Speech and Audio Signal Applications

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INTRODUCTION

With the development of the VLSI technology the performance of signal processing devices have greatly improved making possible the implementation of more efficient systems to storage, transmission enhancement and reproduction of speech and audio signals. Some of these successful applications are shown in Table 1.

BACKGROUND

The signal processing has played a very important role in the development of many speech and audio systems. This section presents a review of some of these successful applications.

Echo Cancellation for Long Distance Transmission

A very successful speech signal processing application is the adaptive echo cancellation used to reduce a common but undesirable phenomenon in most telecommunications systems, called echo. Here when mismatch impedance is present in any telecommunications system, a portion of the transmitted signal is reflected to the transmitter as an echo, which represents an impairment that degrades the system quality (Messershmitt). In most telecommunications systems, such as a telephone circuit, the echo is generated when the long distant portion consisting of two one-directional channel (four wires) is

Table 1. Main audio and speech signal processing applications

- Echo cancellation in telecommunication systems
- Acoustic echo cancellation
- Noise canceling
- Active noise cancellation
- Adaptive equalization
- Narrowband speech coding
- Broadband audio and speech coding

connected with a bidirectional channel (two wires) by means of a hybrid transformer. If the hybrid impedance is perfectly balanced, the two one-directional channels are uncoupled and no signal returns to the transmitter side (Messershmitt). However, in general the bridge is not perfectly balanced because the required impedance to properly balance the hybrid depends on the overall impedance network. In this situation, part of the signal is reflected, producing an echo. To avoid this problem an adaptive filter is used to generate an echo replica that is then subtracted from the signal to be transmitted. Subsequently the adaptive filter coefficients are updated to minimize, usually, the mean square value of the residual echo (Madisetti & Williams, 1998). To obtain an appropriate operation, the echo canceller impulse response must be larger than the longer echo path to be estimated. Thus assuming a sampling frequency of 8kHz and an echo delay of about 60ms, an echo canceller with 256 or more taps is required (Haykin, 1991). Besides the echo path estimation, another important problem is how to handle the double talk, that is, the simultaneous presence of the echo and the near speech signal (Messershmitt).

Acoustic Echo Cancellation

A critical problem affecting speech communication in teleconferencing system is the acoustic echo. When a bi-directional line links two rooms, the acoustic coupling between loudspeaker and microphones in each room causes an acoustic echo perceivable to the users in the other room. The best way to handle it appears to be the adaptive echo cancellation. An acoustic echo canceller generates an echo replica and subtracts it from the signal picked up by the microphones. The residual echo is then used to update the filter coefficients such that the mean square value of approximation error is kept to a minimum (Perez- Meana et al., 2002).

Adaptive Noise Cancellation

The adaptive noise canceller is a generalization of the echo canceller in which a signal corrupted with additive noise must be restored or enhanced. When a reference

Table 2. Digital speech coding standards

Rate Kb/s	Application	Type of Coder	Year
64	Public Switched Telephone Network	Pulse Code Modulation (PCM)	1972
2.4	U.S. Government Federal Standard	Linear Predictive Coding	1977
32	Public Switched Telephone Network	Adaptive Differential PCM	1984
9.6	Skyphone	Multi-Pulse Linear Predictive Coding (MPLPC)	1990
13	Pan-European Digital Mobile Radio (DMR) Cellular System (GSM)	Regular Pulse Excitation Linear Prediction Coding (RPE-LPC)	1991
4.8	U.S. Government Federal Standard	Codebook Excited Linear Prediction Coding (CELP).	1991
16	Public Switched Telephone Network	Low Delay CELP (LD-CELP)	1992
6.7	Japanese Digital Mobile Radio (DMR)	Vector Sum Excited Linear Prediction Coding (VSELP)	1977

signal correlated with the noise signal but uncorrelated with the desired one is available, the noise cancellation can be achieved by using an adaptive filter to minimize the total power of the output of the difference between the corrupted signal and the estimated noise, such that the resulting signal becomes the best estimate, in the mean square sense, of the desired signal. This system works fairly well when the reference and the desired signal are uncorrelated among them. However, appropriate reference signals are not always available. To solve this problem several noise canceling algorithms have been proposed that are resistant to crosstalk situations. A different approach, developed by Dolby Laboratories, is used in the Dolby noise reduction systems in which the dynamic range of the sound is reduced during recording and expanded during the playback (Davis, 2002). Several types of Dolby noise reduction systems have been developed, including the A, B, C, and HXpro. Most widely used is the Dolby B, which allows acceptable playback even on devices without noise reduction. The Dolby B noise reduction system uses a preemphasis that allows masking the background hiss of a tape with a stronger audio signal, especially at higher frequencies. This effect is called psychoacoustic masking (Davis, 2002).

Active Noise Cancellation

A related problem to noise cancellation is the active noise cancellation that is intended to reduce the noise produced in automotive equipment, home appliances, industrial equipment, airplane cabins, and so forth. Active noise is achieved by introducing a canceling antinoise wave through an appropriate array of secondary sources, which are interconnected through an electronic system using adaptive systems with a particular cancellation configuration. Here, the adaptive filter generates an antinoise that is acoustically subtracted from the incoming noise wave. The resulting wave is captured by an error microphone and used to update the adaptive filter coefficients, such that the total error power is minimized. Some successful applications of this technology are the earphone, electronic mufflers, noise canceling in airplane cabins, and so forth (Davis, 2002; Kuo & Morgan, 1996).

Adaptive Equalization

Digital information transmitted through physical communication channels is often distorted due to the intersymbol interference (ISI), which is mainly caused by multipath propagation of the transmitted symbols and by the no ideal characteristics of the communication channels (Proakis, 1985). To overcome this problem, several methods have been reported in the literature such as transversal equalizers (TE), decision feedback equalizers (DFE), maximum likelihood sequence estimation (MLSE), and so forth (Proakis, 1985). Most of them perform fairly well with near stationary communication channels. However, they still present difficulties with mobile communication channels that fluctuate markedly with the motion of a vehicle, ground irregularities and changing environments, because in these situations the communication channels can be assumed to be stationary only within fractions of a wavelength, that is, over 4-5m in the 900-MHz band (Madisetti & Williams, 1998).

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