Adaptive Control of Sound Transmission with Neural Network Algorithms

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Abstract: A neural network-based feedforward adaptive controller using the modified Error Back Propagation Learning Algorithm is presented. The controller is realized with a TMS320C25 DSP Board monitored by an IBM PC compatible, and applied to control the sound transmission through a thin panel between two rooms. Experiments showed that the algorithm was superior in robustness and broad-band performance.

INTRODUCTION

Artificial Neural Network (ANN) is widely interested recently for its self-learning, self-organizing, adaptability as well as non-linearity in nature. Much progress has also been made in the researches of active control of structural sound radiation and sound transmission through panels for its excellent performance in the low frequency range. While most of the progress in experiments was made by using an adaptive controller based on LMS algorithms. Some possible applications of BP-algorithm based ANN in active noise control have been approached before, but the main work on complex situations of structural sound radiation was conducted in computer simulations. In this paper, a neural network-based feedforward adaptive controller applied to control the sound transmission through a thin panel between two rooms is introduced. Experiments were carried out for the active control of sound transmission through a thin panel mounted between two reverberation rooms. Encouraging results were achieved.

ALGORITHMS AND THE CONTROLLER

The algorithm structure of adaptive controller applied in this experiment is a modified traditional multi-layered feedforward neural network (MFNN). Random gradient estimation method is used to update the weights of MFNN. The modification is mainly reflected in 3 respects. (a) The nonlinear transfer function of each neural net is supposed to be in the form of that

\[ f(x) = \frac{1}{\alpha + \beta e^{-x}} \]  

where \(\alpha\) and \(\beta\) are adjustable parameters that are used to change the slope of nonlinearity. (b) An inertia term proportional to the prior variation is introduced in the weight updating subroutine. The term can smooth and cushion the weight variation, and help the weights get out of platforms that might exist during the convergence. (c) To speed the learning rate, the convergence factor can be adaptively adjusted according to the simultaneous power estimation of error signal.

Both filtered-X and filtered-U adaptive filters were designed for the controller to compensate the secondary signal feedback as shown in Figure 1, in which B is the auxiliary filter, NN is the main filter and BP is the back propagation algorithms. The error delay compensation was performed in the main filter and the BP algorithms. Causality analysis of a feedforward active control system shows that, the compensations can essentially improve the system behavior, but it would still be limited by the practical specifications of the digital signal processor, especially the time needed for the calculations.

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EXPERIMENTAL RESULTS

Experiments were carried out in the sound insulation measurement rooms in the Institute of Acoustics, Academia Sinica. The room is 100 m³ each in volume, with rigid walls, floors and ceilings. A 500 × 420 × 3 mm³ steel plate was mounted on the specimen mounting window between the source room and the receiving room. The primary source was simulated with a loudspeaker enclosure placed 1.2 - 1.6 m away from the panel and actuated by a signal generator and a power amplifier. The secondary actuator was a B&K4810 electrodynamic shaker, mounted at the center of the plate. The reference microphone is placed between the primary source and the plate and 0.8 - 1.2 m away from the plate. The error microphone is located at a distance of 0.8 - 1.4 m in front of the plate. The controller was realized with a TMS320C25D digital signal processing board. A 3-layer neural network was applied with a structure of 8 × 6 × 1. The sampling frequency was set to around 4.88 KHz. A 64-step filter was used for the error channel compensation and a 40-step one was used for the feedback channel. The convergence factor was among 0.15 and 0.28. The inertia coefficient of 0.01 to 0.06 was selected. The error signal was generally amplified by 1 to 5 times.

For the pure tones corresponding to the most efficient frequency of sound transmission, the ANN based active control system can achieve an excess transmission loss of more than 20 dB. Figure 2 gives the performance of the system for a broad band noise excitation. The total excess noise attenuation at the error microphone is about 6.9 dB.

![Figure 2](image-url)

**FIGURE 2.** Experimental results of the ANN based active control system for a broad band sound transmission

REFERENCES