THE CHALLENGE FOR TELECOM INDUSTRY IN THE ACOUSTIC AND NETWORK ECHO CANCELLER

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1. INTRODUCTION

The echo cancellation has been a major challenge for the voice quality in telephone conversation since the phone was invented 100 years ago by Alexander Graham Bell. The real progress was made about 50 years ago when the LMS adaptive algorithm was introduced. At first, the algorithm is thought too expensive and complicate to be implemented. In the last few decades, the semiconductor industrial experienced a dramatic technology revolution. The processor speed doubles and memory size halves every 18 months. Now, the adaptive algorithm can be easily implemented in silicon and telephone voice quality improved dramatically. In the mean time, the telecom industrial raises the standard bar again. About 20 years ago, we were starting to talk about full-duplex conversation.

In general, the adaptive algorithm is a linear echo cancellation algorithm; it can model a linear echo path and cancel the echo if both echo and far-end signals are independent Gaussian white signals. However, the echo path is not a linear time-invariant system and speech is strongly correlated signal. The residual echo can still be heard after adaptive echo canceller. How to get rid of the remaining echo while maintain the far end speech and background noise (or music) untouched is still a challenge problem. Today, IP phone is widely used and the conversation delay increases with IP network. This means that a much smaller residual is required for the same speech quality because the human tolerant to the speech distortion is inversely related to the delay between speakers.

On another front, wireless mobile phone is also widely used. In many countries, it becomes mandatory requirement that the phone in the car has to be handsfree. We all know that the loud background noise in the car makes conversation and echo canceller very difficult. These are all challenges we are going to face in today's echo canceller.

With Zarlink Semiconductor TruePlex technology, we are designing echo cancellation chip to give the best speech quality under server environment and with minimum cost.

2. Zarlink Truplex Technology

Zarlink Trueplex Echo canceller technology is shown in Fig.1. Trueplex is patented technology provide a real (True) Full-duplex conversation with both network and acoustic echo cancellation.

The major function blocks are as follows:

Linear adaptive filter is the main filter which emulates the linear echo path to cancel the majority of echo in the received path. This filter can be either LMS based or RLS based linear adaptive filter.

Non-linear echo canceller is to cancel some non-linear components in the echo so that the residual can be further reduced by 10dB+. This will emulate the non-linear behavior of the echo with a non-linear gain being added for different signal level. This can also being done in frequency domain with sub-band decomposition.

Noise Reduction (NR) is to reduce background noise to make speech more understandable. This is especially important in the mobile phone where noise power is much stronger than speech. The common used noise reduction method is spectral subtraction. In the spectral subtraction method, the signal is first being decomposed into different sub-band, the noise power is measured in each sub-band and then subtracted from the signal spectrum.

Equalizer (EQ) is to adjust low end and high end frequency response and also to compensate resonance. In general, the speaker driver is highly non-linear with signal level at some frequency. The Equalizer will identify these frequency and put appropriate gain for different frequency band.

Nonlinear Processor (NLP) is to further reduce echo residual and also maintain far end speech undistorted and background noise uninterrupted. NLP is the last stage of echo canceller and any small amount of echo has to be eliminated and in the mean time, make an intelligent decision which part of speech is echo and which part is incoming speech. The remaining echo residual is very small but also very difficult to cut out without interrupt the incoming speech signal.

Automatic Level Control (ALC) is to automatic adjust speech to a comfortable level no matter the speaker speaks load or soft, or is far away or close to microphone. At low SNR cases (SNR=-20dB), the ALC has to make sure the microphone is not saturated with sudden burst noise such as when window is opened while car runs on high way.
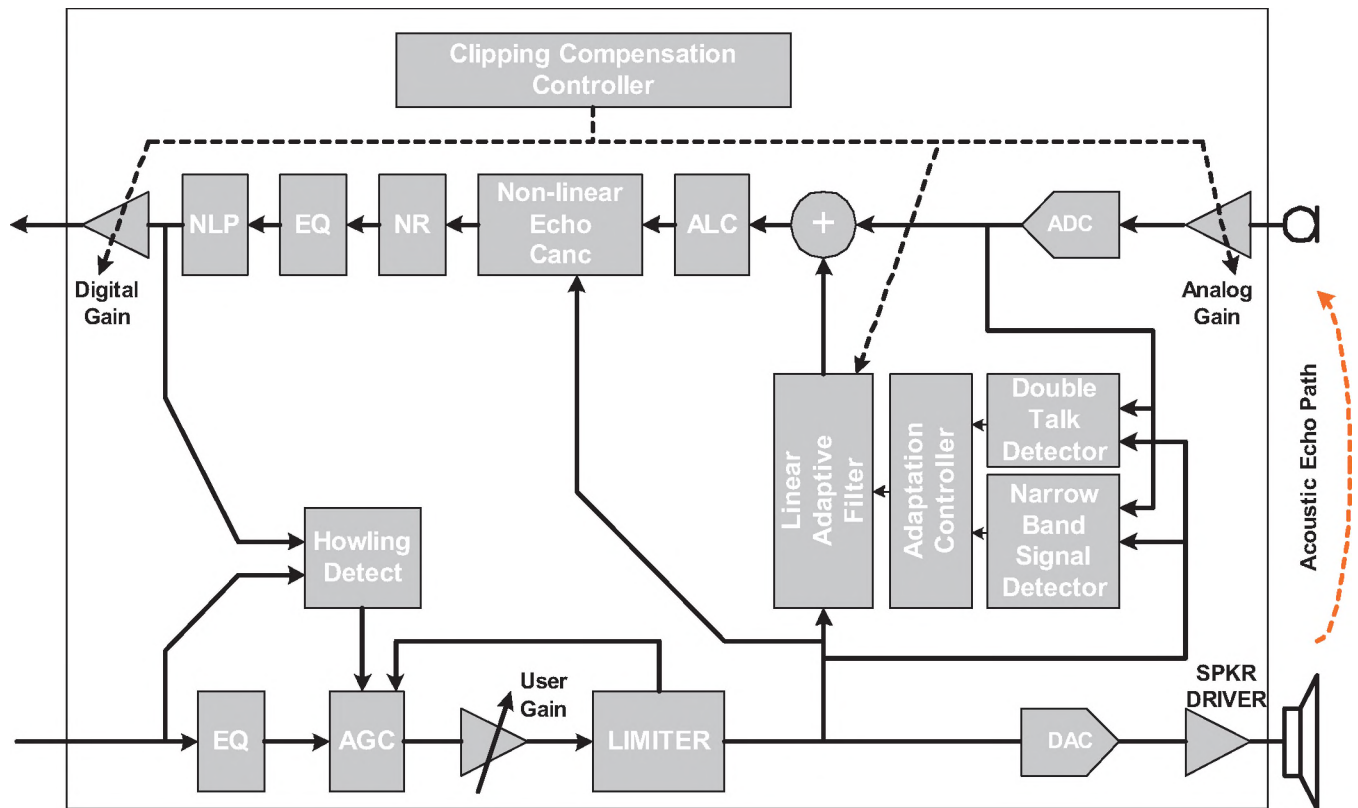
Fig.1 Zarlink Trueplex Technology

Narrow Band Signal Detection is another function block controlling filter adaptation to avoid filter divergence when narrow band tones are sent on either end of communication channels.

Clipping Compensation exchange codec gain with a digital gain so that the microphone will not clip under server condition such as when the car window is opened on the high way during the conversation. It also maintains the original speech level when the background noise is cancelled and environment changes.

Automatic Gain Control (AGC) automatically adjusts the speaker volume for the best conversation and echo cancellation.

Output Limiter emulates DAC clipping so that the adaptive filter still see a linear echo path even the speaker clips.

Anti-Howling controls the loop stability during initial convergence or when echo path changes. It includes two parts: howling detector and howling gain control. When the loop stabilized, the howling gain should be release to 0dB.

Double Talk Detector is to detect whether far end speaker is talking or not. The purpose is to stop adaptation (or control its step size) during double talk to avoid filter divergence. It also controls NLP so that the non-linear processor wouldn't cut a lot of far end speech.

All these function blocks make our chip to fully cancel the echo without affecting the incoming speech quality under server condition such as lower than –20SNR and enhanced echo return of 10dB.

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