Noise Reduction Using an Adaptive Microphone Array in a Car -A Speech Recognition Evaluation

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ABSTRACT

This paper describes an evaluation of an adaptive microphone array with respect to speech recognition performance in a car. The microphone array is compared to two conventional microphones of different types. The speech recognition device is aimed to be a part of a man/machine-interface between the driver and car information services.

1. Introduction ,

In order to enhance traffic safety and keep the drivers' attention to the traffic, it is suitable to use speech synthesis to present information and speech recognition to control the flow of information. Due to the noisy situation in a car it is necessary to employ some noise reducing device in order to make the speech recognition function satisfactorily.

This paper¹ presents an evaluation where recordings from a large speech material have been used in a car in typical traffic situations. Utterances have been recorded with two different conventional microphones and a microphone array. The array recordings have been processed in a signal processing system DSP900, where a spatial filtering technique[1] in conjunction with adaptive filters has been used. The material has been evaluated with respect to speech recognition performance.

2. Spatial Filters for the GSC

A Linear Array Broadband Generalized Sidelobe Canceller[2] (GSC) consists of an array with an upper beamformer G providing the "desired response" to the adaptive algorithm and lower beamformers B_i which provide the inputs to the adaptive filters H_i , see fig. 1. The purpose of the lower beamformers is to protect the target signal from being cancelled by the adaptive filters and 1-D spatial filters have been used for this purpose[1].

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Figure 1: Adaptive beamformer structure

Zero weights in the adaptive filters when no signals are incident on the array are often wanted [3]. This is easily obtained if a leaky algorithm is used to update the adaptive filters, provided the "leaky" noise η_k dominates over the sensor noise. Furthermore, a dominating "leaky" noise implies simple Wiener solutions and straightforward design criteria for the spatial filters B_i . In figure 2 the performance of the adaptive array is illustrated. Utterances are made as five words-pause-five words and the output power level with and without noise cancelling is plotted. A typical 10-15 dB improvement is obtained.

3. The Speech Recognition

The speech-recognition device is a modified version of the Infovox RA-201, marketed by Infovox AB. This is a speaker-adaptive system using template matching of cepstral coefficients[4]. The input is preemphasized by +6 dB per octave between 200 and 5000 Hz, bandlimited for anti-aliasing and sampled with a rate of 10 kHz. The digitized signal is then processed by a NEC 7720 signal processor used as a 16 channel Bark-scale filter bank. Seven of the eight lowest cepstral coefficients are calculated from the amplitudes in the 16 Bark-scale bands with a rate of 40 Hz. The first coefficient contains the overall energy and is not used, and therefore the seven cepstral coefficients describes the shape of the spectrum which is not sensitive to signal amplitude. After endpoint detection, the words are linearly normalized to 32 samples which implies a nominal length of $32 \cdot 1/40$ s. During recognition, the unknown utterance is non-linearly time-normalized for optimal alignment to each

Figure 2: Illustration of SNR-improvement with adaptive array

vocabulary word.

A noise compensation technique has been used, where the recognition system is adapted to environmental noise. The noise is measured just before and/or after the sampled word and is "added" [4] to the reference templates which are recorded in a silent environment. This procedure compensates for the fact that training and recognition have occurred under different environmental conditions.

4. Results

The results presented here are based on six speakers, three women and three men reading fifty-three different utterances repeated ten times each. However, the last two females spoke considerably lower than the other four speakers, making it almost impossible to detect words for the more distant microphones. The material is evaluated with respect to speech recognition, see fig. 3 and fig. 4.

Figure 3: Correctly recognized utterances in percent

The speech material is recorded in a studio and played with a loudspeaker in a car driving 90 km/h. Simultaneous recordings are made in the car with a high quality condenser microphone (AKG), a goose-neck microphone (GN) especially designed for mobile telephony and an eight-element array of medium quality condenser microphones. The array recordings are processed using two particular 1-D spatial filter designs (spatial 1 and spatial 2). One of the array microphones (MID) and the sum of the eight (SUM) are included in the test. The array and AKG microphones are placed at approximately the same distance (40-50 cm) in front of the speaker, while the goose-neck microphone is placed at a distance less than 10 cm.

The results should be interpreted with care since there is a large variation among speakers. In fig. 3, GN is the most robust microphone, while MID, which is the simplest microphone, and AKG, which is the most expensive, perform poorly when the speaker signal is weak (speaker 5 and 6). In fig. 3, spatial 1 and spatial 2 perform significantly better than MID and SUM as expected. The high quality AKG shows the highest detection rate for speaker 1 through 4 (as expected). The GN performs approximately as spatial 1 and spatial 2 despite the fact that it is mounted very close to the speaker.

Further, fig. 4 shows that the "false alarm rate" is significant for the GN microphone due to the commercial mounting. The other microphones are mounted acoustically correctly.

5. Possible improvements and current efforts

The speech recognition algorithm does not take into account the loss of lower frequencies caused by the adaptive array. The array works as an active microphone and

Figure 4: Falsely detected in percent (false alarms)

the frequency response depends on the noise **situation**, i.e the microphone **will have** different frequency **characteristics** at training and at recognition, the **difference** can be **as much as** 20 dB for some **frequencies**. However, the speech recognition **device** is **based** on power measurements in different frequency bands and a **frequency** distortion might therefore be **devastating**. If the algorithms could compensate for the different frequency characteristics at training and recognition, the adaptive **array** would be **better** exploited. In **our** future work **this** will be taken into account. **Further** a **possible** improvement is to use microphones of **better** quality in the **array** (at **least** as good as the **goose-neck** microphone).

A cumbersome part of the spatial filter realization is the calibration of the microphones and analog **channels** at the inputs. A **self** calibrating realization has therefore **been** developed. Apart from being very robust **against** channel **mismatch** **preliminary** results indicate a further **5 dB** improvement of the signal-to-noise ratio, **yielding** even **better** recognition **performance**.

6. Summary and Conclusions

A broadband adaptive microphone **array** has **been** tested and **compared** to conventional microphones with **respect** to speech recognition in a **car**. The **measurements** show that the **array** gives a higher signal to noise ratio **than** the other microphones, and is as good as a **goose-neck** microphone for speech recognition.

References

1. I. Claesson and S. Nordholm, "A Spatial Filtering Approach to Robust Adaptive Beamforming", *IEEE Transactions on Antennas & Propagation*, September 1992.
2. B.D. van Veen, K.M. Buckley, "Beamforming: A Versatile Approach to Spatial Filtering", *IEEE ASSP Magazine*, April 1988.
3. L.J. Griffiths, K.M. Buckley, "Quiescent Pattern Control in Linearly Constrained Adaptive Arrays", *IEEE ASSP Transactions*, July 1987.
4. M. Blomberg, K. Elenius, "Optimizing some parameters of a word recognizer used in car noise", *Speech Transmission Laboratory Quarterly Progress and Status Report*, 4/1990, KTH.