IMPLEMENTATION OF A GRIFFITHS-JIM ADAPTIVE MICROPHONE ARRAY ON A TMS320C25 MULTI-DSP

Sven Nordholm Ingvar Claesson Bengt Bengtsson Per Eriksson

Signal Processing Group, University of Lund
Box 118, S-221 00 Lund, Sweden

1 Introduction

The quality of the sound from a hands-free mobile telephone is often poor, since the desired speaker signal is severely disturbed by several sources, such as wind, engine and tyre noise. One way to eliminate these disturbances are traditional noise cancelling [1]. This works well if there is a possibility to obtain good correlation between the primary and the reference input originating only from the interfering signal. Good correlation is obtained mainly from the engine. Wind noise is however a problem. A way around this is to combine an adaptive microphone,with fixed gain within a focus area, with a noise canceller [2,3]. Such an implementation demands fast hardware in combination with a suitable beamformer structure and adaptive algorithm. The implementation on the multiprocessor computer DSP900 is an attempt to combine these two methods.

2 Description of DSP900

DSP900 [4,3] is a multi-processor computer aimed for digital signal processing use in laboratory environments. The design of the system is done at the dept. of Telecommunication Theory, University of Lund.

The system is built in a frame with space for up to 8 subcomputers called DSPu, see figure 1, each with its own memory, a common memory, a PC-interface card and up to 20 I/O-units. All cards in the frame communicate via a global bus. In this application 5 DSP:s are used together with a IBM/PC as host. The 5 DSP:s and the host work towards a global memory with one of the DSPs as arbiter. The host is used for loading program

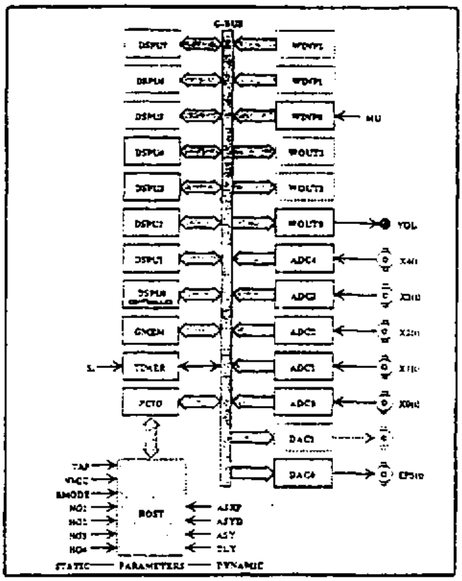

Figure 1: DSP900: block diagram

and presenting results. The arbiter controls the access to the global bus. If a subcomputer does not need the global bus it can work totally independent.

An adaptive microphone array, ARR900, has been implemented on DSP900. The implementation of ARR900 gave some practical problems, such as nearfield and spread sources, imperfections in microphones, amplifiers and filters. These problems must be taken into consideration when choosing filter structure, adaptive algorithm and how to eliminate superresolution.

We chose to use a Griffiths-Jim structure [5], see figure 2, in combination with the LMS-algorithm and to use the "hourglass technique" [2,6], nulling an area of weights around the center, to avoid superresolution. This combination leads to almost perfect parallelism. The updating formula for an arbitrary coefficient is

$$Wi_j(k+1) = Wi_j(k) + 2 * MU * EPS(k) * XiP_j(k) \tag{1}$$

All coupling between coefficients are in $EPS(k)$. The simple weight updating formula hereby facilitates parallel implementations. A 5 microphone array implementation is naturally divided between 5 DSP:s with this type of filter structure and algorithm. The problem with spread sources in the nearfield is solved by focusing to a point where the speaker mainly is

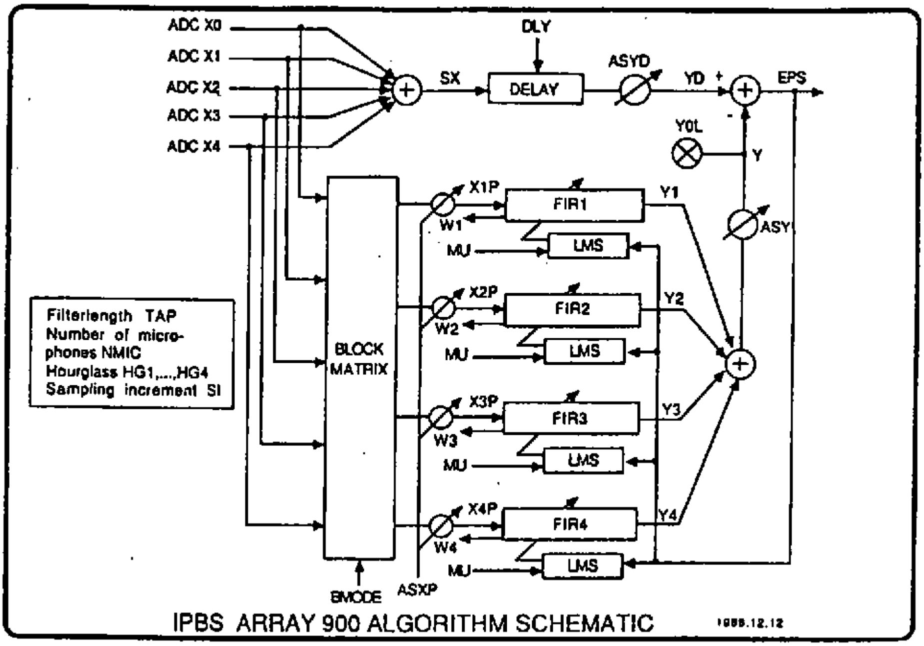

Figure 2: The Griffiths Jim adaptive beamformer structure

situated combined with a speech controlled adaptive algorithm, i.e. adaptation takes only place when the speaker is silent.

The arbiter controls the global bus, handles A/D and D/A conversions and calculates $YD(k)$, $XiP(k)$, $Y(k)$ and $EPS(k)$. It also handles all scaling. The processors DSPu2 to DSPu5 calculate the coefficient updating and the FIR filtering which is the major part of the computational load. For each sample the processors need to communicate only twice to the global memory. First, to leave the output signal $Yi(k-1)$ and next to fetch $XiP(k)$ and $EPS(k-1)$. With this structure and computational power it is possible to implement 128-tap FIR filters behind each microphone at a sampling rate of $8kHz$.

3 Preliminary Measurements

Two types of measurements have been performed in a anechoic chamber, suppression as function of FIR length and directivity diagrams. All measurements have been done at 8 kHz sampling frequency and a single 10 cm loudspeaker broadband source covering 300–1100 Hz. The microphones were placed on a circle with radius 60 cm with 5 cm distance along the circle between successive elements. The measurements show a possible 20 – 30 dB suppression of the jammer, even with reflecting walls inserted around the microphone array to primitively simulate the situation in a car.

3.1 Suppression dependence of FIR length

The jammer suppression measurements were performed with the jammer situated in (-30,60)cm, see Fig. 3, with origo chosen at the center microphone. Fig.4 shows the SINR, Signal-to-Interference-plus-Noise Ratio versus the FIR-length with hourglass length HG=1. The different curves correspond to the following reflection situations in Fig.3:

- Solid line: No reflecting surfaces.
- Dashed line: Reflection surface E.
- Dotted line: Two reflecting surfaces E and F.
- Dashdotted line: Three reflecting surfaces D, E, and F.

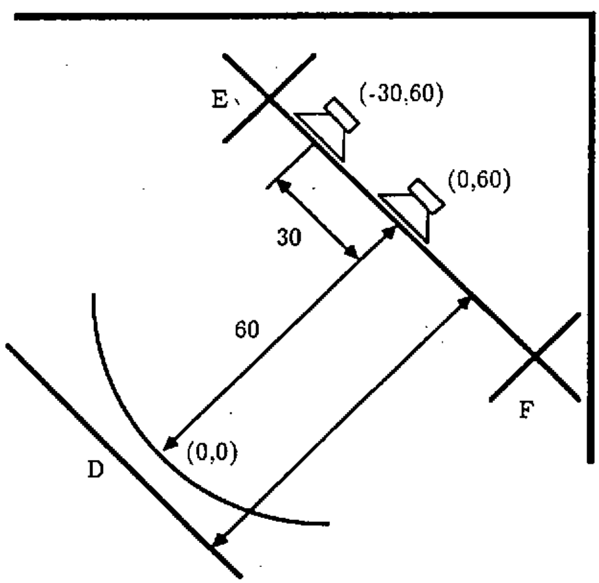

Figure 3: Placement of array, loadspeakers and reflection surfaces.

All curves confirm that the achievable suppression saturates at approximately 50 taps for different reflection situations. In Fig. 4 it is notable that with side walls the suppression can be maintained and even improved if a wall is situated in a direction giving an echo which not is angularly equivalent to the direct wave for the array, see the two wall curve.

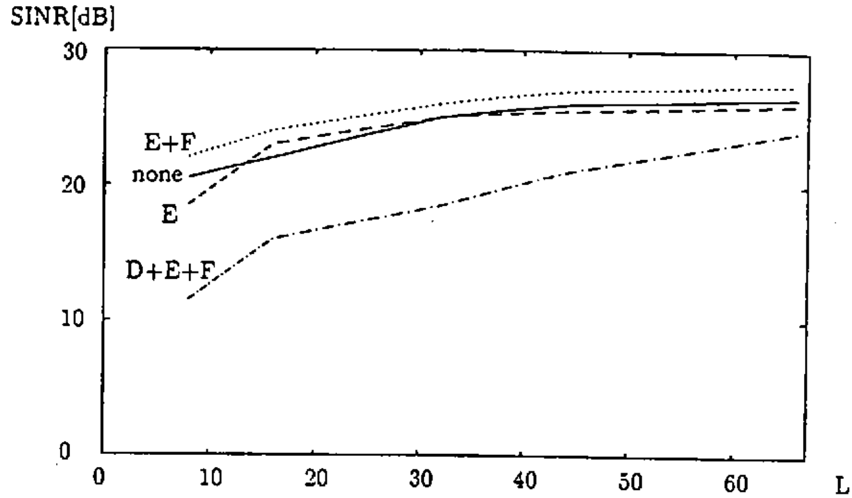

Figure 4: SINR dependence of FIR-length for different reflection situations. One broadband jammer 300 – 1100 Hz. Hourglass length HG=1.

The reason for this improvement is that new cancelling capability is enabled for other filter weight modes, which exceeds the increase in jammer effect that is arriving into the array. However, if a wall is inserted behind the array, more signal power comes into the array from the "same" direction as the direct wave and a degraded performance is obtained, see the three wall situation. This problem can be reduced by using cardiodal microphones in the array.

3.2 Suppression Directivity diagrams

The directivity diagrams have been obtained as follows: A position is selected for the interferer and we let the adaptive filter converge. Then the adaptation is turned off, i.e. the weight values are fixed. The output power is thereafter measured moving the loudspeaker to varying x-coordinates with the y-coordinate fixed, while testing different filterlenghts and hourglasses. The original position for the loudspeaker was chosen to (-30,60)cm, see Fig. 3. A 25 dB suppression can be observed in that direction.

The dashed lines show the directivity of the conventional beamformer, i.e. without the cancelling part. Observe that the spread source and imperfections in the hardware slightly perturb the constraint of 0 dB amplification of signals from the focus point.

REFERENCES

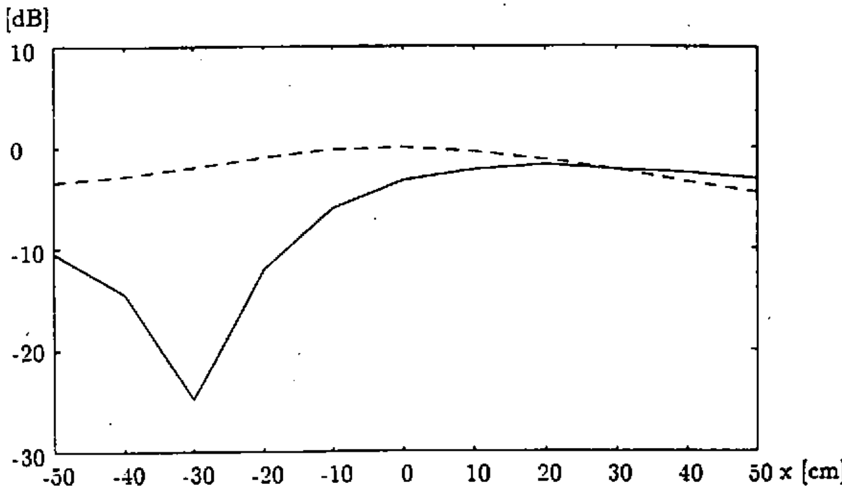

Figure 5: Directivity diagram with filterlength 45 and hourglass HG=1.

4 Conclusions

A multi-DSP implementation has been presented together with an implementation of an adaptive microphone array. A way to handle a spread source in the nearfield is given. The system shows a 25 – 30 dB suppression capability of a wide-band interferer.

Acknowledgements

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References

[1] B. Widrow S. D. Stearns
"Adaptive Signal Processing"
Prentince Hall 1985

[2] Ingvar Claesson Per Eriksson Sven Nordholm
"Adaptive acoustic noise cancelling for a mobile telphone by using a microphone array and secondary sensors"
Signal Processing Group, University of Lund,Technical Report SPR-1, 1988

REFERENCES

[3] S. Nordholm I. Claesson B. Bengtsson P. Eriksson
"A Multi-DSP Implementation of a Broadband Griffiths-Jim Adaptive Beamformer Array"
Department of Electrical Engineering and Computer Science University of Lund, Signal processing Group, Technical Report SPR-7, Okt 1989

[4] Bengt Bengtsson
"MAAS DSP900 Manual and technical description"(in Swedish)
Telecommunication Theory, University of Lund, Autumn 1988

[5] L. J. Griffiths C. W. Jim
"An Alternative Approach to Linearly Constrained Adaptive Beamforming"
IEEE Transactions on Antennas and Propagation, vol. 30, January 1982

[6] B. Widrow J. M. McCool
"A comparison of Adaptive Algorithms Based on the Methods of Steepest Descent and Random Search"
IEEE Transactions on Antennas and Propagation, vol. 24, 1976.