DETECTION OF SHORT DISTANCE WIRELESS TRANSMITTED AUDIO

Rogerio G. Alves and Kuan-Chieh Yen

CSR - Cambridge Silicon Radio
Auburn Hills, MI, USA
rogerio.alves@csr.com, kuan.yen@csr.com

ABSTRACT

With the advance of short distance wireless technologies, it becomes popular to include FM transmission capability to mobile devices such as hands-free car kits, cellular phones with hands-free features, and MP3 players so that in addition to playing audio signal through the on-device loudspeaker, they can relay audio signal to other audio system to utilized their superior audio quality. Traditionally, the switch between the on-device loudspeaker and the FM transmitter is done manually. This paper proposes an automatic and switching algorithm which leverages the AEC capability already exists in many of these mobile devices. The algorithm detects the presence of audio through FM channel and switches off the on-device loudspeaker, which adds to user convenience and power saving efficiency.

Index Terms— Hands-free communication, Acoustic echo cancellation.

1. INTRODUCTION

The popularity of hands-free mobile devices, such as Bluetooth hands-free kits (HFK) and cellular phones with hands-free capability, increases significantly in recent years. Generally, these devices have their own loudspeaker, and considering that one of the most common places to use these devices is in automobiles, an attractive design is to include a FM transmitter to transmit the incoming call to the car radio and utilize its superior audio quality when available. This design is also desirable for the increasingly popular MP3 players (in stand-alone forms or included with cellular phones). Based on these applications, new single-chip solutions with multiple wireless technologies have been developed, such as CSR's BC7 family (www.csr.com), which contains Bluetooth, FM transmitter, FM receiver and GPS capability as well as a digital signal processor (DSP) in a single piece of silicon.

Because power consumption is a major concern for hand-free mobile devices, it is desirable to turn off the loudspeaker on the mobile devices to conserve power when the audio signal is transmitted through FM connection and played through car audio system. Although this can be done manually by users, it would be more efficient and convenient if achieved automatically. On the other hand, hands-free mobile devices must have acoustic echo cancellation (AEC) capability in order to provide full-duplex communication [1]. Embedded in the AEC algorithm is the acoustic path information about the audio transmission from which we can detect whether the FM transmitted signal is presented through the loudspeakers of the car audio system. The purpose of this paper is to leverage the existing AEC algorithm to develop an automatic control mechanism which detects the existence of the FM transmitted audio path and switches on or off the loudspeaker on the hands-free mobile devices accordingly.

The remainder of this paper is organized as follows: Section 2 describes the problem in more details. Section 3 presents the basics of AEC. In Section 4 the proposed idea is described. Section 5 presents the simulation results. And finally, Section 6 shows the conclusion for this work.

2. PROBLEM DESCRIPTION

With the development of the Bluetooth technology, a short distance personal wireless communication network with a universal protocol becomes a reality. The Bluetooth technology has been widely used in hands-free kits (HFK) that free up user's hands for other tasks during cellular phone calls. Due to increasing driving safety regulations that ban the usage of cellular phones while driving, using a Bluetooth HFK becomes necessary in some occasions. One of the interesting HFK designs is a hands-free Bluetooth device that communicates with the cellular phone via Bluetooth link and can play the incoming audio signal in either its own loudspeaker (L1) or in the loudspeakers of the car audio system (L2) via a FM connection. This system is illustrated in the block diagram shown in Fig. 1.

In Fig. 1 the switch S decides whether to relay the received signal to the loudspeakers L1 or L2. Technically, this switch has to be operated manually by the device user, unless the existence of the received signal in L2 can be detected. An intuitive solution to this is to turn off L1 at initialization and transmit the audio signal through the FM transmitter. If the car radio is on and tuned to the proper FM station, the audio signal will be captured by the microphone of the device, thus L1 will stay off. Otherwise, L1 will be turned on.

Although this method is effective, it has some practical issues: First of all, if the car radio is not tuned to the proper FM station initially, the beginning of the phone call will be lost and thus cause inconvenience to the users. Furthermore, once L1 is switched on, there is no mechanism to switch it off even if the car radio is subsequently tuned to the proper FM station. Adding an added audio component to the FM transmitted audio can mitigate these problems, but it is intrusive and can be annoying to the users.
To address the described problems, a novel solution is proposed in this work.

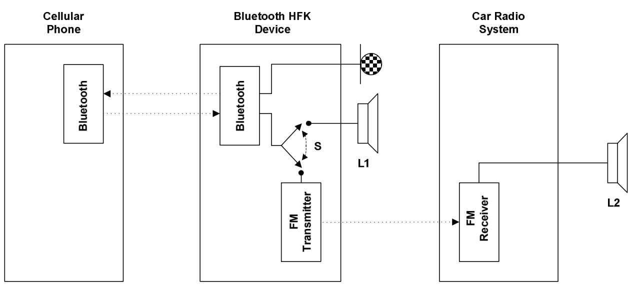

Figure 1 – Block diagram of a short distance wireless communication system.

3. ACOUSTIC ECHO CANCELLATION BASICS

The AEC problem is a well known problem, and it can be described as shown in Fig. 2, where the receive signal (*x(n)*), is sent to a loudspeaker inside an acoustic environment (e.g., car interior), this signal propagates through the acoustic path *q(n)*, and reaches the microphone to generate the echo signal *c(n)*. To cancel the echo signal, an adaptive filter is used, where the objective is to identify the acoustic path *q(n)* using an adaptive filter (*g(n)*), and then subtract the resultant signal (*y(n)*) from the microphone signal. It can be easily observed that if (*g(n)=q(n)*) then (*y(n)=c(n)*), and thus subtracting the output signal of the adaptive filter from the microphone signal will cancel the echo signal.
The adaptive filtering for AEC is well studied in the literature and different types of adaptive filter algorithm can be used such as LMS (Least Mean Square Algorithm), NLMS (Normalized Least Mean Square Algorithm), DRNLMS (Data Reuse Normalized Least Mean Square Algorithm), RLS (Recursive Least Square Algorithm), APA (Affine Projection Algorithm) and others [2, 3], where each algorithm has different trade-offs. Each of these adaptive algorithms can be implemented in time, frequency, or other transform domain with different trade-offs in time-frequency resolutions [2, 3]. This generates a considerably number of choices that system designers need to make. Observe that it is only considering the adaptive filter block (*g(n)*) shown in Fig. 2.
Practical implementation issues further complicate the design options. For example, adaptive filtering algorithms need control mechanisms to prevent them from divergence when receive signal and other send-side signals (user speech, noise, etc.) are present at the same time. Due to it, a double talk detection (DTD) based control is often introduced in AEC systems. Different approaches on DTD control have been considered in research papers [1].

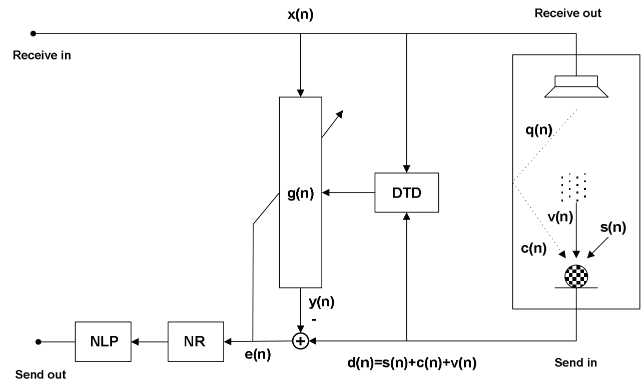

Figure 2 – Acoustic echo cancellation and the hands-free system.

An important quality required for AEC in HFK systems is the capacity to perform well in presence of noise signal (*v(n)* in Fig. 2). A noise reduction algorithm (NR) is often introduced to the system to extract the environmental noise. The information from the NR algorithms can be used by the AEC filter to control its adaptation in order to improve its robustness in noisy conditions. The NR problem by itself is also a challenge application, and approaches based on spectral subtraction, Kalman filter, neural network and others are proposed in the literature [4, 5].
Due to the nonlinearity existent in the system, such as the one from the loudspeakers, and the misadjustment of the adaptive filtering algorithm due to practical issues associated with handling time-varying systems, nonlinear processors (NLP) are often introduced to the complete the AEC function in HFK systems, as shown in Fig. 2. Once again different approaches can be used in the desired implementation [1, 6].
Observing the AEC-based HFK system described in Fig.2, we can say that the initial problem that apparently was a not so difficult task, becomes a complex problem for designers, due to the fact that knowledge in different fields is necessary to design the complete system. It is further complicated by a great number of possible solutions provided in the literature. With it in mind we can say that designing a complete HFK system with low computational complexity, able to work in real time, robust in diverse environmental noise conditions, and providing good performance in various practical environment still is a difficult problem despite intensive research efforts.

4. ACOUSTIC PATH DETECTION AND SWITCHING

Considering the detection problem previously described, the proposed solution is illustrated by the block diagram shown in Fig. 3.

The main idea of the proposed solution is to delay the signal transmitted through the FM transmitter (*x(n)*) by a known period of time (*Delay* ms) which should be greater than the effective tail length of the acoustic path (*q1(n)*) from the mobile device loudspeaker (L1) to the microphone. Because the adaptive filter *g1(n)* is designed to cover only the effective tail length of *c1(n)*, if an echo signal (*c2(n)*), is present at the residual signal *e1(n)*, it has to be due to the presence of the receive signal, *x(n-Delay)*, at the car audio system loudspeaker (L2). This means that the radio is on and tuned to the proper FM station. After the detection, the mobile device loudspeaker (L1) can be turned off through a switch control and the adaptation and application of *g1(n)* can be stopped.

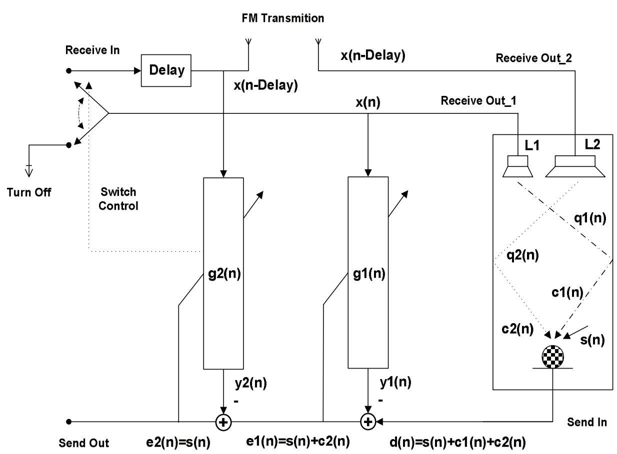

Figure 3 – Block diagram of the proposed solution.

The presence of acoustic echo *c2(n)*, can be determined by the power measurement on the coefficients of the adaptive filter *g2(n)*, as shown in the experimental results section.

Once the presence of acoustic echo *c2(n)* is detected and L1 is turned off, the delay of *Delay* ms should be removed from the system, so that the roundtrip delay of the telephone call is kept at minimum.

After the device loudspeaker (L1) is switched off, the filter coefficient power measurement of *g2(n)* can still be monitored to detect if the user change the radio station, turn off the radio, or in case of external interference on the FM transmission. In this case, the system can return to the initial condition. When this is detected, switch on loudspeaker L1, reintroduce the *Delay* ms delay in the wireless path, and perform the two AEC algorithms to enter the initial detection mode.

Also, in order to reduce the computational complexity of the system, a less complex AEC algorithm can be implemented as *g2(n)* before the detection of *c2(n)*. Once *c2(n)* is detected, a "complete" version of the AEC algorithm can be re-introduced.

Observe that the basic idea proposed here is to detect the presence of the acoustic path associated with the FM radio transmission (*q2(n)*) by exploring the temporal difference between the two acoustic paths (*q1(n)* and *q2(n)*). Alternative implementations are possible to achieve this goal. For example, the same error signal *e2(n)* can be used to drive the adaptation of both adaptive filters *g1(n)* and *g2(n)*. In this case, the two adaptive filters are effectively cascaded into one adaptive filter and each occupies a different time segment of this combined filter, as shown in the block diagram illustrated by Fig. 4.

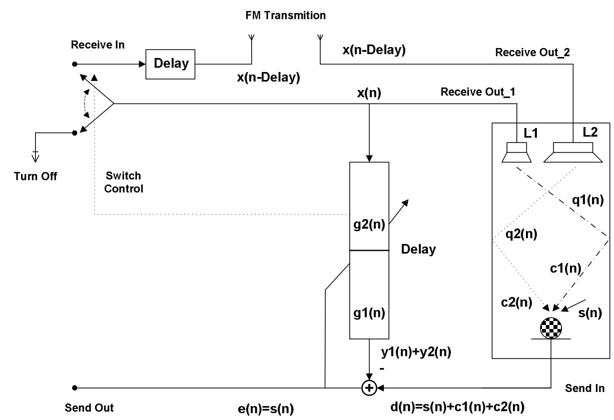

Figure 4 – Block diagram of an alternative algorithm.

Alternatively, the time delay can be inserted into the receive signal to the mobile device loudspeaker instead of the one to the FM transmitter.

In the situations that prohibit inserting delay to the two receive paths (L1 loudspeaker and FM transmitter) independently, it is still possible to apply the invention. In this case, a careful examination of the acoustic environment, the mobile device, and the car audio system can explore the underlying temporal difference between the two acoustic paths. Creative design and physical deployment of the HFK can also help maximize the temporal difference between the two acoustic paths.

5. EXPERIMENTAL RESULTS

The algorithm used to implement the AEC system is described in [6], where a subband DR-NLMS was used with a DFT filterbank of 65 bands (FFT size=128). The audio signal is sampled at 8 kHz, the prototype filter of the DFT filterbank has 256 coefficients [7], the decimation factor is 64, and the data-reusing factor is 2.

The system being analyzed is the one shown in Fig. 4, where the *Delay* used was 64 ms. The FM transmission delay was considered null. The impulse responses *q1(n)* and *q2(n)* have tail length of 16 ms and 32 ms, respectively. The concatenated impulse response is shown in Fig. 5.

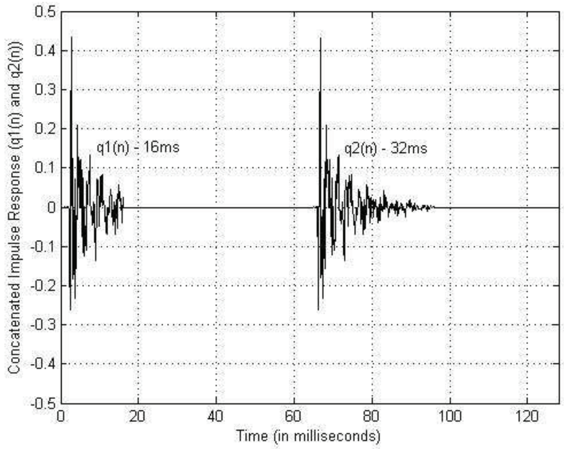

Figure 5 – Concatenated impulse response.

The measurement used to perform the detection was the power of the coefficients of adaptive filters *q2(n)* in frequency domain. Assuming 10 seconds of echo and the signal to noise ratio (SNR) at microphone of 25 dB, the estimated gain (power of the coefficients) of filters *q1(n)* and *q2(n)* in dB are presented in Fig. 6. To show the accuracy of the filter gain estimation, the volume of L2 was set at four different levels relative to that of L1: L2 off, L2 10 dB quieter, equal loudness, and L2 10 dB louder. For L1 the loudness/gain was held constant. Observe that the detection of signal in L2 can be easily done by applying a threshold in the power of the coefficients of *q2(n)*, due to the fact that the curve represented by L2 OFF in Fig. 6 is much lower than the other three curves.

To analyze the robustness of the proposed algorithm in noise condition, the SNR in the microphone was decreased to 5 dB by adding road noise from noisex92 database. The result is shown in Fig. 7. Observe that the main difference between Fig. 6 and Fig. 7 is the "noise floor" of the curve L2 OFF, which is expected.

Observe also that, despite the simulations presented use 10 seconds of echo duration, a visible difference is observed in the curves of the power of the coefficients of *q2(n)* since the beginning of the measurement (for both experiments), which means that a decision of the detection can be done with 2 to 3 seconds of audio signal.

6. CONCLUSION

This paper presents a detection algorithm for short distance wireless transmitted audio signal that can be applied to hands-free devices for power saving. The algorithm leverages the AEC algorithm assuming that it is already present in the target devices. The simulations results show that the proposed algorithm is robust to changes in volume setting of the car radio and the level of environmental noise.

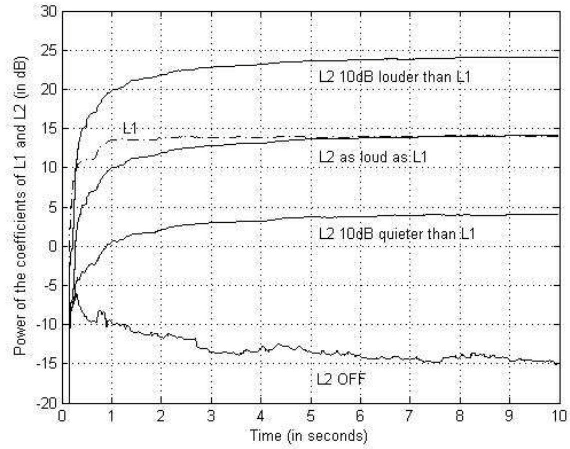

Figure 6 – Power of the coefficients of *q1(n)* and *q2(n)* in high SNR condition.

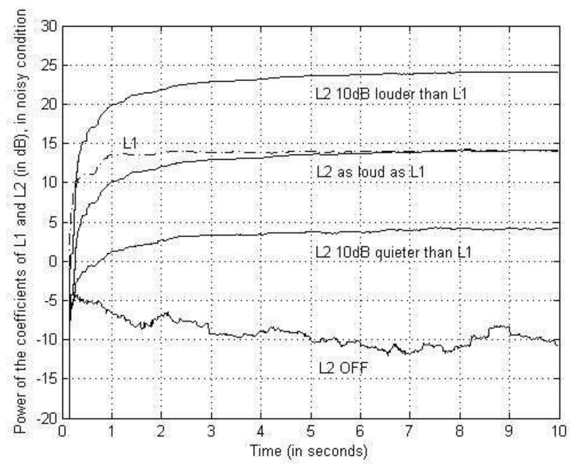

Figure 7 – Power of the coefficients of *q1(n)* and *q2(n)* in low SNR condition.

7. REFERENCES

[1] E. Hänsler and G. Schmidt, *Acoustic Echo and Noise Control: A Practical Approach*, Hoboken, NJ: Wiley, 2004.

[2] S. Haykin, *Adaptive Filter Theory*, 4th edition, Englewood Cliffs, NJ: Prentice Hall, 2002.

[3] P.S.R. Diniz, *Adaptive Filtering: Algorithms and Practical Implementation,* 2nd edition, Dordrecht, The Netherlands: Kluwer Academic Publishers, 2002.

[4] P. Vary and R. Martin, *Digital Speech Transmission: Enhancement, Coding and Error Concealment,* John Wiley & Sons, England, 2006.

[5] P.C. Loizou, *Speech Enhancement: Theory and Practice,* 2nd edition, Boca Raton, FL: Taylor & Francis, 2007.

[6] R.G. Alves, and K.C. Yen, "Echo Cancellation and Noise Reduction Method," U.S. Patent 7,433,463, October 7, 2008.

[7] M.R. Petraglia, P.R.V. Piber, "Filterbank Design for Oversampled Subband Adaptive Filtering Structures," *Electronics Letters*, vol. 39, issue 23, pp. 1685-1687, Nov. 2003.