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ADAPTIVE SYSTEM IDENTIFICATION – THE TOOL FOR ACOUSTICAL MEASUREMENTS

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Abstract − Measurement of the impulse response and anechoic transfer function of electroacoustical systems such as loudspeakers is very important issue. Although impulse response of the test object can be measured in an anechoic chamber, in same cases it is not convenient. A solution for measurement of direct and early sound anechoic spectra in a normally reverberant environment is Heyser's time delay spectrometry, based on linear sine sweep. This paper proposes application of adaptive system modeling and random noise excitation instead. Measurements were performed on a loudspeaker, and computation is shown of the energy-time curve and the cumulative spectral decay plot.

1. INTRODUCTION

The determination of the linear transfer function, defined by the impulse response in the time domain is maybe the most fundamental evaluation of an audio system [1]. There are three established methods of its measurement: periodic impulse excitation (PIE) [2], maximum-length sequences (MLS) [3], and time- delay spectrometry (TDS) [4].

The measured MLS output sequence is cross correlated with the known input sequence in order to obtain the impulse response. The most efficient method of performing the cross correlation is fast Hadamard transformation [1].

TDS yields transfer function information in frequency domain. To reveal impulse response, TDS applies inverse Fourier transform. Using TDS to separate multipath components (or obtain anechoic transfer function) is identical to windowing the impulse response of the multipath system with the window centered at some point in time - axis [5].

The objective of this paper is to examine possibilities of adaptive system modeling on measurement of impulse response measurement of complex electroacoustic system, as well as calculation of energy-time curve and spectral decay plot. The paper is organized as follows: Section 2. describes adaptive system modeling. Section 3. presents computation of energy-time curve from the identified impulse response. Section 4. presents computation of cumulative spectral decay plot. Section 5. provides measurements on a loudspeaker mounted on the acoustic waveguide. The analysis is performed using proposed novel approach. Section 6. summarizes the results.

2. ADAPTIVE SYSTEM MODELING

Adaptive system modeling can be described with the Fig. 1. It is application of adaptive system identification [6], and it can be considered as a discrete-time "black-box" representation of a single-input, single-output dynamic system.

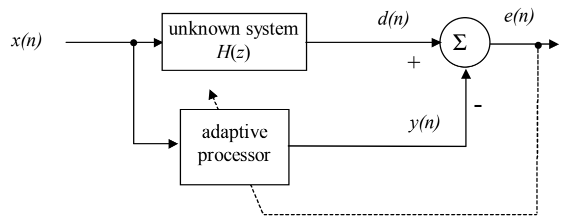

Fig. 1. Adaptive system modeling

The least mean square (LMS) adaptive signal processing algorithm [6] can be applied on adaptive system modeling. Using notations from Fig. 1., *e(n)* denotes error signal, *x(n)* is random noise training signal, *d(n)* is output signal from the plant that is to be modeled, and *y(n)* is adaptive filter output signal, where *n* is sample number.

Filter output is defined by

$$y(n) = \mathbf{X}^T(n)\mathbf{W}(n) \quad , \tag{1}$$

error signal is

$$e(n) = d(n) - y(n) \quad , \tag{2}$$

and adaptive weights updating is

$$\mathbf{W}(n+1) = \mathbf{W}(n) + 2\mu\mathbf{X}(n)e(n) \tag{3}$$

where $\mathbf{W}(n) = [w_n(0) \;\; w_n(1) \;\; ... \;\; w_n(L-1)]^T$ is weight vector of the adaptive finite impulse response (FIR) filter, μ is step size that satisfies stability condition [6], $\mathbf{X}(n) = [x(n) \;\; x(n-1) \;\; ..... \;\; x(n-L+1)]^T$ is vector of the reference signal. L is number of adaptive weights and $[\;]^T$ stands for vector transpose.

3. ENERGY-TIME CURVE

Energy-time curve *e(t),* as defined by Heyser [4] is envelope of the analytic signal *z(t)*

$$e(t) = |z(t)| = \left[g^2(t) + \hat{g}^2(t)\right]^{\frac{1}{2}} \quad (4)$$

where $\hat{g}(t)$ denotes Hilbert's transform of system's impulse response $g(t)$ and *t* denotes time.

Analytic signal is derived from system's impulse response as

$$z(t) = g(t) + j\hat{g}(t) \quad (5)$$

where *j* stands for $\sqrt{-1}$.

The Hilbert transform $\hat{g}(t)$ can be expressed as

$$\hat{g}(t) = g(t) * \frac{1}{\pi t} \quad (6)$$

where * denotes convolution [7].

In the frequency domain, Fourier transform of $\hat{g}(t)$ is then

$$F\{\hat{g}(t)\} = -j \operatorname{sgn} f \cdot G(f) \quad (7)$$

where *G*(*f*) denotes Fourier transform of *g*(*t*), *f* is frequency and sgn*f* denotes signum function. $\hat{g}(t)$ can be obtained by inverse Fourier transform of (7).

4. CUMULATIVE SPECTRAL DECAY

The energy-time-frequency curve (ETFC) is a three-dimensional display of time-delay t_n , frequency and amplitude [5]. Since the energy-time-frequency curve is equivalent to the spectrogram with a modulating term that has no effect on magnitude, it can be derived from the spectrogram

$$S(t_n, f, W) = \left| \int_{-\infty}^{\infty} g(t) w(t - t_n) \exp(-j2\pi f) dt \right|^2 \quad (8)$$

where *w* denotes windowing function.

The cumulative spectral decay plot also shows frequency and time- domain properties of a system. It is computed from the system's impulse response by windowing and Fourier transformation. Unlike the ETFC, the window is not fixed in shape. Its falling edge is fixed, and its rising edge is moving from the time zero to the right. An appropriately apodized cumulative spectral decay plot is defined by following function

$$C_a(\tau, \nu) = F\{h(t)w(t)\} \quad (9)$$

where τ is time variable, and ν is frequency variable of plot, *F* is Fourier operator, *h*(*t*) denotes impulse response of device under test and $w(\tau,\nu)$ is window function [1].

5. EXPERIMENTAL RESULTS

Experimental model was consisted of the Bruel&Kjaer standing waves apparatus type 4002 equipped with the bigger measuring tube (upper frequency limit of 1600 Hz), power amplifier, loudspeaker, dynamic microphone and a personal computer (PC) with acquisition card. Length of the tube is 1 m and diameter is 99 mm. The tube was left with open termination, and the microphone probe was set 8 cm inside the tube, measured from the termination. Training signal was low-pass filtered white noise. The low-pass filter was 20th order Buttherworth, and cut-off frequency was 1.5 kHz, below the first cut-off mode of the duct. Sampling frequency was 8 kHz. The training and error signal were transferred to MATLAB environment, and floating point arithmetic was used for adaptive weights updating.

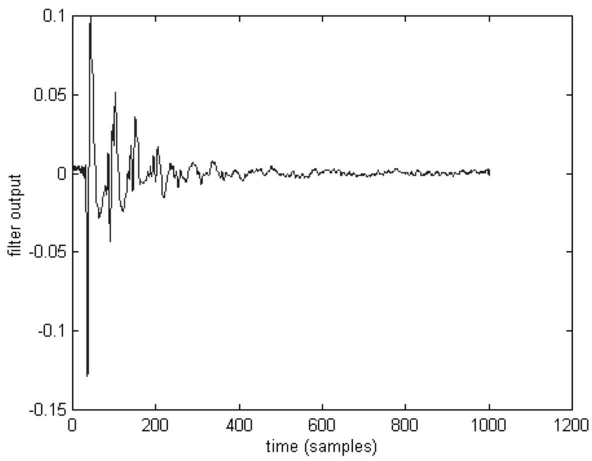

Fig. 2. Impulse response

Number of the adaptive weights was varying, and Fig. 2. presents impulse response for 1000 adaptive weights. It is obvious that this filter length allowed very accurate estimation of overall impulse response. For comparison, Fig. 3. presents system's frequency response calculated from estimated transfer function and system's frequency response derived from fast Fourier transform (FFT). The curves are almost the same, and this qualifies proposed method as efficient and useful in acoustic measurements.

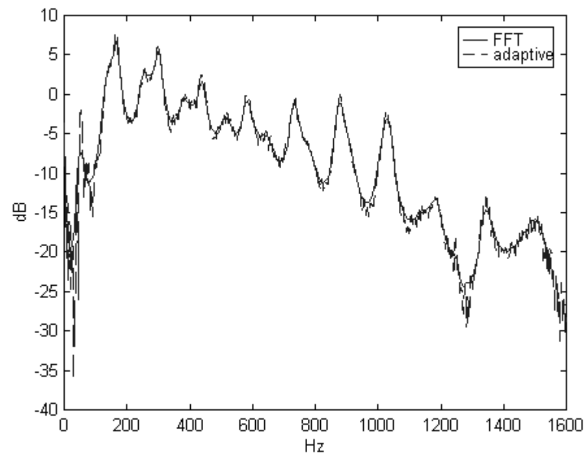

Fig. 3. Frequency response

Fig. 4. presents ETC plot of the same system, and Fig. 5. presents cumulative spectral decay (CSD) plot. ETC and CSD are calculated from estimated impulse response. Falling and rising edge of the apodizing window were derived from Hamming window. CSD is derived from 700-point discrete Fourier transform.

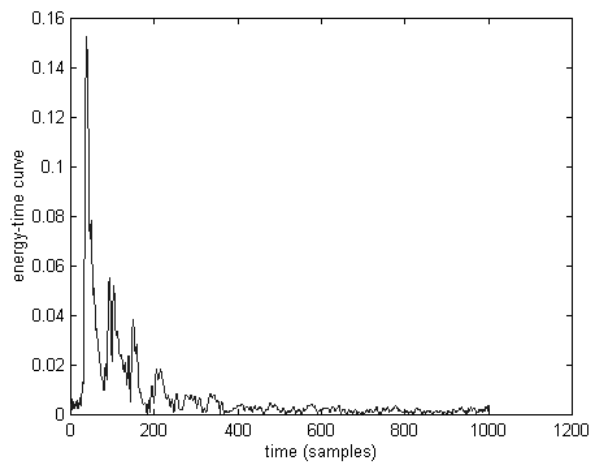

Fig. 4. Energy-time curve

6. CONCLUSIONS

Adaptive system modeling is efficient tool for impulse response measurement of complex acoustic systems. In postprocessing, it can be used as basis for computation of energy-time curve and cumulative spectral decay plot. Unlike TDS, proposed method can be very easy used for determination of system's response on various test signals.

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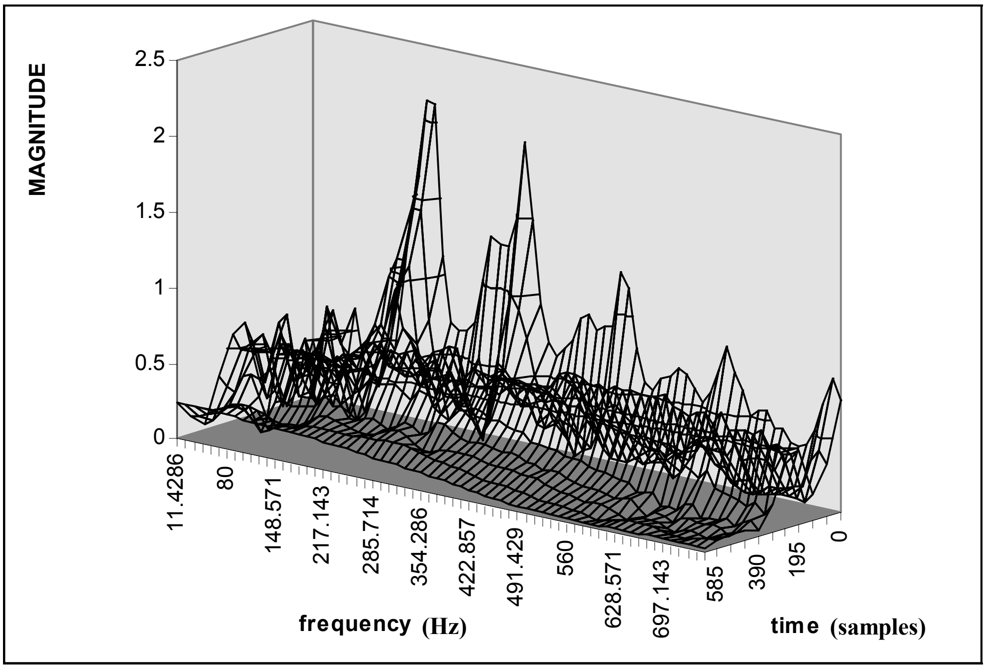

Fig. 5. Cumulative spectral decay plot

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