ANALYTICAL EVALUATION OF A SELF-CALIBRATING MICROPHONE ARRAY.

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ABSTRACT

This paper gives an analytical description of an adaptive microphone array which facilitates a simple built-in calibration to the environment and instrumentation. The scheme offers several advantages, such as a simple calibration procedure and reduced target signal distortion. The analysis employs noncausal Wiener filters yielding compact and effective theoretical suppression limits.

1. INTRODUCTION

Increased use of mobile telephones in cars has created a greater demand for hands-free in-car installations. The filtering required to obtain a similar sound quality as for hand held telephony must suppress the loudspeaker, as well as background noise and room reverberation, without causing severe speech distortion. This paper analyses a proposed calibrated adaptive microphone array as a means of addressing this problem [1].

Three major tasks must be addressed in order to improve the quality of hands-free mobile telephones: noise suppression, blind equalization (i.e., room reverberation suppression) and acoustic feedback suppression of the loudspeaker.

Some of the existing methods of increasing the signal-to-noise ratio in hands-free mobile telephones are spectral subtraction [4, 5], temporal filtering, noise cancellation and a variety of different array techniques [6]. Room reverberation is most effectively handled with array techniques or with microphone design and placement. Acoustic feedback for hands-free mobile telephony is usually addressed by conventional echo cancellation techniques [2]

The main idea behind the proposed approach is to employ calibration signals recorded on site [1, 3]. The microphone elements and their placement can be chosen arbitrarily. The proposed algorithm indirectly calibrates the array for the speaker's location, microphone positions and lobe gains, amplifiers and for the acoustic environment in the car, see Fig. 1. In order to keep discrepancies between the total transfer function during training and operation small and under control, the proposed beamformer manipulates the training signals. The calibrated adaptive microphone array analysed in this paper addresses all the given tasks in a single, concise concept.

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2. ANALYTICAL SIGNAL MODEL

The simple signal model is general in the sense that microphone elements and sources can be placed arbitrarily with any spectral content. The M different point signal sources $s_m(t)$, $m = 1...M$ with spectral densities $R_{s_m s_m}(\omega)$ are assumed to be mutually uncorrelated, i.e. the cross power spectral density $R_{s_l s_m}(\omega)$ is zero if $l \neq m$. All sources impinge on an array of N microphone elements, each corrupted with mutually uncorrelated noise $n_l(t)$. The transfer function between source no. m and an array element no. n is denoted $G_{m,n}(\omega)$ and is either measured, or modeled. In the model, a spherical source in a free field and homogeneous medium has been assumed.

Using this signal model(described in Fig.6) it is straight forward to find expressions for the infinite length Wiener filters:

$$\mathbf{W}_{opt}(\Omega) = \frac{R_{s_1}(\Omega)F(\Omega)\mathbf{G}_{s_1}^H(\Omega)\mathbf{R}_{\nu\nu}^{-1}(\Omega)}{R_{s_1}(\Omega)\mathbf{G}_{s_1}^H(\Omega)\mathbf{R}_{\nu\nu}^{-1}(\Omega)\mathbf{G}_{s_1}(\Omega)+1}, \quad (1)$$

where $\mathbf{G}_{s_1}(\Omega)$ denotes a column vector with dimension N containing all the corresponding digitised signal frequency transfer functions from a source m to all elements. The matrix $\mathbf{R}_{\nu\nu}$ consists of all the undesired signals, including measurement noise. The corresponding expression for the error power spectral density is

$$R_{\varepsilon\varepsilon,opt}(\Omega) = \frac{R_{s_1}(\Omega)|F(\Omega)|^2}{R_{s_1}(\Omega)\mathbf{G}_{s_1}^H(\Omega)\mathbf{R}_{\nu\nu}{}^{-1}(\Omega)\mathbf{G}_{s_1}(\Omega)+1}. \quad (2)$$

The array output signal $y(k)$ is closely related to the desired signal $y_r(k)$, i.e. $F(\Omega)$.

Once the optimum filters $\mathbf{W}_{opt}(\Omega)$ are found for a given situation, we can investigate the total transfer functions from any spatial point. In particular, the transfer function from each source to the output of the array can be determined. The total transfer function $H_m(\Omega)$ expresses how each signal is affected by the optimum beamformer and is given by

$$H_m(\Omega) = \mathbf{W}_{opt}(\Omega)\mathbf{G}_{s_m}(\Omega) = \frac{F(\Omega)R_{s_1}(\Omega)\mathbf{G}_{s_1}^H(\Omega)\mathbf{R}_{\nu\nu}^{-1}(\Omega)\mathbf{G}_{s_m}(\Omega)}{R_{s_1}(\Omega)\mathbf{G}_{s_1}^H(\Omega)\mathbf{R}_{\nu\nu}^{-1}(\Omega)\mathbf{G}_{s_1}(\Omega)+1}. \quad (3)$$

These expressions are used to define a target distortion ratio and jammer suppression ratio.

3. RESULTS

The transfer function $H_1(\Omega)$, i.e. $H_m(\Omega)$ from the target direction, can be used to calculate the Target signal Distortion Ratio, defined as

$$TDR = \frac{\frac{1}{2\pi}\int_{-\pi}^{\pi} R_{s_1}(\Omega)|H_1(\Omega)-F(\Omega)|^2 d\Omega}{\frac{1}{2\pi}\int_{-\pi}^{\pi} R_{s_1}(\Omega)d\Omega}. \quad (4)$$

The distortion is strongly dependent on the amplitude of the calibration signal and the jamming signal, see Figs. 2 and 3, showing TDR versus the amplifying parameter α, for different jamming directions and TDR versus the β i.e. the size of the jamming signal. The parameters are given in figure 1.

The suppression of a jammer for varying α and β has also been investigated in relation to the jammer level at a single array element. The Jammer Suppression Ratio is defined as

$$JSR = \frac{\frac{1}{2\pi}\int_{-\pi}^{\pi} R_{s_2}(\Omega)d\Omega}{\frac{1}{2\pi}\int_{-\pi}^{\pi} R_{s_2}(\Omega)|H_2(\Omega)|^2 d\Omega}. \quad (5)$$

In Figs. 4 and 5, the suppression ratio JSR is presented for different jamming directions with α and β as parameters.

4. CONCLUSIONS AND FUTURE WORK

A cumbersome part of microphone array realization is having to calibrate the microphones and analog channels at the inputs. A self-calibrating realization has thus been developed. This beamformer has been studied from an analytical point of view by using noncausal Wiener solutions. The study has gained insight into which parameters affect the jammer suppression and give target signal distortion.

An interesting application is where the jammer is a hands-free loudspeaker. Placing and design of the hands-free loudspeaker provide different performance.

The study has shown that in order to achieve good echo suppression and small target distortion, the target signal and jammer signals should be well-separated in the spatial domain. This will make the inner product between their transfer functions small.

Further work is also needed on this basic beamformer type. We will employ measured transfer functions, and study finite length FIR filters, as well as combining the beamformer with a conventional echo canceller. The design and placing of microphones and loudspeakers in a real environment will also be studied.

5. REFERENCES

[1] M. Dahl, I. Claesson "Acoustic Noise and Echo Cancelling with Microphone Array", submitted for publication in IEEE Transactions on Vehicular Technology, Dec 1996.

[2] M. Sondhi, W. Kellermann, "Adaptive Echo Cancellation for speech signal", Chapter 11 in Advances in Speech Signal Processing, Edited by S. Furui and M. Sondhi, Dekker, New York, 1992.

[3] I. Claesson, S. Nordebo, S. Nordholm, M. Dahl "An "in situ" Calibrated Adaptive Microphone Array" Submitted to Journ. Acoust. Soc. Am., March 1995

[4] S. F. Boll "Suppession of acoustic noise in speech using spectral subtraction", IEEE Trans. on Acoustics, Speech and Signal Processing, vol 27, Apr. 1979.

[5] J. Yang "Frequency Domain Noise Suppression Approaches in Mobile Telephone System" Proc. ICASSP-1993, pp II-363

[6] Y. Kaneda, J. Ohga, "Adaptive microphone-Array System for Noise Reduction", IEEE Trans. on Acoustics, Speech and Signal Processing, vol. 34, no. 6, Dec. 1986

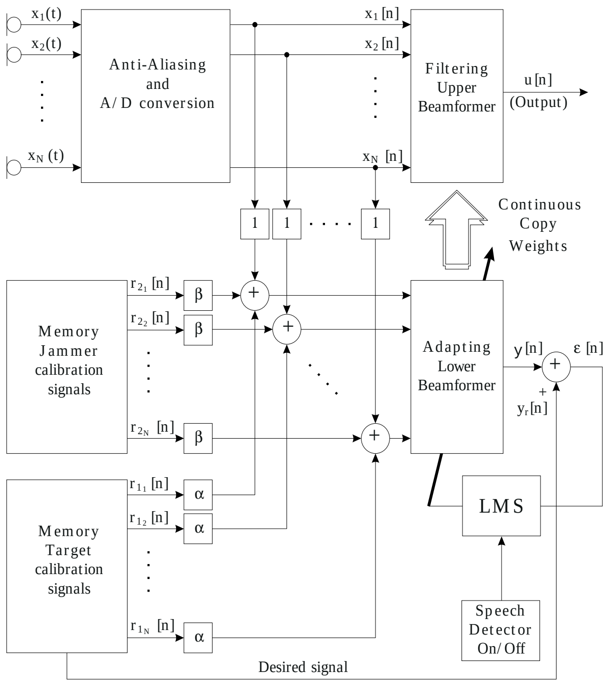

Figure 1: Adaptive Microphone Array in Operation Phase

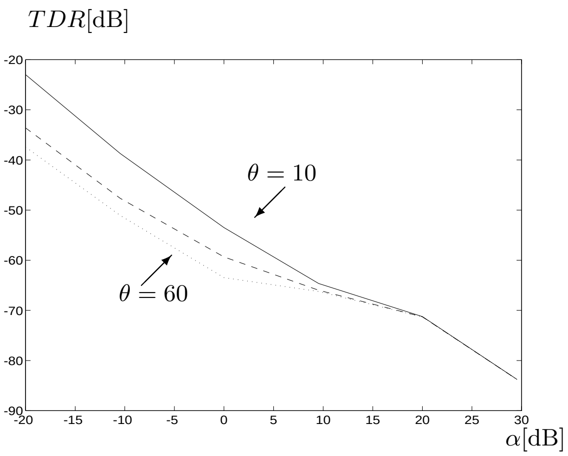

Figure 2: Target Distortion Ratio, TDR, versus α. $\alpha = 1$ (0 dB) corresponds to SNR=30dB and SIR=0dB. Linear array with d=0.05 m, N= 7. Jamming directions $[10, 45, 60]$.

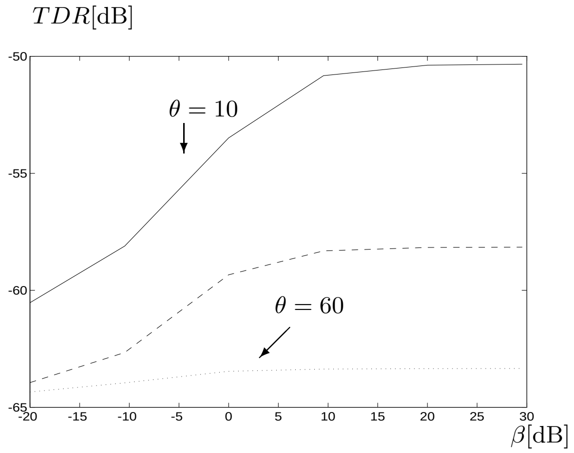

Figure 3: Target Distortion Ratio, TDR, versus β. $\beta = 1$ (0 dB) corresponds to SNR=30dB and SIR=0dB. Linear array with d=0.05 m, N= 7. Jamming directions $[10, 45, 60]$.

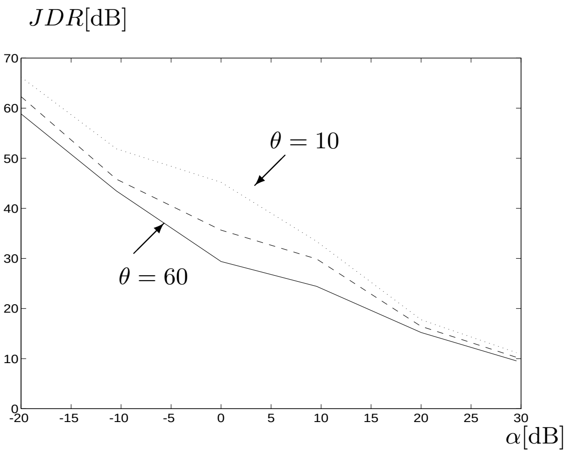

Figure 4: Jammer Suppression Ratio, JSR versus α. $\alpha = 1$ (0 dB) corresponds to SNR=30dB and SIR=0dB. Linear array with d=0.05 m, N= 7. Jamming directions $[10, 45, 60]$.

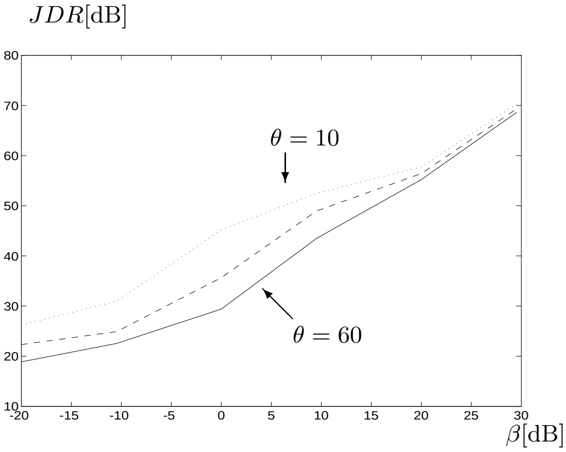

Figure 5: Jammer Suppression Ratio, JSR versus β. $\beta = 1$ (0 dB) corresponds to SNR=30dB and SIR=0dB. Linear array with d=0.05 m, N= 7. Jamming directions $[10, 45, 60]$.

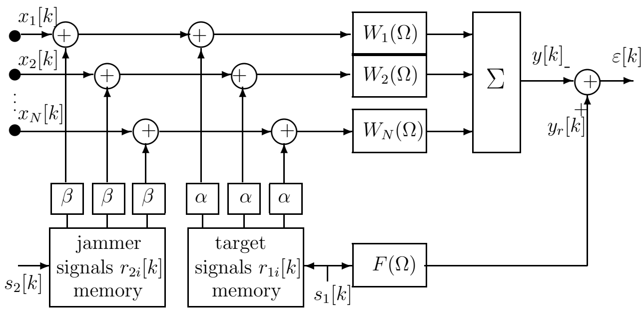

Figure 6: Signal situation for the Wiener analysis