INTRODUCTION

Speech and audio signal processing has been a topic of active research during the last two decades, given as a result of the development of several efficient algorithms for echo and noise cancelling, active noise cancelling, speech enhancement, etc., most of these applications depending on transversal adaptive filters. Although, recently, the multirate implementations of those systems have proved to be a desirable alternative when large filters order with fast convergence rates are required.

In multirate adaptive filtering the input signals are divided into \( N \) subbands signals using a filter bank or an orthogonal transformation. Subsequently, an \( N_f/M \) order adaptive filter is inserted in each subband, where \( N_f \) is the full band adaptive filter order, whose coefficients are updated to minimize either the output errors in each subband given by the difference between the subband adaptive filter output and the subband reference signal, or a full band common output error. Finally, some of the signals involved in the adaptation process are used to synthesize the adaptive filter output signal.

Multirate adaptive filtering presents several advantages over the conventional full band filter structure: a) Because each band is downsampled by a factor \( M \), the computational complexity of the subband system is approximately reduced by a factor \( M \), compared with the full band one. This fact allows the development of more sophisticated and efficient adaptive algorithms, as well as
a relatively easy implementation of high order gradient-search-based algorithms, such as those required for acoustic echo cancellation; b) For speech and audio signals the correlation between consecutive samples decay as the separation among them grows. This fact allows increasing the convergence rate of gradient search-based algorithms achieving, for any kind of signals, convergence rates similar to those obtained when the input signals are white. The main problem present in subband adaptive filtering is the processing delay and the distortion introduced during the analysis and synthesis process, although these problems have been reduced during the last years (see Chapter 1).

This chapter reviews the basic principles of multirate adaptive filters together with some of their most successful applications, as well as some recently proposed schemes that improve the subband adaptive signal processing systems.

ADAPTIVE FILTERING

An adaptive filter (Figure 1) is a time-varying system that consists of a digital structure, \( W(n) \), with adaptive parameters that are updated to minimize some given criterion of the output error, \( e(n) \), which is given as the difference between the reference, \( d(n) \), and adaptive filter output, \( y(n) \). The filter parameters are adjusted repeatedly, in accordance with an adaptive algorithm, which explicitly or implicitly uses all signals shown in Figure 1. The details of two of the most widely used adaptive algorithms are given in the next two sections.

Figure 1: Adaptive filter configuration
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