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A Real-time Method for Underwater Noise Time Series Generating

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ABSTRACT

Fourier synthesis is the most commonly used method for broadband simulation in underwater acoustics. The receiver time series could be obtained via Fast Fourier Transform, where the time window should be long enough to avoid the aliasing in time domain. Therefore, it is optimal for solving the non-real time pulse propagation problem. A new filter method is presented to generate the underwater noise time series real-time by using normal mode propagation model. The simulation process is realized by two FIR filters. The first one has a frequency response approximates to that computed by normal mode selection. The second one is a time-delay filter to achieve the propagating delay. Convolutions in time domain are applying to the source signal with a short time window, which is very suitable for Digital Signal Processor. The ship radiated noise time series emulated by the filter method above is compared with that by Fourier synthesis. The results show that the filter method could generate receiver time series accurately and effectively.

Keywords: Noise, Real-time, Generating I-INCE Classification of Subjects Number(s): 74.6

1. INTRODUCTION

Underwater acousticians have traditionally favored spectral analysis techniques for the high temporal variability of the ocean (1). However, with the development of low frequency sonar, time series analysis and generating techniques have attracted increasing attentions.

Time domain signals acquired by hydrophone array are always the direct input of sonar. Most underwater acoustic localization methods process the time domain signals to extract the information of the relative position between the source and the receiver array, such as TDOA (Time Difference of Arrivals), cross correlation, cross-spectral density and so on. Therefore, the propagation characteristic is of equal importance as the band-averaged energy distributions in sonar simulation methods.

Generally, receiver time series are simulated based on the signal and system theory (2). The oceans are modeled as underwater acoustic propagation channels. For convenience, it is widely accepted to treat the channel as a linear system. The input of the system is the source signal, and the output is the receiver signal. The benefits of the hypothesis is that most signal processing methods, such as convolutions, superposition, and Fourier Transform, could be used directly to describe the relations between input and output.

In signal and system theory, a channel could be determined by its impulse response function in time domain or frequency response in frequency domain. Correspondingly, the output signal could be computed by convolving the input signal and the impulse response function, or multiplying the frequency response by the source spectrum to obtain the receiver spectrum. Ray-based techniques have been used successfully to model acoustic time-series (3,4). The impulse response function could be acquired by combining all the eigenrays into a FIR (Finite Impulse Response) filter.

Another commonly used method is Fourier synthesis (1,5). The receiver time series could be obtained via Fast Fourier Transform, where the time window should be long enough to avoid the aliasing in time domain. A new filter method is presented to generate the underwater noise time series real-time by using normal mode propagation model. The simulation process is realized by two FIR filters. The first one has a frequency response approximates to that computed by normal mode selection. The second one is a time-delay filter to achieve the propagating delay

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2. Theory

2.1 Fourier Synthesis

The solution of the time-dependent wave equation can be obtained via a Fourier transform of the frequency-domain solution as (1)

$$p(r,z,t)=\frac{1}{2\pi}\int_{-\infty}^{\infty}S(\omega)H(r,z,\omega)e^{-j\omega t}d\omega \tag{1}$$

Where $S(\omega)$ is the source spectrum and $H(r,z,\omega)$ is the spatial transfer function at frequency ω. Given the special position (r,z), the received time series could be calculated by fast Fourier transform (FFT). In Eq. (1), the key problem is to get the spatial transfer function $H(r,z,\omega)$. Methods based on rays, spectral integrals, normal modes, or parabolic equations all could be used to determine $H(r,z,\omega)$.

In this paper, the normal modes code is used to compute the spatial transfer function. Suppose that the frequency band is $[\omega_L, \omega_H]$, the source depth is z_s, the receiver depth is z_r, and the range between source and receiver is r, then the spatial transfer function at frequency ω_0 is given by

$$H(r,z,\omega_0)\simeq\frac{i}{\rho(z_s)\sqrt{8\pi r}}e^{-i\pi/4}\sum_{m=1}^{\infty}\Psi_m(z_s,\omega_0)\Psi_m(z_r,\omega_0)\frac{e^{ik_{rm}(\omega_0)r}}{\sqrt{k_{rm}(\omega_0)}} \tag{2}$$

$\Psi_m(z)$ is called eigen function and k_{rm} or k^2_{rm} is an eigen value. By changing the frequency ω, we could get the frequency response in band from ω_L to ω_H.

When generating time series on a digital computer, the best method must be chosen on the basis of numerical feasibility and efficiency. Suppose that the sound speed profile and geoacoustics parameters are invariable in within a short time, the eigen function and eigen value corresponding to different frequencies could be calculated in advance and stored. The frequency response could be computed very fast using Eq.(2).

If the time series length is limited, such as acoustic pulse signal, the generating processing as shown in Eq.(1) could be done by FFT. But when the difference in travel times of two modes is greater than the total time interval of the FFT, the later arrival will wrap around to the beginning of the interval. Furthermore, the frequency response contains the time delay τ_0 caused by long range propagation. The total time interval of the FFT must be greater than $\tau_0+\tau_p+\Delta\tau$, where τ_p is the pulse signal length and $\Delta\tau$ is the maximum of difference in travel times of two modes. This means that the FFT length will be very large when the source is far away from the receiver. As known in digital signal processing theory, great FFT length leads to high frequency resolution, the frequency step in calculating $H(r,z,\omega)$ is small and large amount of eigen function and eigen value data should be stored.

2.2 Travel Time Compensation of Frequency Response

Rewrite the frequency response $H(r,z,\omega)$ in another form, which we are familiar with

$$H(\omega)=\left|H(\omega)\right|e^{j\varphi(\omega)} \tag{3}$$

Where $|H(\omega)|$ is the amplitude-frequency response and $\varphi(\omega)$ is the phase-frequency response. Without loss generality, the ocean acoustic channel could be regarded as a linear phase filter. Therefore, the phase-frequency response $\varphi(\omega)$ is given by

$$\varphi(\omega)=-k\omega \tag{4}$$

k is a constant, called group delay of the filter. By using the properties of Fourier transform, we have

$$F\{x(n-k)\}=x(e^{j\omega})e^{-jk\omega} \tag{5}$$

That means the time delay in time domain is equivalent to phase-shift in frequency domain. Then, the travel time could be compensated by multiplying $e^{jk\omega}$ with $H(\omega)$.

$$G(\omega)=H(\omega)e^{jk\omega} \tag{6}$$

$G(\omega)$ is the a new frequency response function that has the same amplitude-frequency response without the travel time. The FFT length could be reduced greatly by using $G(\omega)$ instead of $H(\omega)$ as the channel frequency response. In this paper, the travel time is determined as

$$\tau_0 = \frac{r}{u_{\max}} \quad (7)$$

Where u_{max} is the maximum of the group speed in the frequency band.

2.3 Normal Mode Selection

Phase compensations of frequency response only reduce the time delay of the normal mode with the greatest group speed. The time length of the impulse response function depends on the time difference of arrival between the fastest mode and the slowest mode, which could be give as

$$\Delta\tau = \frac{r}{u_{\min}} - \frac{r}{u_{\max}} \quad (8)$$

Where u_{min} is the minimum of the group speed in the frequency band.

From Eq.(6), is easy to see that the time length will increase with the source range. For some simulation systems with time constraint, the length of impulse function is limited to achieve real-time series generating. It is necessary to take a mode selection from all modes computed by normal mode codes. If the time length limit is τ_h, the modes that meet the condition in Eq.(9) will be selected to form the impulse response function.

$$\frac{r}{u_m} - \frac{r}{u_{\max}} \le \tau_h \quad (9)$$

Indeed, modes selection may lead to accuracy loss. We must take a tradeoff between accuracy and efficiency according to specific parameters of the ocean.

3. Simulation Results

3.1 Impulse Response Function with Travel Time Compensation

In this paper, the normal mode code Kraken is used to compute the frequency response. The sound speed profile is shown in Fig.1. The source depth is 10m and receiver depth is 30m. The source range is 10km. Fig.2 shows the impulse response function before travel time compensation with a travel time of 6.5549s and τ_h of 0.5561s. Therefore, the total time interval of the FFT must be greater than 7.111+τ_p to avoid the aliasing in time domain.

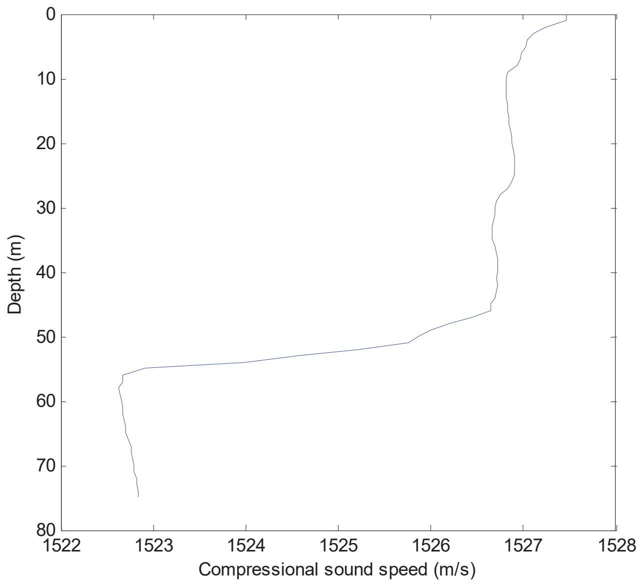


Figure 1 – Sound speed profile

If we compensate the travel time in frequency domain as shown in section 2.2, the total time interval of the FFT will be reduced to 0.5561s+τ_p as shown in Fig.3.

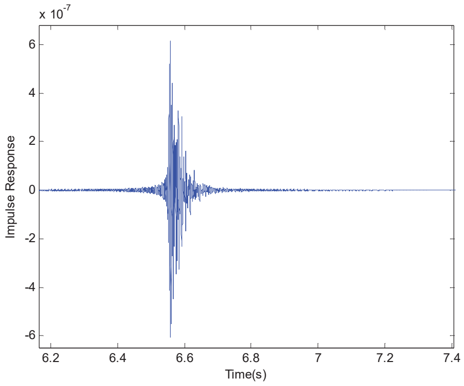


Figure 2 –Impulse response function before travel time compensation

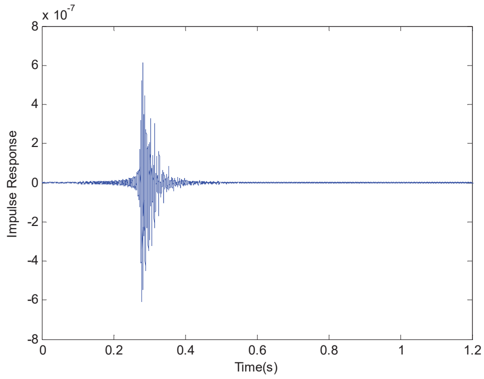


Figure 3 –Impulse response function after travel time compensation

3.2 Frequency Response with Normal Mode Selection

In this paper, we suppose that the time length limit of impulse response function is 2048 samples with a sample rate of 12000 samples per second. The mode selection result is shown in Fig.4. The blue line is the amplitude response versus frequency using all modes, while the red line is that after taking mode selection.

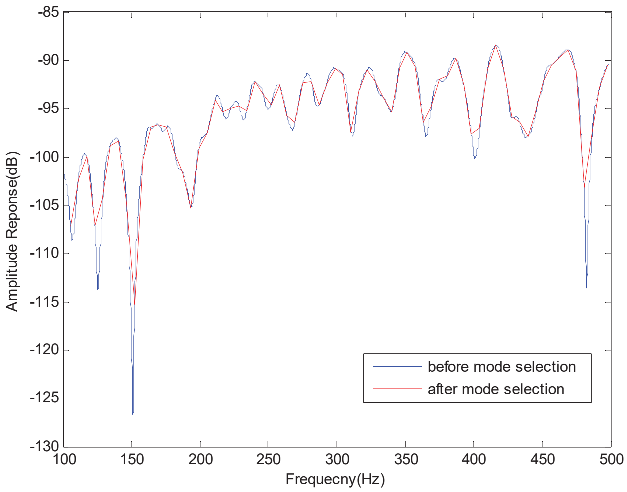
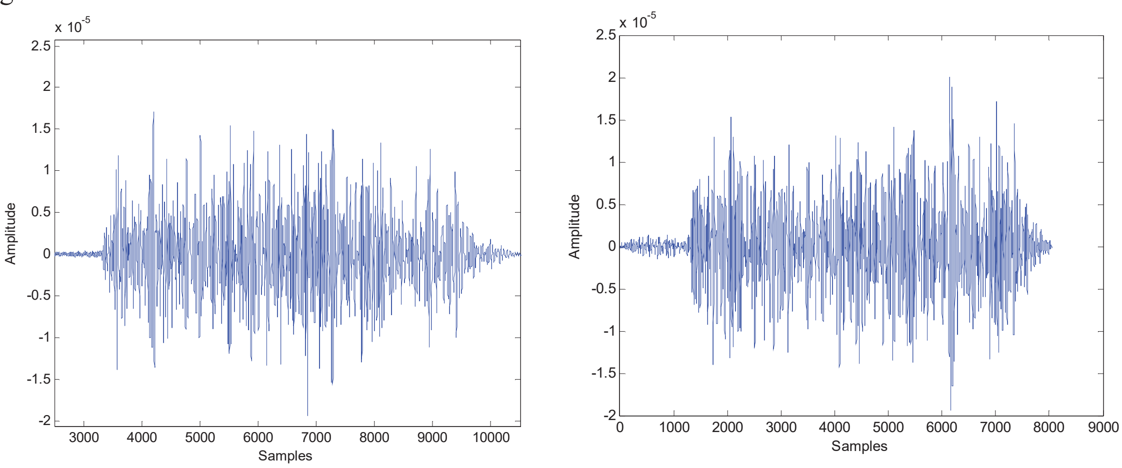


Figure 4 –Amplitude response versus frequency comparison

We use random noise following Gauss Distribution to simulate the ship radiated noise. Fig.5(a) shows the noise time series only by taking the travel time compensation. Fig.5 (b) gives the noise time series by taking mode selection with a 2048-point FIR filter. Fig.6 shows the power spectrum of the generated noise.



(a) time series with time compensation (b) time series with mode selection

Figure 5 – Noise time series

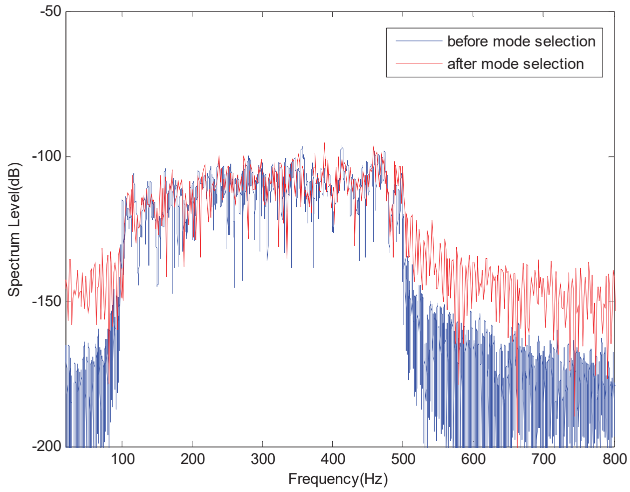


Figure 6 – Noise power spectrum

It should be note that the time delay of the time series in Fig.5(b) is not included because of travel time compensation and normal mode selection. If the travel time cannot be omitted for the underwater acoustics systems, such as active sonar, another FIR filter, called a time-delay filter, is needed to achieve the propagating delay.

4. CONCLUSIONS

A new filter method is presented to generate the underwater noise time series real-time by using normal mode propagation model. The simulation process is realized by two FIR filters. The first one realizes travel time compensation and normal mode selection to reduce the time window for signal generating. The second one is a time-delay filter to achieve the propagating delay. Simulation results prove the validity of the method. The frequency response of the first FIR filter matches well with the result by using all modes. And the power spectrum of time series generated by FIR filters after mode selection also matched well with that of classical methods.

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