Proceedings of the Institute of Acoustics

HEARING AID EMULATION

R D Wright

Research Group, Royal National Institute for the Deaf,
105 Gower Street, London WC1E 6AH, UK

1. SUMMARY

The RNID have developed a system for research on signal processing, with particular reference to hearing aids. The basis of the system is a digital signal processing unit and analogue interfaces, within a host computer.

Varieties of hearing aid processing can be programmed for the signal processing unit, where they can run in real time to emulate the operation of a hearing aid. The emulation system can work as an exact replacement for an aid, with real time input and output from the same transducers as used in conventional aids. Processing strategies can be compared transparently, without the user removing and replacing devices.

The system can also work (still in real time) on digitised samples held in the host computer, allowing adaptive testing strategies on word list or other materials. Full facilities of the host can be used to prepare stimuli, and analyze signals before and after processing. The programmable processing controlled by a host computer lends itself to adaptive processing, including learning networks within the (emulated) hearing aid.

Finally, for longer use outside the laboratory, a pocket-sized, battery powered version of the digital processor is available.

2. INTRODUCTION

Digital signal processing (DSP) is a technology which has evolved over the last thirty years, progressing from a few large research computers through mini and microcomputers, to the development in the 1980's of the DSP chip. The technology has moved from the exotic research of the 1960's to the consumer products of the 1990's. Many applications areas have followed this technology trend, including development of hearing aids. A collection of state-of-the-art research papers using signal processing and digital technology is provided in Levitt (1987), and a general review is given in Wright (1989).

Although a digital hearing aid is commercially available, digital processing is especially useful for research work. Various types of processing can be tried without each time building new hardware. The signals in and out of the processing can be stored (because they must be digitised to be DSP'd) and analyzed in detail. Data and methods are exactly specified, and can be exchanged between laboratories. Finally, the processing of an actual device such as a hearing aid can be exactly modelled (emulated) on general purpose computers before any hardware implementations are considered.

However, not all parties interested in hearing aid research have general DSP workstations. The RNID have developed a set of tools - software and hardware - to allow emulation of hearing aids, including real time emulation, using a personal computer and a small amount of additional equipment. The hope was that these tools would not only be useful in the RNID's own research, but could be used by other small research projects.

The RNID DSP Research System is shown in Figure 1. It has two main parts, the desktop emulation system for development and laboratory testing, and the wearable system for field tests.

<u>Emulation System</u>	<u>Wearable System</u>
PC controller	----------
DSP card (co-processor)	*DSP chip (microprocessor)*
analogue signals card	*codec or wideband codec*
control programme	----------
DSP frame programme <--equivalent-->	*DSP frame programme*
DSP code <--------identical-------->	*DSP code*

Figure 1. The RNID DSP Research System

In microelectronics, the system used to develop and test programmes for a microprocessor is called a development system. The device or unit which would actually use the microprocessor is usually called the target system. The same division exists in Figure 1, but with a different terminology to indicate particular departures from a standard configuration.

The Emulation System is certainly a development system, but includes the capability of executing programmes in real time, using data either (analogue) from the external world or (digitised) from the PC controller. Thus the development system can be treated as a real system, and used directly on subjects for experimentation with DSP approaches. Many DSP workstations do not have a real-time capability, limiting their use in interactive test strategies.

The Wearable System must of course run in real time, as its signals come from one or more microphones (or other analogue inputs) and go to analogue output transducers. However, in a sense the wearable system is still an emulation. It embodies an approach to dealing with signals in hearing aids, using a digital implementation. There might be an analogue method of performing equivalent processing. In such cases the Wearable System can be viewed as a programmable emulation of an analogue system.

The next two sections will describe the Emulation and Wearable systems in greater detail.

3. EMULATION SYSTEM

The ultimate purpose of the personal computer based system shown on the left in Figure 1 is two-fold:

A- Development of signal processing techniques;

B- Interactive tests to optimise or fit the processing to an individuals particular requirements.

In order to accomplish these tasks, the system provides the following capabilities:

A- Development of DSP programmes using standard editors, compilers and assemblers;

B- Debug of DSP programmes in hardware using a monitor;

C- Real time execution of DSP programmes with analogue or digitised signals as inputs or outputs;

D- Interactive experiments on subjects using real time DSP;

E- Off line processing of signals and experimental results.

The equipment used is based on a standard personal computer (PC or clone), with a hard disc for signal storage. In addition there are two special cards which fit inside the PC:

A- a DSP microprocessor (Texas TMS320C25) with memory and digital interface circuits (co-processor card; Loughborough Sound Images PC Short Card);

B- an analogue-to-digital convertor (and converse) (Loughborough Sound Images 4-Channel Analogue Card).

The DSP card acts as a co-processor, running independently but capable of remote control by the host PC. The memory on the card is shared between the PC and the DSP device, allowing signals and parameters to be passed, as well as allowing the PC to load DSP programmes into the co-processor. The analogue interface card provides four input and two output channels, sampling at up to 200 kHz. Both cards are standard commercial product.

In addition to the equipment, the system includes two programmes:

A- the host PC runs a Controller: a programme (in C) which controls the co-processor, the analogue and digital signals, and interacts with the user;

B- the DSP co-processor has a Frame: an assembly language programme which responds to the Controller and accepts sampled signals either from the analogue interface card or the host.

The Controller is meant to support most tasks for which the entire system might be used. It selects which DSP function the co-processor is to perform (and loads it into the co-processor and then starts and stops the processing). It selects which signals are to be used - either analogue signals from the real world, or digitised signals from the hard disc, sampled at up to 16 kHz. In both cases both the input and the result of the processing can be saved on the host's disc. It allows the user to intervene between buffers of data (holding up to 64K samples, or 4 seconds at the maximum sampling rate), which supports interactive tasks such as adaptive speech audiometry or adaptive psychophysical tests.

The Frame is equally important, as it does everything that a DSP programme needs to do except the actual processing. In the heart of the programme a 16-bit word named SAMPLE is stored in a word named PROCSS, implementing a 'null process' or piece of wire. For each application the system developer need only add the relevant code for processing an individual sample, with all the messy details already taken care of by the Frame.

4. WEARABLE SYSTEM

The wearable device (Mark I) uses a simpler version of the DSP chip (the TMS320E17), which is packaged in a way which minimises size and power consumption. The device can run the same code as used in the emulation system, providing that code refrains from using certain instructions and other capabilities. The system runs for several days from one PP3 sized battery, which could be a rechargeable battery.

This wearable system was not exactly an RNID design. The circuit board (approx 5 cm square) was produced by Ensigma as a demonstration of speech recognition. The RNID have been allowed to use the Ensigma board, with some hand modifications (and of course none of the Ensigma speech recognition software) as the first UK digital hearing aid.

However that system has an audio bandwidth limited to the telephone band of 300 - 3300 Hz, because the analogue to digital conversion (ADC) is handled by a codec chip. For greater bandwidth (and greater power consumption - something less than a day on one PP3) the RNID have made their own design (Mark II), using a 14-bit linear ADC with up to 10 kHz bandwidth.

The Mark II also has expanded user control, with two 16-way switches mounted on the top, and a further eight bits of switching mounted internally. These switches allow for parameter setting, and also allow a user to select amongst alternative processing strategies.

Although the power consumption makes the Mark II somewhat impractical, it can still fulfil its main role of wearable device for field tests. It is not expected to be an actual hearing aid, but a device for gaining experience with processing approaches in real situations.

As with the emulation system, there is a standard software Frame programme to handle the input/output. For each application it is only necessary to add code to convert SAMPLE into PROCSS. The use of FRAME programmes allows the same processing code to be used in all the parts of the system: the emulation, the wearable Mark I, and the wearable Mark II. Further, this processing code should be largely transferable to future systems - and provide a starting point for code for other brands of DSP chip.

5. EMULATIONS

A practical use of DSP in audiology is to optimise hearing aid selection and parameter adjustment using a single, highly controllable device, rather than using lots of individual actual hearing aids - and a screwdriver to adjust their parameters. DSP has been used in this way for some years (Levitt, 1982; Jamieson & Raftery, 1989). The RNID DSP Research System could be viewed as such a DSP hearing aid fitting system.

For research, DSP justifies itself when the processing is in some way adaptive. One example of such processing is adaptive filtering, a general approach to the adjustment of a linear system to minimise (on a quadratic error surface) the difference between desired and obtained results (Widrow et al, 1975). Adaptive filters have proven remarkably effective for certain cases of signal detection and signal enhancement. With an error criterion, perhaps even a subjective one provided by the wearer, a DSP system can be made to adapt its own processing - thus eliminating the conventional hearing aid fitting process altogether! This idea could be extended to a network of adaptive devices, a neural net. One use of a 'large but wearable' DSP system would be to support the learning phase, with the resultant programme embodied in a system of more practical size.

Another large class of processing approaches is adaptive non-linear systems. The RNID have begun to use the DSP Research System for work on adaptive clipping. Automatic Gain Control has been studied for two decades (Vilchur, 1973; Moore, 1987), but always as a form of quasi-linear processing with the specific goal of reducing the distortion caused by clipping. Clipping could itself be seen as a form of gain control, given an adaptive clipping level. It is hoped that results of RNID research on adaptive clipping will be reported at a future meeting.

6. REFERENCES

Jamieson, DG & E Raftery (1989), 'A General Purpose Hearing-Aid Prescription, Simulation and Testing System', ICASSP-89, p1989.
Moore, BCJ (1987), 'Design and Evaluation of a Two Channel Compression Aid', Jr Rehab R&D 24:181-192.
Levitt, H (1982), 'An Array-Processor, Computer Hearing Aid', Jr Amer Speech, Hng & Lang Assoc 24:805.
Levitt, H, Ed (1987), Sensory Aids for Hearing, Jr Rehab R&D 24.
Vilchur, E (1973), 'Signal Processing to Improve Speech Intelligibility in Perceptive Deafness', JASA 53:1646-57.
Wright, RD (1989), Signal Processing Hearing Aids, RNID Res Rpt.