SYSTEM IDENTIFICATION WITH ADAPTIVE LATTICE FILTERS FOR SPEECH DATA

Wenxia Shi, Vijay Parsa, and Jagath Samarabandu
Dept. of Electrical and Computer Engineering, The University of Western Ontario
London, Ontario, Canada N6A 5B9, wshi7@uwo.ca

1. INTRODUCTION

In this paper, we investigate the application of adaptive lattice filter structures in modeling the response of hearing aids to speech signals. Adaptive lattice filters are a class of linear adaptive filters whose designs are based on algorithms that involve both order-update and time-update recursions. Although the popular transversal structure is easy to implement, the lattice structures have their own advantages and are attractive in several adaptive filtering applications. Some of the highly desirable properties of the lattice-based filters include: modularity, computational efficiency and statistical decoupling of the individual stages. The lattice structure inherently has the orthogonalization property between the backward prediction errors which helps in faster convergence rates [1].

There are two approaches for lattice-based adaptive filter implementation: the stochastic-gradient approach known as the gradient adaptive lattice (GAL) filter, and the least-squares approach known as the least squares lattice (LSL) filter.

In this paper, the performance of LSL and GAL algorithms is evaluated in the context of adaptive modeling of hearing aids. Speech signals processed through modern digital hearing aids are analyzed using the GAL and LSL algorithms. The performance of these algorithms is compared with the classical Least Mean Square (LMS), Recursive Least Square (RLS), and Affine Projection Algorithm (APA) in terms of computational complexity and modeling performance.

2. METHOD

2.1 Adaptive Lattice Filter Structure and Algorithm

In this section, we outline the structures and algorithms of the GAL filter and the LSL filter. Figure 1 shows the block diagram of the multistage lattice predictor that performs both forward and backward predictions. Here the desired response *d(n)* is estimated by the lattice filter using the input signal *u(n)*. The coefficients in the lattice stages are updated using either the GAL or LSL algorithm. The GAL algorithm is simple to implement, but is approximate in nature due to the fact that each stage of the lattice predictor is characterized by a single reflection coefficient. In contrast, the LSL filters are exact but more complicated due to the fact that each stage of a least-squares lattice predictor requires two different reflection coefficients for its characterization—one for forward prediction and the other for backward prediction [1].

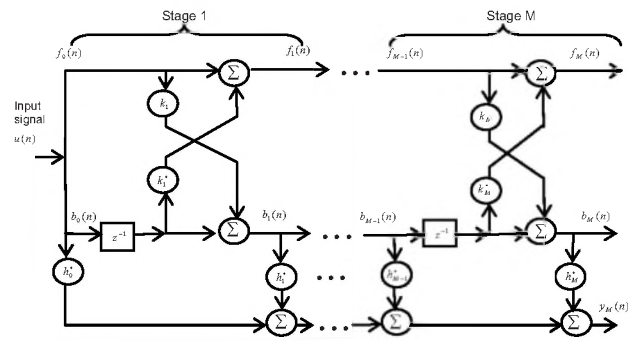


Figure 1. Lattice-based structure for joint-process estimation

2.2 Adaptive Modeling of Digital Hearing Aids

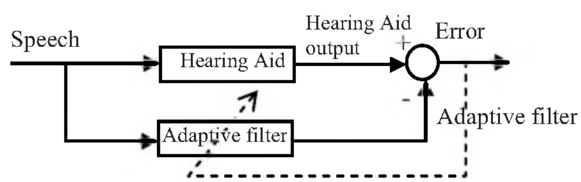


Figure 2. System identification for speech dataset

Figure 2 shows the block diagram of adaptive modeling of digital hearing aids. This system facilitates electroacoustic measurement of hearing aid performance using natural speech and music signals. The adaptive filter models the time-varying behaviour of the hearing aid, leaving the noise and distortion components in the error residual. These components can be used to quantify the quality of the hearing aid. Hearing aid data were collected using a custom Hearing Aid Test System (HATS) developed at the National Centre for Audiology (Figure 3). The speech signal is played back through the speaker in a portable anechoic test box and the response of the hearing aid and a reference microphone are recorded and stored in the computer. The reference microphone input and the hearing aid output are then used to drive the GAL and LSL algorithms.

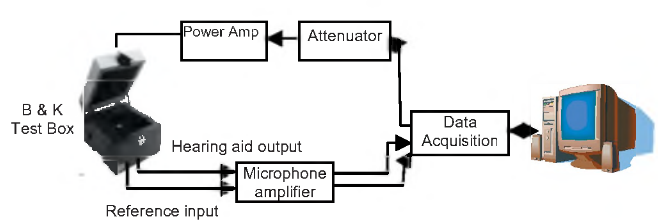


Figure 3. Block diagram of the Hearing aid test system (HATS)

2.3 Relative Performance Comparison

Previous hearing aid modeling studies have exclusively used transversal filter structures and LMS based algorithms [2]. The LMS-based algorithms included the Normalized Least Mean Square (NLMS), and the Affine Projection Algorithm (APA) [1]. In this paper, we have undertaken a preliminary investigation of the relative performance of transversal and lattice filter architectures and algorithms in the context of hearing aid modeling. In particular, the performance of GAL, LSL, LMS, RLS, and APA algorithms was compared in terms of modeling performance, i.e., signal to noise ratio and computational time.

3. RESULTS AND DISCUSSION

3.1 System Identification for Hearing Aid Data

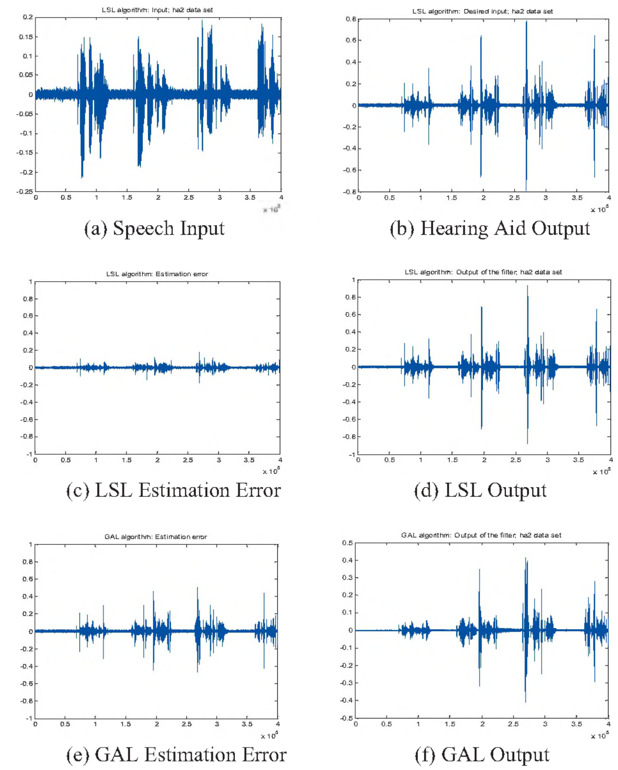
The hearing aid output and the reference input were given to the adaptive filter algorithms. All five adaptive filter algorithms were implemented in MATLAB. Figure 4 shows the results of system identification using the LSL and GAL algorithms for data obtained from a commercial digital hearing aid. The filter length was set 50 and 400000 samples were used for both the LSL and GAL filters. Figure 4(a) shows the speech input and 4(b) displays the corresponding hearing aid output. The predicted hearing aid responses are shown in Figures 4(d) and 4(f) for LSL and GAL algorithms respectively with the corresponding modeling residuals in Figures 4(c) and (e).

3.2 Performance Comparison

Modeling performance and computational complexities of different adaptive algorithms are compared in Table 1. The modelling performance was measured as the ratio of the hearing aid output and error residual powers. Computational time for each of the algorithms was measured in MATLAB as an average of about 50 runs for each algorithm.

Comparison results show that LSL and RLS can obtain very good performance results. LSL is a bit better than its transversal counterpart RLS, and computational time of LSL is less than that of RLS. If the filter length and data length are further enlarged, these differences between LSL and RLS will be increased correspondingly.

From Table 1, we observe that the GAL and NLMS algorithms display poor performance. Although the NLMS is omputationally the most efficient, its modelling performance is quite poor in the context of speech-based modeling of hearing aids.



(a) Speech Input (b) Hearing Aid Output

(c) LSL Estimation Error (d) LSL Output

(e) GAL Estimation Error (f) GAL Output

Figure 4. System identification of speech data for LSL and GAL

TABLE 1. COMPARISON OF DIFFERENT ALGORITHM

	LSL	GAL	NLMS	RLS	APA
SNR(dB)	12.9149	2.9019	0.7143	12.8692	9.9465
Running time (s)	25.75	26.95	13.63	26.92	28.59

4. CONCLUSIONS

In this exploratory study, the relative performance of various adaptive filtering algorithms and structures was investigated in the context of hearing aid system identification using speech stimuli. The Least Squares Lattice (LSL) algorithm provided the best performance and computational efficiency. Our future work is to develop a subband LSL algorithm in order to model the performance of multichannel compression hearing aids better [3].

REFERENCES

[1] Haykin,S. (2002). Adaptive Filter Theory. 4th edition, Prentice Hall, NJ, ISBN-0-13-090126-1.

[2] V. Parsa & D.G. Jamieson, "Hearing Aid Distortion Measurement Using the Auditory Distance Parameter," Audio Engineering Society, Convention Paper 5434, September 2001.

[3] M.R. Wirtzfeld & V. Parsa, "On Subband Adaptive Modeling of Compression Hearing Aids," in ICASSP 05', Volume 3, Page(s): 45-48, Philadelphia, PA, March 18-23, 2005.