GENERATION OF AUDIO SIGNALS FOR SONAR SIMULATION/STIMULATION WITH DIGITAL TECHNIQUES

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ABSTRACT

Sonar signals are conventionally divided into two categories: (1) shaped, broadband components, such as ambient, ship flow and propeller cavitation noises, and (2) narrowband components, including machinery noise, echoes, and reverberations. This paper will explain how Honeywell uses digital data processing to generate these signals for use in sonar trainers.

BROADBAND SPECTRUM SIGNAL GENERATION

For the purpose of this discussion, the broadband spectrum addressed is the bandwidth of 10 Hz to 20 kHz, a spectrum of interest in generating audio signals used in sonar simulation and stimulation. The ocean spectrum is known to have certain predominant shaping in this bandwidth due to wind, shipping, and ocean lift.

The results of recent ambient-noise investigations have been published. The Knudsen curves show the ocean spectrum shaping over the frequency spectrum ranging from 1 Hz to 10^5 Hz. This spectrum may be viewed as a shaped, broadband spectrum. This section discusses an approach used in digital signal processing to synthesize this spectrum. The shaped, broadband spectrum can be subdivided into smaller bands, from which filter networks can be designed, and the overall filter outputs summed to create the shaped broadband spectrum.

One type of filter that can be used to develop the broadband shaping required is a n-pole/m-zero recursive digital filter of the general expression:

$$Y_{out} = \sum_{i=0}^{n} a_i x_i - \sum_{i=1}^{n} b_i y_i \qquad (1)$$

where: Y_{out} = One filter output

a_i = Zero coefficients

x_i = Filter inputs

b_i = Pole coefficients

y_i = Delayed outputs

The first stage in developing this broadband spectrum is to develop a flat response over the bandwidth of interest. This is achieved by using a digital pseudo-random number generator of sufficient length. The pseudo-random generator must then be sampled at a frequency sufficiently high to obtain the desired bandwidth, and also eliminate the unwanted aliasing foldover due to the Nyquist constraint.

The Read-Only Memories provide a table look-up for the filter coefficients, Figure 1. These coefficients also could be a Random Access Memory (RAM), which could be computer-loaded for added filter flexibility. The filter input samples, Figure 2, are then multiplied by the necessary coefficient using a parallel digital multiply algorithm utilizing MSI digital multipliers. These partial products are then summed over the number of poles and zeros in the digital parallel adder to meet the requirements of equation 1. The output of the filter is then sent to a gain adjustment network, which also can be a digital multiplier.

The same constraints in implementing low-pass, band-pass, and high-pass digital filters also apply in this form of spectrum shaping, i.e., the sampling frequency chosen must be sufficiently high to prevent unwanted aliasing and foldover.

The coefficient range must be sufficient to allow proper pole and zero placement. The number of bits carried through the filter must be sufficient to achieve the dynamic range and resolution required by the desired spectrum shape.

Once the desired spectrum shape has been determined, a model can be formed from which a computer simulation can be utilized to determine the coefficient sizes. The attractiveness of this type of digital filter implementation is the flexibility afforded in the selection of the number of poles and zeros required for a particular filter network. The same digital multipliers and adders also can be used for all the filters. This type of implementation is a sampled filter timesharing the multipliers and adders.

The recursive filter is considered to be the most cost-effective approach that will achieve the desired spectrum shape.

NARROWBAND SPECTRUM SIGNAL GENERATION

The basic requirements for narrowband signals used to simulate or stimulate sonar equipment is to generate lines with the following controlled characteristics:

1. Frequency
2. Frequency slide
3. Amplitude
4. Time on/off
5. Line spread
6. Delta phase.

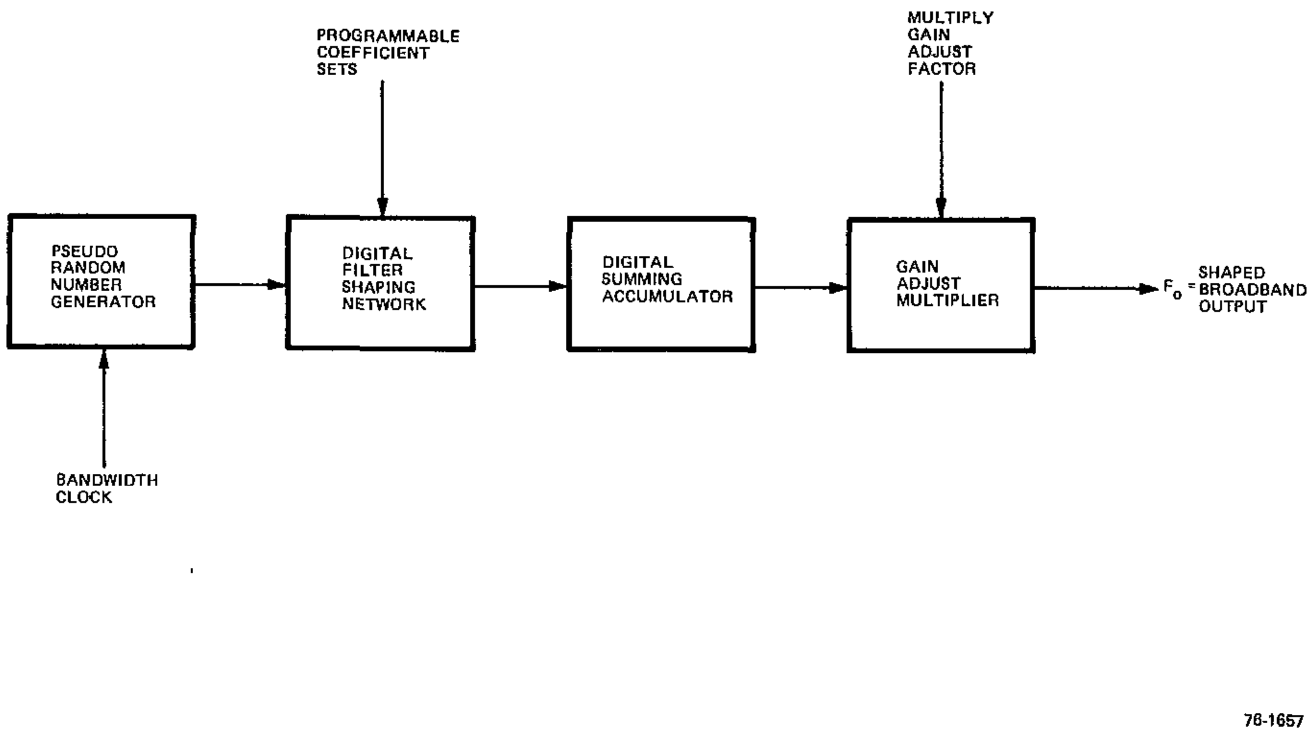

Figure 1. Broadband Digital Audio Model

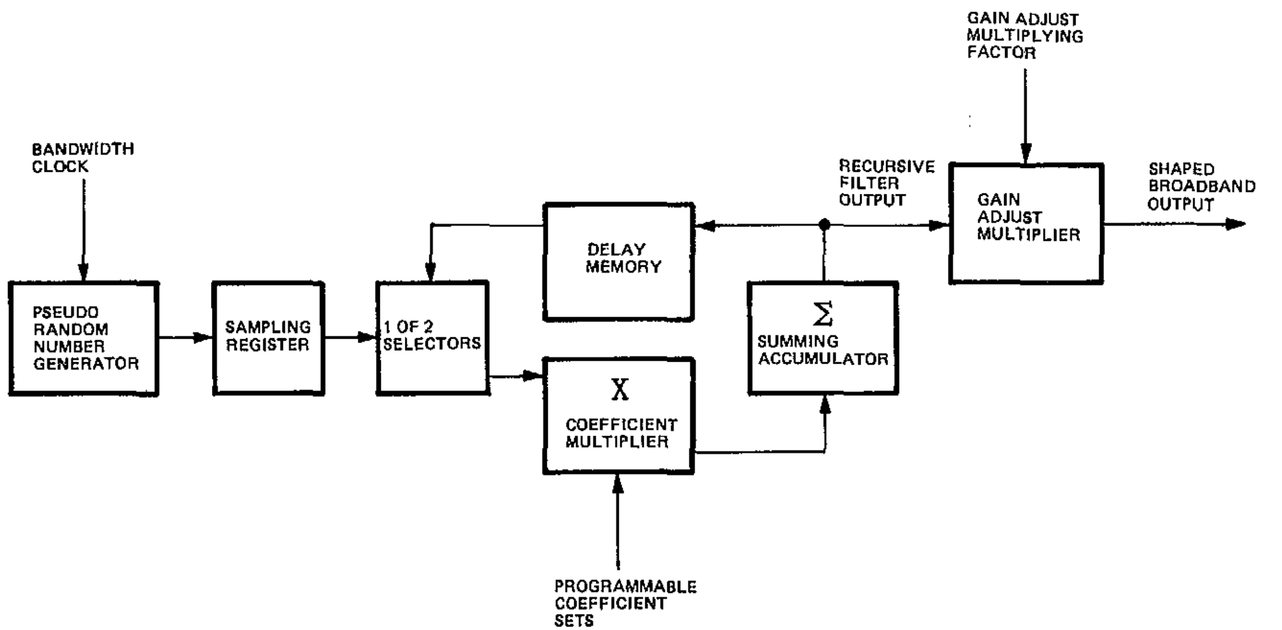

Figure 2. Broadband Digital Audio Model Filter Shaping Network

As may be seen from Figure 3, the approach illustrated in this paper is to use digital techniques to generate and control these signals. First, a sine wave is stored in a Read Only Memory (ROM); second, a step-by-step approximation of a sine wave is generated by extracting sequential values from the ROM. Third, the resulting sine wave is attenuated to the desired value with a digital multiplier and summed with other sine wave points to provide a composite waveform comprising several sine waves of varying frequency and amplitude.

There are two basic methods for generating sine waves with varying frequencies; each involves stepping through a ROM table. The first is to step through a table with a variable clock rate. The second uses a fixed clock rate and step with different step sizes. We have chosen to use the latter. The fixed clock method provides the beneficial side effect of more steps per cycle at lower frequencies. This will be explained later.

To keep high frequency resolution, in addition to the basic count or step size, fractional parts causing a non-integer count, are carried. This results in counts running, for example, 7, 7, 8, 7, 7, 8. Using this technique required the analysis of two questions:

1. What is the harmonic distortion caused by carrying non-integer counts?

2. What is the harmonic content of the wave, based on the fact that the sine wave is being approximated by discrete steps?

Analysis of both problems discloses that (1) the harmonic distortion due to the fact that we count 7, 7, 8, 7, 7, 8 is 44 dB down; this will not create a problem; (2) the harmonic content resulting from discrete point samples, approximating a sine wave, requires a more complex explanation. A computer simulation shows that the first set of bad harmonics are n-1 and n+1, with n being the number of samples per cycle. This means that the sample rate must be selected to ensure that the n-1 and n+1 harmonics are outside the band of interest. Solution of this problem is aided by the fact that, as frequency decreases the number of steps per sample increases. Consequently, this yields a bad harmonic content at a constant frequency. As an example, we are currently generating lines up to 5 kHz with the lowest sample rate being 32 kHz. The frequency of interest, in this particular case (5 kHz), results in the harmonic distortions showing up outside of the band of interest.

Another important effect to be simulated is a frequency slide. For this simulation, the count or frequency of the generated wave is updated once each second by computer control. If it becomes desirable to change frequency during this one-second period, such as a doppler rate change for a target, or a frequency modulated pulse transmission, an additional control is provided; this technique is called delta count. With delta count, a number is added to the count number every iteration for selection of a point in a sine wave. In this manner the frequency can slide up or down. This control also is updated once each second.

After generation, the sine wave is fed to a digital multiplier, Figure 3, with a control word representing amplitude, which is updated once each second from the computer. It is then multiplied (i.e., attenuated) in order to form the final desired amplitude and frequency shaped sine wave.

These are the basic controls by which narrowband generation is accomplished. There are other controls, however, that are necessary as well as some other events that take place (e.g., an active echo) that require on times of less than one second. For these reasons, the computer outputs another control word that, in essence, will be counted down so that once a signal is turned on it can be turned off in increments of one millisecond.

Another important factor of narrowband simulation is that the waves should not be pure sine waves but should have some spreading and fading. This effect is simulated by amplitude modulation of the wave with random numbers that have a Rayleigh distribution.

This, in turn, will cause both spreading and fading. By controlling the rate of the random number selection, the amount of spread and fade may be bordered.

The techniques discussed above have been utilized to generate the narrowband noise. Obviously some of the effects have impact on the system design. For an example, as the sine waves must be any frequency, ranging from a very low frequency to higher frequencies, there can be no further filtering of the output of the narrowband audio generator. The only filter allowed will be a final system filter limiting the bandwidth to that of interest to the sonar being simulated. This, in turn, sets the number of points that must be used per cycle to ensure that the harmonic content in the band of interest is low by placing the higher amplitude harmonics outside of the band of interest.

Another consideration is the resolution required on the lines. Consider, for example, that lines, controllable in 0.1 Hz, are desired; this dictates the number of fractional bits that must be carried with the count word. A potential problem existed previously, because a non-integer number of samples per cycle is used, with the possible net effect of creating subharmonics. If this were so, then the effect would show up on spectrum analyzers and cause operator distraction. A computer analysis, however, definitely establishes that there is no subharmonic generation.

System considerations must take into account the required step size to move the undesired harmonics out of the range of interest, thus setting

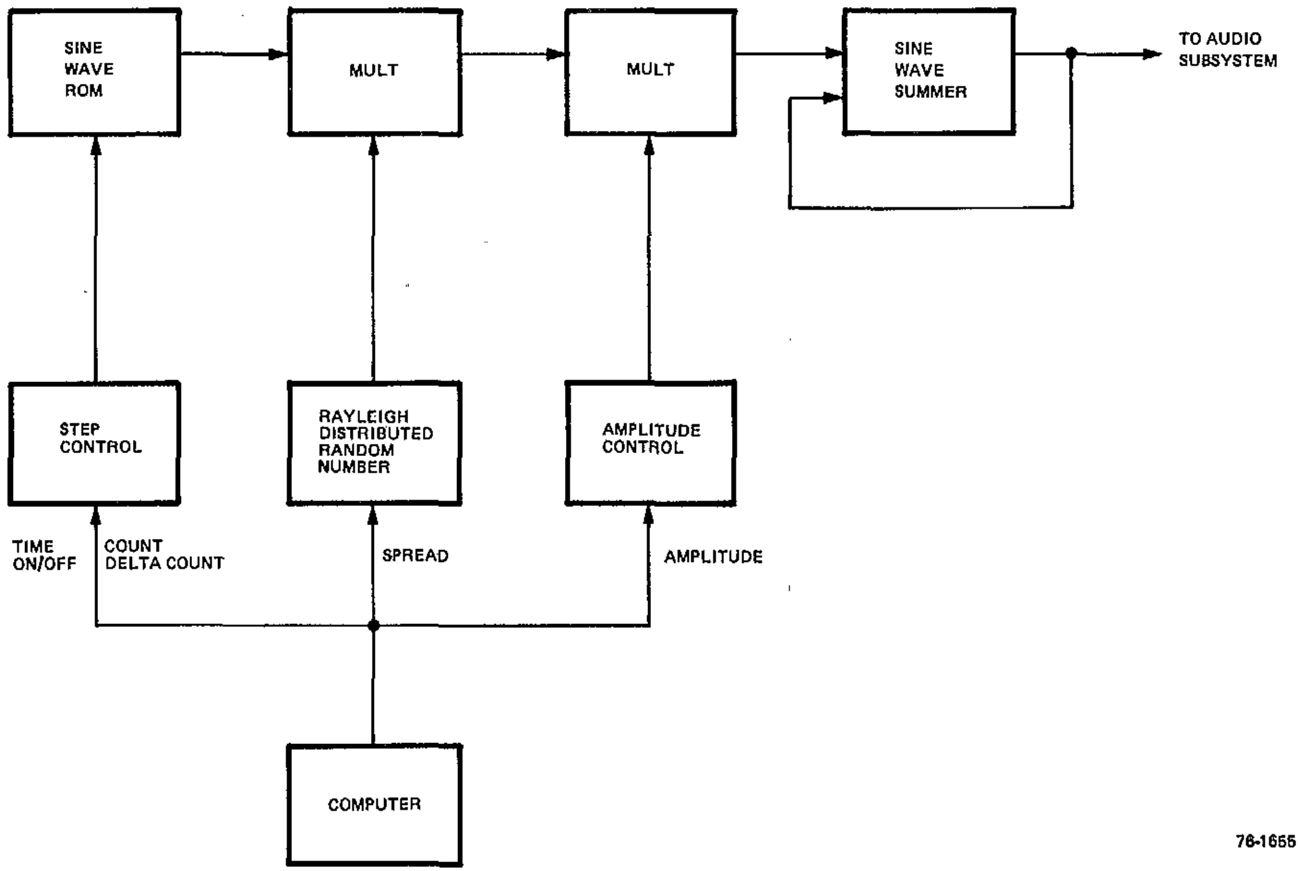

Figure 3. Narrowband Generator Block Diagram

the number of samples per second required. The controllable amplitude range and the desired amplitude steps will, in turn, establish the number of bits required to obtain range.

One final consideration, transcending that of the simple line generator, is the fact that in the undersea environment a distinct possibility exists that the same line from a target will arrive at the own ship point over two vertical paths. These will then sum together to produce interference effects. These effects are simulated by generating a second set of lines, at the same frequency but differing in amplitude, to represent the second path received by the sonar; the second path is then phase-adjusted relative to the first in accordance with the math model that generates path length or time of travel on the path.

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