Subband Adaptive Filters

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1 Overview

The motivation of using subband adaptive filters (SAFs) based on least mean squares (LMS) algorithm is to allow better convergence rate as well as computational savings [5]. Instead of working directly on the input signal, a subband decomposition allows the modeling error signal beging projected on an orthogonal basis, and thus separates the convergence into decorrelated components which would give an improvement of convergence rate [2]. Intuitively, the better convergence rate can be achieved by matching the energy of the subband signal to the adaptation step size in each subband. Specifically, as we increase the number of bands in the filter, the convergence rate increases and approaches the rate that can be obtained with a flat input spectrum [7].

For computational savings, transposing an adaptive filter behind the analysis filterbank turns the problem of adapting a single long finite impulse response (FIR) filter into that of adapting several short FIR filters which operates at a lower rate [2, 6]. For example, in the context of acoustic echo cancellation (AEC), SAF is particularly useful in terms of several advantages including computational savings and better convergence behavior for the real-time identification of long impulse responses [4].

Furthemore, SAFs have been widely used in a lot of fields such as AEC [4], speech enhancement [1] and speech dereverberation [3], etc. In this project, our goal is to not only understand the theoretical difference between SAFs and adaptive filters (AFs) but also apply them on several speech-related problems including AEC, speech enhancement and speech dereverberation. Fig. 1 and 2 are illustrations of using an AF and a SAF.

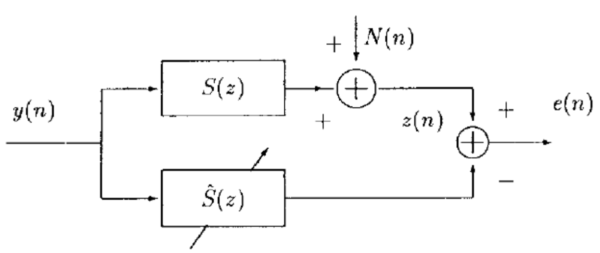

Figure 1: An AF for system identification problem [7].

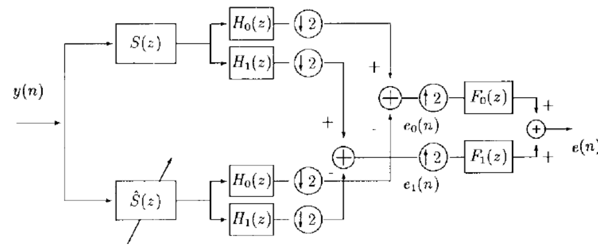

Figure 2: A SAF for system identification problem [7].

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