.

Deviations of the amplitude response from linear within the dynamic range give rise to nonlinear signal distortions. These are evaluated by the coefficient of nonlinear distortions

$$k_{\text{н.и}} = \frac{\sqrt{u_2^2 + u_3^2 + \ldots}}{u_1} \cdot 100,$$

where u_1, u_2, u_3,... are the effective values of the first, second, third, etc. harmonics of the signal being studied. The coefficient of nonlinear distortions of sound pressure meters must not exceed 1-2%.

electronic voltmeters incorporated into the instrument amplifier are used as the displays in sound pressure meters. The voltmeter circuit must provide capabilities for measuring voltages of both sine-wave and noise processes. This requirement is met by rms voltmeters with readings corresponding to the effective value of the measured voltage regardless of waveshape.

Pulsed sound pressures are usually measured by an electronic oscilloscope connected to a special output of the instrument amplifier.

"IZD" [sound pressure meter] and "Voltmeter" sets are most extensively used in practical operation of hydroacoustic facilities.

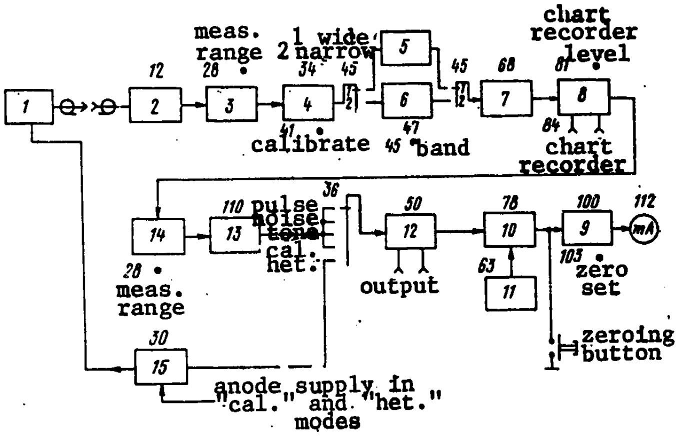


Fig. 3.8. Block diagram of "IZD" set: 1--hydrophone; 2, 4, 7, 12, 13--amplifiers; 3, 14--attenuators; 5, 6--filters; 8--cathode follower; 9, 10, 11--voltmeter circuit; 15--calibration oscillator

The "IZD" set (Fig. 3.8) is used for measuring both continuous and pulsed sound pressures. This set includes a spherical hydrophone, instrument amplifier and VTVM unified into a single instrument, and eight interchangeable bandpass filters.

The first amplification stage is based on a tube with high transconductance, providing a dynamic range of up to 120 dB. The lower limit of measurable sound pressure is 1 dyne/cm^2. A step attenuator provides signal attenuation by 120 dB in 10 dB steps.

The frequency response curve of the amplifier is shaped by interchangeable bandpass filters that are connected in a break in the signal circuit in the "narrow" mode. In the "wide" mode the amplifier passes a wide frequency band that is limited on the low side (300 Hz) by high-frequency filters. The amplifier circuit includes two special outputs for connecting a chart recorder, an oscilloscope or other instruments. The signal goes to these outputs through cathode followers; the level of the signal sent to the chart recorder is adjustable.

The voltmeter circuit includes a diode rectifier, an integrating circuit and a measurement bridge with meter indicator connected in one of the diagonals. The voltmeter characteristics are determined by the values of the components that make up the integrating circuit; the combination of components can be varied by switching the instrument to the different measurement modes: "pulse", "noise" and "tone". The amplitude of singly recurring sound pressure pulses can be measured in the "pulse" mode. In this mode, a storage capacitor is nearly instantaneously charged through the voltmeter rectifier, and the voltage across the capacitor is measured by the voltmeter. After each measurement, the capacitor must be discharged by pressing the zeroing button. In the absence of a signal, the measurement bridge is balanced by the "zero set" control.

The display scale is graduated in dynes/cm^2 and dB. The zero level of the (logarithmic) scale is taken as a sound pressure of 1 dyne/cm^2. The instrument readings taken from the logarithmic scale are summed with the "meas. range" index of the step attenuator. Due to considerable nonuniformity of the "IZD" frequency response, the readout results must be corrected by a calibration curve given in the instructions for the instrument. The calibration curve is plotted with the hydrophone connected, and therefore no other hydrophone can be used than the one included with the set.

The gain of the "IZD" amplifier is 100 (40 dB). To compensate for the change in gain resulting from aging of components, changing tubes and so on, provision is made for calibrating the instrument by a special heterodyne. In the "het." mode a calibration signal is sent directly to the voltmeter, where it is measured. In the "cal." mode, a signal from the heterodyne goes to the amplifier input through the connected hydrophone. The position of the 40 dB attenuator corresponds to gain of unity, and the readings of the instrument in the "het." and "cal." modes should coincide. Otherwise, the amplification of the set must be adjusted by the calibrate control until the readings in both modes agree exactly.

In contrast to the "IZD", the "Voltmeter" set can be used as a sound pressure meter, rms AC voltmeter and instrument amplifier. In addition to the measuring instrument, the "Voltmeter" device includes a set of three spherical hydrophones for measurement in different frequency bands.

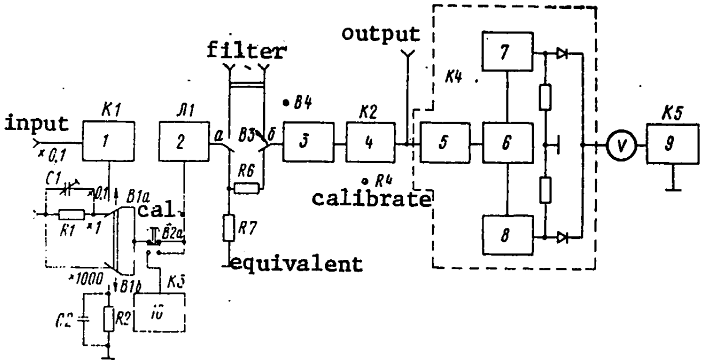


Fig. 3.9. Block diagram of "Voltmeter" set: 1--preamplifier; 2, 7, 8--buffer stages; 3--attenuator; 4, 5--amplifiers; 9--squaring circuit; 10--calibrator

The instrument (Fig. 3.9) has three measurement scales: "×0.1", "×1" and "×10". Measurements on the "×0.1" scale use a preamplifier with gain of 10.

For operation on the "×10" scale, a voltage divider based on resistors R1 and R2 is connected in the signal circuit.

Provision is made in the amplifier circuit for connecting bandpass filters that are unified into a separate "Filter block". The main amplifier of the "Voltmeter" set has a gain of 1500, which can be regulated over a wide range by the "calibrate" control. An oscilloscope can be connected to the output plug when making pulsed measurements. An rms voltmeter (units K6 and K7) is used in the set.

The amplifier is calibrated by a pulse signal from a calibration oscillator (unit K3) that generates square-wave voltage with stable amplitude of 100 mV. The signal is sent to the amplifier input by pressing the "cal." button.

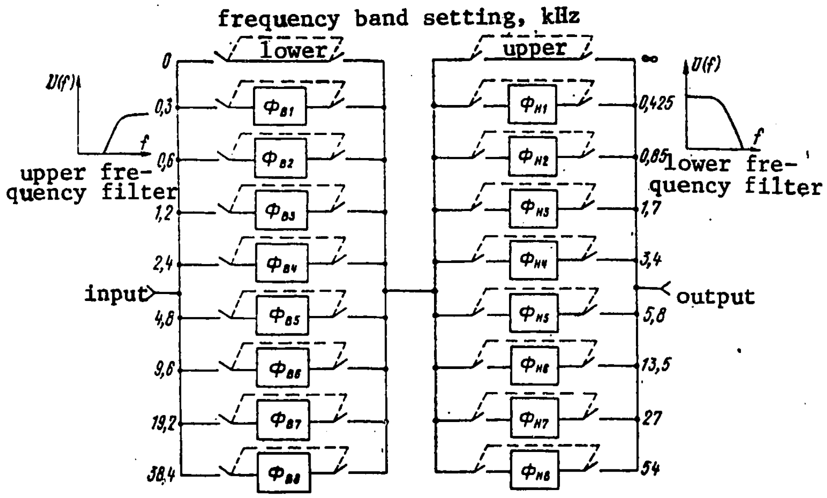


Fig. 3.10. Block diagram of "Filter block"

The "Filter block" is used in combined operation with the "Voltmeter" or "Generator" (see §3.2). This device (Fig. 3.10) is a set of series-connected lower-frequency and upper-frequency filters with cutoff frequencies in each group differing by 1 octave, and the cutoff frequencies of the groups staggered by ½-octaves. By switching filters in the necessary combination, the passband of the amplifier can be restricted to a ½-octave.

§3.4. Display and recording devices

Display and recording devices are used for observing, measuring and recording processes at the output of hydroacoustic stations.

In the operation of hydroacoustic facilities, electronic voltmeters and oscilloscopes are extensively used as displays, and signal-level chart recorders and tape recorders are widely used for purposes of registration.

Electronic voltmeters are used to measure voltages resulting from conversion of acoustic processes to electrical, and also for monitoring the working state of hydroacoustic stations by taking voltage measurements at control points. Electronic voltmeters are used in combination with with hydrophones for measuring sound pressure (see §3.3).

A specific requirement to be met by voltmeters used in hydroacoustic work is the capacity for measuring voltages of complicated waveshapes. This requirement is satisfied by rms voltmeters. In addition, in order to be capable of measuring voltages at different points of hydroacoustic stations, the voltmeters used must be universal, i. e. suitable for measuring both DC and AC voltages.

In §3.3 we considered the specialized "Voltmeter" set that is an electronic AC voltmeter. Monitoring and measurement instrumentation uses a general-purpose type "IU" voltmeter (signal level indicator).

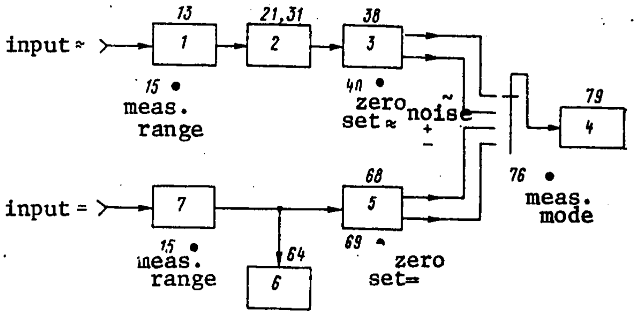


Fig. 3.11. Block diagram of "IU" instrument: 1, 7--output circuits; 2--amplifier; 3, 4, 5--voltmeter circuitry; 6--limiter

The signal level indicator (Fig. 3.11) measures the rms value of sinewave and noise voltages and DC voltages in the circuits of hydroacoustic stations. It has measurement ranges for AC voltages from 0.03 to 100 V, and for DC voltages from 1 to 1000 V. A potential of 10 mV is taken as the zero level of the logarithmic scale.

The functional circuit of the "IU" instrument contains separate DC and AC voltmeters and a common display. The AC voltmeter circuit contains a cathode follower loaded by a voltage divider, a wide-band amplifier and a square-law detector. The use of a cathode follower with low internal impedance ensures linearity of the stage up to 100 V, obviating the need for a high-impedance input divider with large capacitance. The low input capacitance keeps the high input impedance of the voltmeter constant at 0.5 MΩ up to a frequency of 100 kHz. In turn, the frequency response of the low-resistance (30 kΩ) voltage divider that is used is nearly independent of the capacitance of the following stage.

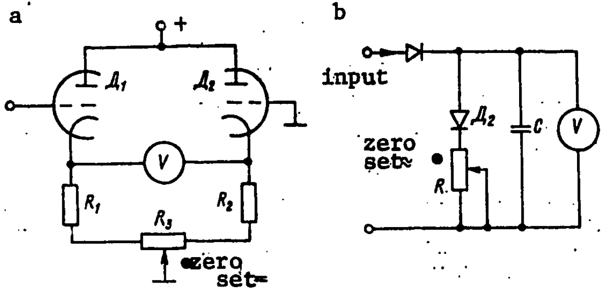


Fig. 3.12. Diagram of voltmeter in "IU" set: a--AC; b--DC

In measuring AC voltage, a square-law half-wave detector is used (Fig. 3.12) that is based on one half of duodiode $Д_1$. The second half $Д_2$ is used for compensating the initial current of $Д_1$ under no-signal conditions. Meter V is connected to the output of the instrument through the voltmeter "meas. mode" selector in positions "≈" or "noise" (see Fig. 3.11). In the "noise" position, capacitance C (Fig. 3.12) is multiplied by six, giving a display time constant of 4-6 s.

A bridge measurement circuit is used in the DC voltmeter (Fig. 3.12b). To limit the current through the display meter, a diode limiter is included that shunts the input of the bridge circuit when the voltage across it exceeds 3.8 V. For convenience in using the instrument, a selector is provided for switching polarity of the display meter in accordance with the polarity of the input signal.

Electronic oscilloscopes are widely used for measuring amplitude and duration of pulse signals, for determining the frequency of signals by Lissajou figures, and also for on-the-spot monitoring of signal waveshape to detect nonlinear distortions, clipping distortions and the like.

Hydroacoustic measurement practice uses general-purpose oscilloscopes that must meet special requirements, and in particular the frequency responses of amplifiers must permit operation at comparatively low frequencies (from DC to hundreds of kHz), the CRT is selected to have long persistence, and a horizontal scanning range that is matched to the time parameters of pulses to be measured (durations from a fraction of a millisecond to 10 seconds).

These requirements are met most completely by the S1-4 and S-19 instruments, and by other low-frequency oscilloscopes. Monitoring and measurement instrumentation includes the "OI" (oscilloscope meter).

The "OI" instrument (Fig. 3.13) enables observation of waveshape and measurements of amplitude and duration of AC pulses; observation of waveshape and measurements of amplitude of periodic signals; frequency measurement by the Lissajou figure method in conjunction with the "GV-1M" instrument or some other audio-frequency oscillator.

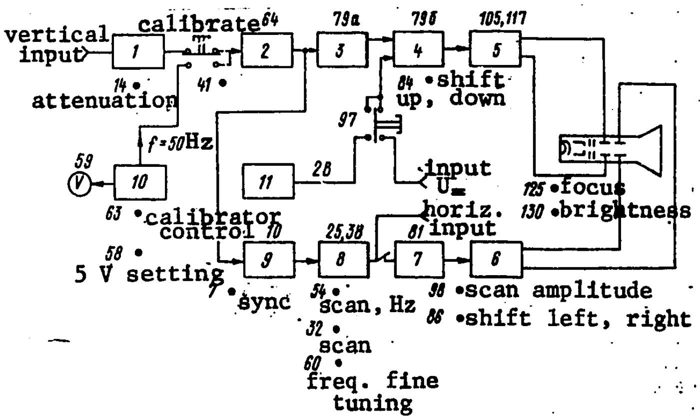


Fig. 3.13. Block diagram of "OI" instrument: 1--attenuator; 2, 3, 5, 6, 9--amplifiers; 4, 7--cathode followers; 8--scan oscillator; 10--AC calibrator; 11--DC calibrator

The process to be studied is sent to the "vertical input" terminals. The input attenuator ["attenuation"] permits signal reduction by factors of 10, 100 or 200. First-stage amplification in the vertical deflection amplifier can be controlled by adjusting a fine-tuning gain potentiometer.

The amplitude of the signal arriving at the "vertical input" terminals can be measured by an amplitude meter. When the "calibrate" button is pressed, the investigated signal going to the amplifier input is replaced by a voltage of controllable magnitude on the line frequency, the amplitude being indicated on a built-in meter. By using the "calibrator control" potentiometer to adjust the calibration voltage, the image on the screen is made to coincide with the investigated signal, after which a meter reading is taken. The voltage fed to the instrument input is determined by multiplying this reading by the input attenuator index.

The amplitude of DC pulses is measured by using a special input on a side panel of the instrument from which a signal is sent to the cathode follower of the vertical deflection amplifier. The measurement is taken by comparing the magnitude of beam dflection under the action of the investigated signal with the calibration voltage of +2 V that is sent in place of the signal when a button is pressed under the input jack.

The horizontal scanning generator provides five modes depending on type of scanning: continuous, driven-sweep with variable duration, and driven-sweep with three fixed durations. The frequency of continuous canning and the duration of the driven sweep can be regulated in stages by using the "scan, Hz" selector.

Horizontal scanning is synchronized by a signal sent from the first stage of the vertical deflection amplifier through a synchronization amplifier with gain that can be controlled by the "sync" potentiometer. The gain of the horizontal deflection channel is adjusted by the "scan amplitude" potentiometer. When a scanning voltage is sent from an external source, the scanning generator is automatically disconnected by a microswitch installed in the "horiz. input" jack.

It should be borne in mind that the "shift left, right" potentiometer acts only in the continuous and variable driven-sweep scanning modes. In the fixed driven sweep mode the spot goes off-screen, and the leading edge of the observed pulse is brought into registration with the zero point of the scale by the "scan amplitude" potentiometer.

Signal level chart recorders are used in case of necessity for prolonged recording of the envelope of an investigated process (i. e. the signal level). The advantage of chart recorders is the capability for keeping a graphic record directly during measurement, enabling on-the-spot monitoring of research results. In operating hydroacoustic stations, the use of signal level chart recorders in conjunction with other instruments enables automating a number of labor-intensive jobs such as measuring directivity patterns, reception interference level, frequency and spectral characteristics and the like.

The most widely used chart recorders in hydroacoustic measurements are the Soviet N-110, and the Danish Bruel and Kjor instruments types 2304, 2305 and 2307. The design of these instruments is essentially the same, reducing to an automatic control circuit that enables continuous tracking of changes in the level of voltage sent to the input of the instrument.

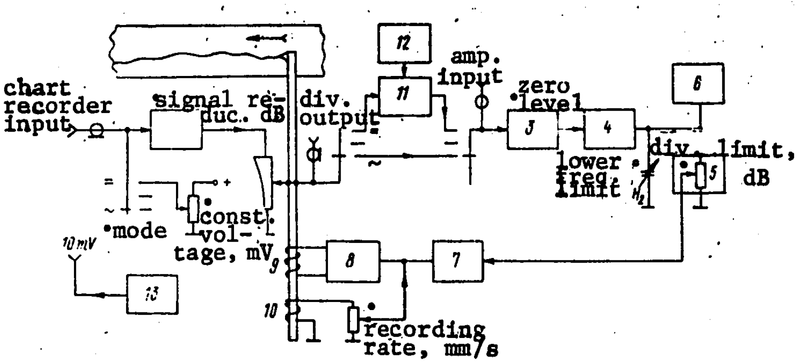


Fig. 3.14. Block diagram of N-110 chart recorder: 1--input attenuator; 2--functional divider; 3--amplifier; 4--detector; 5--adder; 6--reference voltage source; 7--limiter; 8--DC amplifier; 9--stator winding of motor; 10--rotor winding of motor; 11--vibrotransducer; 12--400 Hz oscillator; 13--10 mV calibrator

The N-110 chart recorder (Fig. 3.14) has three operating modes determined by the position of the "mode" switch:

--recording voltage level on audio and ultrasonic frequencies (from 20 Hz to 200 kHz) (selector in "~" position);

--recording instantaneous voltages that vary slowly in the range from 0 to 5 Hz (switch in "=" position);

--mode of automatic stabilization of the level of physical quantities such as sound pressure that have been converted to voltage (selector in middle position).

As a rule, only the first mode is used to measure parameters of hydroacoustic stations.

In all modes the signal goes to the attenuator input that attenuates the signal to 50 dB in 5-dB steps. In all attenuator positions the input impedance of the chart recorder remains constant at 40 kΩ. The voltage from the attenuator output is sent to functional voltage divider 2, which is a wire-wound tapped potentiometer that acts as the sensing link in the automatic control system. The voltage taken by the sliding contact from part of the functional divider is amplified by AC amplifier 3, rectified by detector 4 and sent to adder 5 in negative polarity; the adder also receives DC reference voltage in positive polarity from reference voltage source 6.

The difference signal taken from the adder output is amplified by DC amplifier 8 and sent to motor winding 9. Fastened to one side of the motor armature is the recording pen, and on the other side is the sliding contact of the functional divider. The armature shifts the sliding contact in the direction that reduces the voltage difference at the adder output. Motion stops when the rectified signal voltage becomes equal in absolute magnitude to the constant reference voltage. Any change in the input voltage causes a motor control signal to appear at the adder output, shifting the sliding contact to the point of the functional divider where the signal voltage corresponds to the compensation voltage. In this way, the voltage across the sliding contact is held constant during operation. This property of the chart recorder is used in the mode of stabilization of physical quantities; the "div. output" and "amp. input" plugs are provided in the circuit for realization of this mode. Obviously in the absence of an input signal the sliding contact will be moved by the reference voltage to the end position (the upper position in the diagram), i. e. it will lock out, while application of the compensation voltage to the functional divider will shift the pen to the zero point of the scale. For the N-110 recorder, this voltage is 10 mV, and readings of the voltage level are taken from this point in decibels on a logarithmic scale. In the "=" mode, the contacts of vibrotransducer 10 driven by oscillator 12 that generates voltage on a frequency of 400 Hz are switched into the signal circuit. Thus in this mode a DC signal is converted to AC voltage. The initial recording level is selected by feeding DC voltage from a special source through the "const. voltage" control.

The nature of the scale is determined by the law of resistance distribution lengthwise of the functional divider. The chart recorder is equipped with a set of interchangeable dividers: three logarithmic having a dynamic range of 25, 50 and 75 dB, and one linear with dynamic range equal to that of [vibrotransducer] 11. The dynamic range D_u for the logarithmic scales is defined by the expression $D_u = 20\,\lg(u_{max}/u_{min})$, where u_{max} and u_{min} are the voltages corresponding to the final and initial readings of the scale ($u_{min} = 10$ mV). For the linear divider, the dynamic range T is defined as $T = u_{max}/u_{min}$. The level of the voltage fed to the chart recorder input relative to $u_{min} = 10$ mV is defined as the sum of the reading taken from the chart recorder strip and the reading corresponding to the selected position of the input attenuator. To convert the result to a zero level equal to 1 V, 40 dB must be subtracted.

It should be emphasized that normal operation of the chart recorder depends primarily on the state of the commutator field and the sliding contact of functional divider 2. Corrosion, contamination and mechanical damage of these components cause an abrupt increase in measurement error, and disrupt stable operation of the follow-up system.

Amplifier 3 of the chart recorder has a "zero level" control that can be used for gain calibration of the amplifier. For calibration, a voltage of 10 mV is fed to the chart recorder input from an internal source. Calibration is done by setting the pen to zero on the scale with the input attenuator completely removed.

Capacitor C is connected in parallel with the detector load, and realizes time averaging of the rectified signal. A change in value of this capacitor shifts the lower limit of the chart recorder passband. An increase in capacitance expands the frequency response in the low-frequency region and reduces the static error of measurement. On the other hand, increasing the capacitance increases the time lag of the follow-up system, resulting in an increased dynamic tracking error, smoothing of level fluctuations, limiting pen displacement velocity (recording rate) and impairing stability of the system. Thus the position of the "lower freq. limit" control must be chosen to conform to the nature of the process.

Potentiometer R is the detector load, and at the same time performs the function of adding the signal and reference voltages. The "div. limit" control has been introduced to compensate for the change in transfer factor of the chart recorder when functional dividers are changed. The transfer factor of the chart recorder increases with expansion of the dynamic range of the divider. An extreme increase in the transfer factor may lead to intensification of self-oscillations of the system, while reduction smooths out the processes being recorded. Therefore the position of the "div. limit" control must be matched to the rating of the working functional divider.

The last link of the follow-up system--the actuating motor with DC amplifier 8 --is covered by negative velocity feedback. The feedback circuit includes the tachogenerator motor winding 10, and the "recording rate" feedback depth control. The velocity of armature movement increases as feedback weakens. An increase in recording rate is advisable only if it has not been previously restricted by capacitance C. An extreme increase in recording rate may put the system out of the stable state, and therefore the positions of the "lower freq. limit" and "recording rate" controls must be matched. The maximum possible recording rate is determined by control signal limiter 7.

The N-110 chart recorder leaves a record on paper tape by scratching a magnesium coating with a needle. The paper is 50 mm wide. Paper of two types is used: A--with black backing and white coating to get a contrasty image, and B--with transparent backing and red coating to produce photocopies by contact printing. The paper is moved at constant velocity by a tape-transport mechanism driven either by its own electric motor, or by an external device through a special shaft. The tape transport speed can be varied from 0.003 to 100 mm/s. The tape transport mechanism can be used to activate a variety of mechanisms used in conjunction with the chart recorder such as a frequency spectrum analyzer, audio-frequency oscillator and the like. To do this, two rollers with right-hand and left-hand rotation are provided with speeds that can be varied from 0.108 to 3600 rev/hr. The set has a cam switch driven by one of the rollers and enabling two processes to be sent alternately to the chart recorder input.

Tape recorders, or instruments for magnetic recording and playback of acoustic waveforms are extensively used for recording signals of complex shape. Their advantage is the capability for repeated playback of a stored process without repeating a complicated experiment. This is particularly convenient in cases where the environment in which the recording is done does not permit the use of cumbersome analyzing and recording equipment. In hydroacoustic measurement practice, tape recorders are used to record the noises of ships, hydroacoustic interference, and signals with the aim of subsequent processing and analysis of the recording in the laboratory.

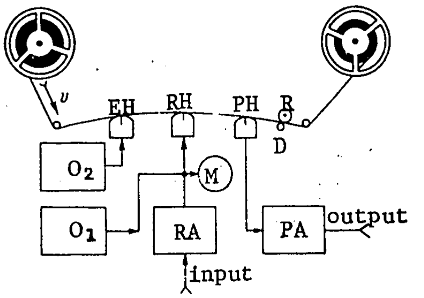


Fig. 3.15. Block diagram of tape recorder

The principle of magnetic recording (Fig. 3.15) is based on altering the magnetic state of a sound-recording medium by using an alternating magnetic field. The medium -- magnetic tape -- is moved at constant velocity past three heads: EH--erasure; RH--recording; PH--playback. The tape is moved by a drive roller D against which the tape is pressed by pressure roller R. Alternating current from the output of recording amplifier RA flows through the winding of the recording head. The voltage of the signal to be recorded goes to the input of the recording amplifier. The magnetic field of the recording head magnetizes the tape. Thus a time change in current is recorded on the tape as a a change in residual magnetization with respect to length. To ensure linear dependence between the signal current and the residual magnetization, current of higher frequency is simultaneously sent to the recording head from magnetizing oscillator O_1.

Residual magnetization forms a magnetic field around the tape, and as this tape passes the playback head, the field induces a flux in the core of the head that is proportional to the residual magnetization. The emf induced in the winding of the head in proportion to the rate of change of the flux is amplified by playback amplifier PA and transmitted to reproducers as a copy of the signal recorded on the tape.

Before recording, the tape must be demagnetized, i. e. the previous recording must be removed. The function of tape demagnetization is performed by the erasure head. Current is sent to the winding of this head from erasure oscillator O_2 that generates a waveform with frequency much higher than the upper frequency of the signal to be recorded and with amplitude sufficient to saturate the tape.

Thus the following principal components can be distinguished in any tape recorder: sound-recording medium, tape-transport mechanism, magnetic heads, recording and playback amplifiers, magnetizing and erasure oscillators and reproducers. As a rule, the recording and playback functions in commercial tape recorders are performed by general-purpose heads and amplifiers, and the erasure and magnetizing currents are generated by the same oscillator.

The sound-recording medium for most present-day tape recorders is powder-oxide magnetic tape. Most extensively used are two-layer tapes consisting of a base and a working layer. The base material is acetylcellulose or Mylar film. Ferric gamma-oxide and iron-cobalt ferrite ($CoFe_2O_3$) are used for making the working layer. Particle size is no larger than 0.4 µm. The powder is applied to the surface of the backing by a special lacquer. The working side of the tape is polished to reduce head wear due to abrasive action of the film, and also to enhance sensitivity and improve lengthwise uniformity of sensitivity. The nonworking side of the tape is dull to improve winding quality.

Existing standards provide for production of magnetic tapes with total thickness of from 12 to 55 µm. Thickness of the working layer ranges from 4 to 16 µm. The trend toward decreasing tape thickness results from the attempt to maximize the duration of continuous recording. The limiting factor in this trend is the tensile strength of the tape, and also extensibility under load. The standard width of magnetic tape is 6.25 mm. The use of old tapes 6.5 to 6.35 mm wide may be detrimental to the directional properties of tape recorders. Cassette tape recorders use tapes 3.81 mm wide.

Magnetic tape quality is characterized by the following principal electro-acoustic parameters: sensitivity and nonuniformity of sensitivity, frequency response, level of nonlinear distortions and noise level.

Sensitivity (efficiency) characterizes the ratio of the residual magnetic flux to the intensity of the low-frequency field of the recording head. In practice, relative sensitivity is taken as the ratio of sensitivity of the tape that is used to the sensitivity of a standard tape, expressed in decibels. Variation in uniformity of sensitivity by ±1 dB is heard as a change in loudness, and is unacceptable for using tapes in measurements. Tape sensitivity increases with increasing thickness of the working layer.

Frequency response of tape depends on many factors. Chief among these are the "penetration effect" and "self-demagnetization", which show up as a drop in frequency response in the high-frequency region. In the self-demagnetization effect, elementary sections of the magnetic tape are demagnetized by an external field with lines of force directed contrary to the internal flux.

The intensity of this effect increases with decreasing ratio of the length of an elementary magnet to its thickness, i. e. the shorter the wavelength of the signals being recorded. The penetration effect shows up as a reduction in depth of penetration of the magnetic field into the working layer of the tape with increasing signal frequency. This effect is compensated by reducing the thickness of the working layer.

In practice, the frequency response of the tape is defined relative to some standard tape as the difference between the ratios of efficiency on the upper cutoff frequency to efficiency on the reference frequency, which is taken on a linear section of the frequency response curve for both tapes.

Nonlinear distortions of the magnetic tape make the greatest contribution to the overall nonlinear distortions that arise during magnetic recording. These distortions are due to nonlinear dependence of remanent induction on magnetic field strength. Linearity of this dependence is considerably improved by sending a high-frequency magnetizing current with amplitude considerably greater than that of the signal being recorded to the recording head simultaneously with the signal. Tape sensitivity during magnetization is considerably improved together with the increase in linearity of the remanent magnetization curve [Ref. 31]. The magnetization at which tape sensitivity is maximum is called optimum magnetization.

The magnitude of distortions is usually evaluated with respect to the level of the third harmonic expressed in percent relative to the level of the fundamental frequency.

Tape noises show up as an emf that appears at the output of the playback head when a tape passes that contains no recording. They are caused principally by nonuniformity of the structure of the working layer.

Tape noise can be reduced by shielding the tape and heads from the action of external magnetic fields, balancing the magnetizing current and setting current at the optimum value.

One special kind of tape noise is print-through. The essence of this effect is in mutual magnetization of tape layers in contact on the reel. During playback, the signal will be accompanied by weaker copies of itself that may lead and trail. The relative level of print-through on present-day tapes is from -50 to -56 dB.

In measurement practice, consideration must be taken of such a tape property as reduction of the recording level after the first few playbacks. For example after the first playback the recording level decreases by about 30%; the falloff in recording level stops after about ten playbacks. Therefore, in especially important cases a control signal of known level should be recorded on the tape.

The tape-transport mechanism ensures high constancy in the rate of movement of the magnetic tape past the heads in the recorder. Standards provide for the following nominal tape speeds: 38.1, 19.05, 9.53, 4.76 and 2.38 cm/s.

The instability of tape motion due to different mechanical components of the tape recorder is characterized by wow and flutter defined as

$$Dr = \frac{\Delta v_{max}}{v} \times 100\%,$$

where Δv_{max} is maximum drift of tape speed from nominal, and v is nominal tape speed.

Wow refers to fluctuations of tape speed of less than 10 Hz frequency, and flutter refers to fluctuations at a frequency greater than 10 Hz. Wow causes periodic shifting of the frequency spectrum and is heard as "wavering" of the sound; flutter causes frequency "chopping" and is perceived as "warbling" of the sound. The wow and flutter of laboratory tape decks should not exceed ±0.1%.

In addition to handling its main job, the tape transport mechanism also provides fast-forward and fast-rewind modes, high-quality reeling and fast stop.

The recording head is a ring-type electromagnet made up of two half-rings. The forward or working gap contacted by the moving magnetic tape is filled with a thin layer of nonmagnetic material. The width of the gap is made approximately equal to the thickness of the working layer. A relative reduction in gap width leads to frequency distortions due to the "penetration" effect, and an increase in gap width increases nonlinear distortions. The possibility of overloading the recording head is reduced by a rear gap in the core that increases reluctance. In addition, the rear gap reduces the level of noises that arise as a consequence of remanence of the core.

The playback head is analogous in design to the recording head. The difference is absence of a rear gap since the low currents in playback cannot cause saturation of the core.

The playback head is the component that makes the greatest contribution to nonlinearity of frequency response of the record-playback channel.

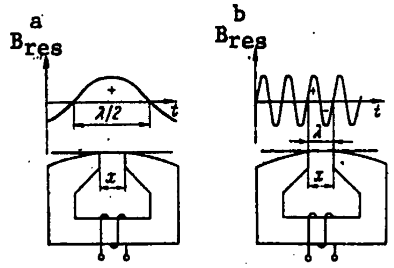


Fig. 3.16. Effect of gap distortions: a--wavelength greater than gap width; b--wavelength equal to gap width

One of the major causes of frequency distortions in playback is the finite size of the head gap. It can be seen from Fig. 3.16 that as the wavelength λ of the recorded signals decreases, the emf induced in the head decreases, and vanishes when a whole wave fits into a section of the tape equal to the width x of the gap. This can be attributed to cancelling of the oppositely directed magnetic fluxes of elementary sections of the tape of length λ/2. These are called gap distortions. The position of the first minimum that occurs on the frequency response curve as a result of gap distortions is determined by the ratio of the width of the working gap of the head to the wavelength of the signals recorded on the tape x/λ. Wavelength is defined as $\lambda = v/f$, and consequently the upper frequency limit of magnetic recording can be raised by reducing the width of the working gap and increasing tape speed. In the general case, the width of the gap should not be greater than half the wavelength of the upper frequencies of the signal. Frequency distortions similar to gap distortions arise when the working gaps of the heads are skewed; they should be perpendicular to tape motion and parallel to each other.

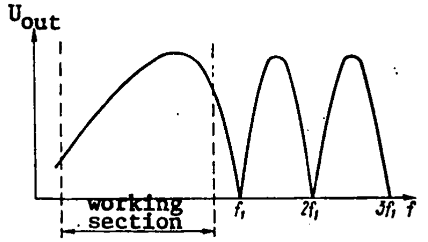


Fig. 3.17. frequency response of the record-playback channel

A second cause of frequency distortions is direct dependence of head efficiency on frequency $E = \omega\Phi_m \sin \omega t$, where ω is angular frequency, and Φ_m is the amplitude value of magnetic flux.

The resultant frequency response of the record-playback channel is shown in Fig. 3.17. The falloff of the curve at low frequencies makes it difficult to play back infrasonic signals.

The erase head differs in a much wider working gap (100-250 μm) and higher current (tens of mA). During travel past the head, each section of the magnetic medium is subjected to the alternating magnetizing action of a magnetic field that rises smoothly to saturation and then falls to zero. The frequency of the erasure current is selected so that the magnetizing force changes by no more than 1% during a period.

In addition to their usual function of signal amplification, the record and playback amplifiers also handle the no less important job of correcting the frequency response of the channel comprising the recording head, sound-recording medium and playback head (Fig. 3.18). The amplifiers in tape recorders used for measurement purposes must be equipped with calibration devices, step attenuators for amplification control, and meters for measuring record and playback levels.

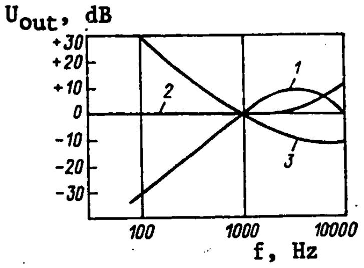


Fig. 3.18. Amplifier frequency responses: 1--record; 2--playback; 3--resultant

It should be borne in mind that the same tape recorder must be used for recording signals and for playing them back during analysis since each recorder has its own frequency response peculiarities.

§3.5. Equipment for analyzing phase and spectral characteristics

The basis of directed action of hydroacoustic stations consists in accounting for and making use of the phase structure of the acoustic signal field. Calculation and design of receiving and sending acoustic antennas and devices for directional action boils down to calculating phase relations in the electric circuits of equipment and antenna structures. Therefore rather severe requirements are imposed on the phase characteristics of hydroacoustic equipment, and particular attention must be given to checking them. Phase meters are used for analyzing the phase characteristics of hydroacoustic equipment.

Phase meters measure the phase shift between two voltages, and in hydroacoustic measurement practice they are used to verify phase identity of multichannel receivers and amplifiers. Besides, phase meters may be incorporated into sound speed meters based on using the phase method of measurement. And finally, nearly all hydroacoustic direction finders are devices that in one way or another measure the phase difference of acoustic waveforms received by separate components or reception groups of an acoustic antenna.

Two techniques are used for measuring phase difference in hydroacoustic technology: the compensation method, and the method of converting a phase shift to a proportional time interval.

The compensation method involves using a graduated phase shifter to cancel the phase shift between two voltages. Phase equality is determined by a null indicator, and the sought phase shift is read out from the scale of the phase shifter. The method is realized in hydroacoustic stations for determining the direction to a signal source, where the phase shifter is a compensator coupled to the course angle scale.

The method of converting a phase shift to a proportional time interval is used in low-frequency phase meters for direct measurements in the circuits of hydroacoustic stations. One of these, named the "Phase shifter", is included in a special set.

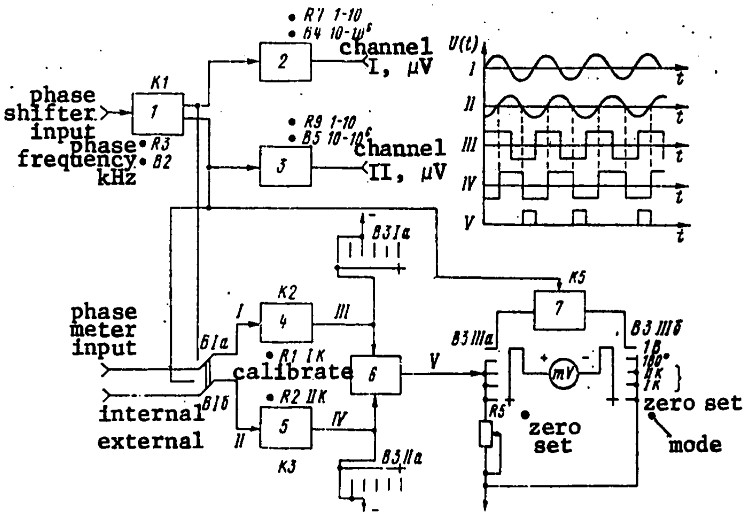


Fig. 3.19. Block diagram of "Phase shifter" set: 1--phase shifter; 2, 3--output attenuators; 4, 5--limiter amplifiers; 6--coincidence gate; 8--voltmeter circuit

The "Phase shifter" set (Fig. 3.19) measures the phase shift between two sinusoidal waveforms from external sources, and splits the sinusoidal signal from an external oscillator into two voltages phase-shifted through a continuously variable angle. The phase shift is monitored by the scale of the phase meter in the phase shifter.

The phase shifter (K1) is made in an RC circuit. To get the required frequency band, the values of the circuit components are selected by the "frequency kHz" switch that has three positions. Output voltages phase-shifted by 10-90° are sent to the "channel I" and "channel II" plugs through calibrated attenuators that are analogous in construction to those used in the "Generator" and "GV-1M" devices. The voltage sent to the attenuators -- 1 V -- is monitored by built-in voltmeter K5 in selector position B3-"IV". The phase shift is checked by the phase meter with selector B1 in the "internal" position.

The phase meter of the instrument contains two identical limiter amplifier channels K2 and K3 that convert the phase-shifted sine-wave voltages sent to their inputs into square pulse sequences that are time-shifted relative to one another. The time of coincidence of negative polarity of both series of pulses is fixed by coincidence gate K4. Obviously, pulse duration at the output of the coincidence gate is proportional to the phase shift. A meter connected to the output of the coincidence gate in the "180°" position of the B3 selector measures the average current of the pulse series, enabling calibration of the instrument scale in degrees of phase shift from 0 to 180°. Zero phase shift of the input voltages corresponds to complete time coincidence of negative-polarity pulses, and the average current of the coincidence gate is maximum; therefore the zero of the meter is on the right.

The phase meter has provisions for three auxiliary modes switched by selector B3. In the "zero set" position, both inputs of the coincidence gate are open and the millivoltmeter needle is zeroed by the "zero set" potentiometer. In calibration modes "IK" and "IIK', voltage from an external source can be sent to the input of the device. Calibration involves setting phase identity of both channels by simulating zero phase shift alternately for each channel.

An exceptionally important place in hydroacoustic measurements belongs to spectral analysis of complex signals, ship noises and interference to reception of hydroacoustic information. Theoretically, a complex function can be represented by a Fourier series consisting of separate harmonic components. The procedure of resolving a complex waveform into components of definite frequency and amplitude is called spectral analysis. Spectral analysis can be used to solve such problems as objective classification of signals, locating sources of elevated noise and interference and the like.

Modern methods of spectral analysis can be divided into two groups: filtration methods and Fourier transform methods. Filtration methods, which have been more widely used in practice, have three varieties: direct filtration methods, methods with preheterodyning and methods with time compression of the signals being studied. Instruments for spectral analysis are divided into harmonic analyzers for sequential analysis of the spectrum, and spectrometers for simultaneous or parallel analysis.

Instruments for sequential analysis may be based on using the method of direct filtration and the method with preheterodyning.

The former method is used in selective amplifier circuits with working principle that consists in sequential tuning of a frequency-selective negative feedback circuit of a wide-band amplifier to separate frequency components of the process. As a result of action of the feedback circuit, all frequency components of the signal will be suppressed except those that correspond to a dip in the frequency response curve of the feedback circuit at the given instant. The width of the passband of selective RC amplifiers is constant relative to the tuning frequency, i. e. $\Delta f/f_0 = \text{const}$ throughout the frequency band. Readout is usually by a meter with scale graduated in decibels or percent of the measured frequency component with respect to the fundamental harmonic of the signal.

In connection with expansion of the absolute band of analysis in the high-frequency region, the use of selective amplifiers is limited to the infrasonic and audio ranges. The frequency band of analyzers can be expanded into the high-frequency region by using heterodyning to convert the frequency spectrum.

The working principle of the heterodyne analyzer consists in continuous shifting of the investigated process relative to the narrow passband of a filter with fixed tuning. The spectrum is shifted by a heterodyne with frequency f_h mixed with the signal frequency f_s. The narrow-band filter isolates one of the side frequencies $f_h + f_s$ or $f_h - f_s$ at the mixer output. The level of the isolated frequency component is measured by a display. Since the width of the filter passband does not change, the absolute width of the passband of heterodyne analyzers does not vary throughout the frequency band. This limits the use of heterodyne analyzers in the low-frequency region.

Instruments for parallel analysis are designed on the basis of using a direct filtration method. The working principle of devices of this type is based on using a system of narrow-band filters with inputs connected in parallel, and outputs alternately switched to the display by an automatic commutator. The overall frequency response of the spectrometer is a "comb" made up of the frequency responses of individual filters intersecting on the level of 70% decline of the curve from the maximum value of the transfer factor of the filter. When the investigated process is sent to the filter inputs, all frequency components of the spectrum of this process that are isolated by the filters will be observed simultaneously at the outputs. The spectrometer filters have a constant relative passband width measured in fractions of an octave.

The principal characteristic of the spectrum analyzer is its resolution defined as the capacity of the analyzer to distinguish adjacent components with near frequencies. Quantitatively, resolution is evaluated by the smallest frequency interval within which two adjacent components are observed with a dip equal to 50% of the maximum value. Resolution is primarily determined by the parameters of the analyzing filters, and corresponds approximately to the interval between average frequencies of spectrometer filters or to the analysis bandwidth of the harmonic analyzer.

An advantage of harmonic analyzers over spectrometers is high resolution of the instruments with fairly simple design. The cumbersome selective system of spectrometers on low audio frequencies precludes the use of filters with bandwidth of less than $1/3$ of an octave. A considerable disadvantage of sequential-action devices is the limitation on rate of analysis that depends on the duration of transient responses that arise in the selective part of the instrument during retuning.

It can be shown [Ref. 53] that the time necessary for rise of a signal to 95% of the steady-state value for a tank circuit with passband width Δf is defined by the relation

$$t_1 = 1/\Delta f, \tag{3.6}$$

i. e. t_1 is the time necessary for analyzing an individual frequency component. The time required for analyzing an entire spectrum by the sequential-action instrument with absolutely constant analysis band Δf in the case of continuous and uniform tuning of the analysis frequency can be found as

$$T_a \geqslant \frac{4(f_{upper} - f_{lower})}{\Delta^2}, \tag{3.7}$$

where f_{upper} and f_{lower} are the limiting frequencies of the band being studied.

Thus for values of $\Delta f = 6$ Hz, $f_{lower} = 20$ Hz and $f_{upper} = 20$ kHz the minimum permissible analysis time is 38 minutes.

In instruments of simultaneous analysis used for studying standard processes, the rate of analysis is nearly unrelated to parameters of the filters used in the selective system of the spectrometer since the investigated process is continuously sent to all filters during the entire time of observation, and the waveforms in the filters can be taken as steady-state for steady-state processes. In this case, the maximum permissible rate of parallel analysis will be determined by the time constant of the display that is used. Spectrum analysis may be practically instantaneous when electronic displays are used in combination with high-speed commutators.

Use of the method of time compression of the signal in sequential-analysis devices increases the rate of analysis almost to the level of parallel-action devices while retaining high resolution. It can be shown [Ref. 55] that time compression of the process leads to multiplicative transfer of its spectrum as a result of multiplication of the frequencies that make up the spectrum by the time-scale transformation factor, or compression coefficient k_t (Fig. 3.20):

$$\frac{T_{i.r}}{T_{c.r}} = \frac{f_{c.r}}{f_{i.r}} = k_t, \tag{3.8}$$

where $T_{i.r}$, $T_{c.r}$ are the durations of the investigated and converted realization, $f_{i.r}$, $f_{c.r}$ are the frequency components of the investigated and converted spectrum.

Using (3.8) to transform (3.7) we get

$$T_a \geqslant \frac{4(k_t f_{upper} - k_t f_{lower})}{k_t^2(\Delta f)^2} \quad \frac{4(f_{upper} - f_{lower})}{k_t(\Delta f)^2}. \tag{3.9}$$

Expression (3.9) implies that the time of analysis of a compressed signal decreases in proportion to the increase in the time-scale transformation factor. Physically, acceleration of the process of analysis can be attributed to the fact that a k_t wider analysis band can be used for the expanded signal spectrum while retaining the former resolution by reducing the duration of transient processes in the filter k_t times. For example, in accordance with (3.6) the time of analysis of an individual frequency component in a band of 0.025 Hz should be 40 s; after signal compression with $k_t = 400\,000$, the band of analysis can be increased to 10 000 Hz, and the time of analysis reduced to 0.1 ms.

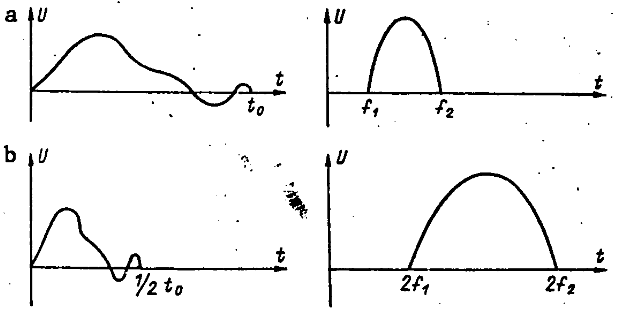


Fig. 3.20. Deformation of spectrum in time compression: a--realization and spectrum of process before conversion; b--after conversion

Any time-scale converter includes a recording device, storage unit and readout. Signal compression is realized by increasing signal readout rate by k_t times over the recording rate. Miniature equipment most frequently uses a delay line as the storage unit (Fig. 3.21).

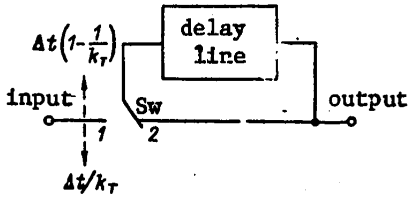


Fig. 3.21. Block diagram of time compressor

The input of the device receives a signal that is converted by switch Sw to a sequence of individual samples (Fig. 3.22). According to Kotel'nikov's theorem, the sampling period Δt is determined by the condition

$$\Delta t \leqslant \frac{1}{2f_{upper}}, \tag{3.10}$$

where f_{upper} is the highest frequency contained in the spectrum.

If duration $T_{i.r}$ is given, the number of samples is defined as

$$N \geqslant \frac{T_{i.r}}{\Delta t} = 2f_{upper}T_{i.r}. \tag{3.11}$$

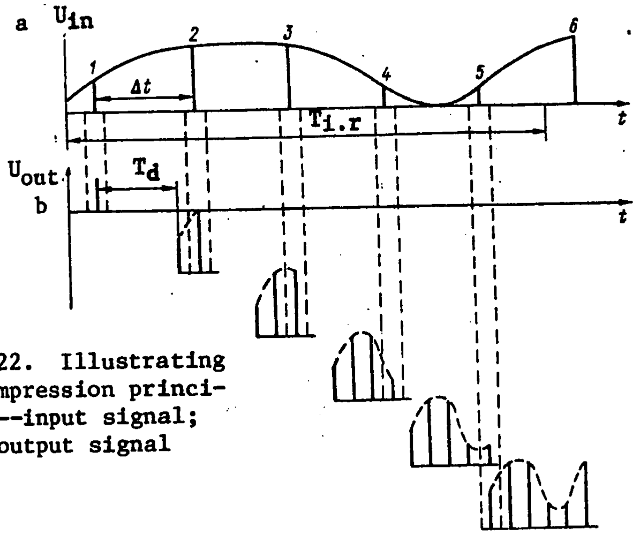


Fig. 3.22. Illustrating time compression principle: a--input signal; b--output signal

Each sample reading goes simultaneously to the output of the device and to the input of the delay line where it is shifted at a finite rate. During the time between two adjacent sample, switch Sw (see Fig. 3.21) is in position 2 ensuring circulation of the given sample in a closed ring with period T_d equal to the time of the delay line. After time T_d the sample shows up at the output of the delay line and is fed to the output of the device and to the input of the delay line. The idea of the method is that the secondary occurrence of sample 1 at the output of the device should precede the instant of switching of Sw to position 1 by a time $\Delta t' = \Delta t/k_t$. This is possible if $T_d = \Delta t - \Delta t' = \Delta t(1 - 1/k_t)$. If switch Sw is in position 2 for time T_d, and in position 1 for time $\Delta t - T_d = \Delta t'$ (switching times are indicated by the broken lines on Fig. 3.22), a compressed copy of the signal that is gradually accumulated in the recirculation ring is sent to the output of the device. If the signal is presented in the form of N samples, after N cycles a complete copy of the signal will appear at the output of the device that consists of the $N - 1$ samples stored in the delay line, and one last sample coming directly from the input. Excluded from further work in this cycle is the first reading that goes to the output of the device but does not enter the delay line input, since at this instant switch Sw will be in position 1. In all subsequent cycles the last reading will be replaced by the next to last, and the compressed copy of the signal will "slide" in time. Obviously, the necessary capacity of the delay line is numerically equal to $N_d = N - 1 = k_t - 1$.

This arrangement has come to be used in spectrum analyzers of sequential type (Fig. 3.23). As the resolution of the instrument increases, analysis is done on a fixed rather than a sliding time interval. For this purpose, a second recirculating ring is added to the time-scale converter circuit. This additional recirculator stores the compressed copy of the signal produced in the main compressor ring. Switch Π_2 is in position 1 at the instant corresponding to termination of shaping the complete signal copy in the compressor, and it

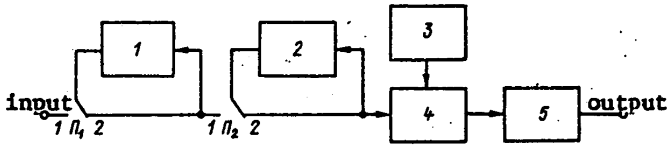


Fig. 3.23. Block diagram of heterodyne spectrum analyzer with time compressor: 1--time compressor; 2--buffer store; 3--mixer; 4--tunable heterodyne; 5--narrow-band filter

remains in this position for time Δt during which the signal copy passes to the mixer and to the delay line of the buffer store, after which switch $П_2$ is changed to position 1. Delay time in the buffer store is taken as Δt, the compressed signal circulates there N - 1 times, and at each occurrence of a copy of the signal one frequency component of the spectrum is analyzed. Thus the total analysis time is ΔtN, i. e. it is equal to the time of signal observation. During this time, the next copy of the signal has been formed in the compressor, and the procedure is repeated. The heterodyne is automatically tuned by steps before each successive signal copy appears at the mixer input. It can be concluded that operation of the device diagrammed in Fig. 3.23 is equivalent to operation of an analyzer of parallel type that contains N filters with absolutely constant passband width $\Delta f = 1/\Delta t$, i. e. analysis is done in real time.

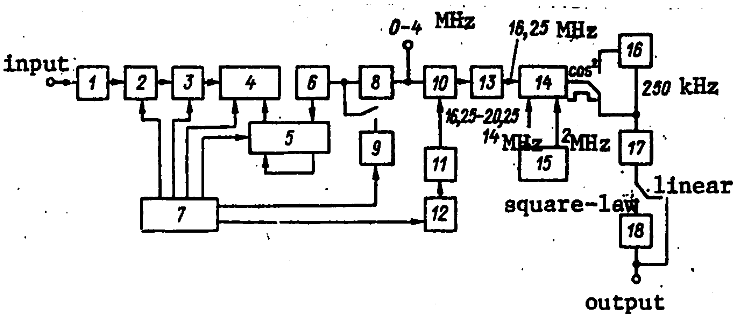


Fig. 3.24. Block diagram of 2030 analyzer: 1--attenuator and filters; 2--commutator; 3--analog-digital converter; 4--compressor; 5--buffer store; 6--digital-analog converter; 7--clock unit; 8--low-frequency filter; 9--marker-frequency generator; 10--mixer; 11--voltage-controlled oscillator; 12--linearly variable voltage oscillator; 13--bandpass filter; 14--dual mixer; 15--heterodyne; 16--correction link; 17--tank circuit; 18--squaring circuit

The time compression principle is used in the 3348 spectrum analyzer made by Bruel and Kjor. The instrument consists of three modules: the 2030, which is the analyzer proper, the 6701--a conversion device, and the 4710--an electronic display. The spectrum analyzer 2030 (Fig. 3.24) operates as a heterodyne analyzer with time compressor on a fixed time interval. A distinguishing feature of the circuit is use of a shift register instead of a delay line as storage element. This makes it possible to change the compression factor for different frequency bands. The input signal must be presented in digital code [Ref. 55].

The analyzer has 11 frequency bands (Hz): 0-10, 0-20, 0-50, 0-100 and so on to 0-20 000. The input signal goes to an attenuator and to eleven low-frequency filters (one for each band) with cutoff frequencies corresponding to the upper limits of the bands. Commutator 2 samples the signal in accordance with (3.10) and (3.11). The frequency of the samples is stipulated with some reserve, and is equal to $3f_{upper}$. In analog-digital converter 3, each value of the sample is converted to an 8-place binary word and fed to compressor 4 where a compressed copy of the signal is formed.

The total number of samples (3.11) depends on the minimum required time for analyzing one frequency component (3.6) and on the number of frequency components that can be distinguished in analysis. Regardless of range, the analyzer has the capacity for distinguishing 400 discrete components, which is equivalent to parallel connection of 400 narrow-band filters with resolution defined as $\beta = f_{upper}/400$. Taking the frequency of samples as $3f_{upper}$ and minimum time for analysis of one component $T_{i.r} = 1/\Delta f \approx 1/\beta$, we can get from (3.11)

$$N = \frac{3f_{upper}}{\beta}. \qquad (3.12)$$

Substituting $\beta = f_{upper}/400$ in (3.12), we can find that to analyze one frequency component it is necessary to accumulate 1200 samples, and therefore buffer store 5 has a capacity for retaining 1200 8-place binary words.

Data readout is at a rate of 12 MHz regardless of band, which is determined by the crystal-controlled oscillator in clock unit 7. Since the data input rate is $3f_{upper}$, we can find the compression coefficient for each band as $k_t = 4\cdot 10^6 f_{upper}$. Thus for the end bands of 0-10 and 0-20 000 Hz the compression factor is 400 000 and 400 respectively. The digital information taken from the buffer store is converted by digital-analog converter 6 to analog form. The analog signal goes to low-frequency filter 8 with cutoff frequency of 4 MHz, which corresponds to a shift in the f_{upper} of the signal spectrum regardless of band $f_{upper}/k_t = 4\cdot 10^6$ Hz. The signal with spectrum 0-4 MHz goes to mixer 10 of the heterodyne analyzer, in which heterodyne 11 has a frequency in the range of 16.26-20.25 MHz under the influence of linearly variable voltage oscillator 12. At a transcription rate of 12 MHz, circulation of 1200 samples in the buffer store is completed within 100 μs, the time necessary for analyzing one component; therefore the complete cycle of heterodyne tuning is 100 μs × 400 = 40 ms.

The difference frequency of 16.25 MHz is isolated by bandpass filter 13 with passband width of 10 kHz, corresponding to βk_t for any band. The signal is then subjected to double frequency conversion, reducing the frequency to 250 kHz. In the tank circuit tuned to the signal frequency, the positive half-period is sampled every 10 kHz of frequency tuning of heterodyne 11, i. e. 400 times in a 40 ms cycle. The resultant 400 smaples corresponding to the separate frequency components are sent to the analyzer output. The complete cycle of analysis, including resetting the heterodyne, is 45 ms.

Conversion module 6701 averages the signal, and also stores the resultant spectra in two parallel memories. This enables on-the-spot comparison of two spectra: the running spectrum and one stored in memory, or two in storage.

The display of the device is based on a 12" CRT. The 45 Hz horizontal scan frequency is synchronized with the heterodyne tuning frequency. The display also provides digital readout of the levels of the 400 components.

The 3348 instrument is for matching to a chart recorder. Provisions are also made for data output to a 7504 digital computer.

CHAPTER 4: METHODS OF MAKING HYDROACOUSTIC MEASUREMENTS DURING OPERATION OF THE HYDROACOUSTIC STATION

§4.1. Basic principles of making hydroacoustic and electronic measurements

All technical parameters that are measured in the course of operation of hydroacoustic stations can be divided into two groups: electroacoustic, and special electronic parameters. Measurements of parameters in the first group are associated with the necessity of doing an experiment in water, i. e. they can be classified as hydroacoustic measurements; measurements of the second group involve only electrical parameters. For example, the first group might include the overall sensitivity of the station, directional characteristics, sound pressure on the axis of the radiator and the like, while the second group would include gain, frequency and phase responses of different components, level of electrical interference and so on. Hydroacoustic measurements are the more difficult, and as a rule involve complex checking of hydroacoustic stations.

Practically all hydroacoustic measurements done under shipboard conditions are aimed at determining sound pressure in some point of the acoustic field that is excited either by the measurement emitter in measuring parameters of the station in the passive mode, or by the station's own antenna when measuring its parameters in the active mode. A generalized diagram of hydroacoustic measurements is shown in Fig. 4.1. The essence of the measurements boils down to determining the sound pressure p set up by the emitter at a distance r, i. e. at the hydrophone location. As was remarked in §3.1, sound pressure is measured by converting it to electric voltage, and therefore the process of hydroacoustic measurements also reduces to measurement of electric quantities: the voltage applied across the emitter, and the emf developed by the receiver. Nonetheless, hydroacoustic measurements are specifically typified

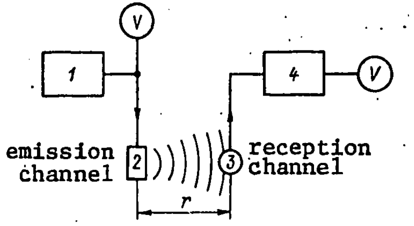


Fig. 4.1. Generalized diagram of measurement of acoustic parameters: 1--oscillator; 2--emitter; 3--receiving transducer; 4--amplifier

by the characteristics of the measurement transducers and by the conditions of their operation under water.

Hydroacoustic measurement errors are minimized by satisfying certain requirements on the measurement site, relative placement of transducers and the stability of their characteristics, linearity of the reception part of the equipment and selection of signal-to-noise ratio. Let us consider these requirements in more detail.

The measurement site is chosen with consideration of the necessity of setting up conditions of sound propagation as close as possible to a free field. Such conditions can be brought about in a space of large dimensions sufficient for eliminating reflections from its boundaries. The principal reflecting surfaces are the bottom and the surface of the water, and therefore the critical dimension of the body of water used for measurements is local depth. The maximum permissible depth depends on the frequency band of the measurements: depths of 8-15 m are considered acceptable for measurements on frequencies no lower than 1 kHz; the depth should be increased for lower frequencies [Ref. 5]. The bottom at the measurement site should be even and have good absorbing properties. If hydroacoustic station [HS] parameters are being measured in a harbor, consideration must be taken of reflections from horizontal boundaries of the body of water as well: wharves, piers and other port structures, and also nearby ships.

The water at the measurement site must be free of inhomogeneities that may cause refraction or scattering sound (temperature inhomogeneities, currents, marine organisms and the like).

The level of acoustic interference or the noise background at the measurement site must be low. If the measurements are being done on a special test site, its noise background is determined by sea noises due mainly to surface waves and the noises of passing ships. Therefore the location of such a test site must be chosen in a region shielded from winds and far away from heavy traffic. The noise background of harbor waters is due in large measure to industrial activity of port enterprises and heavy boat traffic, and it may be so high that measurement of the parameters of the HS becomes impossible.

The relative placement of transducers is determined by the minimum permissible distance between them and the necessity of lining up their acoustic axes. Here the acoustic axis is taken to mean the direction corresponding to maximum sensitivity of a transducer.

The distance between the transducers being used in measurements is a quantity that appears in computational formulas, and therefore it must be measured axactly. It should be borne in mind that the distance must be determined with respect to the acoustic center of the transducer, i. e. the imaginary point from which the radiated sound waves emanate. The acoustic centers of spherical and cylindrical measurement transducers coincide with the center of their geometric symmetry. The acoustic center of flat antennas corresponds to the center of their active surface. For circular antennas the acoustic center coincides with the center of the virtual active surface corresponding

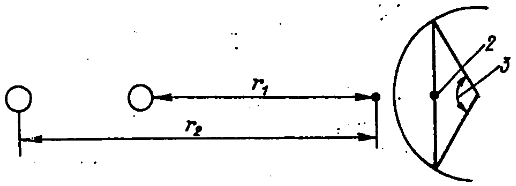


Fig. 4.2. Diagram of verification of choice of the acoustic center: 1--selected center; 2--true center; 3--compensated sector

to the compensated sector. The acoustic center position on antennas of any configuration can be experimentally checked by measuring the way that the sound pressure created by the antenna depends on distance (Fig. 4.2). If the position of the assumed acoustic center coincides with the true position, then the sound pressure under free-field conditions varies in inverse proportion to distance, i. e.

$$p_1r_1 = p_2r_2. \quad (4.1)$$

In the case $p_1r_1 > p_2r_2$ the assumed center is shifted backward relative to the true position, and if $p_1r_1 < p_2r_2$, the displacement is forward. Similar arguments hold for passive antennas in which the acoustic center is defined as the center of the surface (real or imaginary) that receives the waveforms of the acoustic front of the signal. Experimental verification of coincidence of the assumed and actual acoustic center of the antenna in this case is done by shifting the measurement emitter along the acoustic axis of the antenna, criterion (4.1) being replaced by the expression

$$u_{out1}r_1 = u_{out2}r_2, \quad (4.2)$$

where u_{out1} and u_{out2} are the voltages measured at the output of the station and corresponding to distances r_1 and r_2 of the radiator from the antenna.

The minimum permissible distance between the acoustic antenna and the hydrophone must be selected on the basis of the condition of hydrophone location in the field of the spherical wave radiated by the antenna. It can be shown [Ref. 5] that the sound pressure on the acoustic axis of a flat radiator corresponds to a spherical wave at some distance from its surface that depends on the ratio of the largest dimension of the radiator d to wavelength λ. Closer to this distance one observes alternation of pressure maxima and minima due to interference of waveforms set up by different sections of the radiator surface. This part of the acoustic field of the emitter is called the Fresnel zone, or the near field. The distance of the last pressure maximum from the emitter surface is d^2/λ. Beyond this distance begins the far field, or Fraunhofer zone, in which sound pressure falls off in inverse proportion to distance, i. e. in accordance with the law of a spherically diverging wave. Based on the principle of reversibility, analogous arguments are valid for a passive antenna located in the sound field of a point radiator. Shifting the emitter at distances from the antenna exceeding d^2/λ causes a change in the sound pressure, and consequently in the voltage at the output of the station that is inversely proportional to distance. This behavior is not observed when the emitter is shifted in the near zone of the zntenna. In practice, the distance between the acoustic antenna and measurement transducers is chosen on the basis of the relation

$$r \geqslant \frac{2d^2}{\lambda}, \qquad (4.3)$$

where d is the greatest dimension of the active part of the antenna.

Relation (4.3) is also valid for the case of location of the hydrophone in the field of a cylindrical measurement emitter that has vertical dimension d when measuring the sound pressure developed by this emitter.

A necessary condition of relative placement of transducers during measurements is coincidence of their acoustic axes since the major parameters of hydroacoustic stations -- overall sensitivity and sound pressure -- must be determined in the direction of their maximum values. Inaccurate orientation of the acoustic axis of a narrow-band antenna relative to the position of measurement transducers may lead to appreciable errors in measurement. In some cases, deviation of the hydrophone from the acoustic axis of the emitter may cause an apparent effect of pressure increase with increasing distance due to a reduction of the angle between the direction to the hydrophone and the acoustic axis of the antenna (Fig. 4.3).

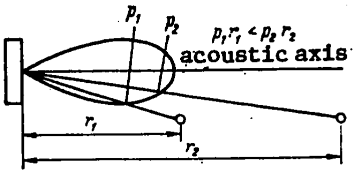


Fig. 4.3. Choosing depth of immersion of hydrophone

Bringing the acoustic axes into line, or establishing coaxiality is done first in the vertical, and then in the horizontal plane. Coaxiality in the vertical plane is established by selecting the depth of immersion of measurement transducers. In doing this, the acoustic axis of the antenna must first be established horizontally (Fig. 4.4). The instant of coincidence of the acoustic axes is

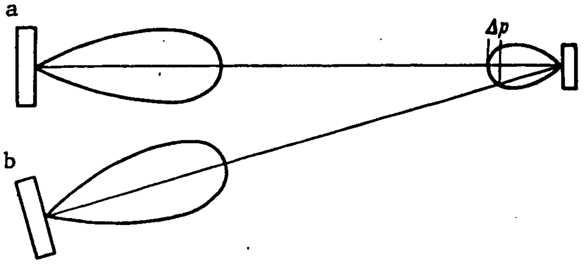


Fig. 4.4. Establishing coaxiality of the reception antenna and emitter direction: a--correct; b--incorrect

determined from the maximum signal level at the output of the hydroacoustic station and the hydrophone operating in the mode of reception of the sound field of the radiator, or from the maximum voltage level at the output of the hydrophone that receives signals from a hydroacoustic station operating in the active mode.

Establishment of coaxiality in the horizontal plane is done by rotating the directivity pattern of the antenna until the maximum signal is attained at the output of a station operating in the passive mode, or at the output of a hydrophone that receives signals of an actively operating station. if the station has a stationary directivity pattern, coaxiality in the horizontal plane is established by shifting the measurement transducer relative to the acoustic center of the antenna at the same distance away.

Consideration should be taken of the fact that spherical and cylindrical transducers in the horizontal plane as a rule have a real directivity pattern that does not correspond to circular. Therefore in such transducers the direction of the acoustic axis must be chosen beforehand in the sector that corresponds to the smoothest change in sensitivity. This direction is indicated by a reference line that is made on the transducer housing and is used to establish coaxiality of the transducers. Immersion of the measurement transducer in water and orientation of its acoustic axis should be done by using a rigid bar.

The stability of characteristics of measurement transducers to a considerable extent determines the error of the entire measurement. For example, the frequency response of the sound pressure meter is used to correct the readings of an instrument in the measurement process. Practice has shown that hydrophone characteristics change with time, which must be taken into consideration in doing calculations, and in this connection, measurement hydrophones should be calibrated at least once a year.

The stability of measurement results also depends on preparation of the measurement transducers. Before immersing in the water, the surface of the transducers must be carefully cleaned and degreased to ensure better wetting of the trnasducer. Before doing measurements, the transducer is held in water for the time required to equalize the transducer and water temperatures.

Particular attention must be given to secure fastening of the transducers to prevent them from shifting relative to the acoustic antenna of the station during measurement. Even slight shifting of the transducer (as a result of rocking of the ship) in the field of an antenna with sharp directivity characteristic relative to the platform to which the transducers are secured causes considerable fluctuation of the signals being measured. If it is not possible to avoid signal fluctuation, the number of readings must ensure the capability of statistical processing of measurement results.

Linearity of reception and amplification channels of a hydroacoustic station is one of the necessary conditions for accuracy of measurements of parameters that have the physical sense of the transfer ratio: overall sensitivity, frequency response and directivity pattern. Measurement of these parameters involves determining the ratio between the output and input quantities, i. e. between the voltage u_{out} at the output of the channel and the sound pressure p accing on the antenna. In the given case, the requirement of linearity of the reception and amplification channel is formulated as the condition

$$u_{out}/p = \mathrm{const}, \tag{4.4}$$

i. e. the output quantity is proportional to the input, and the ratio of these quantities does not depend on the absolute value of either of them. Graphically, the concept of linearity of a channel is illustrated by a linear segment of its amplitude characteristic. Linearity of a channel is lost if the input signal quantity goes beyond the limits of the linear section of the amplitude response curve (Fig. 4.5a). This leads to nonlinear distortions of the the output signal, and increases measurement error. And at the same time, violation of conditions of nonlinearity is the most frequent and worst mistake of beginning specialists.

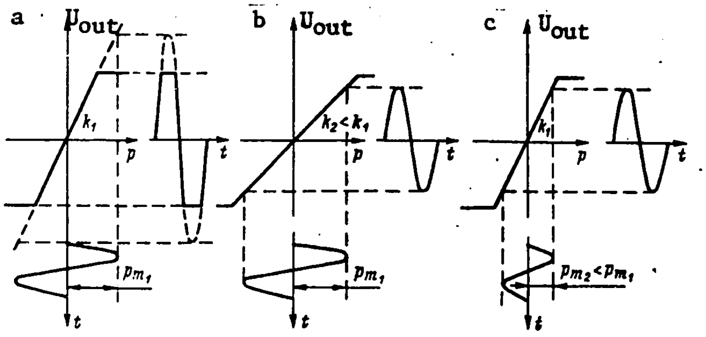


Fig. 4.5. Illustrating linearity of transmission: a--nonlinear signal distortions; b--selection of gain; c--selection of signal magnitude

There are two ways to ensure conditions of linearity: selecting the magnitude of the input signal while keeping the gain of the channel constant (Fig. 4.5c), or selecting the gain of the channel while maintaining the signal magnitude constant (Fig. 4.5b).

The first method is used if there is no calibrated gain control in the reception channel. in this case the fine-tuning control is set at the position of maximum gain, and the sound pressure is selected so that variation over a small range causes a proportional change in the voltage at the output of the station.

The second method is used if there is a calibrated stepwise gain control in the reception channel. The fine-tuning gain control is also set in the position of maximum gain, and the magnitude of the input signal and position of the step control are sequentially selected so that a change in position of the control by one step causes a certain change in output voltage. For example if a change in control position by one step produces a change of 6 dB in the gain, then the output voltage should be changed by a factor of two. On the amplitude response curve, the change in gain corresponds to a change in the angle of slope of the linear section. The signal-to-noise ratio at the point being checked should be chosen fairly high for certain indication and measurement of signal parameters (at least 5).

Two kinds of interference are most typical in measurement of the parameters of hydroacoustic stations: wave noises or interference resulting from reflections from the bottom surface, harbor structures and the like, and the noise background of the body of water. Wave interference is difficult to recognize if the signal is radiated directly. One of the methods of recognizing wave interference is to determine the way that voltage depends on frequency at the output of a hydrophone located in a field of sinusoidal waveforms [Ref. 5] (Fig. 4.6b). The graph of Fig. 4.6b shows the interference pattern resulting from summation of the direct wave and the wave reflected from the surface. If the paths traveled by these waves differ by Δx, the phase difference at the reception point will be $\Delta\phi = k\Delta x = 2\pi\Delta x/\lambda$. If the frequency shift Δf corresponds to a change in phase shift by 2π, i. e. $2\pi\Delta x/\lambda = 2\pi$, and

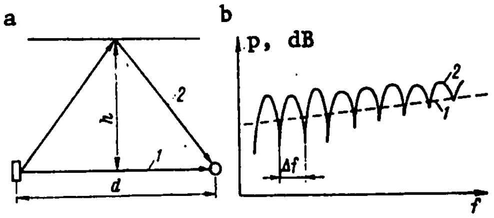


Fig. 4.6. Interference scheme: a--path of direct signal (1) and signal reflected from surface (2); b--frequency dependence of hydrophone output for direct signal (1) and the sum of the direct and reflected (2) signals

$$\Delta x = \frac{c}{\Delta f}. \tag{4.5}$$

Comparing the quantity found from (4.5) with distances to assumed objects of reflection, one can find the source of interference. It has been shown [Ref. 5] that for the first reflection from the surface the path difference Δx is determined by the expression

$$\Delta x = 2\sqrt{h^2 + (d/2)^2} - d$$

(h and d in the symbols illustrated by Fig. 4.6).

Wave interference can be reduced by selecting optimum placement of the transducers relative to reflecting boundaries. The greatest effect is from using a pulsed mode of operation that enables separation of the direct signal and wave interference in time, i. e. that realizes time selection of the signal. Time selection or gating of the signal can be realized by using a special component -- a selector.

The selector is an electronic switch in the pulse signal transmission circuit (Fig. 4.7) that is opened by a strobing pulse for a time equal to the received

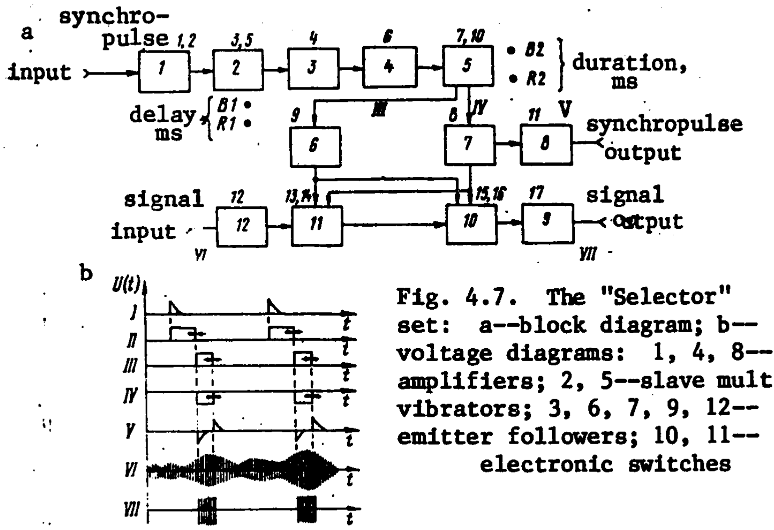


Fig. 4.7. The "Selector" set: a--block diagram; b--voltage diagrams: 1, 4, 8--amplifiers; 2, 5--slave multivibrators; 3, 6, 7, 9, 12--emitter followers; 10, 11--electronic switches

signal duration. The strobing pulse delay time and its duration can be continuously varied over a range from 0.5 to 50 ms. The delay multivibrator is triggered by a synchronizing pulse from a pulse generator. The leading edge of the gating pulse is used to synchronize scanning of an oscilloscope; the voltage of the signal isolated by the selector is fed to the vertical input of this oscilloscope. Thus only the direct signal free of wave interference can be observed on the oscilloscope screen.

Noise interferences, including the noise background of the body of water, electrical interference proper and pickups act directly, in contrast to wave interferences. The level of noise interference can be reduced by connecting bandpass filters in the signal reception circuit. The filter passband in the case of pulse measurements is limited by $\Delta f \geqslant 2/\tau$, where τ is the duration of the pulse. The signal-to-noise level is established by an indicator connected to the output of the station by regulating the measurement oscillator signal amplitude, which must remain within the limits of the linear section of the amplitude response of the reception channel of the hydroacoustic station. The fact of normal signal transmission must be established by periodically switching off the measurement oscillator signal, in which case the readings of the output display drop off sharply.

Summing up, we can conclude that in preparation for hydroacoustic measurements it is necessary to verify the following rules:

1) there must be no reflections from boundaries in the water;

2) the distance between measurement transducers and the acoustic antenna of the station must be chosen in accordance with condition (4.3);

3) the acoustic axes of the measurement transducers and antenna must be made to line up;

4) the signal amplitude and gain must be chosen on the basis of the condition of linearity of the reception channel;

5) the signal-to-noise ratio at the point being checked should be at least 5.

The most universal method of checking satisfaction of these rules is to change the distance between the acoustic antenna station and the measurement transducers while verifying that relations (4.1) and (4.2) are met, which correspond tc the law of attenuation of a spherical wave. Violation of any of the enumerated rules will have an effect on satisfaction of these relations.

§4.2. Methods of measuring parameters of the reception part of the HS

A. Acoustic measurements

The reception part of the hydroacoustic station is rather well characterized by the set of technical parameters measured as diagrammed in Fig. 4.8. This set includes: station sensitivity; amplitude and frequency responses; directivity pattern; threshold sensitivity.

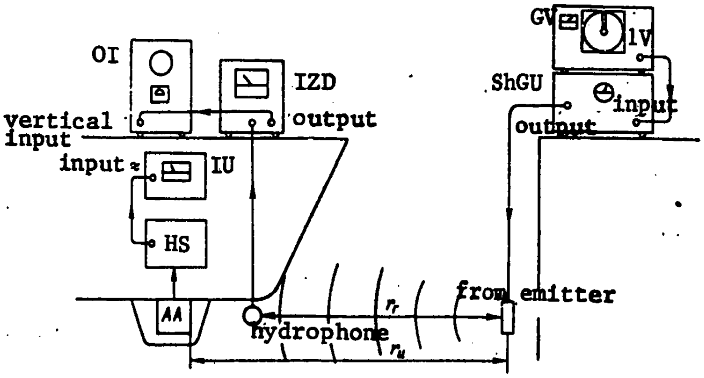


Fig. 4.8. Diagram of instrument hookup for measuring the acoustic parameters of the reception part of the HS

Let us consider the technique for measuring these parameters.

The sensitivity ζ_Σ of the station is defined as the ratio of the voltage u_{out} measured at the output of the reception-amplification channel to the sound pressure p perceived by the antenna,

$$\zeta_\Sigma = \frac{u_{out}}{p}. \qquad (4.6)$$

In contrast to determination of hydrophone sensitivity (see §3.1), when measuring station sensitivity the sound pressure on the surface of the antenna will be greater than that in the free field as a consequence of summation of the direct and reflected waves on the antenna surface. The sound pressure on the antenna surface is set up by the measurement emitter excited by the power amplifier to which an input signal is sent from the instrument oscillator. As a rule, a sine-wave signal is used to measure overall sensitivity with frequency corresponding to the average frequency of the station working band. Direct measurement of the sound pressure on the antenna surface is nearly impossible, and therefore pressure measurement is done with an "IZD" set in which the hydrophone is placed on the acoustic axis of the antenna at a certain distance from its surface. Distances r_r and r_u must satisfy condition (4.3). Attention must be given to meeting all other requirements of §4.1, and in particular (4.4).

The voltage at the output of the station is determined by a voltmeter that measures the rms value. Usually the headset output of the HS is used in doing this. If the measurements are done in the pulse mode, an oscilloscope is connected to the output of the IZD sound pressure meter and the station.

Sensitivity of the HS is calculated by the formula

$$\zeta_\Sigma = \frac{u_{out}}{p_m} A^{N-m} \frac{r_u}{r_r}, \qquad (4.7)$$

where p_m is the pressure measured by the IZD, A is the coefficient of reduction of the signal in one step of the station gain control, N is the total number of control steps, m is the number of the step on which the measurements are made, r_r and r_u are the distances from the emitter to the hydrophone and to the antenna respectively.

The need for coefficient A is explained by the necessity of meeting rule (4.4). This coefficient is used to adjust measurement results to the maximum possible overall sensitivity, which is usually given in the station specifications. Cofactor r_u/r_r is necessary to normalize the sound pressure measured by the hydrophone with respect to the antenna surface in accordance with (4.1).

We should emphasize the importance of the maximum possible accuracy in measuring overall sensitivity since this parameter, as will be demonstrated in §5.2, appears in formula (5.2) for calculating the acoustic interference level that has a direct effect on the range of action of the station, and hence the confidence of predicting the range of action will be determined to a considerable extent by the error of measuring overall sensitivity.

Having measured the sensitivity of the station, we can calculate the sensitivity of the acoustic antenna which is defined as the ratio of the no-load output voltage E at the output of the antenna to the sound pressure at its input. Direct measurement of antenna sensitivity is practically impossible in virtue of the smallness of E. The sensitivity $\zeta_{a.a}$ of the acoustic antenna is calculated by the formula

$$\zeta_{a.a} = \frac{\zeta_\Sigma}{k_{max}}, \qquad (4.8)$$

where k_{max} is the maximum throughput of the reception-amplification channel (see §4.2B). The acoustic antenna sensitivity is expressed in μV/Pa.

The amplitude response of the reception-amplification channel is defined as the dependence of voltage measured at the channel output on the sound pressure at the active surface of the antenna. The amplitude response is determined by measuring the voltage at the station output while slowly varying the voltage fed to the measurement emitter at constant signal frequency. The measurement hookup is shown in Fig. 4.8. Sound pressure is measured at the same time. Measurement results are plotted on a curve (Fig. 4.9a) that usually consists of three sections: a central linear segment, a lower section determined by the action of acoustic interference, and an upper section formed by processes of saturation of components of the reception-amplification channel. The slope of the linear section is numerically equal to sensitivity, and its length is determined by the dynamic range of the channel: $D_p = 20 \lg p_{max}/p_{min}$ dB.

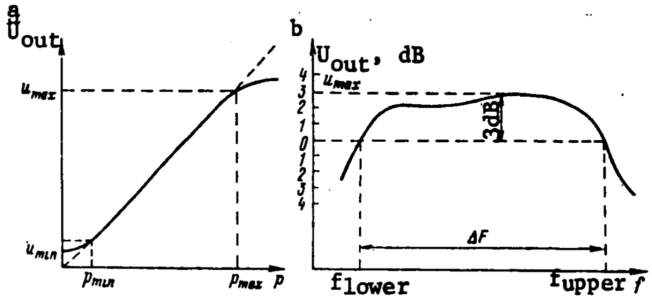


Fig. 4.9. Amplitude (a) and frequency (b) response of HS

It must be understood that excess of the signal amplitude over p_{max} inevitably leads to nonlinear distortions, and therefore when operating the station care must be taken to see that u_{out} does not exceed u_{max}, which must be determined beforehand for every station. Signal reduction below p_{min} may lead to missing the signal because of masking by interference. The nature of the lower section is closely tied up with the threshold sensitivity of the station. Considerable nonlinearity of the central section may also lead to nonlinear signal distortions.

Frequency response is defined as the dependence of voltage measured at the channel output on the frequency of the acoustic signal while the sound pressure on the antenna surface is held constant. The frequency response is determined by measuring the voltage at the station output while slowly varying the signal frequency of the measurement oscillator. The "IZD" set is used to monitor constancy of the sound pressure (see Fig. 4.8). The result is plotted on a graph (Fig. 4.9b). A logarithmic scale should be used for wide-band channels, and a linear scale for narrow-band channels.

The frequency response curve can be used to determine the passband width in the form of the frequency interval between output voltages differing by 3 dB from the maximum value (or what amounts to the same thing, the components $0.7u_{max}$). A change in passband width may take place as a result of malfunctions in the frequency-selection circuits of the station. Expansion of the passband leads to a reduction of the interference suppression in reception, while narrowing of the passband leads to loss of signal due to straying beyond the passband limits, e. g. due to the Doppler effect. A shift of the passband along the frequency axis as a rule takes place due to drift of heterodyne frequencies, and leads to distortion or total loss of the signal spectrum.

Distortion of the shape of the frequency response within the limits of the passband may disrupt conditions of optimum reception and cause frequency distortions of the signal.

The directivity pattern is defined as the dependence of voltage measured at the channel output on direction of incidence of the wavefront on the surface of the antenna relative to its acoustic axis at constant frequency of the signal and intensity of its source.

The directivity pattern is measured in the horizontal and vertical planes. The technique of measurement for hydroacoustic stations that have antennas with movable acoustic axis is different from that for stations with static acoustic axes of the antenna. For stations of the former type, after measuring sensitivity the acoustic axis is slowly rotated relative to the initial position that corresponds to maximum output voltage. Over certain intervals that depend on the sharpness of directional action of the antenna, the output voltage readings are recorded. The results are plotted on a graph, usually in polar coordinates (Fig. 4.10), enabling evaluation of the accuracy of direction finding, the magnitude of additional maxima and other characteristics.

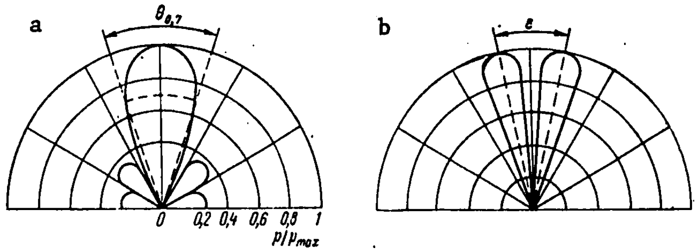
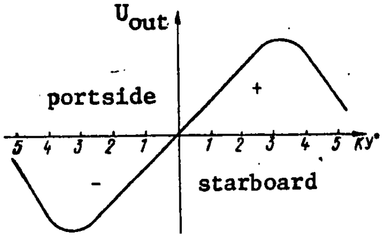


Fig. 4.10. Polar patterns: a--for maximum method of direction finding; b--for phase method of direction finding at difference channel output

Fig. 4.11. Direction-finding characteristic at output of phase-sensitive detector



At the output of difference channels in stations that realize the phase and phase-amplitude methods of direction finding, the directivity pattern contains two main lobes, and the angle ε between these lobes characterizes the accuracy of direction finding. Constant voltage is taken from the output of the phase-sensitive detector, and the directivity pattern measured at the output of the channel that realizes the phase-amplitude method of direction finding is usually plotted in rectangular coordinates and is called the direction-finding characteristic (Fig. 4.11).

The method for measuring the directivity pattern of stations that have antennas with fixed acoustic axes is much more complicated. Essentially, the technique consists in shifting the measurement emitter around the antenna at a fixed distance away, and taking voltage readings at the station output that correspond to certain directions toward the emitter relative to the acoustic axis. This method requires the use of a motor launch or rowboat for making the measurements. The main difficulty is in keeping the distance constant between the measurement emitter and the acoustic center of the antenna.

Threshold sensitivity is a parameter that characterizes reception-amplification channels containing threshold circuits. The threshold sensitivity is defined as the minimum value of the sound pressure that acts on the antenna (under the previously considered conditions of measuring overall sensitivity) to cause operation of the threshold device. By definition and in essence, this parameter is the closest thing to a recognition coefficient.

Measurements are done in the arrangement diagrammed in Fig. 4.8. By analogy with measurements of the amplitude response, the sound pressure set up by the emitter is gradually increased from zero. The instant of operation of the threshold circuit is determined from the response of the corresponding display, and is noted by the "IZD" reading. The result is normalized in accordance with (4.1) to the antenna location point. This pressure value corresponds to the threshold sensitivity of the station.

B. Special electronic measurements

The electroacoustic parameters considered above give a rather complete characterization of the state of the reception part of the channel, and if they did not go beyond set tolerances, this would constitute the sum total of the measurements. However, in view of the complexity of the equipment and the difficulties of organizing hydroacoustic measurements in practice, this is rarely the case, and special electronic measurements must be made. Among the electrical parameters of the reception part of the station that are subject to periodic measurements are the throughput, amplitude and frequency responses, amplitude and phase identity of multichannel amplifiers, electrical interference, resistance of antenna insulation and power circuits.

The throughput of the reception-amplification channel is defined as the ratio of voltage measured at the channel output to the voltage fed to its input through an equivalent acoustic antenna:

$$k = u_{out}/u_{in}. \tag{4.9}$$

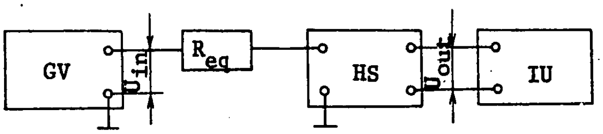


Fig. 4.12. Diagram of equipment hookup for measuring the throughput

Measurements are taken by a voltmeter connected to the channel output and by an instrument oscillator with output connected to the input of the preamplifier through the antenna equivalent R_{eq} (Fig. 4.12). As noted in §3.2, the output impedance of the oscillator must be considerably less than the external circuit impedance; otherwise, the oscillator will have to be connected through a voltage divider, and the resistor from which u_{out} is taken must be much less than R_{eq}. Signal frequency is taken as equal to the average frequency of the channel passband. Before measurement, in accordance with the requirements presented in §4.1 it is necessary to check the signal-to-noise ratio and the working state on the linear section of the amplitude response. The maximum possible gain is calculated by the formula

$$k_{max} = kA^{N-m}. \quad (4.10)$$

Connection of the oscillator through the antenna equivalent reduces the output voltage, and consequently the throughput calculated by (4.10) is less than the true value. This enables us in calculations by formula (4.8) to get the no-load sensitivity of the acoustic antenna. In direct connection of the oscillator to the amplifier input, the throughput calculation will give a large value, and the calculated value of antenna sensitivity will be less than the real value. The most convenient method is to connect the oscillator in a break in the "neutral" wire of the antenna (Fig. 4.13). In this case the control signal is sent to the preamplifier inputs through all transducers of the antenna.

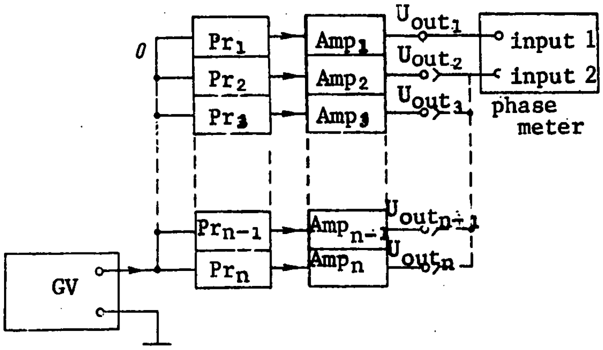


Fig. 4.13. Diagram of equipment hookup for measuring gain and checking phase identity of multichannel amplifiers

Amplitude and frequency responses are measured by the same arrangement. The measurement technique is analogous to that considered in section A.

Verification of identity is essential for multichannel preamplifiers. The scatter of gain and phase responses of preamplifiers determines the distortion of the shape of the station's directivity pattern. When checking the amplitude identity, the gain of each of the preamplifiers is measured in the hookup shown in Fig. 4.13. This same arrangement is used for checking the phase identity of multichannel amplifiers. The first input of the phase meter is connected to any amplifier, whose output voltage is taken as the reference, and the second input is connected alternately to the outputs of the remaining amplifiers.

The technique for measuring the level of electrical noises will be considered in §5.1.

The isolation resistance of the acoustic antenna is an electrical parameter that can appreciably influence antenna sensitivity. Conditions of operation in sea water necessitate special attention to systematic measurement of this parameter. Damage to the hermeticity of transducers results in sea water getting into the works, i. e. "inleakage", and consequently to shunting of the input of the amplifiers.

The isolation resistance is measured with a megohmmeter (Fig. 4.14). The isolation resistance R'_{is} of the entire antenna disconnected from the HS is measured between the "neutral" wire and the housing; in doing this, the jumper that connects the "neutral" wire to the housing must be removed. The isolation resistance R_{is} of the individual receivers is measured between the signal terminal of the receiver disconnected from the preamplifier, and the housing.

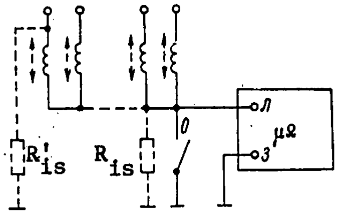


Fig. 4.14. Diagram of measurement of isolation resistance of acoustic antenna

To measure cable isolation (Fig. 4.15), steps must be taken to compensate for the influence of surface currents I_s on the measurement result. To do this, terminal E

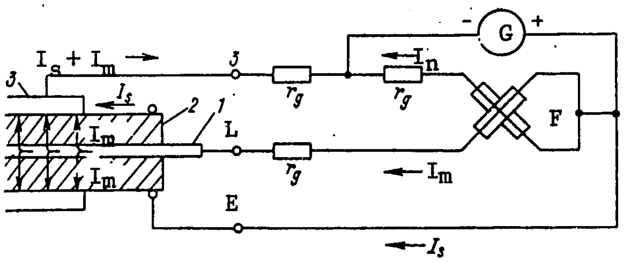


Fig. 4.15. Diagram of measurement of cable insulation resistance

of the megohmmeter is connected to a loop wound around the cable insulation. As a result, surface currents flow past the measurement coil F of the ohmmeter. A tester cannot be used to measure the isolation resistance of receivers and antenna since the influence of electrolysis when sea water gets into the transducers can completely distort the measurement result.

§4.3. Methods of measuring parameters of the transmitting part of the HS

A. Acoustic measurements

The acoustic parameters of the transmitting part of the hydroacoustic station are the sound pressure on the emitter axis and the directivity pattern of the antenna. The hookup of instruments for measuring these parameters is diagrammed in Fig. 4.16.

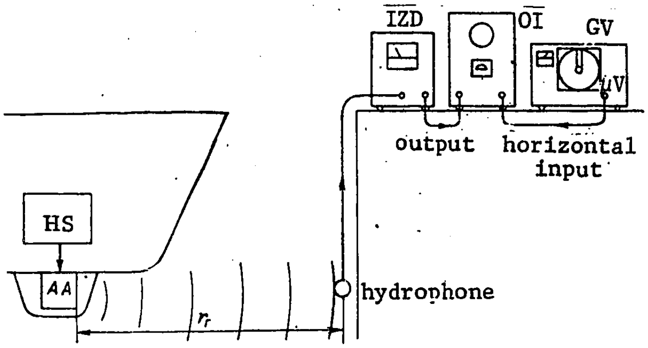


Fig. 4.16. Equipment hookup for measuring acoustic parameters of transmitting part of the HS

The sound pressure on the acoustic axis of the antenna characterizes the acoustic power of the station. Measurements are made by the "IZD" and an oscilloscope connected to its output. The hydrophone of the "IZD" is located on the acoustic axis of the antenna at the distance determined by (4.3). The measurement result must be normalized to a distance of 1 m from the antenna by a formula obtained from (4.3):

$$p = p_m r_r. \tag{4.11}$$

The method of measuring the directivity pattern of the antenna and processing the measurement results is analogous to the technique explained in §4.2 for the reception part of the hydroacoustic station.

B. Special electronic measurements

The transmitting part of the HS is characterized by electric power of the oscillator, working frequency, pulse shape and duration.

Electric power is determined by measuring the voltage at the output followed by calculations by the formula $W = U^2/R$, where R is the value of the resistor across which the voltage measurement has been taken. A resistor of low value is connected in series in the signal circuit.

To measure the working frequency and determine pulse parameters, use is made of a voltage divider or a special instrument transformer, which are usually incorporated into the oscillator circuitry. The working frequency is measured by the "GV-1M" wave meter or an oscilloscope using Lissajou figures.

An oscilloscope is used for determining the parameters of radiated pulses by a conventional procedure.

CHAPTER 5: METHODS OF MEASURING THE LEVEL OF INTERFERENCE TO OPERATION OF THE HYDROACOUSTIC STATION

§5.1. Electrical interference to operation of HS and methods of measurement

The electrical interference to HS operation is defined as voltage measured at the station output when the acoustic antenna is disconnected, with normalization of the measurement result to the input.

Electrical interference can be divided into three major components according to the nature of origin: set noises, supply line background, and spurious pickups.

The set noises are caused by the noisiness of tubes, transistors and other active electronic components. As a rule, these have a continuous spectrum and a comparatively low level.

The supply line background shows up at the station output as a voltage with frequency that is the same as the line voltage of the station, or a multiple of the line frequency. This background depends on the quality of the smoothing filters in the rectifiers.

In a general sense, spurious pickups refer to transmission of electric voltage from one component to another that is not intended by circuit design, but is due to the specific construction and configuration of the equipment. Spurious pickups are due to stray coupling, i. e. coupling that appears between separate components of the equipment [Ref. 11]. Spurious pickups are divided into internal and external. In the former case, they may lead to self-excitation of amplifiers on a frequency determined by the parameters of the components with stray coupling. Spurious pickups of this kind are eliminated on the stage of equipment design, and as a rule are not observed in operation. The greatest difficulties under conditions of shipboard power line operation are encountered in controlling external spurious pickups, i. e. spurious background pickups. These spurious pickups show up at the outputs of equipment as voltages and currents that do not correspond to its intended purpose, and that have frequencies equal to or multiples of the line frequency. Spurious background pickups are the result of stray coupling between the electronic part of the equipment and the power line in conjunction with the overall system of electrical equipment on shipboard.

External spurious pickups represent a hazard to hydroacoustic equipment since their spectrum lies in the audio frequency range, i. e. it is in the passband of the stations. This reduces threshold sensitivity and is detrimental to the classifying capabilities of the stations.

By their nature, spurious background pickups are harmonic components of the line frequency that arise as a consequence of deviation of the current waveshape in the power line from sinusoidal. The primary sources of background pickups are the electric generators that power the equipment, and this does not necessarily mean the particular equipment that shows the pickup at its outputs.

It is nearly impossible to give the supply voltage the ideal waveshape, and therefore the existence of harmonic components that are multiples of the frequency of the supply voltage is inevitable in the supply circuits. It has been demonstrated [Ref. 45] that voltage of a generator working without load contains harmonic components of odd multiplicity (e. g. 50, 150, 250 Hz, etc.), with amplitude that decreases with increasing number of the harmonic (Fig. 5.1).

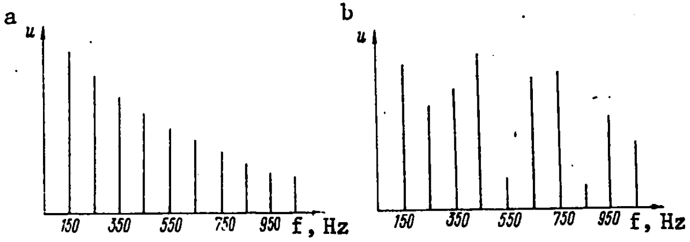


Fig. 5.1. Harmonic components in line voltage: a--generator without load; b--generator under load

However, the pattern of amplitude distribution changes sharply when the generator is connected to a load, and each change in the load, e. g. switching the working mode of the station, changes the spectrum of harmonics in the supply circuit. The selective properties of equipment make their own contribution to the resultant spectrum of possible pickups, and may cause the occurrence of even harmonics of the funcamental frequency at the station output.

In the process of operation, the level and spectrum of spurious background pickups may change appreciably even with a slight change in wiring, relative placement of individual instruments or malfunction of interference-rejection units. Only service personnel who have a good idea of the physical essence of the problem effect can deal with the influence of sources of interference and the variety of types of stray coupling that carry spurious background pickups.

Three kinds of stray coupling can be distinguished that give rise to background pickups: coupling through a common resistance, capacitive coupling, and inductive coupling.

The common resistance of different equipment components may be the internal resistance of the power supplies, common connecting wires of supply, control and monitoring circuitry, the common section of the instrument chassis or grounding lines. This kind of coupling shows up frequently as "non-equipotentiality" of the housings of different instruments, i. e. as a certain potential difference between the grounding points of the instruments. This effect is eliminated by selecting the grounding points of the instruments and of various points of the wiring.

Capacitive stray coupling is equivalent to connecting a parasitic capacitor between the source and receiver of the pickup. This capacitance serves as a source of action of the electric component of the electromagnetic field of the pickup source on the equipment components. Capacitive stray coupling is reduced by using electrostatic shields in the form of cable brading or special metal cans. Electrostatic shielding can be treated as closure of the electric field to the metal surface of the shield and transfer of the induced electric charges to the housing of the instrument.

Inductive stray coupling is characterized by the mutual inductance between circuits of the source and receiver of the pickup, and is a cause of action of the magnetic component of the field of the pickup source on the equipment. Inductive stray coupling is reduced by using magnetostatic shields with working principle based on closure of the magnetic field within the shielding wall made of high-permeance metal. Shielding from the magnetic component is more complicated than for the electric component since it is necessary to make the shield fairly thick to reduce reluctance.

It should be borne in mind that the shielding action of protective devices of both types is considerably reduced when breaks are present. Particular attention should be given to the reliability of bridging between the housing of the instrument and cable braiding, between pipes and the hull of the ship, and between the separate parts of the shielding.

Capacitive and inductive stray coupling can be considerably reduced by separating the interference-sensitive devices from power-supply units and power cables. The level of pickups depends on the placement of the signal strands in the cable relative to the neutral wire. Experience has shown that it is optimum to use the central strand of the cable as the neutral wire.

To prevent harmonics from penetrating into the instruments through the supply circuits, filtration is often used by connecting large-value capacitors between the wires of the power supply line. However, this technique has considerable disadvantages: when harmonics from one part of the spectrum are reduced, there may be an increase in the level of harmonics from another part, as well as an increase in the magnetic component of the field of the harmonics with an increase in their currents through the low resistance of the capacitor. A much more effective means is to determine the paths of penetration of the pickups into the equipment, and to provide shielding and filtration inside the instruments.

The overall level of electric interference is measured by an rms voltmeter connected at the output of the hydroacoustic station (Fig. 5.2). The action

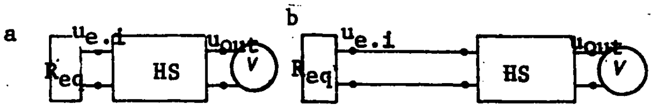


Fig. 5.2. Block diagram of measuring the level of electric interference: a--measurement of set noises; b--measurement of overall noises

of acoustic interference is eliminated by substituting a dummy load for the station antenna. The gain of the reception channel is selected so that the voltage measured at the station output is within the limits of the linear segment of the amplitude response.

The measurement technique depends on the kind of interference to be measured. For example, interference caused by internal sources (set noises and line background) is measured under conditions that minimize the action of external pickups. To do this, as many electrical mechanisms as possible are turned off on the ship. The artificial antenna is connected directly to the preamplifier input to prevent the influence of pickups on the cable connecting the antenna to the station. The overall interference level including external spurious pickups is measured under conditions where all shipboard electrical mechanisms are in operation. The dummy load is connected as close as possible to the acoustic antenna.

The interference voltage u_{out} measured at the station output is normalized by reference to the station input according to the formula

$$u_{n.e} = \frac{u_{out}}{k} \times 10^6, \tag{5.1}$$

where $u_{n.e}$ is the level of electrical interference normalized to the station input, μV, and k is the gain at which the measurements have been taken.

If the interference level found from (5.1) exceeds the value stipulated in the station specifications, steps must be taken to find the sources of the excess interference. The percentage of each kind of interference can be found by subtracting the level due to internal sources from the overall interference level, and then determining the region of search for the source of interference.

To find internal interference sources, which might be noisy tubes, transistors, self-excited amplifiers, malfunctioning smoothing filters and the like, it is sufficient to use an oscilloscope, since interference of this type either has a continuous spectrum or exists on a single frequency. The interference spectrum from external sources is more complicated and is discrete in nature (see Fig. 5.1), and location necessitates the use of a spectrum analyzer that enables measurement of the level of each harmonic component of background pickup.

According to technical data, a V6-2 selective voltmeter is the most suitable device for use under shipboard conditions. The instrument makes direct

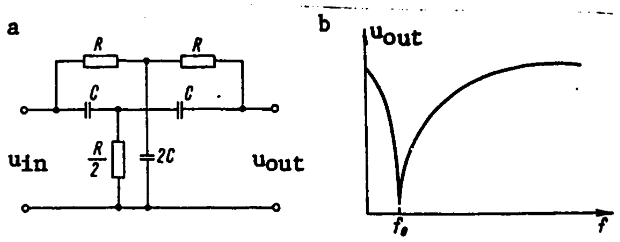


Fig. 5.3. Test filter: a—schematic diagram; b—frequency response

measurements of the useful signal in the transmission circuits. Harmonic components in the supply line can be measured only when a test filter is connected to the input of the instrument. This filter is calculated for the fundamental harmonic frequency (Fig. 5.3) by formula

$$f_0 = \frac{1}{\pi RC},$$

where f_0 is the fundamental harmonic frequency (i. e. the line frequency).

For example, for a frequency of 50 Hz, the parameters of the filter components must be $R = 150$ kΩ, $C = 0.02$ μF. A single such filter section suppresses the fundamental harmonic by a factor of 400. Harmonic components measured by using a test filter must be amplitude-corrected with consideration of the known frequency response of the filter.

The instrument can be used to measure the magnetic component of the pickup field, which is most hazadardous for hydroacoustic equipment. To do this, it must be equipped with a circular coil made of insulated copper wire 0.3-0.6 mm in diameter. The coil has 100-1000 turns with average diameter of 80-20 mm respectively. The coil must have an electrical shield made of metal foil with a circular slot 1-2 mm wide. The coil is connected to the input of the selective voltmeter, and is moved around the investigated instrument to look for the pickup source. The coil has directional properties, and therefore it must be aimed for the maximum voltmeter reading during measurement.

In case no small selective voltmeter is available, spectral analysis of electrical interference can be done in a laboratory provided with the necessary equipment. To do this, the interference must be recorded on magnetic tape. To record harmonic components directly in the supply line, it is recommended that a test filter be used (see Fig. 5.3) that is made up of two identical sections. Interference recording should be preceded by recording of an audio frequency oscillator signal calibrated with respect to both frequency and amplitude.

§5.2. Acoustic interference to HS operation and measurement methods

As was pointed out in §1.1, interference to signal reception is one of the most important factors that are detrimental to the output characteristics of hydroacoustic facilities. Their main component is shipboard acoustic interference or set noises [Ref. 3, 14], meaning the background noise produced by the carrier vessel and received by the acoustic antenna of the station, interfering with signal reception. Acoustic interference shows up as voltage measured at the station output under conditions of minimum sea noise in the absence of useful signals.

Acoustic interference should be distinguished from the shipboard or radiated noises that constitute the primary acoustic field which is defined as the sound pressure measured in the free acoustic field of the ship and normalized to a unit field of 1 m [Ref. 14].

The main sources of acoustic interference are the screws, mechanisms and fairing with associated part of the hull plates. Depending on the source and the paths of its action on the antenna, we can differentiate principal components of acoustic interference: noise, hydrodynamic, vibrational and structural.

The noise component of interference is the result of action on the antenna by the nearby acoustic field produced by operation of the screws and mechanisms. The level of noise interference increases with increasing velocity, and with particularly great intensity starting at some critical velocity where screw cavitation begins (Fig. 5.4). Especially strong interference results from "singing" of the screw on discrete frequencies that are a consequence of resonant excitation of the screw blades. The spectrum of the noise component at velocities greater than critical is continuous. At slower velocities, discrete components are observed in the low-frequency part of the spectrum. The frequency of these components is determined mainly by the "blade" frequency of rotation of the screw [Ref. 69] $f_{bl} = mnv/60$, where m is the number of screw blades, n is the number of the harmonic of the blade frequency, and v is the screw speed in rpm.

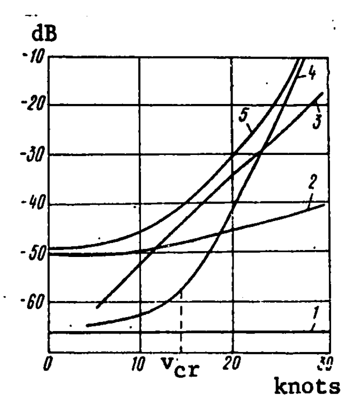


Fig. 5.4. Level of shipboard acoustic interference as a function of ship velocity: 1--level of electrical interference; 2--interference from mechines and mechanisms; 3--hydrodynamic interference; 4--noise component; 5--overall interference level

The level of noise interference can be reduced by using special screw design, by putting as much distance as possible between the antenna and screws, and by using a sound-absorbing cover to shield the stern side of the antenna. To prevent "singing" of the screws, a constant watch must be kept on the state of the blades. It is recommended that practical use be made of hydroacoustic stations at ship speeds below critical.

The hydrodynamic component of interference is caused by excitation of the fairing and adjacent hull plates by counterflow of water. The spectrum of the hydrodynamic component is continuous, and the level increases uniformly with increasing ship speed. Deterioration of the state of the fairing surface may cause a sharp increase in the level of the hydrodynamic component of interference, and therefore care should be taken in the operating process to see that the fairing surface is kept free of cracks, dents, fowling and so on.

The vibrational and structural components are the result of action of vibrations of operating shipboard mechanisms and devices, including the gearing and shafts of the screws. In this connection, the vibrational component is due to the action of vibrations on the fairing, and thence through the water on the antenna, while the structural component is caused by the action of vibrations directly on the antenna transducers. The level of these components has comparatively little dependence on the ship velocity, and the spectrum is discrete (Fig. 5.5) [Ref. 69]. A reduction in vibrational and structural components is attained by improving the design of mechanisms, isolating them from the hull be special shock absorbers, acoustic decoupling of antenna components from the hull, and locating auxiliary mechanisms as far as possible from the antennas. During the operation of stations, it is necessary to monitor the state of acoustic decoupling of mechanisms, devices, pipe clamps and so on. A watch should also be kept on the state of the mechanisms themselves since imbalance of rotating parts, wear of gear teeth and the like can sharply increase vibrations. When stations are in use, it is recommended that mechanisms be shut off that are not in use at a given instant, especially if they are situated close to acoustic antennas.

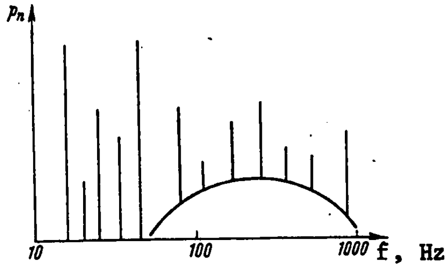


Fig. 5.5. Exemplary spectrum of shipboard acoustic noises

In addition to the enumerated components, random acoustic interference can arise from local sources, e. g. as a result of crew activities, blows from objects that are not fastened down and the like, which must also be taken into consideration in implementing steps aimed at reducing the level of acoustic interference.

Practice has shown that acoustic interference is markedly directional (see Fig. 1.2). In the sternward direction, noise level increases as a rule due to the screw noise component, while in the direction of the bow there is an increase in the action of the hydrodynamic component when the ship is making headway. In other directions there may be an increase in interference level due to the action of localized isolated sources. Best reception conditions are found in sectors that correspond to minimum interference levels, which should be considered by the pilot during maneuvers.

Acoustic interference is a parameter that is subject to considerable variability due to the action of many factors that are difficult to account for. Therefore, thorough periodic monitoring is necessary. Acoustic interference should be checked in two stages. The first stage consists in measuring the integrated level of interference as a function of direction (headway) for different ship speeds, and has the goal of verifying that the acoustic interference level is within set norms. The second stage includes analysis of acoustic interference, and is carried out if the first stage has shown elevated interference levels. The purpose of the second stage is to locate sources of elevated interference on the basis of fine spectral analysis that reveals discrete components caused by operation of specific mechanisms and devices.

Measurements on the first stage reduce essentially to measuring the sound pressure of interference acting in the vicinity of the acoustic antenna, i. e. the measurement algorithm should conform to the arrangement shown by the sound pressure meter in Fig. 3.7. In practice, the acoustic station itself is used as the sound pressure meter since in this case the working conditions of the given station are taken into account automatically with consideration of all components of shipboard acoustic interference.

The acoustic antenna of the station, whose sensitivity is known from specifications or can be determined during measurement of overall sensitivity (see §4.2), is used as the measurement hydrophone. The role of instrument amplifier with bandpass filter is played by the reception-amplification channel of the station, the voltage across the output (usually a headset) being measured by a voltmeter or by a signal-level chart recorder.

Since the reception-amplification channel is used as the instrument amplifier, it must be calibrated before measurement, i. e. its gain (see §4.2) and level of electrical interferences (see §5.1) must be determined. The measurements should be done on a body of water that meets the requirements specified in §4.1.

Voltage measurements are made at anchor and at various ship velocities; in each state of motion, the output voltage is found as a function of the course angle by rotating the directivity pattern in the scanning sector. Readings are taken over certain angular intervals depending on the required degree of detail. The position of the gain control should ensure measurement on the linear section of the channel amplitude response (see §4.1).

In the case where an rms voltmeter is used, the interference p_i is calculated by the formula

$$p_i = \frac{u_{out}}{\zeta_\Sigma} A^{N-m}. \tag{5.2}$$

The quantity calculated on the basis of equation (5.2) includes the level of electrical interference. To find the value of acoustic interference alone, the level of acoustic interference calculated by formula (5.1) must be normalized to units of sound pressure measurement as

$$p_{e.i} = \frac{u_{e.i}}{\zeta_{a.a}}. \tag{5.3}$$

The level of acoustic interference $p_{a.i}$ can be found as

$$p_{a.i}=\sqrt{p_i^2-p_{e.i}^2},$$

Measurement results are tabulated, and can be graphed in polar coordinates (Fig. 5.6) or rectangular coordinates (see Fig. 1.2).

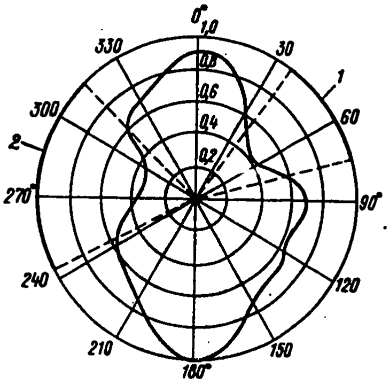


Fig. 5.6. Shipboard acoustic interference as a function of course angle: 1, 2--sectors of optimum signal reception

Expression (5.2) enables evaluation of reception conditions for a specific station carried on a given vessel. To compare working conditions of hydroacoustic stations on different vessels independently of station parameters, the results are normalized, i. e. they are referenced to a band of $\Delta f = 1$ Hz, average frequency $f_0 = 1$ kHz, and non-directional reception by the formula

$$p_{i0} = p_i\frac{\sqrt{\gamma}f_{av}}{\sqrt{\Delta f}}, \tag{5.4}$$

Where f is measured in kHz, and Δf -- in Hz.

Expression (5.4) can be used only for interference that is isotropic in direction as otherwise the concentration coefficient γ loses meaning as a parameter for evaluating interference resistance [Ref. 59]. Therefore if the sources and paths of interference propagation are undetermined, it is recommended that data obtained from (5.2) be used for calculations.

The measurement process is considerably speeded up if a chart recorder is used instead of a voltmeter. In this case the directivity pattern of the station should be rotated at constant velocity to get a uniform scale of course angles lengthwise of the recorder chart.

Voltage level readings u_t taken from the chart recorder tape in dB must be converted to sound pressure interference levels relative to a reference pressure of $2\cdot10^{-4}$ dyne/cm^2 or $2\cdot10^{-5}$ Pa. To carry out the conversion, it is recommended that a formula be used that is obtained by taking the logarithm of expression (5.2):

$$p_i = u_t + (N - m)A - 20\lg\zeta_\Sigma + 34, \tag{5.5}$$

where $u_t = 20\lg\frac{U_{out}}{0.01} = 20\lg U_{out} + 40$; $-20\lg(\zeta_\Sigma/2\cdot10^{-4}) = -20\lg\zeta_\Sigma - 74$.

If $2\cdot10^{-5}$ Pa is taken as the reference pressure, the last term is replaced by 54 dB.

To carry out the second measurement stage, the acoustic interference voltage is analyzed by a spectrometer with fairly narrow analysis band. Analysis in a ⅓-octave band precludes isolation of discrete components belonging to individual mechanisms (Fig. 57), [Ref. 3]. Therefore it is necessary to use

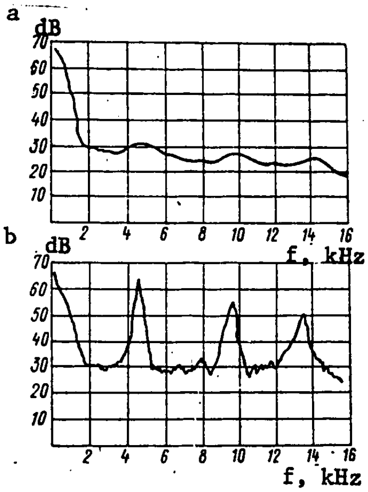


Fig. 5.7. Results of spectral analysis: a--using ⅓-octave filter; b--fine spectral analysis

heterodyne harmonic analyzers, or analyzers with time compression of the signal. In case portable instruments are not available, the interference is recorded on magnetic tape together with a calibrated control signal for subsequent laboratory analysis.

CHAPTER 6: LOCATING AND ELIMINATING MALFUNCTIONS IN MODULES AND COMPONENTS OF HYDROACOUSTIC STATIONS

§6.1. Factors that determine restorability of hydroacoustic stations

The increasing complexity of modern electronic equipment, and in particular hydroacoustic equipment, is to a great extent outstripping the improvement of system reliability. This reduces the mean time between failures and increases the down time of equipment during restoration of operability. The problem of hardware recoverability is of considerable interest for both makers and users. On the production stage, it is resolved by attaining a given level of hardware repairability typified by patterns of prevention and elimination of failures, i. e. by the technical aspect of repairability. According to

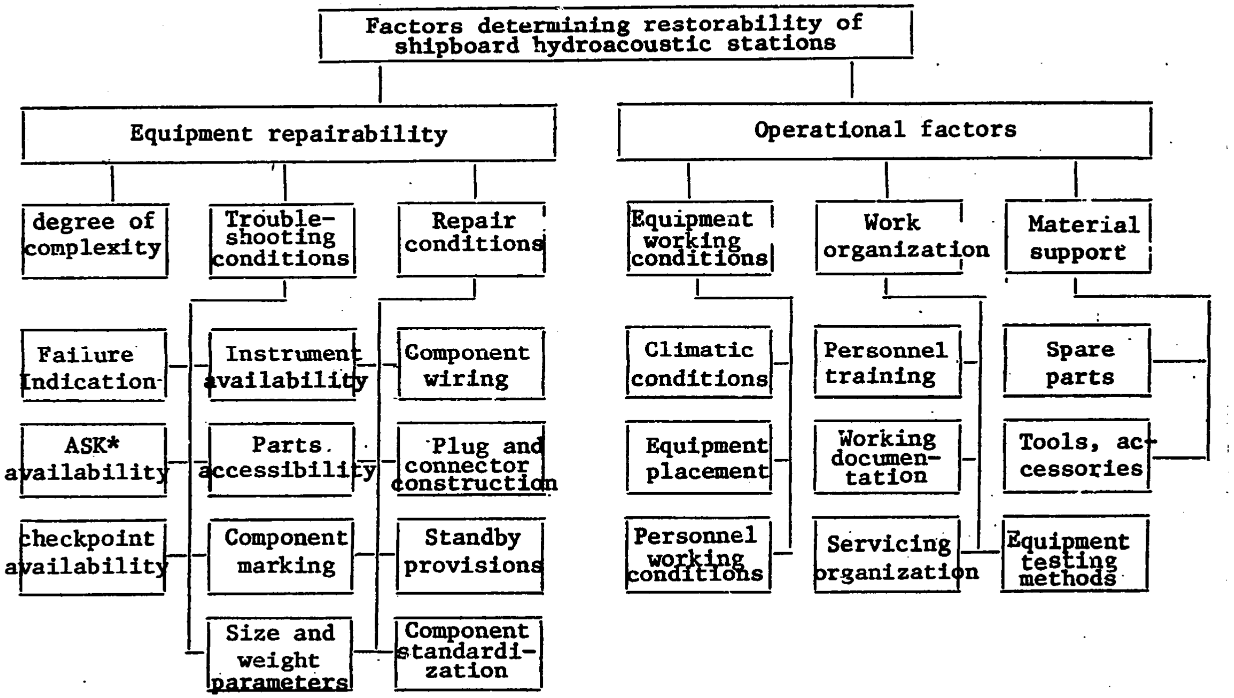


Fig. 6.1. Classification of factors that determine restorability of hydroacoustic stations

*ASK = avtomatizirovannaya sistema kontrolya [automated monitoring system]

GOST specifications [Ref. 36], repairability is the term given to the property of a system that consists in adaptability to prevention and detection of the causes of failures, damages and elimination of their consequences by carrying out technical servicing and repairs. The organizational aspect of the problem is more in evidence on the operational stage, i. e. administrative and material support and training of service personnel. Fig. 6.1 shows the classification of factors that determine restorability of equipment.

The influence of various factors on operability of equipment enables us to use operational experience to determine the necessity of carrying out so-called planned repairs, i. e. repairs provided for in normative documents [Ref. 48]. Elimination of sudden failures that arise usually when the equipment is being used for its purpose is handled immediately after detection by unplanned repair.

Depending on the particulars of operation, degree of wear and the technical state of equipment, as well as the labor inputs for regulatory work, a distinction is made between navigational repair, routine maintenance and overhaul.

Navigational repair, which is sometimes called preventive maintenance, is done during preparation for the voyage, and consists in upgrading the level of equipment operability by replacement or restitution of individual components utilizing the efforts of service personnel over periods that usually do not exceed a few days.

Routine maintenance involves restoring the output functional characteristics of a system by carrying out repairs or replacing malfunctioning components with elimination of detected problems. Routine maintenance is handled by repair facilities -- electronics shops or the electronics departments of ship repair plants.

Overhaul has the purpose of restoring operation of equipment and total or nearly total restitution of the work life of the system with replacement or restitution of any components, which may amount to more than 50% of the entire equipment. This kind of repair is done by land-based repair enterprises.

After routine maintenance and overhaul, a complete check is run, and the equipment is aligned with measurement of electroacoustic parameters immediately after repair, and measurement of output parameters on a shakedown cruise. Guaranteed time of trouble-free operation of equipment after repairs is usually indicated in the repair-release documentation; otherwise the warranty extends to three months after routine maintenance, and to six months after overhaul. Times between repair for equipment that has undergone routine maintenance or overhaul are shortened by 20-30% [Ref. 2].

Unplanned repair is also provided for by normative documentation, only without stating times. Depending on the extent of damage to the equipment, a distinction is made between running and emergency unplanned repairs. Running repairs can be handled by service personnel using spare components, units and modules. As a rule, emergency repairs require the facilities of repair organizations and more components than are avaialble in the ship's inventory.

If equipment has failed during the warranty period, it is the right of the users to make a claim against the manufacturing enterprise. In doing this, a claim sheet is filled out stating the complaint against the manufacturer regarding deviation of working quality of a hydroacoustic station from technical specifications during the warranty period.

Flaws that are cause for a claim are: premature wear or breakage of parts, components and modules of the equipment, and also malfunctions of the device as a whole that cause it to fail; considerable deviation of parameters from the norm that cannot be eliminated by following operating instructions for adjustments and alignments.

The claim sheet is to be compiled by the manufacturer's representative; if this is not done within 10 days of notification of the defect by telegram, the claim is compiled by the chief of the electronics service on the vessel, signed by the captain and presented to the SERP [expansion not given] which transmits the claim to the manufacturer. The plant must eliminate the defect at no charge by repairing or replacing the failed component.

The level of repairability of marine hydroacoustic stations is assumed to be characterized by quantitative indices that can be grouped into two categories [Ref. 2]: operational (temporal) and economic. Operational indices include: average recovery time T_B, probability of restoring operability within a predetermined time $P(t_B)$, and operability restitution parameter $\mu(t_B)$. The time of restoring operability of equipment is comprised first of all by the active repair time, and secondly by the time for ensuring repair. Calculation of the active repair time for any facility is based on knowledge of the failure rate of its components and the time necessary for repairing these components. On this basis, the average repair time T_a is found from the relation

$$T_a = \sum_{j=1}^{n} q_j t_{a\,j} = \sum_{j=1}^{n} \lambda_j t_{aj} / \sum_{j=1}^{n} \lambda_j,$$

where q_j is the conditional probability of failure of elements of the j-th group; t_{aj} is the active repair time upon failure of elements from the j-th group; n is the number of groups of elements.

As we can see the quantity that determines the average time of restitution of operability is time t_{aj}. This time depends to a considerable degree on the type of failed component, the complexity of the equipment and its design peculiarities.

In the case of a large volume of statistical data, such a point evaluation can be used, but in the case of few data it is necessary to determine interval estimates by known methods of probability theory. For a rough estimate of repairability, particularly on the design stage, use can be made of statistical data obtained in the operational process or on tests of equipment of similar types or purpose. Some of these data for individual electronic components are summarized in Table 2 [Ref. 7].

Analysis of operational experience with marine hydroacoustic stations has enabled us to define the reliability of restitution time among other

TABLE 2

Statistical indices of restoration of operability of electronic equipment by types of failed components

Type of failed component	Detection time, hr	Correction time, hr	Average repair time, hr	
			min	max
Vacuum tubes	0.72	0.22	0.23	0.96
Oscillator tubes	0.4	0.44	-	-
Resistors	0.4	0.27	0.3	0.98
Capacitors	4.4	1.8	0.4	1.7
Tuning controls	3.0	5.2	-	-
Switches	-	-	0.25	1.06
Motors	-	-	1.26	5.13
Tanks circuits	-	-	0.65	2.8
Wiring	1.26	1.37	-	-
Fuses	-	-	0.75	3.2

quantitative indices. For example, for the Paltus-M hydroacoustic station this index is 2.4 hr, for the most reliable Soviet fish-locating hydroacoustic station -- the Kal'mar -- it is 0.75 hr, and for the sonar unit in the Pribor-101 fish-locating set with little cumulative operating experience, it is 3.4 hr.

The probability of timely restoration $P(t_B) = \text{Prob}|\tau_B \leqslant t_B|$, i. e. the probability that the running recovery time τ_B will not exceed a predetermined time t_B, is found from the expression

$$P(t_B) = \int_0^{t_B} f(\tau_B)\, d\tau_B,$$

where $f(\tau_B)$ is the probability density function of recovery time. For example in the case of an exponential law

$$f(\tau_B) = \frac{1}{T_B} \exp(-t_B/T_B)$$

probability $P(t_B) = 1 - \exp(-t_B/T_B)$.

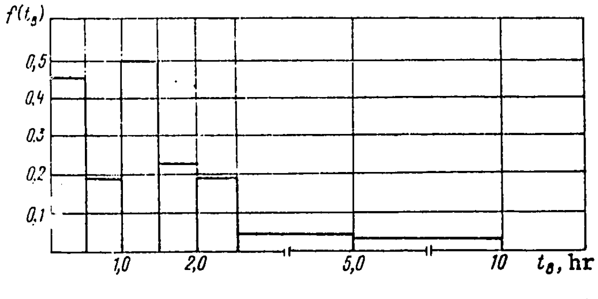


Fig. 6.2. Histogram of time of recovery of operability of the Paltus-M hydroacoustic station

Statistical data on the times of recovery of operability for the Paltus-M hydroacoustic station have been used to plot a histogram of the distribution of these times (Fig. 6.2) in accordance with the relation

$$f(t_B) = n(\Delta t_{Bi}) / N(t_B)\,\Delta t_{Bi},$$

where Δt_{Bi} is the i-th interval of grouping of values t_B; $n(\Delta t_{Bi})$ is the number of values of t_B in the i-th interval Δt_B; $N(t_B)$ is the total number of recoveries.

The flow of recoveries μ_B characterizes the recovery rate, or the number of repairs that have been made in a unit time. This quantity can be determined from statistical data for n failures as the reciprocal of the average recovery time $\mu_B^* = N(t_B)/t_B$. In the case of an exponential law of distribution of the probability density of recovery time, $\mu(t_B) = 1/T_B = \text{const}$.

Generalization and dissemination of experience with equipment operation as well as a number of technical measures are improving $\mu(t_B)$ with a corresponding reduction in t_B, which can be illustrated by the way that this last quantity (yearly average for the Paltus-M hydroacoustic station) depends on operating time:

Year of operation	1970	1971	1972	1973	1974
t_B, hr	2.3	1.8	2.1	1.6	0.5

Economic quantitative indices characterize the expenditures of labor and material resources on restoring operability of equipment. These include [Ref. 2]: repair cost, average repair cost, and also the coefficient of recovery cost. The repair cost depends on a large number of factors, among which are: cost of components, materials, electric energy, depreciated cost of equipment, payments to repair agencies. Obviously this is a random quantity, but the average cost of repair can be deduced from operational experience as the mathematical expectation of the cost. Depending on expenditures for restoring operability in the case of a specific failure, equipment can be categorized as either repairable or non-repairable. Advisability of repair can be established from the ratio of expenditures for repair C_r to the cost of making and installing the given equipment C_{pr}, i. e. from the coefficient of recovery cost $k_{rc} = C_r/C_{pr}$. Depending on remaining service life and other factors, the critical or threshold value of this parameter that corresponds to making a decision about advisability of repair may range from very small values to nearly unity. In the latter case, the decision about advisability of repair is made when there is no possible way to replace the failed component, unit or module.

Let us note possible ways of improving the repairability of technical systems: improvement of the method of locating failures; using automated monitoring systems; optimizing the spare parts inventory; improving the skill of service personnel; improving repairability of equipment; improving the method of predicting failures; improving technical documentation.

§6.2. Troubleshooting methods

The process of running repair of equipment can be divided into four stages: determination of failure; establishing the nature of the failure and localizing the malfunction; correcting the malfunction; post-repair check of operability.

These stages are common to all known methods of repair regardless of the method of recovery of operability--automated or manual. Analysis of the steps taken, and also experience in operating hydroacoustic equipment, have enabled determination of the average proportion of time expenditures with respect to each of the stages. According to Ref. 4, the average time of preparing monitoring and measurement equipment and localizing malfunctions in electronic equipment takes up about 77%, correcting the malfunction -- 15%, and post-repair check -- 8% of the technical time of recovery, which in turn takes up only about ¼ of the total recovery time. More than 70% of this time goes for nonproductive expenditures, e. g. various steps to organize repair, including down time due to lack of spare parts. Analysis of statistical data over five years of operation shows that proportion of required time expenditures for finding and correcting malfunctions in hydroacoustic stations, for example in fish-locating gear, is 31% and 69% respectively for the Paltus-M hydroacoustic station, and 42 and 58% for the Priboy-101 set. As we can see, a considerable part of the active repair time goes to locating the malfunctions. This fact makes it necessary to optimize the algorithm for locating malfunctions so as to reduce the pinpointing time.

The location of malfunctions in modern hydroacoustic equipment is increasingly difficult because of growing complexity, which increases the time for checking the working state and determining the causes of a failure. The search algorithm can be simplified if certain a priori data are available on the properties of the equipment: probability of failure of given components, time for checking operability of these components and so on.

Analytical determination of the optimum check sequence in locating a failed component involves the following assumptions: the equipment consists of n components with independent failures; q_j is the probability of failure of the j-th component; τ_j is the time spent in checking the component.

It can be shown [Ref. 7] that in the case of equipment failure, the conditional probability q_j^* of failure of the j-th component leading to equipment failure is determined by the relation

$$q_j^* = (q_j/p_j) \sum_{j=1}^{n} q_j/p_j,$$

where q_j and p_j are the a priori probabilities of failure and trouble-free operation of the j-th component, and n is the number of components. When the number of the component coincides with the number of the check over time τ_j, the time for locating a malfunction when the i-th component has failed is

$t_{\text{п. н} i} = \sum_{j=1} \tau_j,$ and the average time for pinpointing the failure is defined as

the mathematical expectation in the form

$$t_{\text{п. н}} = \sum_{i=1}^{n-1} \left[(q_i/p_i) \sum_{i=1}^{i} \tau_j \right] \Big/ \sum_{j=1}^{n} q_j/p_j.$$

The optimum sequence for checking operability of components, or the optimum algorithm for pinpointing a failure, is found by solving the variational problem on minimizing $t_{\text{П.Н}}$, giving condition $\tau_1 p_1/q_1 < \tau_2 p_2/q_2 < \ldots < \tau_{n-1} p_{n-1}/q_{n-1}$.

If the time for checking operability of components is not assigned, or is approximately the same for all components, then the criterion for selecting the optimum sequence may be the condition for getting maximum information on the state of the system, which is satisfied in the case of equiprobability of events. Consequently the most information, and hence the optimum check sequence, can be obtained in the case of verifying the operability of the system at the point that divides the section of a circuit being checked into parts that have equal probabilities of failures, i. e. with so-called binary division (Fig. 6.3). This method gives a gain in the number of checks as compared with the method of unorganized trials that is estimated by the expression

$$k = N'_{\max}/N''_{\max} = (n-1)/1{,}43 \ln n = 0{,}7\,(n-1)/\ln n,$$

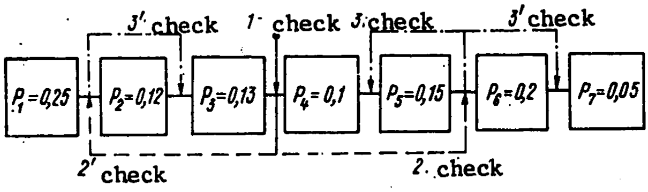


Fig. 6.3. Sequence for pinpointing malfunction by binary division

where $N'_{\max}$, $N''_{\max}$ represent the maximum number of checks of a failed component among n by the method of unorganized tests, and by binary division respectively.

This gain increases with an increase in the number of components: at $n = 8$, $k = 2.35$, and at $n = 15$, $k = 4$.

In the practice of using hydroacoustic equipment for locating malfunctioning components upon occurrence of a failure, sometimes tables of typical malfunctions are used, although the increasing complexity of equipment precludes inclusion in these tables of all possible malfunctions, and the ambiguity of reasons for failures makes it difficult for service personnel to use such tables.

The ways mentioned above for determining the optimum sequence in checking for operability of components when localizing a failure, i. e. when searching for a failed component, are used for the main reliability hookup, and are

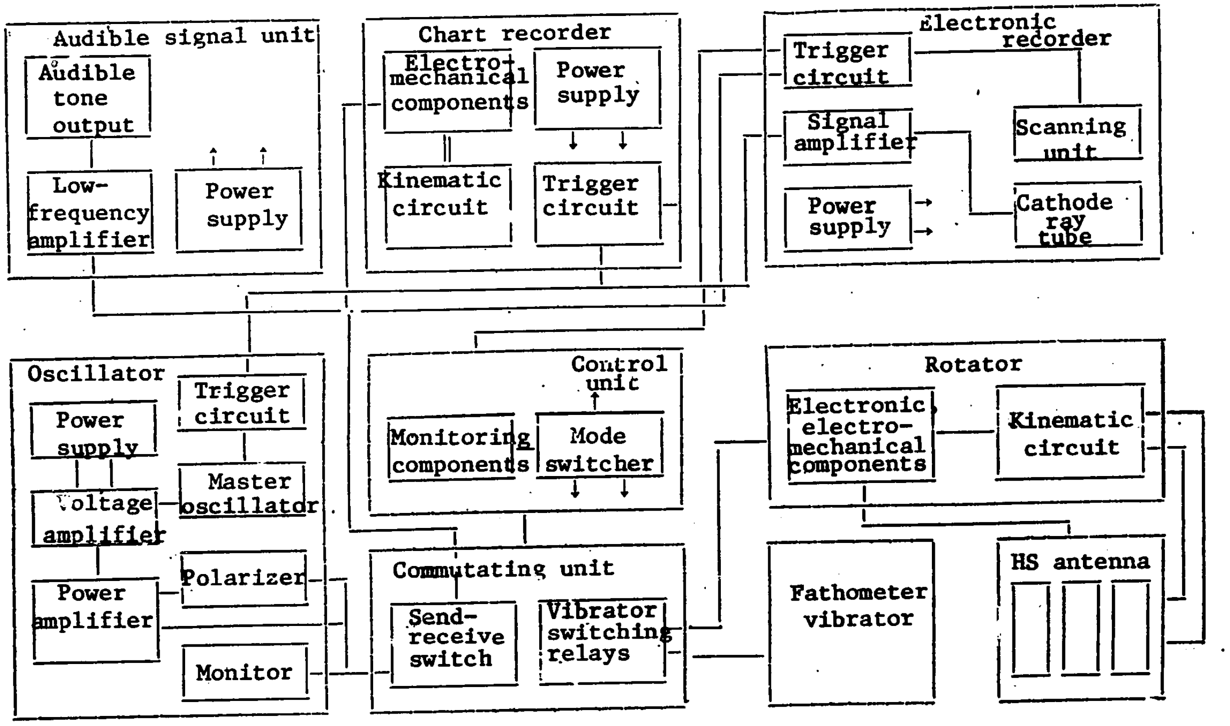


Fig. 6.4. Block diagram of typical shipboard hydroacoustic station

unsuitable for the kind of systems with branched structure representative of most present-day hydroacoustic stations (Fig. 6.4) [Ref. 29]. It is this factor that has determined the choice of a sequence for locating a failed component based on logic, and designed both with regard to the structure of the system and the results of verification of components on each step.

Thus while the search algorithm may already be defined before checking starts for the main reliability hookup, each step of the check sequence must be based on the results of the check on the preceding step for a system of complex branching structure, and not only that, it must be based on the functional relation between the component being checked and the adjoining components. Such a method of verification when looking for malfunctions in electronic systems is known as the method of functional tests [Ref. 27, 18]. In accord with this method, the components of the hydroacoustic station are divided into two groups -- functional and power components -- the former being those that receive, convert or transmit information, while the latter are sources of energy. Depending on number of inputs and outputs, functional components are divided into information sources (output only), information converters (one input and one output), data-gathering components (several inputs and one output) and information sharing components (one input and several outputs). Functional and power components are interconnected, and depending on the functions that they perform these connections may also be either functional or power connections.

Analysis of the components and the connections between them enables formal representation of the hydroacoustic station as a structure that can also be of the functional or the power type.

The logic that defines the structure of the algorithm of localizing a malfunction is based on the following information concerning malfunctions of functional and power components:

--when a functional component fails, the signal at the output is absent or distorted with normal input signal, while failure of a power component causes a deviation of output power from the norm;

--as a consequence of failure of these components, there is an interruption in functioning of components that receive power or information from the failed components;

--the generalized index of failure is the change in output parameters of the system that shows up as abrupt deterioration or cessation of functioning of the system, and also indication of a failure.

Use of the method of functional tests to find a failed component in a system with series functional structure can be illustrated by the example considered above (see Fig. 6.5). Determination of the sequence for locating the failed component is based on the probability density function for occurrence of a failure f(n) as well as the probability distribution function F(n). If a functional test has given a positive answer ("signal present", "signal amplitude within the norm", etc.), the formal search diagram places the point of the next test further to the right or closer to the output, while a negative

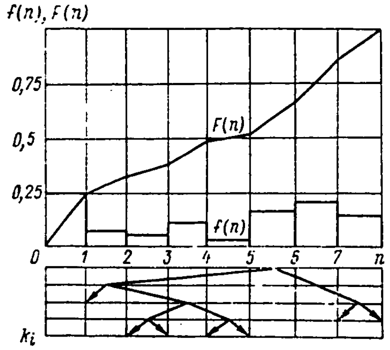


Fig. 6.5. Diagram of search for a failed component with respect to laws of failure probability distribution

response shifts the next test point further to the left or nearer to the input than the point of the preceding test. As can be seen from the figure, a tree-like formal diagram is obtained for combined search by developing the points transferred to the axis of abscissas by test numbers, and joining them together. Each branch of the diagram, i. e. the sequence of tests corresponds to only one state of the channel, and leads to one failed component.

The difference between search in systems containing nodal elements and in systems with series structure consists in the fact that in the former case it is impossible to use the principle of binary division, and a priori information can be taken into consideration only for renumbering the components of a group in descending order.

Systems or channels that contain data-gathering components are analyzed when locating malfunctions with consideration of the condition of impossibility or very low probability of simultaneous failure in two or more parallel independent circuits. An example of realization of the method of functional tests in checking a data-gathering component 3 with two inputs from units 1 and 2 is shown on Fig. 6.6 [Ref. 27].

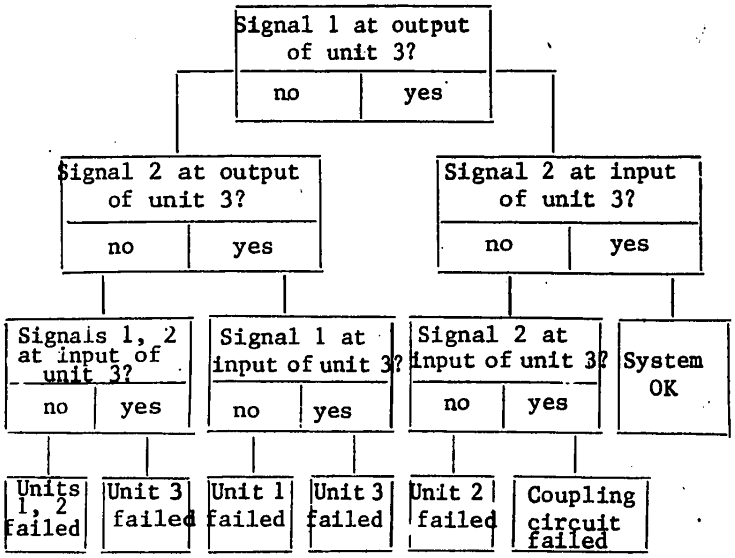


Fig. 6.6. Realization of method of functional tests in localizing a malfunction (example)

Localization of malfunctions in power components has a certain peculiarity that consists in the fact that these components are connected to practically all the functional elements, and operation of at least one of the latter is an indication that a power component is functional. At the same time, functional connections have little influence on power components, being determined mainly by secondary failures. Besides, the strong influence of these components on operability of systems as a whole predetermines their saturability with external monitoring devices. The simplicity of power components permits easy prevention of component failures by using fuses or other more complex protectors to disconnect the load.

Pinpointing the failure of a power component usually entails no difficulties since the number of parts in these components is ordinarily not large. The probable causes of failures of power components for different states of the load protector and readings of monitoring devices are summarized in Table 3.

TABLE 3

Causes of power component failures

State of load protector	Readings of monitoring devices		
	Input voltage within norm		No input or output voltage
	Output voltage within norm	No output voltage	
Normal	Break of power or functional connection in the load	Break of power connection in secondary voltage circuits	No voltage in supply circuit
Energized	Overload in functional circuits	—	Overload in power circuits of load and power system

A block diagram of locating malfunctions by the method of functional tests for a power structure is shown in Fig. 6.7.

The proposed technique can be easily realized when locating a malfunction with respect to external signs and test results in the power supply of the Paltus-M hydroacoustic station diagrammed in Fig. 6.8.

After selecting the optimum sequence for checking operability of the parts and components of the system, the checking method itself must be worked out. In hydroacoustic equipment operational practice, component operability is verified by methods of external inspection, comparison, intermediate tests and replacement.

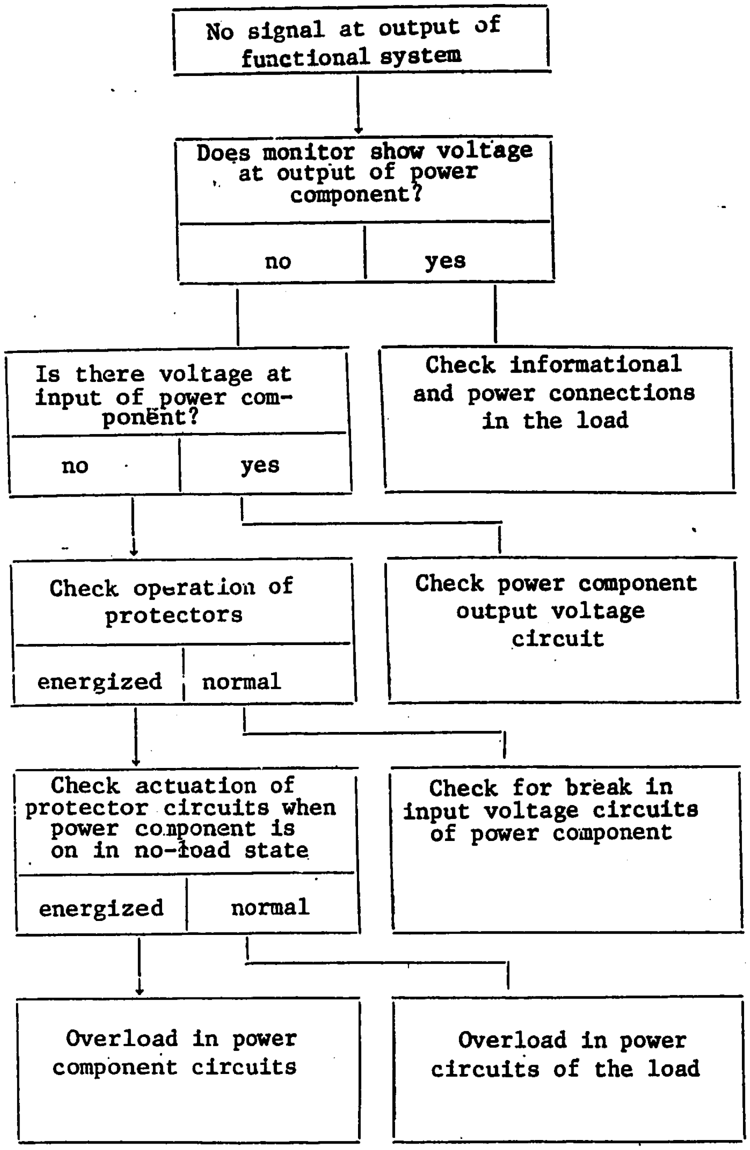


Fig. 6.7. Diagram of localizing malfunction by method of functional tests for power structure

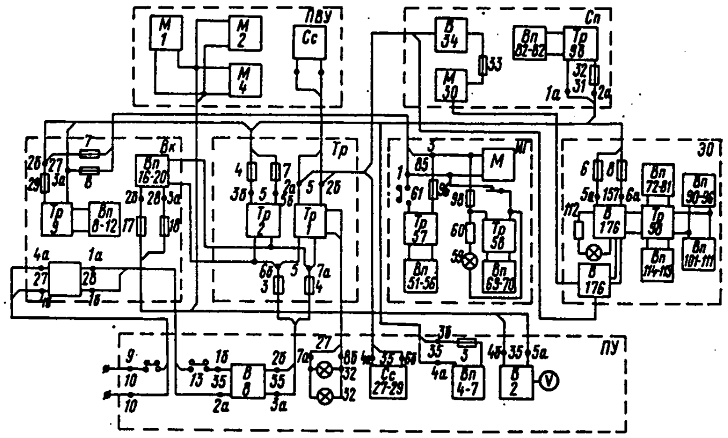


Fig. 6.8. Schematic of power supply for the Paltus-M hydroacoustic station

External inspection reveals malfunctions with causes that show up on the outside, and not just visual, e. g. from the change in color of coating of the components, deformation of the casings, absence of filament glow in tubes, but also by sound, touch and smell, e. g. overheating of transformer and coil housings, change in flexibility of wiring at a break point, or the small of overheated insulation coverings, paint, rubber, etc. This method can be quite effective, although realization necessitates considerable practical experience on the part of service personnel.

The essence of the method of comparison is identification of working conditions of components and modules classified as malfunctioning with units of the same kind operating in the same or similar conditions. Portable measuring instruments -- multimeters and oscilloscopes -- can be of considerable assistance here. The former are used to check electrical conditions at like points of the circuits being compared, and the latter are used for visual comparison of voltage waveforms. The method can be used to find malfunctions in multichannel lines of hydroacoustic stations.

The method of intermediate measurements consists in measuring the electrical and acoustic parameters of components or circuits in the equipment. The operator should be firmly convinced by such measurements that the parameters measured are within the necessary allowances. Although this method takes the longest time, it gives maximum information on the technical state of the equipment.

The method of replacement is used when there is no chance of measuring the parameters of components, subassemblies and modules by instruments, and at the same time there is a high probability that the malfunction has arisen in this particular component, subassembly or module. In addition to locating malfunctions, this method simultaneously corrects them by replacement of the failed component.

It should be pointed out that it is practically impossible to say beforehand which method should be used to locate malfunctions in a specific device. In practice, these methods are generally used in combination adapted by service personnel to the particular operating conditions, but in pursuit of a common goal -- minimizing the troubleshooting time.

Let us note the principal stages of designing search plans by the method of functional tests [Ref. 27]:

--grouping elements by the functional and power label, and unifying them into a sequence of channel sections;

--choosing the method of checking the elements that comprise separate sections;

--making up functional search plans;

--passing on to elements of search on lower levels.

§6.3. Methods of correcting principal malfunctions of components and subassemblies in shipboard hydroacoustic stations

The circuitry of present-day hydroacoustic stations contains a large number of electrical and electronic subassemblies, components and parts. The diversity of their working principles, operating conditions, and responses to external stimuli predetermines diversity both in causes of failures, and in the methods used to restore operability. Let us consider typical features of malfunctions and methods of correcting them in the most extensively used components and parts of hydroacoustic stations.

1. Electroacoustic Transducers. A peculiarity of using electroacoustic transducers as active components of acoustic antennas is that they are subject to the continued action of sea water, the salts dissolved in it, considerable differentials of hydroacoustic pressure and variable mechanical loads (vibrations and accelerations), as well as the action of oils and petroleum products, and encrustation of surfaces with a variety of microorganisms. This situation makes severe demands on the construction and installation of antenna components with respect to strength, hermeticity and effectiveness of shock mounting. These requirements are satisfied in most cases, which determines the comparatively high reliability of the elements of acoustic antennas, and failures of these units account for 3% of the total number of failures in the Paltus-M hydroacoustic station, and 4% of the total for the Priboy-101 fish-locater. However, as opposed to other components and modules of hydroacoustic equipment, in the overwhelming majority of cases electroacoustic transducers, especially piezoceramic elements, cannot be restored, and therefore repair is handled by replacement, which requires drydocking in the case of a stationary antenna located below the waterline. Particular attention during replacement must be given to observing polarity when installing leads, and ensuring that current-carrying components are watertight.

2. Resistor. The main parameters of resistors are: nominal value; permissible deviations of resistance from the nominal value in percent; working temperature range; permissible relative humidity; temperature coefficient of relative change in resistance; maximum working voltage.

According to construction and purpose, resistors are divided into fixed, variable and adjustable. According to the type of current-carrying material, resistors are divided into carbon, borocarbon, metal-film, wire-wound and so on.

Malfunctions of resistors in hydroacoustic stations are of two kinds: electrical and mechanical. Electrical malfunctions show up as a break in continuity of the conductive layer due to burn-through, an increase in resistance due to current overload or aging, and a reduction of resistance due to partial or total breakdown through the current-conducting layer.

Mechanical malfunctions that lead to failure of resistors show up as damage of lead-ins, abrasion of the current-conducting layer by contacts, and also breakage of the stops on the rotating sector in variable resistors when permissible forces are exceeded.

It should be noted that the percentage of failures of hydroacoustic stations by reason of resistor failure is low, and such failures occur chiefly in modules with a high load factor. For example, failures of the Paltus-M hydroacoustic station for this reason amount to 7.5%, of which 6% applies to resistors in the oscillator (oscillator failures account for 61% of the total number of failures).

When resistors fail, operability of equipment is restored by replacing them, and only in a few cases can variable and open wire-wound resistors be repaired. The state of resistors is checked and they are replaced with the equipment turned off, and mandatory disconnection of the shunting circuit from monitoring. A failed resistor is replaced by soldering in one of the same type with at least the same power dissipation rating and precision. Malfunctioning variable resistors are replaced with identical units. In case a spare resistor of the same rating is not available, the failed resistor can be replaced by a parallel or series hookup of two or more resistors with consideration of the power dissipation of the resultant circuit.

3. Capacitors. Capacitors show the greatest diversity of types among electronic components. For example with respect to change of capacitance, they may be fixed, variable and trimmers; with respect to characteristics of the dielectric -- paper, mica, ceramic, film, electrolytic, air, glass-enamel, etc.; with respect to purpose -- isolating, feedthrough, filtering, storage, etc. In addition, capacitors differ with respect to capacitance rating, precision, breakdown voltage, temperature coefficients, fastening, etc. Within one class there may be several types, for example electrolytic capacitors are produced in the following types: KE, EGTs, EM, ET, ETO, etc.

Malfunctions of capacitors, like resistors, may be of electrical or mechanical origin. Electrical malfunctions are caused by internal breaks and shorts between the separate parts of the capacitor. Mechanical malfunctions show up as damage to the housing, bends and breaks in the leads. Data of operational results with the Paltus-M hydroacoustic station have revealed that the failures of this station by reason of capacitor failures average 13% of the total number of failures, and most of these are in the oscillator unit.

Capacitors are checked for operability by measuring the electric circuit for a short (in the case of breakdown) or an open, and also for leakage current. This check is done with an ohmmeter or megohmmeter, which will show a reading of zero or infinity for a malfunctioning capacitor in the absence of a shunting circuit. Typically, an operable capacitor shows initial deflection of the meter needle to zero with a gradual increase toward infinite resistance, the rate of change in the reading increasing with decreasing capacitance. The equipment is repaired by replacing the failed capacitor with an operable one. It should be remembered that each of the types of capacitors has a preferred field of application, and violation of this rule when replacing capacitors may change the electrical characteristics of devices. For example, electrolytic capacitors are used in circuits with direct or pulsating current, and they have polarity. Ceramic capacitors are used chiefly in high-voltage circuits with low working voltages. Sealed capacitors with paper dielectric are used in scanning circuits and anode filters, and are not recommended for use in intermediate circuits because of considerable leakage currents. Mica capacitors are most frequently used in intermediate circuits, shapers and delay lines.

When replacing a failed capacitor with an operable one, it is necessary to pay attention to the capacitance value and the breakdown voltage. If the required rating is not available, compensation can be made by parallel or series connection of capacitors of other ratings, using the following relations for computing the capacitance C_Σ and breakdown voltage $U_{n\Sigma}$:

--for a series hookup $1/C_\Sigma = \sum_{i=1}^{n} (1/C_i)$; $U_{n\Sigma} = \sum_{i=1}^{n} U_{ni}$;

--for a parallel hookup $C_\Sigma = \sum_{i=1}^{n} C_i$; $U_{n\Sigma} = U_{n\,\min}$.

4. Transformers, Chokes, Inductances. A common feature of these electronic components is the presence of a winding with a core laminated from transformer sheet iron, a solid core, or no core at all. The parameters of these elements are determined by their purpose, the power to be transferred or converted, and functional peculiarities.

For example, transformers may perform functions of signal conversion (functional component) and functions of converting and redistributing energy (power component). In the former case, the transformers are mostly low-power units, with the exception of transformers in the output stages of oscillator devices, where failures may arise due to a break in the winding, or breakdown of the anode voltage. In the latter case, the transformers are components of rectifier units, and though they are generally protected, nevertheless they do malfunction due to opens and shorts. The latter is more common; for example during operation of the Paltus-M station about 10% of the failures occurred by reason of failure of the TR-58 transformer in the oscillator unit.

Power transformers are usually not repaired under shipboard conditions because of the considerable difficulties of repairing, but rather are replaced with operable units from spare parts. In replacement, it is necessary to adhere strictly to the correspondence between lead numbering and connection point since otherwise the transformer may fail along with other components of the device, or there may be a change in the functional characteristics of the hydroacoustic station. For example improper connection in the primary and secondary circuit of the beginning and end or centertap of the sum and difference transformer of the two-channel amplifier in a hydroacoustic station causes a sharp change in the functional characteristics of this unit and of the station as a whole. Transformer operability is checked by measuring the continuity of the circuit and the resistance of windings with elimination of the shunting action of other circuit elements, and checking shorts between windings and also between the chassis and the windings. When data are not available on the winding resistance, it must be approximately evaluated from the number of turns of wire, its cross sectional area and the resistivity of the material since an ohmmeter can usually tell resistance starting from units and tens of ohms, and the resistance of some windings, for example filament windings, may be a fraction of an ohm, and an incorrect decision may be made on the basis of measurement results that the winding has a short circuit.

The principal malfunction that leads to failure of transformers is shorting between turns, producing an internal circuit in the form of a shortened coil with an abrupt increase in current. Usually transformers with this kind of trouble are not repaired, but are replaced with a good transformer from spare parts. In this case as well, it is mandatory to observe the marking of the output ends of the windings.

Transformers and chokes are repaired by rewinding, beginning with the windings situated on the outside of the coil form and ending with the winding that has failed if there is no doubt as to the operability of subsequent windings. In doing this work, it is necessary to observe conventional electrical engineering rules on coil form manufacture, coil winding, finishing and marking leads, cable-paper and varnish insulation between layers and between windings, shielding and final installation of the coil in the housing. The work is usually done on special coil-winding machines, but may be done manually in the case of a small number of turns of wire with large diameter (more than 0.4 mm).

The post-repair check is done first without current, when an ohmmeter is used to check the internal resistance of windings and absence of shorts between turns and between windings, as well as between the windings and the chassis. With current switched on, a check is made of correspondence of voltages to predetermined or calculated values, electrical strength of the insulation, and no-load power consumption by checking for overheating of windings. Overheating Δt is calculated from the change in resistance of the windings R_0 (before energizing) to R_f (after two hours of operation) in accordance with the relation $\Delta t = \frac{R_f - R_0}{0.004R_0}$, and then compared with the value given in the technical specifications.

Verification of chokes is done by measuring their inductance with special devices that simulate working conditions in the actual unit.

As contrasted with other components of the hydroacoustic station, repair of inductances under shipboard conditions and subsequent alignment of channels is difficult because of the lack of special measuring equipment; however, in an emergency inductances can be made by the service personnel.

Analytically, the inductance of coils L (in μH) can determined from the following relations [Ref. 25]:

--single-layer coil

$$L=\frac{D_{cp}h^{2}\cdot 10^{-3}}{L/D_{cp}+0{,}44};$$

--multilayered coil

$$L=\frac{8D_{cp}^{2}h^{2}\cdot 10^{-3}}{3D_{cp}+9l+10t},$$

where l is the length of the winding in cm, h is the number of turns, D_H and D_{BH} are the outer and inner diameters of the winding respectively,

$$D_{cp}=\frac{1}{2}(D_{\text{н}}+D_{\text{вн}});\qquad t=\frac{1}{2}(D_{\text{н}}-D_{\text{вн}}).$$

When a core is used, the inductance L_C of the coil will be defined in terms of the actual permeability of the core μ_C in the form $L_C=\mu_C L$, the Q of the core with the coil being approximately $\sqrt{\mu_C}$ times greater than that of a coil with the same inductance but without the core.

5. Switches, Relays. The principal trouble that leads to failure of switches is wear of rubbing parts, deformation of locators, breakage of nonmetallic components. These faults usually occur as a result of long service, but sometimes are due to violation of rules of technical service by personnel. The set of spare parts usually includes most of the types of switches that are used in the equipment, and failures that arise in these components are gradual in nature, which makes it important to carry out preventive maintenance so that they can be forestalled. In this connection, particular attention should be given to switch contacts since charring increases the resistance of the circuit being switched, and this circuit may be broken. In this case the contacts must be cleaned, and then coated with gold or silver by rubbing with an object made of these metals. When replacing switches, particular attention should be given to proper marking and installation of the wires to be switched.

Relay malfunctions are mechanical and electrical. Mechanical malfunctions are determined by damage to the moving and stationary relay components. Electrical malfunctions are differentiated as follows according to their nature:

--burnup of contacts and increase in resistance or disruption of switched circuits;

--welding or sticking of contacts during making and breaking of circuits with large currents;

--opens or shorts in relay windings.

In most cases failed relays are not repaired, but are replaced by operable units from spare parts, and attention must be given to post-repair adjustment of the mechanical strength of the moving system in the working range of controlling currents. Relay leads are generally soldered at a distance of 1.5-2.0 cm away from the surface of the board by a gun with power of no more than 100 W. In doing this, it is necessary to see that the flux and solder are kept off the board. When heating and soldering one of the relay leads, the gun must not touch any of the remaining leads.

6. Electrical Machines. These include selsyns, rotatable transformers, tachometer generators, electric motors and converters of electrical energy. These elements of hydroacoustic stations are characterized by the presence of electrical, magnetic and mechanical connections and circuits. Depending on the functions to be performed in shipboard hydroacoustic stations, electric machines may be either functional or power components. For example, functional elements with the purpose of transforming type of information are selsyns, rotatable transformers, while power components designed for energy transformation are electric motors and generators.

The principal malfunctions attendant on use of these components are: breaking of electric circuits; breaking of electric contact between moving and stationary components of machines; breaking of electric circuits; breakdown to the housing and overheating of windings of generators and motors; breaking of mechanical connections.

The major cause of failures of electrical machines is damage to insulation under the action of external factors, mainly temperature and humidity. The distribution of failures by components in electric machines is not uniform. For example, according to Ref. 44, most of the failures (91%) in machines without commutators occur in the stator winding, while 6.7% of the failures take place in the rotor winding and brushes, and the remaining failures apply to the bearings and other structural elements. In commutator machines it is mainly the rotor winding and brush assembly that fail (64%), more than 10% of the failures occur in the stator, 9.1% in the commutator, and the remaining 10.6% in the bearings and other structural components. In the process of operating the Paltus-M hydroacoustic station, cases of failure of the ADP-262 motor in the chart recorder have been noted (1.5% of the total number of failures), and also the SL-521 motor in the antenna rotator (4.5%).

7. Kinematic Components. Modern marine hydroacoustic stations have fairly complicated kinematic systems in their components and modules to perform the functions of raising and lowering antennas, deploying them in the proper direction with respect to course angle and angle of inclination, rotating the deflecting systems in electronic recorders, chart recorder range scanning, etc.

The principal malfunctions of mechanical components in these systems are jamming, slipping and unnecessary play. Jamming occurs when there is a shift in the shafts of rotating parts or when foreign metallic objects get into the kinematic system. Slipping, or the absence of the expected effect at the output of systems when mechanical forces are applied at the input, occurs in the case of worn gear teeth and loose engagement, displacement of gear centers, sliding of couplings, damage to stop pins. Increased play is also caused by component wear, weakening of spring elasticity, increased backlash of control mechanisms.

Failures of hydroacoustic stations by reason of malfunctions of the antenna rotator take second place after failures due to oscillator units. For example, malfunctions of the antenna rotator of the Priboy-101 fish locater amount to about 38% of the total number of failures. Among such problems have been: shearing of the rotator shaft pin, deformation of the worm drive, breakage of shaft pinions and so on. For the Paltus-M hydroacoustic station, malfunctions of the antenna rotator make up 28.5% of the total. Repair of mechanical components of kinematic systems of modern hydroacoustic stations on board ship is very difficult because of the lack of necessary spare parts, complexity of disassembly, assembly and subsequent adjustment of the system. Therefore repair of kinematic systems with rare exceptions is done on shore or in floating shops by skilled specialists.

Wires and Cables. An important part in hydroacoustic equipment production is played by installation of wires and cables as connecting components between outboard units and devices located inside the ship, and also for modular construction.

All cables and wires can be classified into the following major groups: low-voltage, high-voltage, watertight, and special (towing cables, guy cables and the like).

Like most components of hydroacoustic equipment, cables are also subject to failure: leaky seals, broken strands, abrasion of the sheath. Failures occur most frequently in cables that are used for moving installations of hoisting and rotating units and systems between devices.

The principal grades of wires and cables used in the manufacture and operation of hydroacoustic equipment according to GOST specifications have the following designations and purposes:

--PGSh, PGESh -- rubber-covered wires with copper conductors in rubber sheathing and shielded wires used for moving and stationary installations that may be either in open air or in sea water; these are repaired by using 8615 rubber or PI-35 insulation and S-572 hose;

--PGVO and PVDN -- watertight wires with copper conductors and rubber insulation in a rubber oil-resistant sheath designed for the windings and wiring in electric devices to operate in sea water and MVP oil; they are repaired by using 8615 rubber or PI-35 insulation and hose material according to specifications;

--KShG, KShEG -- watertight cable with copper conductors in rubber insulation with overall rubber cover for stationary wiring between instruments in systems with components located in open air and in sea water; repaired by using PI-35 rubber, or 8615 insulation and S-572 hose;

--KVD, KVDE (when conductors have individual shielding -- KEVD, KEVDE) -- watertight cable with copper conductors in rubber insulation and rubber-coated overall, designed for intraship and outboard stationary installations for wiring between systems operating in open air and in sea water; repaired by using PI-35 rubber or 8515 insulation and S-572 or S-576 hose;

--MPVEG, MPEVEG -- watertight cable with copper conductors in polyethylene insulation in overall PVC sheath designed for intraship and outboard installations and wiring between instruments operating in open air and in sea water; repaired by using P200ZKA polyethylene and RS-1-NT rubber.

The repair technique depends on many factors, including the type of wire or cable and the extent of damage, and also the working conditions (on-board or outboard connection, working voltage, climatic conditions and the like). For example, breaks and burns in wiring between components are repaired by cleaning off the wires to be spliced, soldering them and insulating the splice with grade 230 or 230T PVC tubing. High-voltage wires and rf cables are usually totally replaced unless they are included in a harness and the wiring gets in the way of repair.

Power cables and special cables between devices usually have spare conductors that are inactive in the wiring, and when a breakdown or rupture occurs in an active conductor the problem is corrected by connecting a spare conductor in place of the failed one.

In addition to the problems already noted, there may be cases in operation of hydroacoustic stations when the hose covering is damaged on cables going to outboard components by covers torn from the fastening and by foreign objects that get into the cage or enclosure. In these cases, outboard water may enter under pressure along the cable into the junction box inside the ship, or into the antenna. Such a cable may be replaced during routine maintenance in drydock if the antenna is below the water line and is not raised. In case it is necessary to use a hydroacoustic station having this kind of problem, first the damaged section is sealed off by vulcanization. If vulcanization is impossible, a temporary seal can be made by winding the damaged place with an overlap of 8-10 cm using several layers of PKhL-030 or PKhL-040 (TU MKhP 2898-55) PVC tape alternating with layers of 00 gauze. When the insulation of the cable has deteriorated, it can be improved after sealing by sending a direct current of 0.1-0.3 mA between the coneuctor with poor insulation and the hull from a low-voltage source ($U \leqslant 24$ V).

When doing most kinds of repair on hydroacoustic equipment, service personnel have to replace failed electrical and electronic components with operable ones, which involves deinstallation of the former, and installation of the latter. The work entails the use of welding, soldering, cementing, as well as fastening elements like screws, bolts, nuts and the like.

Welding is a process of joining parts made of metallic and some nonmetallic materials by local fusing of contacting surfaces. A certain percentage of the structural elements and units of marine hydroacoustic stations are made by welding, but the need for welding arises comparatively rarely in the process of operating hydroacoustic stations.

Soldering is a process of joining two or more metallic components after preliminary mechanical fastening by using molten solder in the presence of flux. The flux is intended for removing the oxide film from the surface of the metals being joined and from the solder, which enhances wetting of the metal surface by the solder and reduces the surface tension of the molten solder on the metal-to-solder interface. Depending on the melting point of solders, fluxes are divided into two classes: for solders with melting point lower than 450°C, and for those with a higher melting point. Besides, fluxes may be resinous, based on borax or boric acid, and also based on chlorides and fluorides. The following fluxes are used in repair of hydroacoustic equipment:

--FKSP (10-40% rosin, 60-90% ethanol) -- for soldering wiring components of copper and other metals plated with tin, silver, and also for tinning and to protect surfaces;

--protective flux (3-5% rosin, 0.05-0.55% wax, 3-5% ethanol, hydrazine dihydrochloride 0.25-0.31%, and solvent No 646 -- 86.5-90%) applied by dipping in the flux or by brushing.

Solders are alloys of tin, antimony and lead. Depending on requirements for strength of solder joints, as well as peculiarities of the materials, soft (less than 450°C) and hard (more than 450°C) solders may be used. For example POS-61 (tin--60%, antimony--0.8%, lead 39.2% in accordance with GOST 1499-54) is the most widely used solder; POS-40 (tin--40%, antimony--2%, lead--58%) is used for soldering items not to be heated above 150°C, e. g. for hot-tinning polarized piezoelectrics with heating no higher than 65°C. Among the hard solders are silver-copper-zinc alloys PSR-45 and PSR-25 (25% silver, 40% copper, 35% zinc). These have high strength and ductility but require a high temperature for fusion -- up to 750°C.

When cementing certain materials used in the electronics industry, glues and compounds with and without electrical conductivity are used. For example, to secure active piezoelectric components to steel, brass and so on, current-conducting epoxy adhesive compound DS-65 is used. Components are secured without maintaining contact by DM-5-65 cement. DS-65 adhesive is a composite based on ED-5 epoxy resin (100 parts by weight), silver vitriol (350 parts by weight) and triethanolamine (14 parts by weight). Gluing temperature is 65°C, time 24 hours, resistivity $5.6 \cdot 10^{-3}$ Ω·m.

The following materials are also used in repairing hydroacoustic equipment [Ref. 25]: acetone GOST 2768-69 and ethylene dichloride GOST 1942-74 as solvents; BF-4 adhesive, MKhP 1367-49 and AK-20 adhesive, MKhP 720-41 for securing filament banding, casings, components on boards; KhVK-2a cement, MKhP KU463-56 for fastening PVC tubes; cellulose nitrate varnish 5236-50 for covering solder joints; SB-1S varnish, MKhP 2875-54 for moisture-proofing wiring; STU57-224-64 hydrolytic alcohol for cleaning solder joints and removing char; varnished cambric isulation GOST 2214-78 for winding and wiring materials; TsIATIM-201, 221 lubricant for rubbing parts; cotton thread GOST 6309-73 for applying banding, tying up wiring and harnesses.

CHAPTER 7: ERGONOMIC FACTORS AND POSSIBILITIES FOR TAKING THEM INTO CONSIDERATION IN THE PROCESS OF OPERATING HYDROACOUSTIC STATIONS

§7.1. Role of the operator in the man-machine complex

Designing engineers and operators are now coming to the conclusion that special recommendations of psychologists, physiologists, hygienists are no longer sufficient, since a need is being felt for a unified system of requirements with consideration and evaluation of the effectiveness of ergatic systems, i. e. systems in which one of the components, most frequently the final one that makes decisions and performs control functions is a human operator. In the technical literature, such systems are called "man-machine systems" or MMS.

The problem of taking account of human influence on the success of problem handling by a technical system that is controlled by a human operator has given rise to the following research areas:

--describing the characteristics of the human operator as a component or link in the technical system;

--discovery of the peculiarities of designing facilities with which the human being interacts;

--optimizing the distribution of functions between the human operator and automatic technical facilities.

The first area involves processes of human perception of information, memory, decision making, the dynamic and frequency band of motor functions and the like. The second area involves dealing with such questions as determining the optimum characteristics of such facilities as displays, control units, computer input units and so on. The third area entails only the distribution of functions between the human operator and the automatic device so as to maximize the effect of the MMS as a whole.

The approach to the given set of problems as a system is the basis of ergonomics (ergon--work, nomos--law) -- the science dealing with "the study and planning of labor activity to optimize tools, conditions and the work process, as well as professional mastery" [Ref. 18]. This science is intimately related to others that in one way or another involve people as a topic of investigation such as engineering psychology, systems analysis, psychology, physiology, work hygiene and so on.

A system with a human operator, in contrast to automatic devices has a number of distinguishing features, and therefore its description requires more parameters that influence operational efficiency than for automatic systems. In accordance with the definition introduced in §1.4, we will take the efficiency of the MMS to mean the degree of correspondence of the results of using the system to its goal designation under predetermined conditions. The indices of efficiency depend appreciably on how well the characteristics of technical devices are matched to human perception, motor activity and memory, what conditions of habitability typify the work site, what psychological state the operator is in, etc.

We now have a large volume of experimental data that allow us to assign given functions to the operator or to automatic equipment. Some of these functions are given in Ref. 18. In particular, in virtue of their greater flexibility and adaptability to the signal processing algorithm during detection and classification, lower threshold values of the signal-to-noise ratio, greater volume of storage for informalized information and so forth, the human sense organs (eyes and ears) are utilized in observation systems as the final decision making element.

As already noted in §1.4, quantitative evaluation of effectiveness of using observation systems is possible when their goal designation and results of functioning can be quantatively represented -- range, accuracy of determining coordinates, etc. Consequently, to evaluate the efficiency of a hydroacoustic station as an ergatic system necessitates quantiative evaluation of the efficiency of human performance of working functions in the process of the designated use of the hydroacoustic station.

§7.2. Some human operator characteristics with regard to receiving and processing displayed information

The conditions of human performance of working functions, or the factors that influence the functional efficiency of ergatic systems are unified into the following groups (Fig. 7.1) [Ref. 2]: design peculiarities of the system; conditions of functioning of the system; organization of functioning; peculiarities of human perception.

The design peculiarities of ergatic systems have as their purpose the attainment of maximum matching between the output characteristics of display and other devices and the characteristics of the human operator. In this connection, one of the important peculiarities that determine functional sucess of the system is the volume and rate of access of information. As it turns out, with either too little or too much information the operator makes more mistakes than for an intermediate amount.

Mistakes may be of various types: missing a signal; delay in transmitting different signal components; reduction in the amount of perceived information; failure to handle a job.

The numerous attempts of researchers to establish exact quantitative characteristics of the optimum information load for operators to perform various functions run up against difficulties in formal representation of the significance (value or risk of a decision) of messages. At the same time it is known

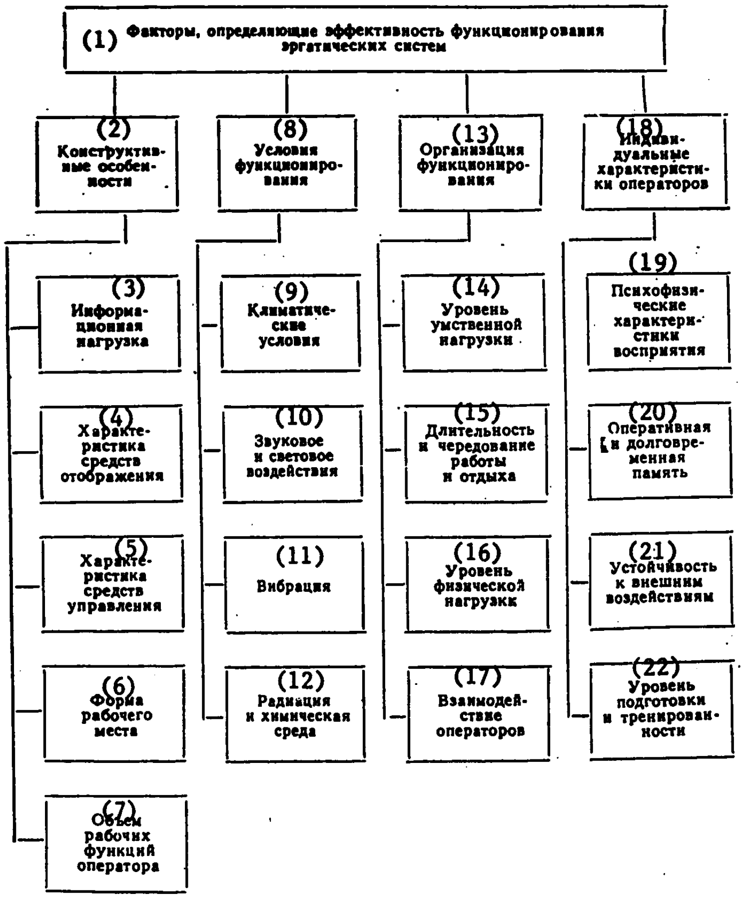


Fig. 7.1. Classification of ergonomic factors

KEY: 1--Factors that determine efficiency of functioning of ergatic systems
2--Design peculiarities
3--Information load
4--Characteristics of imaging facilities
5--Characteristics of control facilities
6--Type of work place
7--Volume of operator's working functions
8--Functioning conditions
9--Climatic conditions
10--Sound and sight stimuli
11--Vibration
12--Radiation and chemical environment
13--Organization of functioning
14--Level of mental load
15--Duration and alternation of work and rest
16--Level of physical load
17--Interaction of operators
18--Individual characteristics of operator
19--Psychological characteristics of perception
20--Short-term and long-term memory
21--Resistance to external stimuli
22--Level of training and breaking in

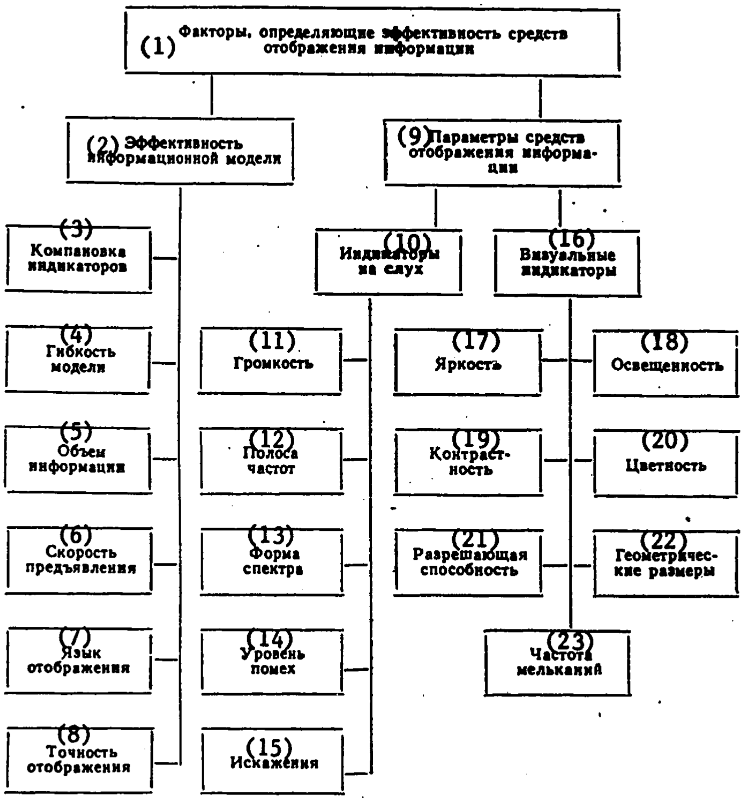


Fig. 7.2. Classification of factors that determine efficiency of information representation facilities

KEY:

1--Factors that determine efficiency of data imaging facilities
2--Efficiency of information model
3--Display configuration
4--Model flexibility
5--Data volume
6--Presentation rate
7--Language of representation
8--Accuracy of representation
9--Parameters of data imaging facilities
10--Audible signals
11--Loudness
12--Frequency band
13--Spectrum shape
14--Interference level
15--Distortions
16--Visual displays
17--Brightness
18--Illumination level
19--Contrast
20--Color
21--Resolution
22--Geometric dimensions
23--Flicker rate

[Ref. 2] that man receives 87% of his information through the visual channel, 12% by hearing, and about 1% through the other channels. Some authors propose that in this case the operator load factor η be used, which is defined by the relation

$$\eta = 1 - \tau_p / T,$$

where T is the total time that the operator is at the work place, and τ_p is the pure time of processing incoming information.

The optimum value of the operator load factor lies in a range of 0.4-0.75, and outside these limits there is insufficient or excess information.

There is a firm relation between the degree of information loading and optimization of the construction and technical characteristics of representational facilities, which in hydroacoustics use two channels of perception: sight and sound. Fig. 7.2 shows the classification of factors that influence the efficiency of functioning of information display facilities. Unfortunately, not all imaging facilities can be quantitatively evaluated at present. For example the signals used in audio indicators require two characteristics for description -- energy and frequency -- whereas more such characteristics are required for most visual displays. This complicates their formal or analytical description, and determination of the quantitative characteristics of imaging facilities.

With respect to the way that the human operator interacts with control facilities, they can be represented mainly by characteristics of motor functions: range of mechanical efforts with respect to switching, moving and rotating controls, the spatial limitations on them, tactile memory and so on. Some of these characteristics in qualitative form are presented in widely known reference papers [Ref. 12, 18]; however, in the general case our knowledge in this area is sketchy.

Under conditions of functioning of ergatic systems, we include all external non-informative stimuli that act on the human organism as an element of the system. Physiologists long ago established the range of characteristics of climatic and other conditions of habitability under which there is no impairment of performance of working functions by an operator. These data have been given by many authors, and specifically in Ref. 18.

Organization of functioning of a hydroacoustic station, as shown by operational experience, has an appreciable effect on the efficiency of utilization. For example, no solution has been found for the problem of determining the optimum duration of a shift when the hydroacoustic technician is working with different displays under different travel conditions, even though some data are known on duration of the period of exhaustion or adjustment, constancy of working capacity and its decline. By way of specific recommendations used in operational practice with hydroacoustic stations, we can mention a shift duration ranging from 2 to 4 hours (depending on the strain or the number of targets being simultaneously tracked, and the interference level) with a 10-20 minute overlap when two operators are changing shift.

Practice has shown that individual peculiarities of information perception by different operators considerably extend the limits of parameters when averaged over a large number of operators. Therefore the chief of the electronics service of the ship should know the quantitative characteristics of information perception by the operators in his jurisdiction when they are performing given working functions so as to optimize the distribution of such functions among operators, and to maximize the efficiency of utilization of observational facilities. This applies both to the thresholds of perception, i. e. the minimum detectable increment of a signal against a background of interference on various indicators, and to the probability of a false alarm -- an erroneous decision that a target has been observed when such is not the case. For example the spread in signal detection thresholds by ear may reach 12 dB at a confidence coefficient of 0.8, while a change in probability of a false alarm depends on the specific conditions under which an experiment is being done, but in some cases reaches values of 0.1-0.15 [Ref. 18]. In some measure these values can be attributed to the considerable difference in signal detecting efficiency by different operators that is well known from experience in using hydroacoustic stations. In addition to these quantities, there are certain other human psychophysical characteristics that should be known in operating and designing the displays of hydroacoustic stations. Let us briefly take up the major ones.

When working with audible signals, hydroacoustic station operators use headsets or loudspeakers. The strong dependence of threshold intensity on interference when working with loudspeakers makes headsets preferable in most cases, reserving loudspeakers for alarm use. When working with a headset, the operator uses the gain control to set the loudness level corresponding to normal audibility and minimum fatigue. The optimum level lies in a range of 60-80 dB.

Differential sensitivity with respect to frequency and intensity, i. e. the maximum [sic] change perceived by the operator in the parameter, enables the operator to detect an echo signal from a moving target against a background of reverberation interference due to frequency difference, as a consequence of different Doppler signals, and also intensity. Masking of adjacent tones within the limits of the so-called critical band considerably impairs the capability for recognizing wide-band signals by ear in which the informative components might be beyond the limits of this band. When designing audio indicators consideration must be taken of the fact that the masking effect is more pronounced for higher signals. Spatial pinpointing of audible signal sources is determined by the binaural hearing effect -- the capacity to compare the phase shift between signals that arrive in both ears within the cerebral cortex. This effect was used at the dawn of hydroacoustics in the first noise locater stations.

Visual perception of displayed information can be most completely characterized by the criterion of visibility. In this case, the signal is a change in brightness, shape, color of the display, and the interference is the background against which this change takes place, and also the brightness, color and toher stimuli on the visual analyzer from the surrounding space. The level of signal intensity during the process of its perception by the visual analyzer is characterized by visual brightness: the greater the brightness of the object, the more visible it is. This perception involves the dimensions and shape of the light source: it increases with an increase in the ratio of candlepower to area. Quantitatively, the perception of brightness can be represented by a certain range of values -- from the threshold of luminous sensitivity to the level that causes the maximum sensation -- blinding. The range of brightness fluctuations within which satisfactory visual observation is possible is limited by values from $6.36 \cdot 10^{-7}$ to $1.59 \cdot 10^{5}$ cd, i. e. about 120 dB [Ref. 18]. Such a range is determined by the capability of vision to adapt to the background, i. e. to suppress or enhance the perception of background intensity of luminous flux depending on brightness. The degree of difference in brightness (color) of two or more objects is called luminous (color) contrast. Determination of the presence of contrast with respect to brightness and shape is the main job of the hydroacoustic station operator when detecting signals with respect to brightness of a pip on a CRT screen and on chart-recorded tapes. So far we do not have adequately precise recommendations on optimum brightness characteristics of visual displays, and the operators themselves set the brightness level, adapting to it during performance of their functions.

Visual perception of space is characterized by angular dimensions of the field of view of 120-160° in the horizontal direction, and 55-60° upward and 65-72° downward in the vertical directions. In the case of color perception, this sector is considerably narrowed. Consequently, in hydroacoustic station operational practice the operator cannot be asked to keep track of objects outside of this range of angles. Spatial perception of the characteristics of objects is represented among other parameters by the visual acuity, i. e. the minimum angle at which two points can be distinguished as separate. For example at an illumination level of 0.01 lx, visual acuity is 7'9", at 0.1 lx -- 3'45", at 1.1 lx -- 1'13" and so on [Ref. 18]. Contrast sensitivity is considerably influenced by extraneous objects that draw the operator's attention, and light sources that enter the lateral field of view. It is for this reason that electronic displays are generally equipped with tubes, and as a rule no signal lights or other visual indicators are located next to the screen.

Visual perception is also characterized by the unique capacity for distinguishing image color, i. e. the visual analyzer performs spectral analysis of signals, or responds differently to signals that differ in frequency. With consideration of this parameter, i. e. the color of the screen and the pips from targets, an attempt is made to choose a color that minimizes the signal-to-noise ratio at which the operator detects a signal, and also to reduce visual fatigue when the operator is working with a given type of instrument.

Of considerable interest for designers and operators are the time characteristics of perception. The dynamics of the change in brightness sensation during the time of action is characterized by the time constant of "visual inertia" which is equal to 0.1-0.2 s for central vision, and 0.2-0.32 s for peripheral vision. Among the time characteristics of visual perception is dependence on the kinematics of the object of observation, in our case for the pips from targets on screens and displays, scans, pointer deflection and the like. The threshold of perception of rate of displacement of an object of observation lies in a range of from 1-2 to 15-30 ang. min/s.

The psychophysical characteristics of perception are closely related to the characteristics of short-term (operational) and long-term memory. Despite some known data, the mechanism of memory is not completely understood, and is the subject of careful research by scientists.

The level of training and breaking in is characterized by the degree of development of skills by the operator in performance of working functions. Operator training starts with vocational selection when people are chosen from a given contingent for their psychophysical characteristics corresponding in the greatest degree to qualities needed by the hydroacoustic station operator. Further instruction includes theoretical studies of equipment and rules for using it, and also acquisition of skills in performing practical functions on the part of the station operator. Perfection of skills in technical servicing and utilization of the equipment for its purpose is completed in study halls and on hydroacoustic stations installed on ships. In this training, it is forbidden to break in the operator on troubleshooting with artificial malfunctions (simulation).

§7.3. Accounting for ergonomic factors during operation of hydroacoustic stations

Ergonomic factors in the process of operating the hydroacoustic station have an effect on nearly all operations of technical servicing of facilities and utilization for the specified purposes. It is natural that accounting for ergonomic factors should be based on analysis of the working functions to be performed by the operator. Therefore, let us analyze steps for operating the hydroacoustic station from the standpoint of quantitative or qualitative evaluation of the role of the human operator.

1. Analysis of measures in technical operation of hydroacoustic facilities shows that the principal working functions performed by the servicing personnel are:

--investigation of the layout of technical systems, monitoring facilities, rules of servicing them and restoring operability when failures take place;

--storing, protecting and installing the facilities on the vessel or in land-based stations and testing facilities;

--monitoring, preventive maintenance and repair of technical facilities.

It should be noted that the working functions are performed by a collective of operators rather than by just one, and are distributed among them either with flexibility or on a rigid basis.

It was noted above that the influence of ergonomic factors on performance of working functions by an operator can be quantitatively evaluated from results that are themselves quantitatively represented. For example the mastery of devices and rules of servicing them by operators is evaluated qualitatively or subjectively -- quantitatively (in points); steps taken to support storage, installation and tests can be quantitatively evaluated only in part. Inspection, preventive maintenance and repair are characterized, as pointed out above, by a rather large number of parameters that in turn permit quantitative evaluation of ergonomic factors.

An example of ergonomic evaluation in this case might be evaluation of the influence of ergonomic factors on effectiveness of monitoring the technical state of equipment. Determination of the necessary accuracy of measurement devices is based on the way that the probability $P_{\text{ош}}$ of error in measurement depends on its permissible value in the form

$$P_{\text{ош}}=\int_{a_\mu}^{a_\nu} f_1(a_1)\left[\int_{-\infty}^{a_\mu-a_1} f_2(a_2)\,da_2+\int_{a_\nu}^{\infty} f_2(a_2)\,da_2\right]da_1+$$

$$+\int_{-\infty}^{a_\mu} f_1(a_1)\left[\int_{a_\mu-a_1}^{a_\nu-a_1} f_2(a_2)\,da_2\right]da_1+\int_{a_\nu}^{\infty} f_1(a_1)\left[\int_{a_\mu-a_1}^{a_\nu-a_1} f_2(a_2)\,da_2\right]da_1.$$

In carrying out a quantitative analytical or experimental evaluation of the percentage of the error $P_{\text{ош.оп}}$ introduced by the human element into the measurement process relative to the total error, we can determine the efficiency of inspection $Э_{\text{к.оп}}$ of the technical system by a human operator as a a monitoring component in the form of the probability of no operator error, i. e.

$$Э_{\text{к.оп}}=1-P_{\text{ош.оп}}.$$

2. Steps on using hydroacoustic facilities include (see Fig. I.1) first of all analysis of the conditions of application and selection of optimum working conditions of the facilities and maneuvering of the ship, and secondly direct servicing of the facilities in the process of application.

The influence of ergonomic factors on handling jobs in the first group can be evaluated in the main qualitatively in virtue of the incomplete correspondence of mathematical models to the algorithms that they describe -- the methods of decision making.

Solution of problems in the second group is more amenable to quantitative evaluation since the problems in this case involve the relation between psychophysical and psychomotor characteristics of data perception and behavior on the one hand, and the physical characteristics of displays and control on the other, i. e. in either case with characteristics that can be quantitatively represented.

Let us consider an example of evaluation of the influence of ergonomic factors on the efficiency of detection $Э_D$ and direction-finding $Э_{\Delta\phi}$ by the noise-direction finding channel in a hydroacoustic station with the operator as the final link that makes a decision as to the presence of a target, taking as criteria of success the detection range D and the accuracy of determining direction $\Delta\phi$ no worse than predetermined values, i. e.

$$Э_D=\text{Prob}\{D\geqslant D_{\text{pred}}\};\quad Э_{\Delta\phi}=\text{Prob}\{\Delta\phi\leqslant\Delta\phi_{\text{pred}}\}.$$

The probability of satisfaction of the given conditions in turn can be found by integrating the probability density distribution of range f(D) and direction finding error f(Δϕ) within given limits, i. e.

$$Э_D = \int_{D_{зад}}^{\infty} f(D)\,dD; \qquad Э_{\Delta\varphi} = 2\int_{0}^{\Delta\varphi_{зад}} f(\Delta\varphi)\,d\Delta\varphi.$$

[subscript зад = predetermined]

Consequently, ergonomic evaluation of efficiency of the noise-direction finding channel of the hydroacoustic station necessitates analytical expressions that relate the range of action and accuracy of direction finding to properties of operator perception of displayed information in the station and statistical characteristics of these properties. As the latter we take the probabilistic characteristics of signal detection against a background of interference as a function of the signal-to-noise ratio k_δ at the input of the display device $f(k_\delta)$, and detection of deviation $\Delta\psi$ of an electronic target pip from the sighting line $f(\Delta\psi)$. These characteristics can be experimentally determined for all computational operators of the hydroacoustic stationby using simulation of signal and noise under controllable conditions, i. e. at known magnitudes of signal and interference.

It was noted above that most psychophysical characteristics of human perception conform to a log-normal law since the properties of perception in the main have logarithmic scales, i. e. response is proportional to the logarithm of the stimulus. Then the probability density function of k_δ according to log-normal law will be represented by the expression

$$f(k_\delta) = \frac{1}{2{,}3\sqrt{2\pi}\,\sigma_{k_\delta} k_\delta} \exp\left[-\frac{(\lg k_\delta - \lg m_{k_\delta})^2}{2\sigma_{k_\delta}^2}\right],$$

where m_{k_δ} and σ_{k_δ} are respectively the mathematical expectation and rms deviation in the normal law of distribution of $\lg k_\delta$.

As the analytical dependence of the detection range as a function of threshold coefficient k_δ we use the expression

$$D = \frac{1}{0{,}23\sqrt[3]{\left(\frac{p_{п0}}{p_0 r_0}\right)^2 \frac{k_\delta}{g b^2 \sqrt{\Delta f T}}}},$$

where $p_{п0}$ is the normalized pressure of interference; p_0 is the normalized pressure of the signal at distance r_0 from the source; Δf, T are the passband and integration time that characterize the reception channel; g is the coefficient of proportionality in the expression for the concentration coefficient as a function of frequency

$$\gamma = gf^2;$$

b is the coefficient of proportionality in the expression for attenuation of sound $\beta = bf^2$.

Then the density distribution function of detection range, assuming that all other factors are deterministic, will be represented by the expression

$$f(D)=\frac{1}{0{,}77\sqrt{2\pi}\,D\sigma_{k_\delta}}\exp\left\{-\frac{[\lg(A^3/D^2)]-m_{k_\delta}]^2}{2\sigma^2_{k_\delta}}\right\},$$

where

$$A^3=\frac{b^2g\sqrt{\Delta fT}}{0{,}22^3\,(\rho_{\text{по}}/\rho_0r_0)}.$$

Substituting into the resultant expression the values of parameters of the hydroacoustic station to be evaluated, and also the values of the level of the primary field of the target ship and the interference at the input to the hydroacoustic station, we get the probability density function of target detection range for the given station under the assumed conditions. And finally integrating this function within the required range, i. e. from $D_{\text{зад}}$ to infinity, gives the operational efficiency of the hydroacoustic station with consideration of the properties of perception expressed by the probability characteristic of the threshold coefficient k_δ in the form of the relation

$$F(D)=0{,}5+\frac{1}{\sqrt{2\pi}}\int\limits_0^U\exp(-D^2/2)\,dD,$$

where $U=\frac{1}{\sigma}\left(\lg\frac{A^3}{D^2_{\text{зад}}}-m_{k_\delta}\right)$.

Analysis of this expression enables us to evaluate the variation of target detection range due only to the probabilistic properties of operator perception of information displayed by the hydroacoustic station. For example, for a coefficient of variation of the threshold coefficient $\sigma_{k_\delta}/m_{k_\delta}=0.15$, the range variation will be $\sigma_D/m_D=0.085$, i. e. about 10%, which cannot be disregarded in operation of the hydroacoustic station. With the purpose of accounting for this, it is necessary to determine the probabilistic characteristics of the threshold coefficient of all operators when they work with different indicators of the station for maximum efficiency of utilizing the station for its purpose by optimum distribution of operators by working functions.

In evaluating the efficiency of direction finding of targets by the hydroacoustic station, e. g. in the noise-direction finding mode, it is necessary to find an expression for the probability density of the error of direction finding due to the probabilistic characteristics of coincidence of target pips with a sighting line corresponding to a definite direction. When we know the relation between errors of such coincidence and errors of target direction finding, and also when we know or can experimentally determine the probability density function of the error of coincidence, we can find an expression for the probability density of the error of direction finding. For example when evaluating the efficiency of direction finding of targets by the phase method with the aid of an electronic display, the accuracy of direction determination $\Delta\phi$ is related to operator error in registering the major axis of an elliptical target pip with a vertical sighting line $\Delta\psi$ by the expression

$$\Delta\varphi = \pi d \Delta\Psi / \lambda.$$

Assuming that the law of distribution $f(\Delta\psi)$ of the error of registration is normal with zero average $m_{\Delta\psi} = 0$, and experimentally determined rms deviation of $\sigma_{\Delta\psi}$, i. e.

$$f(\Delta\Psi) = \frac{1}{\sqrt{2\pi}\,\sigma_{\Delta\Psi}} \exp\left(-\frac{\Delta\Psi^2}{2\sigma^2_{\Delta\Psi}}\right),$$

we find the probability density function of the direction finding error in the form

$$f(\Delta\varphi) = \frac{\lambda}{\pi\sqrt{2\pi}\,d\sigma_{\Delta\Psi}} \exp\left(-\frac{\lambda^2\Delta\varphi^2}{2\pi^2 d^2 \sigma^2_{\Delta\Psi}}\right).$$

Substituting into the resultant expression the parameters of the noise-direction finding channel λ and d, and also the parameters of psychophysical characteristics of the operator working with the given display, and integrating over the predetermined range, i. e. $(0, \Delta\phi_{\text{зад}})$, we get the probabilistic estimate of satisfying the requirements to be met by the direction finder under the assumed working conditions of the hydroacoustic station.

Thus, as operational experience with hydroacoustic stations has shown, the scatter of psychophysical characteristics that represent the properties of perception of information displayed in hydroacoustic stations is quite considerable. Consequently, both the efficiency of the facilities for hydroacoustic observation in which the operator is the final decision-making link, and the successful handling of the job by the ship carrying the station depend appreciably on the characteristics of information perception by the hydroacoustic operator. This necessitates quantitative evaluation, or where there is no such possibility at least qualitative evaluation of the perception properties of each operator on duty for the purpose of optimizing the distribution of operators by the working functions that they perform.

CONCLUSION

Development of hydroacoustic technology, and in particular marine hydroacoustic stations, necessitates improving methods of operation. Solution of this problem requires comprehensive analysis of basic factors that influence the attainment of maximum efficiency of hydroacoustic stations, as well as quantitative evaluation of the extent to which such efficiency is realized. Achievement of quantitative evaluation requires the development and introduction into hydroacoustic practice of an engineering computation technique, one version of which has been proposed in this book. Practical implementation of the given technique will enable optimization of measures on technical servicing and purposeful utilization.

The consideration of steps in checking the technical condition, preventive maintenance, provisioning with spare parts and accessories and routine repair has been based on common features of utilizing electronic equipment as a whole; because of the wide variety of types of hydroacoustic stations it has not been possible to consider the operational peculiarities of every one of them. This situation is compensated by accessibility of the required information in the working documentation of each model of hydroacoustic equipment.

Hydroacoustic measurements occupy an important place in applied hydroacoustics. The considerable number of random factors that influence measurement results have so far precluded an appreciable improvement in reliability, which necessitates further improvement of facilities and methods of measurement.

An examination of the problems of restoring operability of hydroacoustic stations -- methods of finding and correcting malfunctions -- shows that increasing the efficiency of utilization requires designers to correspondingly increase the level of monitorability and restorability of hydroacoustic stations, as well as to optimize the modes of operation under different working conditions.

Analysis of the influence that properties of perception of displayed information by human operators have on the output functional characteristics of hydroacoustic stations is aimed at optimizing their design to maximize efficiency of utilization of the hydroacoustic station. At present, this problem is still far from a final solution. However, research in this area and analysis of the experience of operation of hydroacoustic stations will doubtlessly make an appreciable contribution to the development of hydroacoustic facilities as ergatic systems.

Our presentation leads us to hope that the book will be an impetus to further improvement of operational efficiency of marine hydroacoustic stations as they are used to carry out the important jobs of the ships carrying them.

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