METHOD BASED ON BROADBAND COMPRESSED PULSE TO MEASURE PROPERTIES OF UNDERWATER ACOUSTIC MATERIALS

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1 INTRODUCTION

Usually, the acoustic characteristics of underwater acoustic materials can be evaluated by measuring the reflection and transmission coefficients of thin panels, such as echo reduction, insertion loss, absorption and attenuation coefficients. Using the conventional measurement approaches in a free sound field, the sized of tested samples should be at least five times sound wavelength under water in order to reduce the diffraction due to scattering from the edges of panels. Recently, the working frequencies of sonar are gradually extended towards the low frequencies. For applications of underwater acoustic materials, the measuring frequencies must be reduced accordingly. Some improved measurement techniques had been developed [1,2,3,4].

In this paper, a new free field measurement method is proposed — Broadband Compressed Pulse Method (BCPM). It can be used to extend the frequency range in conventional panel tests, especially to reduce the low-frequency cutoff. Before measuring, the least-square inverse-filter processing is done for the measurement system, so that the prime waveform of radiated acoustic pulse is compressed to a certain degree. Then the broadband tests for the reflection and transmission are accomplished, and the superposition method is used in reflection tests. Finally, other acoustic parameters are computed from the obtained reflection and transmission coefficients. The experimental measurement results for two actual material samples are given in this paper, and the results agree very well with those theoretical values.

2 PRINCIPLE OF BOADBAND PULSE COMPRESSION

Let the output of the signal source be a broadband short pulse, if the transient suppression processing is not performed for the projecting part of the measurement system, then the radiated acoustic pulse waveform must have a certain transient period due to Q factor of the transducer, and its spectrum approximates to the frequency response function $H(f)$ of the system. When properly increasing this short pulse width, the transmitting voltage response curve can be compensated to a certain degree at low frequencies. However, that is very limited. When questioning whether the acoustic signal from the transducer is an ideal short pulse whose spectrum is very flat within the measuring frequency range. The answer is yes.

Assuming that the input signal of the power amplifier from a generator is not a $\delta(t)$ function but a signal whose spectrum is $1/H(f)$, in this way, the spectrum of the sound signal received from a hydrophone is approximately equal to 1, and the waveform shows a sharpen pulse in time domain. This process is called inverse-filtering. The actual effect is that the prime output sound pulse is compressed to a desired measuring signal. The degree of their approximation is commonly judged by the least-square standard, and the mathematical model is given by Fig.1.

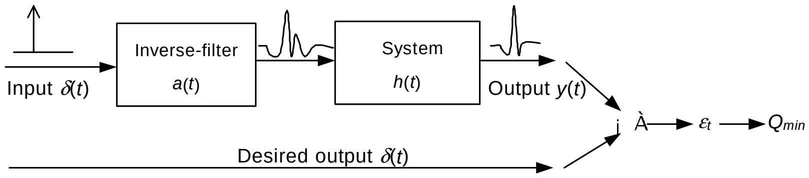


Fig.1 Least-square inverse-filter model

Then, the mathematical relationships are

$$y(t)=\delta(t)*a(t)*h(t)=\hat{\delta}(t) \tag{1}$$

$$a(t)=h(t)^{*^{-1}} \tag{2}$$

Where, $\delta(t)=\begin{cases}1 & t=0\\ 0 & t\neq 0\end{cases}$, $*^{-1}$ is the deconvolution. $a(t)=(a(-m_0),\cdots,a(-m_0+m))$, the factor of inverse-filter is designed to make Q minimum. That is

$$Q=\sum_{t=-\infty}^{+\infty}{\varepsilon_t}^2=\sum_{t=-\infty}^{+\infty}[y(t)-\delta(t)]^2=\sum_{t=-\infty}^{+\infty}[a(t)*h(t)-\delta(t)]^2 \tag{3}$$

Obviously, this is a least-square filter problem. must need

$$\frac{\partial Q}{\partial a(t)}=2\sum_{t=-\infty}^{=\infty}(y(t)-\delta(t))h(t-n)=0\,,\ n=-m_0,\cdots,-m_0+m \tag{4}$$

After being deduced, the least-square inverse-filter function (5) is obtained[5].

$$\begin{bmatrix} r_{hh}(0) & r_{hh}(1) & \cdots & r_{hh}(m)\\ r_{hh}(1) & r_{hh}(0) & \cdots & r_{hh}(m-1)\\ \vdots & \vdots & \vdots & \vdots\\ r_{hh}(m) & r_{hh}(m-1) & \cdots & r_{hh}(0)\end{bmatrix}\begin{bmatrix} a(-m_0)\\ a(-m_0+1)\\ \vdots\\ a(-m_0+m)\end{bmatrix}=\begin{bmatrix} h(-m_0)\\ h(-m_0-1)\\ \vdots\\ h(-m_0-m)\end{bmatrix} \tag{5}$$

where, $r_{hh}(n)$ is the self-correlation function of $h(n)$, $n=-m_0, \cdots, -m_0+m$. Because the measuring system is a physical realized system, so when $n<0$, $b(n)=0$. And the unit-sample response function of system has the feature of minimum phase signal whose energy is concentrated in $[0,m]$, assume $m_0=0$. Let $b(n)$ is the minimum phase signal of $h(n)$, then $b(-1)=\cdots=b(-m)=0$, inverse-filter function (5) can be written in form of Eq.6.

$$\begin{bmatrix} r_{bb}(0) & r_{bb}(1) & \cdots & r_{bb}(m)\\ r_{bb}(1) & r_{bb}(0) & \cdots & r_{bb}(m-1)\\ \vdots & \vdots & \vdots & \vdots\\ r_{bb}(m) & r_{bb}(m-1) & \cdots & r_{bb}(0)\end{bmatrix}\begin{bmatrix} a(0)\\ a(1)\\ \vdots\\ a(m)\end{bmatrix}=\begin{bmatrix} b(0)\\ 0\\ \vdots\\ 0\end{bmatrix} \tag{6}$$

In order to compress the output pulse waveform of the measurement system, the input in Fig.1 may not be $\delta(t)$ but another waveform, for example, a rectangular wave $z(t)$. The desired compressed waveform output can also be obtained. At this time, the $b(n)$ on the right of Eq.6 should be replaced with a cross correlation function of $z(n)$ and $b(n)$.

3 MEASUREMENT SYSTEM AND METHOD

The basic block diagram of the measurement system composed of electronic equipments and underwater units is shown in Fig.2.

Signal generation, radiation and data acquisition in the electronics were controlled through GPIB bus by a computer, which also processed signals in time and frequency domains and accomplished other functions such as acoustic properties calculation, display, storage and printing. The underwater units were arranged in an anechoic tank of 8m×5m×5m, and the acoustic center was at depth of 2m under water. The transmitting transducer was connected to a turning and lifting gear to adjust its direction and depth. So the sound wave beam could be projected on the surface of sample vertically. The tested sample was also fitted to a lifting device to move steadily. The distance L between the tested sample and the surface of the projector was 1.8m to assure the far field condition. The receiver, type B&K8103 hydrophone, was placed in front of the panel for reflection measurements or at rear of the panel for transmission measurements. The hydrophone was away from the surface of the sample d_1 and d_2 respectively and deviated several centimeters from the acoustic axis to eliminate the interference of diffracted wave. The projector was a circular plane transducer array with high directivity and low side lobes that composed of 21 longitudinal composite vibrators, so the disturbance of diffraction from the edge of the sample could be suppressed to a concern degree. Its resonance frequency was about 18kHz.

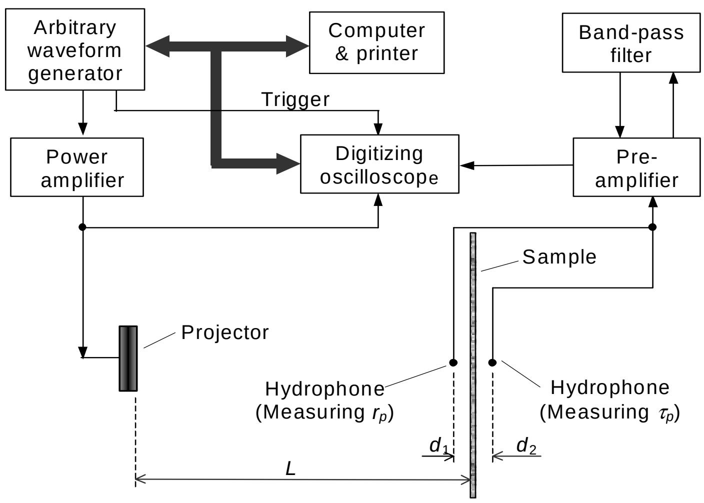


Fig.2 Block diagram of the measurement system

In the tests, because the hydrophone was placed at the position near the surface of the measured sample. So, the reflected signal was separated from the diffraction signal, however overlapped with the directly traveling signal. At that time, the signal received by the hydrophone was $p_d=p_i+p_r$. To gain the reflected signal from the measured sample, the signal p_i was measured lonely without the sample, then those signals are processed in time domain, this was $p_r=p_d-p_i$. Finally, the reflection coefficient r_p and echo reduction E_r were calculated from Eq.7.

The transmission test was as same as traditional method and easy. The reference signal p_i was sound pressure of directly arriving wave without sample, then the measured sample was put down and the signal p_t transmitting through the panel was measured. From Eq.8, the transmission coefficient τ_p and insertion loss I_L were obtained.

$$r_p = \frac{p_r}{p_i} \quad , \quad E_r = -20\lg\left|r_p\right| \quad \text{(dB)} \tag{7}$$

$$\tau_p = \frac{p_t}{p_i} \quad , \quad I_L = -20\lg\left|\tau_p\right| \quad \text{(dB)} \tag{8}$$

Where, p_i, p_r and p_t are the pressure of incident wave, reflected wave and transmission wave of sample, respectively. From Eq.9 and Eq.10, the absorption and attenuation coefficients could also be worked out[1].

$$a = 1 - \left|r_p\right|^2 - \left|\tau_p\right|^2 \tag{9}$$

$$\alpha = \frac{I_L - L_E}{d} \quad \text{(dB/cm)} \tag{10}$$

where, d is the thickness (cm), L_E is the loss due to the acoustic signal reflections and written as Eq.11.

$$L_E = -20\lg\left|(1+r_p)(1-r_p)\right| \quad \text{(dB)} \tag{11}$$

4 MEASURING PROCEDURES

Compared to the conventional measurements for a large area sample in free field, the different procedure in this new method was that the prime sound signal radiated from the transducer which excited with rectangular wave pulse needed to be compressed. The data sampling rate of the arbitrary waveform generator was set to 256kHz, i.e. the address width of one data was approximate 3.9µs. Meanwhile, the sampling rate of the acquisition system was set to 256kHz and the total number of data points was 256. After passing through pre-amplifier and filter, every kind of electric signal is averaged over 128 times in the oscilloscope. So the S/N ratio was raised. The first step to compress pulses was to compute the frequency response function $H(f)$ of the system. Then the discrete values of $H(f)$ and the rectangular waveform of ideal output were substituted into the computer programs for calculation of least-square inverse-filtering. Finally the gained inverse-filter factor was transferred to the arbitrary wave generator through the GPIB bus as signal source data. When this signal was radiated from the transducer, the projected sound signal would become a sharpen pulse and the prime signal was compressed. Taking a rectangular wave (pulse width =3.9×5µs) for an example to illustrate and compare the effects with or without compression. Fig.4 shows a waveform and its frequency spectrum hard copied from the digital oscilloscope with FFT processing function, which was received by the hydrophone, while a rectangular wave pulse was straightly radiated. Fig.5 indicates another waveform and its frequency spectrum received by the hydrophone, while the inverse-filter factor acted as output of the signal source to excite the transducer.

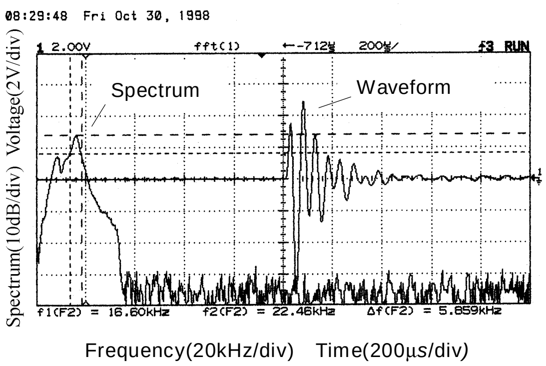


Fig.4 Prime waveform & spectrum

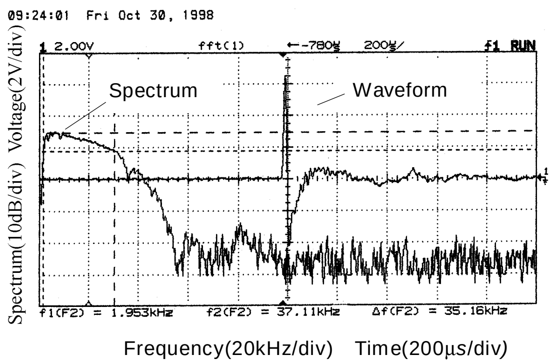


Fig.5 Compressed waveform & spectrum

By comparing Fig.4 with Fig.5, we can see that the prime sound signal was really compressed to a sharpen pulse waveform by the inverse-filter processing, whose frequency spectrum is improved greatly, which the –6dB band is 2-37kHz and -3dB band is 2-25kHz, approaching to the frequency spectrum of an ideal rectangular wave (=3.9×5μs). Hence, to satisfy measurement requirements, the acoustic signals could be well controlled by pulse compression. After the pulse width becomes short, the waveform could isolate the transmission or reflection signal from the diffraction signal in time domain and avoid signals overlapping. With the broadband pulse signal received by the hydrophone being done FFT processing, all the data in the range of measuring frequency could be obtained to realize the required broadband measurements.

In the reflection tests, Utilizing the waveform math function of the digital oscilloscope, the reflected wave p_r is extracted from the overlapped signal p_d. The procedure is as follows:
(1) The sample was hung over the water surface to test the directly arriving signal p_i. After being acquired and averaged over 128 times, this signal was stored in the memory Channel m_1. Neglecting the attenuation of the spherical spreading, this signal was approximate to a reference signal for calculating reflection coefficient, but the distance d_1 should be accurately measured to modify the phase calculation of the reflection coefficient.
(2) The sample was immersed in water, the acquired signal p_d was stored in the memory Channel m_2 as described above. Subtracted the data in Channel m_1 from the data in Channel m_2, i.e. f_1=m_2-m_1, p_r was obtained, and stored in the memory Channel m_3.
(3) The data in channels m_1 and m_3 are sent to the computer, then stored in the hard disk. After FFT analysis, complex frequency spectrums of the signals are obtained. From Eq.7, the reflection coefficient of sample in the range of measuring frequencies is calculated.

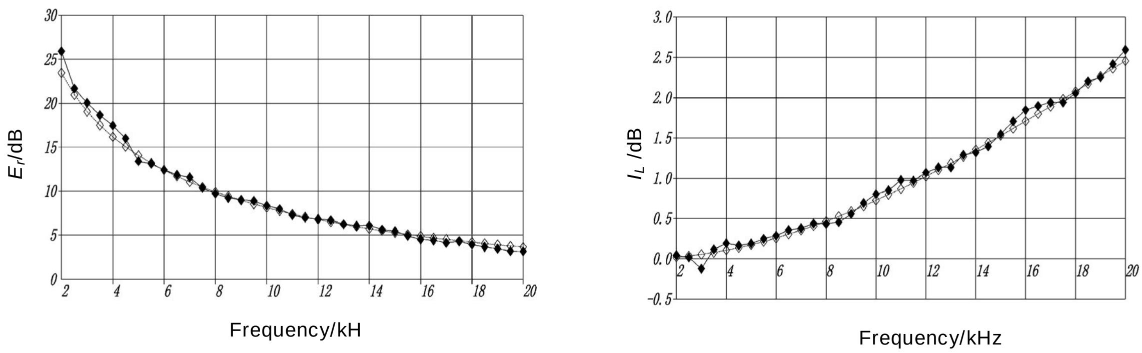
In the transmission tests, the reference signal p_i was a directly arriving wave without any sample. Then the tested sample was put down and the signal p_t transmitting through the panel was acquired and stored in the computer, too. The two waveforms were analyzed by FFT, and their frequency spectrums were substituted into Eq.8 to calculate the transmission coefficient within the measuring frequency range.

5 APPLICATIONS

With the measurement system and programs described above, two actual samples were tested: an aluminum panel (No.1) and a type A-802 rubber panel (No.2)[6]. Their size and thickness were 1.1×0.8m, 8mm and 1×1m, 50mm respectively. Not only the reflection and transmission coefficients were measured, the other acoustic parameters could be also calculated from these coefficients. The results are presented in the Fig.6 and Fig.7, in which "—◆—" is the measuring values and "---◇---" indicates the theoretical values of an infinite large panel model (Their densities are 2700kg/m^3 and 1350 kg/m^3, longitudinal wave velocities are 6260m/s and 900m/s, attenuation factors are 0 and 0.2, respectively). It is found from the Fig.6 and Fig.7 that the measuring curves and theoretical values of the two samples are in very good agreements.

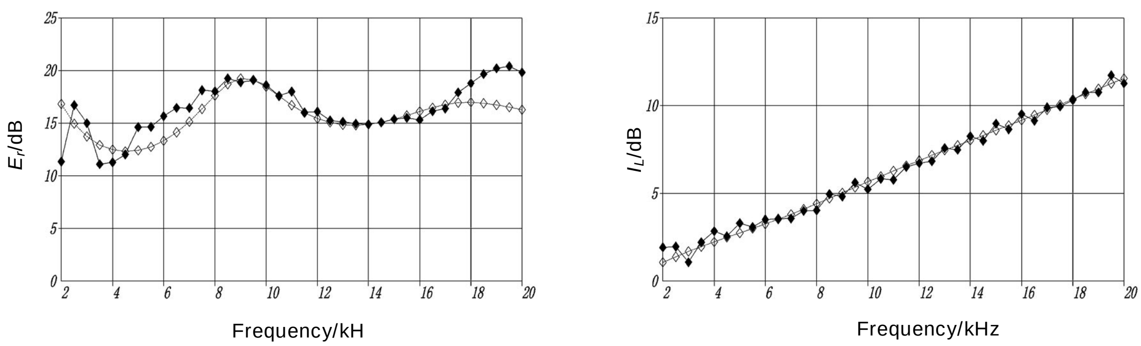
It is known from the Fig.6 that the echo reduction of the 8mm thick aluminum panel is higher at low frequency band, about 25dB at 2kHz, and reduces with the frequencies increasing, about 4dB at 20kHz. its transmission property is so good that the insertion loss is below 3dB.

From Fig.7(a), we can find that the impedance of type A-802 rubber quite matches with water, therefore it doesn't have intensive reflection effect, the echo reductions are above 15dB in the range of 6-20kHz. It is also known from the curves of absorption coefficients in FIG.7(c) that the sample with 50mm thickness has the ability of above 50% sound absorption at above 6kHz. In fact, type A-802 rubber is one kind of materials used for absorbers in underwater acoustic engineering. Since the insertion loss of this panel mainly comes from inside sound attenuation, the measured curves of I_L and α have the same varying tendency. The attenuation coefficient is 1.0dB/cm at 10kHz, 2.0dB/cm at 18kHz and has linear relationship with frequencies. Meanwhile, This sample has the sound attenuation of 5.0dB and 10.0dB at above two frequencies, which approximate to the measured values of I_L.

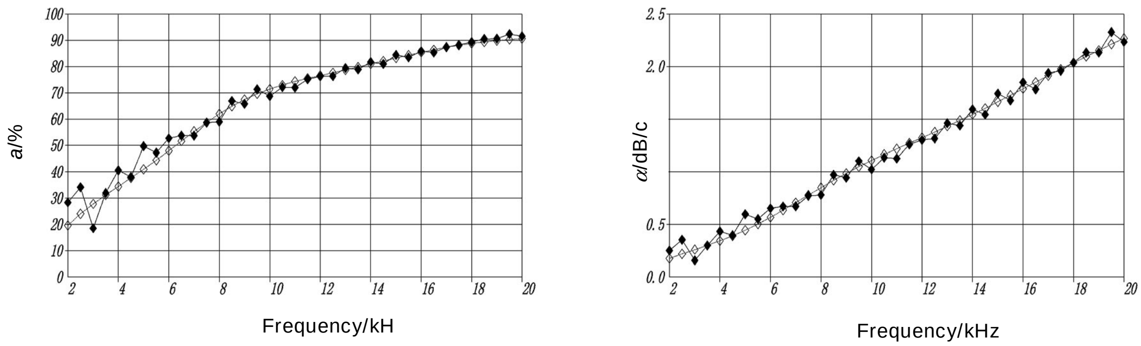


(a) Echo reduction (b) Insertion loss

Fig.6 Acoustic properties of the aluminum panel (—◆— Measuring, ---◇--- Theoretical)



(a) Echo reduction (b) Insertion loss



(c) Absorption coefficient (d) Attenuation coefficient

Fig.7 Acoustic properties of type A-802 rubber panel
(—◆— Measuring, ---◇--- Theoretical)

The measuring errors were mainly caused by the edge diffraction of panels and phase measurements. Because No.2 sample was square form, the influence of diffraction was obviously, as shown in the Fig.7(a). Additionally, the accurate distance between the hydrophone and sample panel under water was difficult to obtain. This measuring error would lead to the phase measuring errors.

6 CONCLUSIONS

The broadband compressed pulse method (BCPM) described in this paper makes a breakthrough at traditional measurement techniques of underwater acoustic materials. The measurement system can be used to test large area material or construction samples in a small tank with better measuring accuracy under automation, the size of tested sample can be lessened. Furthermore, if the low frequency property of a projector system is improved or the resonance frequency of transducer is reduced, this technique could be extended to lower frequency range. It also may be applied to other aspects of calibration and measurement in underwater acoustics.

7 REFERENCES

1. M. Paul Hagelberg & Robert Corsaro
   "A small pressurized vessel for measuring the acoustic properties of materials"
   J.Acoust.Soc.Am. 1985, 77(3), pp1222-1228
2. V. F. Humphrey & H. O. Berktay
   "The transmission coefficient of a panel measured with a parametric source"
   Journal of Sound and Vibration 1985, 101(1), pp85-106
3. C. Audoly & C. Giangreco
   "Improvement of the measurement of the transmission coefficient of panels at normal Incidence using surface receivers"
   J.Acoustique. 1990(3), pp369-379.
4. J. C. Piquette
   "Some new technique for panel measurements"
   J.Acoust.Soc.Am. 1996, 100(5), pp3227-3236
5. CHENG Qiansheng
   "Mathematical principle of signal digital processing(second edition)"
   pp11-488
   Oil Industry Press 1996
6. MIAO Rongxing & GONG Jixiang
   "Underwater acoustic passive materials"
   pp121-122
   Zhejiang University Press 1995