MULTIBEAMS SIMULATOR FOR SONAR ENVIRONMENT

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The multi-beams simulator system is described in which signals from a plurality of hydrophones of array different configuration are simulated by selecting storage addresses of a random access memory whose successive addresses contain samples of an input signals. The system simulates sonar signals as received by an underwater sonar array for test and characterization and/or training. The system offers a wide range of capabilities including: moving targets scenario generation, simulation of propagation path, and ambient and reverberation noise generation. The system is operated from any computer running LabVIEV.

INTRODUCTION

Sonar Environment Simulators (SES) are becoming common tools for the test and debug of signal processor characteristics and target detection/classification algorithms during the sonar development phase. In production, the SES can serve as factory test equipment. In the real field of the sonar, the SES functions as Built-In-Test. Most important has been used of SES for the training of operators.

This paper will emphasize the Computer Subsystem aspects of the simulator. Overall system concepts and special purpose hardware will be discussed only in general terms relating to reasonableness of SES complexity.

A broadband SES is described in which signals from a plurality of hydrophones are simulated by selecting storage addresses of a random access memory (RAM) whose successive addresses contain samples of an input signal. The range of each hydrophone from the target whose signal it is receiving is determined by the address assigned to the hydrophone relative to the address of the input signal in the RAM. In order to produce a Doppler shift (time compression/expansion), the address of the simulated hydrophone signal is changed with respect to the signal, which is being stored in the RAM. The rate at which the address of the simulated hydrophone signal is being changed relative to the input signal determines the Doppler shift. In order to reduce the size of the RAM an interpolation technique is used to obtain signals corresponding delays, which are smaller than the delay between adjacent addresses.

This paper relates to apparatus for the simulation of the acoustic signals produced by moving vessels. This signals received by several receiver locations, each at a different range and bearing to the vessels. Each of the simulated signals must provide time delayed replica of the signals from the vessel corresponding to its range from the receiver, its Doppler shifted signals, and its amplitude, which reflects the propagation path loss. The time delay is necessary because a maneuver by vessel, which changes acoustic (passive/or active) signal will arrive at the different receivers delayed by the propagation times. The time difference between received signals at the receivers can be used to localize the source. Separate Doppler shifts are required because the vessel's velocity will be projected at different angles along the propagation paths to the receivers. The differential Doppler shifts can also be used to localize the vessel location relative to the receivers. Finally, the vessel's signal will experience different propagation conditions along the paths to the receivers. Relative signal levels can be used to localize the signal source.

1. STIMULATION SIGNAL REQUIREMENT

The SES system can be described in terms of three distinct function; namely, generating the targets signals, simulating the propagation path, and adding ambient and reverberation noise [1]. Our basic model of SES system is shown in Figure 1. It contains two main functional blocks, scenario and target control and sonar control. The functional requirements of the first component necessitate the use of a computer, or other processor with a generic programming environment. The second component is specialized hardware, which must be able to simulate the analog signal to each hydrophone from each source in real time.

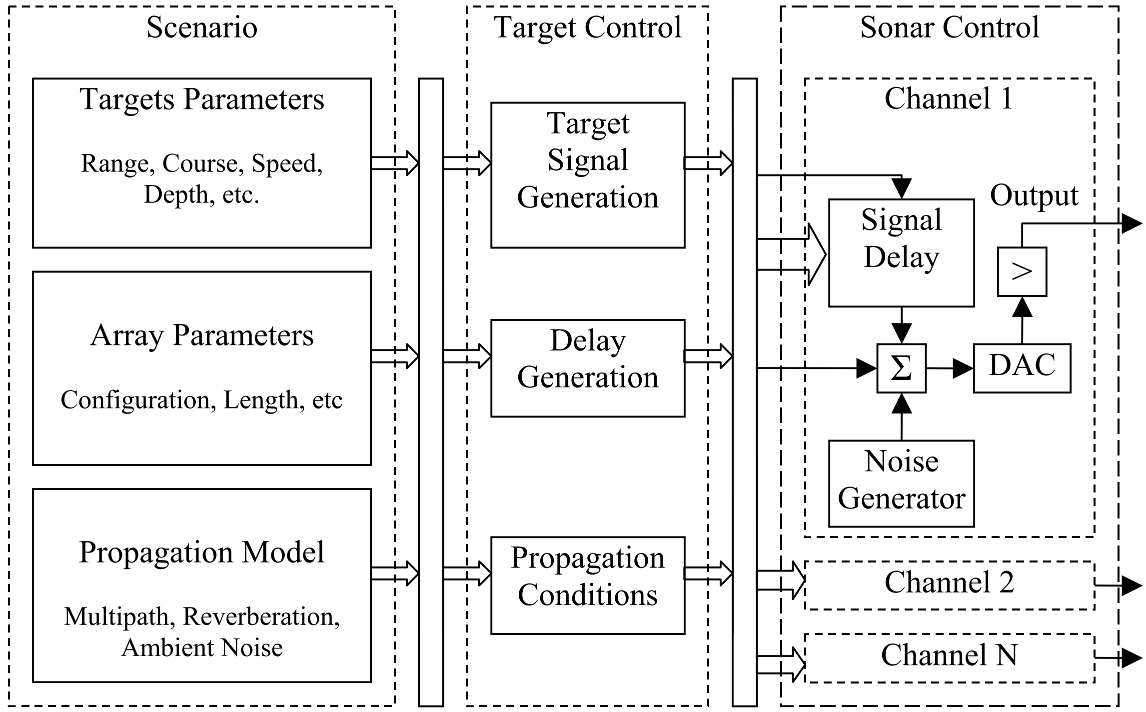

Fig. 1. Sonar environment simulator system.

The detection of underwater targets is normally separated into two modes, active mode and passive mode. With passive detection mode, the sonar simply receives the acoustic energy that is produced by a target in the water. When active mode is employed, acoustic energy is emitted by sonar, which then received any acoustic energy being reflected by the underwater target.

The acoustic energy, which is radiated from a target, can be separated into noise from two sources – machinery noise and propeller noise. These target-radiated noise characteristics comprise many frequency components covering a wide spectrum. Typically there will be several harmonic series, with no simple relationship between the series, as well as unrelated spectral lines. Frequently the vessel noise will exhibit closely spaced frequency "doublets", arising from pairs of shafts running at nearly the same speed, which provide valuable clues and, therefore, need to be reproduced realistically [2]. Machinery noise appears as multiple narrowband signals centered at variety of frequencies. This noise is simulated by a variety of tonal signals. They can be set to any arbitrary frequency and at any arbitrary source level. Noise is generated since a target's propellers in motion and it is more broadband in nature. The propeller noise is also amplitude modulated by the blades of the propeller with the modulation frequency being determined how fast the blades are rotating. This type of noise is simulated by a white noise source passed through band-pass filter with arbitrary cut-off frequencies and an arbitrary source level.

The acoustic energy received by array during active mode operation is initially dominated by reverberation noise, which normally decays exponentially immediately following an active transmission, followed by an echo or copy of the originally transmitted waveform, which is reflected from the target of interest. The different types of transmit waveforms are used, but continuous wave (CW) and coded pulse with modulation (frequency or phase) are most often used. The simulation software calculates the required time delay after a transmission before the echo, generates of signals simulative of echoes having characteristics dependent upon target aspect and target highlights. The Doppler effect also affects the signals generated by the simulation software during either active or passive mode operation. Suffice it to say for this brief description that the Doppler effect generator is responsive to range rate information signals to produce synthetic Doppler shift in the sonar pulse signal wherein the extent of shift for any frequency within the sonar pulse is proportional to that frequency. This is particularly important in simulation of FM or other form of modulated sonar pulse with a large product of bandwidth-duration. As soon both the own-ship and individual target positions change, the simulation software calculates the respective radial velocities. These relative radial velocities are used to calculate the required Doppler shifts (time compression/expansion).

The output of the Doppler effects generator is applied to an aspect dependent echo generator. This generator provides software means for synthesizing sonar echo signals, which are characterized by the effects of target diameter and length. In this regard it is objective provide aspect dependent simulation echoes wherein

$$EchoLength = (2 \cdot s / c) \cdot \left|\cos\theta\right| + (2 \cdot d / c) \cdot \left|\sin\theta\right| + T$$

where s and d - are the target length and diameter, respectively, θ- is the aspect angle of the target, c- is the velocity of sound, and T- is the transmitting pulse length.

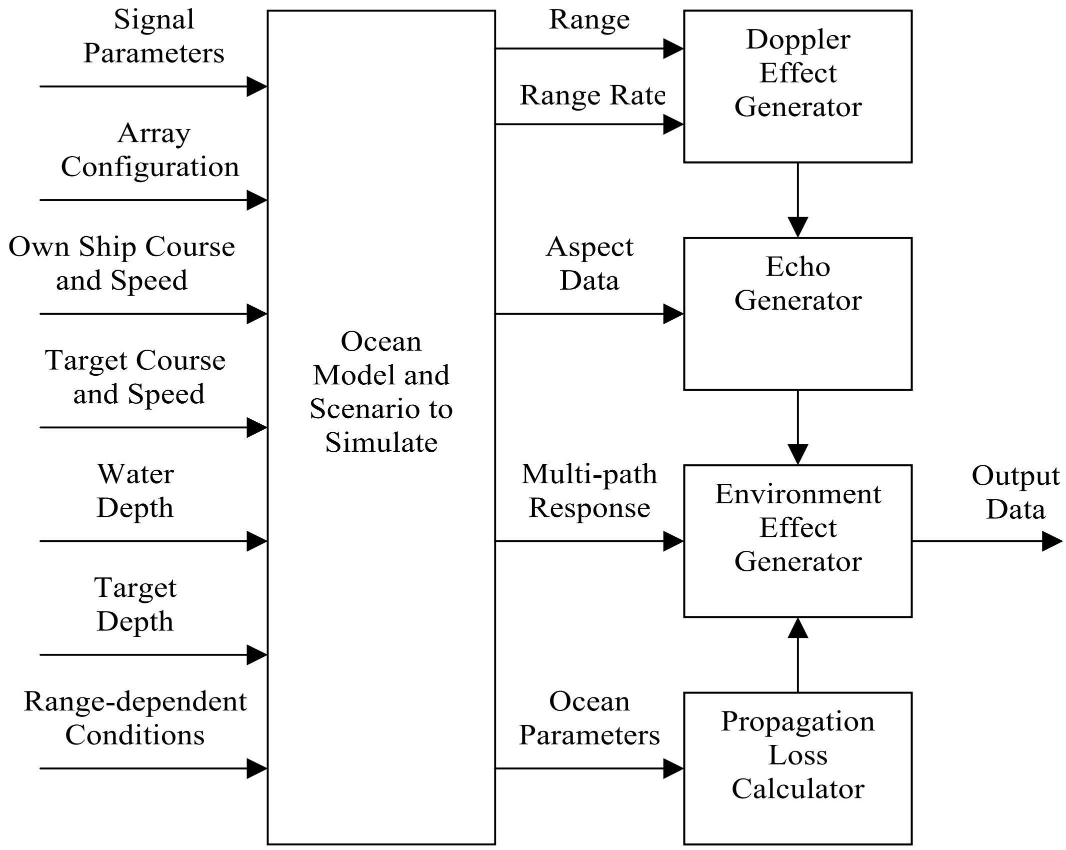

Fig. 2. Signal generation block diagram.

The echo generator output is applied to environmental effects generator, which may introduce factors such as ocean noise, reverberation effects, multi-path condition, and the like. All gain parameters are calculated based on the source level and the transmission loss determined by the distance from the source of the receiving array [3]. The output of the environmental effects generator is fed to the card, which are simulating propagation path for multiple targets and many hydrophones any array type in real time. A graphical representation of the target signal generation for simulating single target in active mode operation of sonar is shown in Figure 2.

2. REAL TIME HARDWARE FOR TARGET STEERING

After the received signals from the target have been calculated, it is necessary for the SES system to simulate the effect position of the target with respect to the position of the own ship sonar array. Each unique target must be treated independently of all other simulated targets to allow for complete freedom of simulated movement during the test scenario.

The multi-channel DSP board, configured as inverse beamformer, easily handled this task. This board will create multiple delayed replicas of each target, with the number of replicas produced equal to the number of receive hydrophones in the sonar array. In this way, the SES system will produce composite signal containing the acoustic energy from each hydrophone, that will result in the sonar receive system seeing each target at unique bearing.

A block diagram of an embodiment of this unit for single channel is shown in Figure 3. There are three major functions involved in this unit of the simulator: Delay Address Generator (DAG), Coarse Delay Generator (CDG), and Fine Delay Generator (FDG).

Delay Address Generator.

On periodic intervals (e.g. one quarter second to one second) the computer provides updated values of the target range and range rate. Constant range rate is assumed between updates. Range rate is thus accumulated into range at the output of delay address generator. Any error with respect to the range, provided with the assumption of constant range rate, is corrected by the periodically updated range value from the computer.

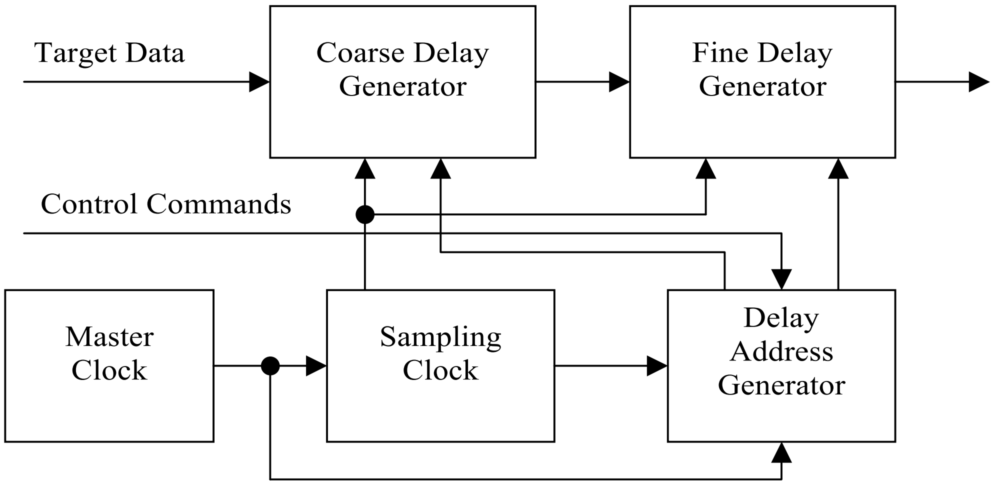

Fig. 3. Unit for simulating multiple delay replicas of signal.

There are four word size (in bits) used in the delay address generator: A1 is the number of bits in the coarse delay address; A2 is the number of bits in the fine delay address; A3 is the number of the least significant bits needed in connection with A1 to resolve the minimum time delays required at minimum range rate; and A4 is the number of bits required to encompass the desired maximum range rate given the minimum range rate (A3).

A1 is determined from the sampling rate f_s and the total time delay T_{max} (range) required. A1 must be equal or grater then $\log_2 (T_{max} \cdot f_s)$ for f_s= 8192, A1=16, T_{max}=8 s.

A2 is determined by the time delay resolution required at maximum range rate R_{max} and maximum signal frequency F_{max} to provide acceptable performance (maximum allowable distortion). A2 must be equal or grater then $\log_2(R_{max} \cdot \Delta T_{min} \cdot f_s/C)$, where ΔT_{min} is update time interval (minimum of two updates each period of maximum simulated Doppler shift, $\Delta T_{min} < 2 \cdot F_{max} \cdot C/ R_{max}$).

Delay resolution DR defined as:

$$DR = R_{\max} / C / \Delta T_{\min}$$

A3 is determined by the minimum time delay increment needed to implement the minimum range rate R_{min}. A3 must be equal or grater then $\log_2 (R_{min} \cdot \Delta T_{min} \cdot fs/C)$.

A4 is determined by the value of R_{min} resulting from the selection of A3 and R_{max}. A4 must be grater then $\log_2 (R_{max}/ R_{min})$.

For example, if $(\Delta T_{min} \cdot f_s)=1$, $R_{max}=90$ knots and $R_{min} \leq 0,02$ knots, than A2,A3,and A4 are equal 5,18, and 13 bits respectively.

Coarse delay generator.

The coarse delay generator uses a conventional RAM memory technique for a delay device. Sampled data from card of digital target data is stored in successive locations of memory. Each sampling interval $1/f_{s,}$ causes the write address to increment by one. Delayed sampling signals are accessed with respect to the write address by subtracting the desired delay from the write address to obtain read address. A read of samples from RAM at the rate “m” times (“m” being greater than one) that of the sampling frequency from each of m addresses are retrieved to be used in interpolation filter to create a sampled signal at rate “m” times of the sampling frequency f_s.

Fine delay generator.

To provide delayed signal output having Doppler shift with respect to source signal, the times delays much finer than one sampling interval $1/f_s$ must be generated. For example, at the highest update rate $(\Delta T_{min} \cdot f_s)=1$ as many as A2=5 bits of fine delay address are required to resolve the time delay. Hence, the interpolation filter is used to compute the values of sampled signals at delay times, which are intermediate to the RAM address delay times. Figure 4 shows sample filter, which utilizes “m “ tap weights per calculation [4].

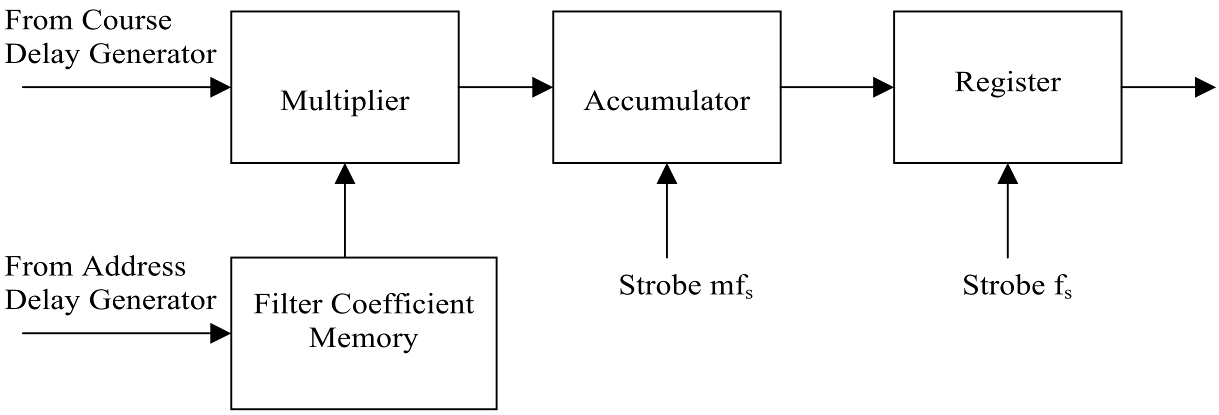

Fig. 4. Interpolator of the Doppler simulator.

Total of $m=2^{A2}$ coefficients are stored in PROM memory, giving the capability of interpolation to 2^{A2}-1 intermediate times delays. A selected group of sampled delayed signals from the delay storage (RAM) of the CDG are provided in time sequence to the fine delay generator (FDG), which contains an interpolation filter. The filter selectively weights each delayed signal and provides a resultant summed signal, which is shifted in delay by a prescribed amount relative to the group of signals received from the CDG. Changing the weighting distribution changes the amount of effective delay. The coarse and fine delay address signals provided to RAM and interpolation filter, respectively, determine the delayed signals and weighting factors, which are formed from the coarse and fine delay signals received from the DAG. The weighting factors of adjacent addresses of the PROM are obtained by choosing the amplitude values of equispaced intervals of a weighting function. The rate of change of range is determined by the amplitude change in the weighting factor applied to each signal read from the RAM. The weighting factors are multiplied with the corresponding signals from each address read from the RAM. Increments in the time delay or range of the interpolated output signal may be a small fraction of the delay between addresses of the storage unit RAM.

The weighting function can be used with eight (m= 0, 1,…, 7) regions, each region having 256 addresses (for A2=8), with a weighting factor distribution determined by multiplying the Hanning Window function $\cos^2(x/m)$ with the sinx/x function. The PROM has $m2^{A2}$=2048 addresses in which 256 weighting factors in each of eight regions are stored. If A2 is equal eight bits, zeros of weighting function will occur at addresses of the PROM corresponding to 0, 256, 512,…, 2048 except for address 1024, where the weighting factor is unity. There are eight regions between these addresses, where the weighting factor is other than zero or unity as determined from the weighting function.

The output signal from the interpolation filter is provided to the output signal conditioner, where the sampled digital signal is converted to a sampled analog signal in DAC converter and then filtered of output filter. This signal can be used for the test and debug of signal processor characteristics without the ambient and reverberation noise.

Ambient and coherent reverberation noise simulation.

An important feature in any SES system is the ability modeling the noise sources that are present in the ocean. These noise sources are crucial, as they are often the limiting factor in the effectiveness of the sonar array to detect an underwater target in the presence of noise.

The ambient noise is essentially the background noise of the ocean environment and can traced to such sources as surface waves, marine animals and shipping traffic. Digital pseudo-gaussian noise generator simulates this noise. Digital pseudo-gaussian noise has basic advantages over analog generated noise, in its repeatability and inherent stability. In practice, the pseudo-gaussian noise sequence length is chosen so that it does not repeat itself during the period of any experiment in which the generated noise is used. This sequence is filtered (spectrally shaped), and than divided into subsequences, one for each hydrophone. The ambient noise is constantly being inserted into the data flow, with the spectral characteristics and noise level set to match the desired ocean environment.

The reverberation noise, in the form of its electrical analog, has been most commonly simulated by band limited random noise. For some applications this approach is satisfactory. However, it is not acceptable where it is desired to simulate reverberation, most of which arises from a relatively few major reflectors. In this case, it may closely resemble a target echo in the sense that it will consist of a relatively small number of overlapping replicas of the original transmitted signal.

The original transmitted signal whose reverberation is to be simulated is digitalized and stored in the random access memory (RAM). To generate many overlapping signals, representative a number of echoes, the RAM is read out by overlapping sequences of digital numbers, generated from a random pulse generator. A logical and gating means are provided the effect individual replica readouts beginning at times corresponding the arrival energy at a receiver scatters of differing distances [5]. Reference amplitude signal generation means is provided of adjust the amplitude of individual replicas, such that by adjusting both replica numbers and amplitudes, one may more closely match natural reverberation arising from different sources.

The Sonar Control hardware contains several modules, which connected with scenario and target control computer by fast Compact PCI bus. Each module includes 32 channels and used SHARC DSP processors for generating the required delays by means of inverse beam-forming. The ambient noise is generated by used a pseudo-random code generation technique to produce, in real-time, a noise sequence. A 32-channel module produces the final analog output to the system under test. The number of this modules required is a function of the number of hydrophones. A module has digital outputs for the system under test is expecting digital inputs.

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