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ON THE USE OF DIFFERENTIAL MICROPHONE ARRAY FOR ECHO CANCELLATION

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Abstract

This paper describes a beamforming technique, the differential microphone array (DMA), used to perform echo attenuation on communication devices such as mobile phones or tablets. Our implementation of DMA uses two microphones and consists in delaying a microphone signal. Considering tablets or mobile devices where the position of the loudspeaker in respect to that of the microphones is known, we propose to use DMA to create a null in the direction of the loudspeaker. Our experiments are based on signals recorded with a tablet device and show that DMA is efficient in attenuating echo even in double-talk periods where the useful speech is not distorted by DMA. By using DMA as pre-processing to an echo postfilter, we observe that use of DMA permits to significantly reduce the amount of distortions introduced by the postfilter during double-talk periods.

1 Introduction

In voice communications, speech quality is degraded by several elements among which ambient noise and acoustic echo. During a voice call, the microphone(s) of the device will capture the voice of the near-end speaker (useful signal) as well as the ambient noise (airport noise, car noise etc…). In addition part of the speech from the far-end speaker played by the loudspeaker of the device can be captured by the microphone(s). If no processing is done, the far-end speaker will hear the voice of the near-end speaker together with a delayed version of his voice (acoustic echo) and ambient noise [1].

To guarantee good speech quality, most of communication devices embed speech enhancement algorithms with the position of the transducers being optimized for some of these algorithms:

- On mobile devices the loudspeaker is often placed further away from the microphone such as to minimize the level of the echo picked by the microphone. This is useful for echo cancellation.
- Tablets and laptops are often equipped with two or more microphones which are used for ambient noise reduction based on beamforming approaches.

In this paper, we use two microphones for echo cancellation through beamforming based on differential microphone array (DMA) [2]. The remainder of this document contains a brief description of DMA and some experiments in the context of echo cancellation for a tablet.

2 Differential microphone array

The block diagram of DMA is depicted in Figure 1. Assuming all sound sources to be far-field and denoting θ the direction of arrival of a given sound, the microphone signal $X_2(n)$ is a delayed version of the microphone signal $X_1(n)$. The time delay between $X_1(n)$ and $X_2(n)$ is equal to $d\cos(\theta)/c$ where d is the distance between the microphones and c is the speed of sound in the air.

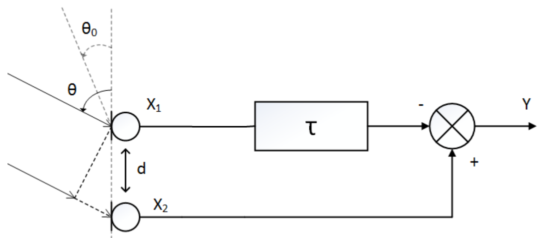

Figure 1 : Block diagram of a differential microphone array with 2 microphones

As showed in Figure 1, DMA consists in delaying and subtracting $X_1(n)$ from $X_2(n)$ to obtain $Y(n)$

$$Y(n) = X_2(n) - X_1(n-\tau) \quad \text{Eq. 1}$$

where τ is a delay in samples. Changing the value of τ permits to control the shape of beam pattern

[2] and the direction of the null. We express τ as follows

$$\tau = -\frac{d\cos(\theta_0)}{c} \qquad \text{Eq. 2}$$

where θ_0 is the direction for which the beam pattern is null. Note that Eq. 2 is valid for values of θ_0 comprised between π/2 and π. In this paper, we exploit the null of the DMA beam pattern for the purpose of echo cancellation by setting θ_0 to match with the direction of the loudspeaker. For given device, our proposal to achieve echo cancellation using DMA consists in defining the position of the loudspeaker in respect to the microphones.

One advantage of DMA is that its beam pattern is independent of the frequency. But this independency is valid as far as spatial aliasing does not occur. If the distance d is greater than half the wavelength, spatial aliasing occurs [2]. For example, if the sampling rate is 16 kHz, spatial aliasing starts from 1.9 kHz. To avoid dealing with unpredictable effects due to spatial aliasing, DMA is not applied above this frequency.

3 Assessment

We assess the usage of DMA for echo cancellation using signals recorded on a tablet device. Recordings have been performed in an anechoic chamber. Our tablet is equipped with two microphones spaced of 9 cm and the loudspeaker is located at $\theta_0 = \pi/2$. The near end speech is generated by an artificial mouth placed at 35 cm in front of the tablet (Skype Lync test setup [3]). Microphone signals contains echo-only, near-end only and double-talk periods. The efficiency of our approach is measured in terms of ERLE (Echo Return Loss Enhancement). ERLE measures the amount of echo attenuation achieved by a processing. DMA is assessed in two steps: first we assess the performance of DMA standalone; secondly we assess its performance when used as pre-processor to an echo postfilter [4].

Our measurements show that DMA can achieve up to 4 dB of echo attenuation during echo-only periods. Figure 2 shows the microphone signal before and after DMA during a double-talk period. In this figure, only the frequency region in which DMA is applied is shown. Figure 2 shows that the echo components are attenuated. In contrast to traditional echo cancellation approaches that can achieve up to 20 dB (at the expense of poor performances during double-talk periods), DMA is able to attenuate the echo even during double-talk periods without distorting useful signal.

Figure 3 shows a comparison of an echo postfilter when used standalone and when used in combination with DMA. This figure corresponds to a double-talk segment. The echo post-filtering (top) performs a good attenuation of the echo, but when used in combination with the beamforming (bottom) the near-end speech signal is less distorted. Informal listening tests show that the improvement brought by the beamforming is audible.

In the full version of this paper, we will provide further details about our implementation of DMA. We will also provide more details about our experimental assessment both using the tablet (as reported in this abstract) and using some additional recording performed using optimal transducers arrangement (smaller distance between the microphones).

Future investigations of the work presented in this paper include studying the influence or interest of using more microphones (3 or 4) in order to achieve more echo attenuation with DMA.

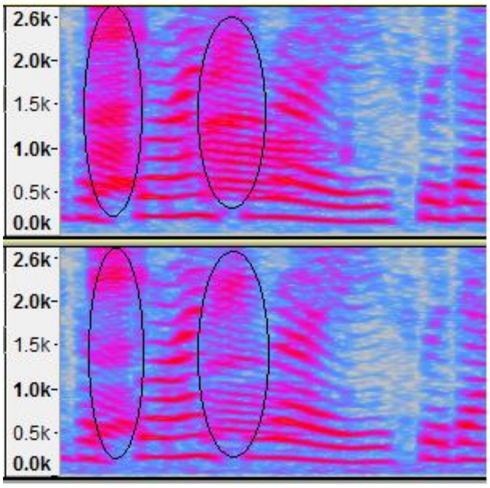

Figure 2 : Double-talk segment before DMA (top) and after DMA (bottom).

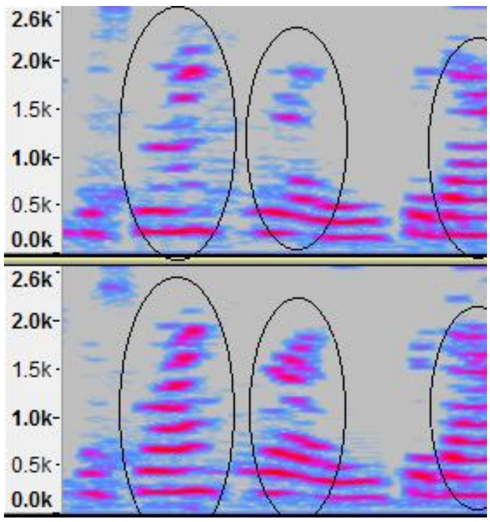

Figure 3 : Double-talk segment processed by the echo postfilter standalone (top) and double-talk segment processed by DMA and echo-postfilter (bottom). The circles show the near-end speech component.

References

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About the Authors

Raphael Cathelain is currently completing his last year of M.Sc. student in electrical engineering at Grenoble Institute of Technology. During his previous experience, he worked on speech transformation and reverberation effects. He currently works on speech enhancement algorithms at Intel Mobile Communications to complete his specialization in signal processing.

Christelle Yemdji received her engineering diploma from ISEN Lille in 2003. She received her M.Sc. and PhD in signal processing from Rennes University (2009) and Telecom ParisTech (2013) respectively. In 2008, she worked in Orange Labs on acoustic echo cancellation. She carried her PhD research studies while been part of EURECOM. She worked as visiting scientific at IND RWTH institute in winter 2012. Since 2013, she joined Intel Mobile Communications as audio research engineer.