Advanced System Identification Techniques for Acoustics Enhancement

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Abstract This paper proposes a novel scheme of adaptive system identification for acoustics enhancement applications. A key contribution of this study is the development of the variable exponentially-weighted-stepsize LMS (VES) algorithm based on the statistics of the overall varying characteristics of the estimate of room impulse responses. The modelling error of the VES algorithm can reduce to -20dB after only 2N iterations (N is the length of adaptive filter), about 4 times faster than the normalised LMS algorithm under the test conditions.

INTRODUCTION

Room acoustics enhancement systems are used to improve and control the natural acoustics of concert halls, auditoria and smaller performance venues. They operate by using microphones, electronic processors and loudspeakers to provide additional sound reflections and reverberations to the naturally occurring reflections and reverberation. System identification is a useful technique for increasing the performance of room enhancement systems. There are a variety of approaches which have been developed. The most common methods, including transient (or impulse) analysis, frequency analysis, the least squares method, the instrumental variable method and the prediction error method, have their own advantages and limitations [1]. The performance of these methods is strongly dependent upon the system structure, measurement signal and the measurement environment. Considering that acoustics enhancement systems are usually operating in a closed loop and any measurement signal must be inaudible, as well as the requirement of real time measurement, it has been found that the transient analysis, spectral analysis and the instrumental variable method are very difficult to be employed in this situation. The least squares method based adaptive identification approach with its modest requirement on memory and reasonable computational load, has attracted attention recently.

ADAPTIVE SYSTEM IDENTIFICATION

Various adaptive algorithms are applicable to acoustical system identification. The recursive least-squares (RLS) algorithm provides fast convergence at the price of a relatively high computational load. The recently developed fast RLS algorithms still require excessive computation. The least-mean-squares (LMS) algorithms, on the other hand, is robust and simple. The normalized LMS (NLMS) algorithm, whose convergence speed is independent of input signal power, is widely used in acoustic system identification for commercial applications. One of the major drawbacks of the LMS and NLMS algorithms is their slow convergence rate under some conditions. A high convergence rate is critical for an acoustic signal processing system with multiple channels, when it is operating in real time.

A time-varying stepsize or time-varying matrix-form stepsize NLMS algorithm based on the principle that the feedback constants vary according to an estimate of the distance to the mean-square-error minimum has been proposed to improve the convergence rate. These variable step adaptive filter algorithms, however, require complicated control of the stepsize, and have received limited application. Recently, an exponentially weighted stepsize NLMS adaptive algorithm based on the statistics of a room impulse response is proposed in [2]. Although this algorithm is simple to implement and increases the convergence rate at the beginning of the iteration, it results in a slower convergence rate at later times and a larger excess mean-squared error.

VES ALGORITHM

Statistically speaking, the correction term in the NLMS algorithm for all weights of the new estimate impulse response vector has the same level. This is suitable for the case where the error of the estimate
has the same level, or is evenly distributed for all weights of the vector. However, it is well known that
room responses are exponentially decaying curves [3] and could be approximately expressed by

\[ h(l) = h_0 e^{-\gamma l/T_0} \quad \text{for} \quad l = 0, 1, \cdots, L - 1 \]  

where \( T_0 \) is the reverberation time, \( h_0 \) and \( \gamma \) are constants, respectively. Obviously, at the initial stage
of the adaptive identification process, the difference between the real impulse response and the estimate
is approximately exponentially distributed over all weights associated with the constants \( h_0, \gamma \) and
\( T_0 \). The evenly-distributed correction term may too large for the weights where \( l \) is close to \( L \) and too
small for indices close to 0. This results in a slow convergence speed of the NLMS adaptive algorithm at the
initial stages of the adaptation. The exponentially weighted stepsize (ES) algorithm proposed [2] is
intended to adjust the coefficients with large errors in large steps and the coefficients with small errors in
small steps by introducing a diagonal stepsize matrix \( \Delta \). The ES algorithm produces a better convergence
speed at the initial stage of the adaptive process in comparison with the NLMS algorithm when used for
room transfer function modeling. However, this advantage of the ES algorithm over the NLMS algorithm
reduces as the iteration continues, because the fixed additional weight matrix \( \Delta \) is only associated with
the estimate error at the initial status.

A variable exponentially-weighted-stepsize NLMS algorithm, called VES adaptive algorithm, has been
proposed in this study to improve the convergence rate over the full iterative process and obtain higher
identification accuracy. Instead of the additional weight matrix \( \Delta \) with time-invariant elements \( \alpha_l = \alpha_0 \beta_0^l \),
the additional weight matrix in the proposed VES algorithm is time-variant and with the elements

\[ \alpha_l(k) = \alpha_0 \beta_l(k) \quad \text{for} \quad l = 0, 1, \cdots, L - 1 \]  

The corresponding weights updating step in the VES algorithm can be expressed by

\[ \hat{h}_{ij}(k + 1) = \hat{h}_{ij}(k) + \frac{e_i(k)}{\|A(k)z_{jL}(k)\|^2} A(k)z_{jL}(k) \]  

where \( \hat{h}_{ij}(k) \) is the estimate of a weights vector which represents the impulse response from the \( j \)th system
loudspeaker to the \( i \)th system microphone at time \( k \), \( \alpha \) is the scalar stepsize, \( e_i(k) \) is the corresponding
error signal and \( \| \cdot \| \) is the Euclidean norm. \( z_{jL}(k) \) is a segment of input signal for the \( j \)th loudspeaker
with the length of \( L \). The variable exponential attenuation ratio \( \beta(k) \) can be calculated from the estimate
error by the following equation, given that the envelope of average room impulse response \( h_{a+}(l) \) (for \( l = 0, 1, \cdots, L - 1 \)) or the reverberation time \( T_0 \) is available.

\[ \beta(k) = \left[ \frac{\Delta h_{lw}(k)}{\Delta h_{uw}(k)} \right]^{1/\Delta t} \]  

where \( \Delta h_{lw}(k) \) is the estimate error calculated by comparing the amplitude of estimated impulse response \( \hat{h}(k) \) with the envelope of average room impulse response at a lower index \( l_{lw} \) at time \( k \), and \( \Delta h_{uw}(k) \) is the estimate error at an upper index \( l_{uw} \) at time \( k \). \( \Delta t \) is the difference between the lower index and the
upper index numbers.

**SUMMARY**

In this study, a set of novel techniques for acoustic system identification which relates to multi-channel
acoustics enhancement systems are proposed. A variable exponentially weighted stepsize NLMS, or VES
adaptive algorithm has been developed to improve the convergency rate at all stages of adaptation. Also
suggested is the identification of the transfer functions group by group to reduce the stimulus signal level,
and by the use of time averaging to further reduce the effects of interference. Results of simulation tests
have shown that the proposed techniques are practical for the applications.

**REFERENCES**

Based on the Statistics of a Room Impulse Response”, *IEEE Trans. Speech and Audio Proc.*, Vol.1,