A NEW DIGITAL AUDIO SIGNAL PROCESSING RESEARCH SYSTEM USING THE MOTOROLA DSP56001

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INTRODUCTION

Digital Filtering is rapidly becoming a cost effective solution to many signal processing problems. In addition to the usual advantages of digital implementation, such as no drift with temperature and age, recent advances allow further improvements. The use of an adaptive filter for tracking and filtering in a interference-cancelling task is a good example. This is one of a large number of applications in which adaptive signal processing is not only appropriate but will achieve results not attainable via analogue signal processing techniques. However, the development of a practically useful adaptive digital system takes a great deal of effort and can be very time-consuming. Because of the real-time nature of a target system, software verification can not be used to completely test the system, and further field tests using a real-time prototype system are usually required.

In general, research and development for a practical real-time digital signal processing system will proceed in the following order:

1. Selecting or developing a suitable model with possible processing algorithm.
2. Software verification using computer simulation.
3. Field trial using real-time system(software/hardware).
4. Implementation of the prototype target system.

In order to assist the first 3 of the steps outlined above, particularly for digital audio processing systems, a new digital signal processing system(REASP) using the Motorola DSP56001, CPU68020 and DSP56200 Digital Filter Chip was developed. This DSP system is referred to as "**REASP**"(**RE**AL-TIME **A**DAPTIVE **S**IGNAL **P**ROCESSOR) for short.

This paper describes the REASP and some examples of digital audio processing using this system. The rest of the paper first describes the architecture and processing functions of the REASP and then discusses some useful applications.

ARCHITECTURE

The REASP is a general purpose digital audio signal processing system designed to perform the computationally intensive tasks associated with digital filtering in the time domain, and furthermore it offers a high speed computation for relevant basic algorithms such as FFT, Hilbert transform or FIR/IIR digital filter designs. Users can develop their own application programs in UNIX or MS-DOS environments and can download them to the REASP through the system monitor in the host computer. The system monitor itself supports many useful commands with a window-based graphic presentation which includes the system identification, and adaptive noise cancellation using the Least-Mean-Squares(LMS) algorithm.

Figure 1 shows the hardware block diagram of the REASP. The CPU68020(+68881) manages interface between the REASP and a host computer through the ordinary RS232C interface or the high speed ARCNET interface. The REASP monitor on the CPU board allows the reception of basic task commands from the host computer. The host computer can designate a specific task by sending a list of successive basic commands such as "download program", "upload processed data" and "execute LMS adaptive filtering". The processing board based on a DSP56001 offers a very high processing speed in combination with the other FIR filter board(based on the DSP56200 FIR Filter Chip), with particular advantages for the high speed filtering needed for real-time adaptive processing of high bandwidth signals. The REASP has a DAT(digital audio tape-recorder) interface and 16 bit ADC and DAC interface using the oversampling technique. The user can specify the sampling frequency from the host computer. The other setup conditions such as, channel selection, buffer gain of ADC and DAC and the number of taps for the FIR digital filter are also controlled by the external host computer.

DSP56001 BOARD

The DSP56001 board consist of the DSP56001, 16KW data memory, 16KW program memory, 256KB boot strap memory, 4KB dual port RAM and a digital interface as shown in **Figure 2**. The DSP56001 is a 24 bit, fixed point digital signal processor. It features single cycle hardware multiplication, arithmetic to 56 bits, and a range of sophisticated addressing modes. In the digital audio application, speed(including parallelism) and precision are important features, as discussed below.

Speed: At 10.25 million instructions per second(MIPS) the DSP56001 can execute a 1024 point complex FFT in 3.39 milliseconds. The data ALU, address arithmetic units, and program controller operate in parallel so that an instruction prefetch, a 24x24-bit multiplication, a 56-bit addition, two data moves, and two address pointer updates using one of three types of arithmetic(linear, modulo, or reverse carry) can be executed in a single instruction cycle. This parallelism allows a four coefficient IIR filter section to be executed in only four cycles, the theoretical minimum for a single multiplier architecture.

Precision: The data paths are 24 bits wide thereby providing 144 dB of dynamic range; intermediate results held in the 56-bit accumulators can range over 336 dB. Apart from ordinary FIR filtering, all of the other signal processing incorporated in the REASP can be executed by the DSP56001 board.

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DSP 56200 BOARD

The DSP 56200 board consists of 8 chips of DSP56200 for each channel(2 channels are available in one board) and control circuitry(for DSP chip and BUS). The DSP56200 chip is an algorithm specific peripheral dedicated to digital filtering. The chip operates in one of three modes: Single FIR Filter, Dual FIR Filter(two independent FIR filters on one DSP56200), and Single Adaptive FIR filter. The DSP56200 contains both a 256 tap FIR filter section and the update circuitry necessary to implement the LMS adaptive algorithm. The input data width is 16 bits, output word width is 32 bits. In order to maximize throughput in applications not requiring a full 256-tap FIR sections, the user may program the number of taps from 4 to 256. The maximum number of taps which can be used in each filter, will depend on the sampling frequency and the mode of operations. For examples, in the case of adaptive filter mode with eight chips in cascade, a maximum sampling frequency 69 KHz would be possible for a total of 512 taps, as shown in **Table 1**. The DSP56001 can communicate to the DSP56200 board through an external special DSP BUS in the REASP. This special BUS configuration yields the flexibility in the manner of data manipulation. Thus the data from the ADC or DAT interface could be processed by the both DSP boards(DSP56001 or DSP56200) independently, and also one DSP board can access the filtered data transmitted from the other DSP board with maximum throughput.

APPLICATIONS

Applications of adaptive digital filtering have been recognized in such diverse fields as speech analysis, seismic, acoustic, and radar signal processing, and digital filter design. We also note that adaptive signal processing is not appropriate for all applications and, when it is applicable, we must design the adaptive system very carefully. Adaptive filters are particularly useful in systems with unknown or slowly varying characteristics. Applications fall into four general classes: Prediction, System Identification, Inverse Filtering, and Interference Cancelling. One popular telecommunications application of adaptive filters is in the area of echo cancellation. There are many applications in the field of digital audio. In this section, some examples of adaptive digital audio signal processing using the REASP will be discussed.

1. Loudspeaker Measurement and Simulation

This is a typical example in digital audio signal processing, in which the system identification technique and conventional FIR digital filtering technique are applied. System identification applications seek to find the impulse response or frequency response of an unknown system, for example, the transfer function defined from the input terminal of a loudspeaker to the output terminal of a measurement microphone. The basic structure for adaptive system identification is illustrated in **Figure 3-a**. **Figure 3-b** show the actual block diagram of the measurement using the REASP. The M-sequence signal take two paths to the adaptive filter section in the REASP - one path directly and one path which is connected to the input of the loudspeaker under measurement. The output acoustic signal emitted from the loudspeaker under measurement is received by a microphone. This received signal by the microphone is used as a desired signal D in the adaptive system identification process. This application of adaptive digital filter is referred to as "adaptive forward modelling" and is often an integral part of an adaptive control system process. The adaptive processor(DSP56200 board is used in this case) adjusts its weights to produce a response x that is as close as possible in the least-squares sense to the desired signal D. When the modelling error E becomes small, then the adaptive processor will have adapted to become a good model of the loudspeaker system.

The user can make this measurement very quickly by using the system monitor in the host computer. **Figure 4** shows an example of the window type graphic presentation used in the monitor, in which the measured impulse response uploaded from the REASP is shown. Furthermore, the user can set back the impulse response to the coefficient memory of the FIR digital filter in the REASP, and create a real-time simulation of the loudspeaker. The user can thus listen to the simulated sounds of many loudspeakers in a variety of setup conditions. The coefficients of the FIR digital filter can be chosen very quickly from the pre-measured impulse response library as stored on a hard disks.

2. Adaptive interference cancelling - one channel case

Adaptive filters can be very effective at noise cancelling, especially when a correlated version of the additive interfering noise is available. However it is often not possible to obtain this second noise which is not only correlated to the interference noise but also free from the desired input signal. An audio example of this situation is where the signal has already been recorded with the noise present. In this case, it is still possible to cancel the noise if the noise is periodic in nature such as engine noise or powerline hum interference. **Figure 5-a** shows the basic structure of a periodic noise cancelling system(also known as an "adaptive line enhancer") with a single input, using an adaptive digital filter. The delay in **Figure 7** is used to decorrelate the desired signal components at both input of the adaptive filter section. Since the noise is periodic, however, the interference components on both inputs will always be correlated. The adaptive processor tries to minimize the internal error by producing an output to be an approximation to the interfering noise. As a result, the error term becomes a good estimate of the desired signal. **Figure 5-b** shows the actual block diagram using the REASP. The DSP56001 works as an delay unit and also controls the adaptation sequences of the adaptive digital filter(DSP56200) in this configuration.

CONCLUSION

As shown in the above mentioned examples, the DSP56001 can be liberated from the intensive tasks associated with the ordinary FIR filtering routine, therefore the DSP56001 is made available for more sophisticated tasks; such as execution of variable step LMS algorithms; control of adaptive digital filter networks for a multi-channel system; and an implementation of IIR filtering using FIR adaptive digital filters. The REASP is also designed to perform batch processing using the extension memory. In this case, more complicated data flow of processing can be formed with two DSP boards. The REASP has been used in a wide range of research fields; active noise and vibration control, digital simulation of room acoustic including mobile cabin, digital audio recording and reproducing system, and digital control for Hi-Fi loudspeaker reproduction. The next stage of development will include multi-REASP network control and further reinforcement of the existing applications software base.

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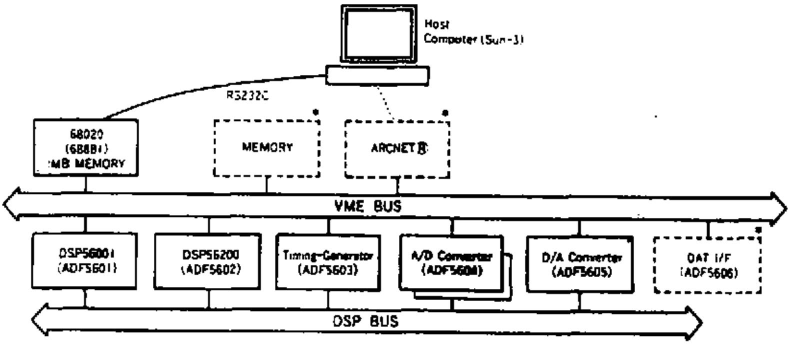

Fig.1 Block diagram of the REASP

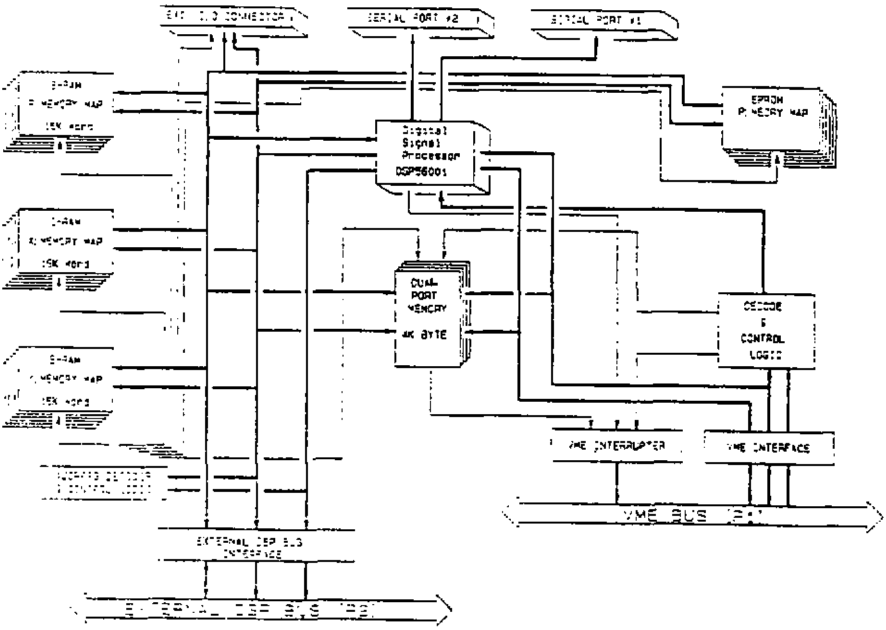

Fig.2 The DSP56001 digital signal processing board

Table 1 The DSP56200 performance(Eight Chips in Cascase)[2]

Maximum Sampling Frequency (kHz)				
Mode	Total Number of Taps			
	256	512	1024	2048
FIR Filter	204	132	71	37
Adaptive Filter	115	69	37	19

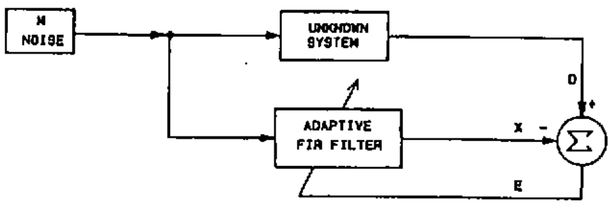

Fig.3-a The basic structure for adaptive system identification

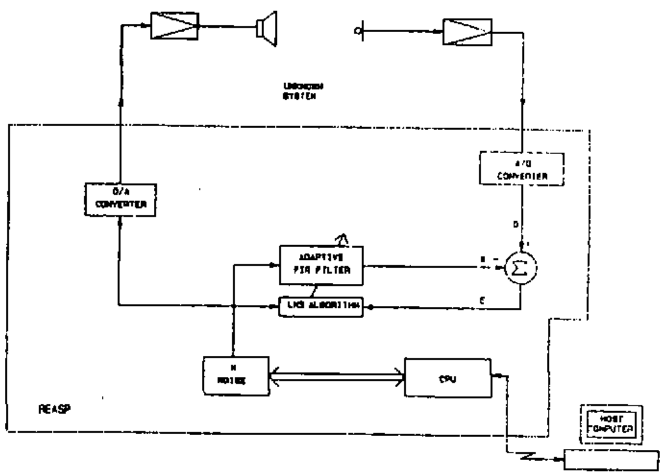

Fig.3-b Block diagram of the measurement using the REASP

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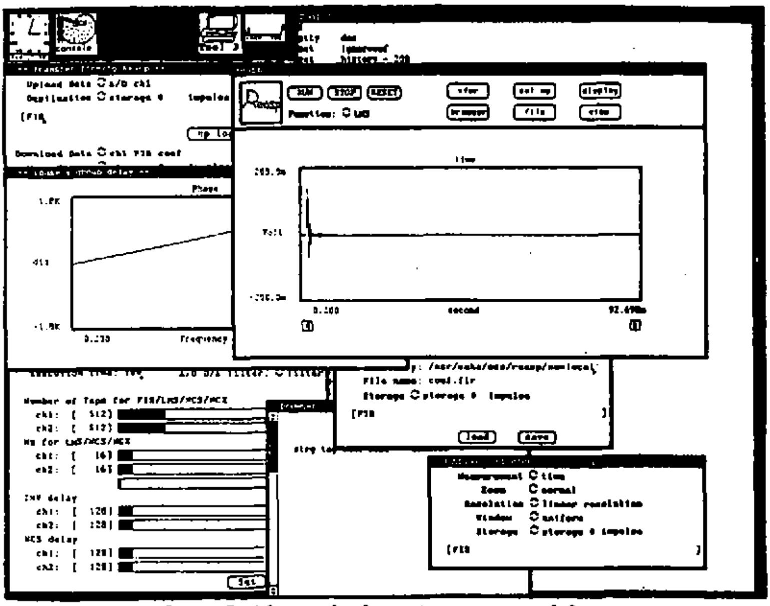

Fig.4 An example of the window type graphic presentation used in the monitor

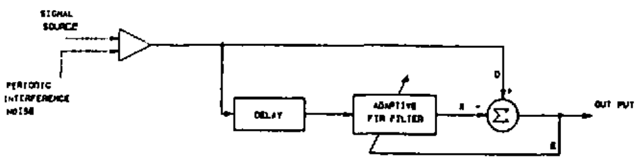

Fig.5-a The basic structure for adaptive interference cancelling

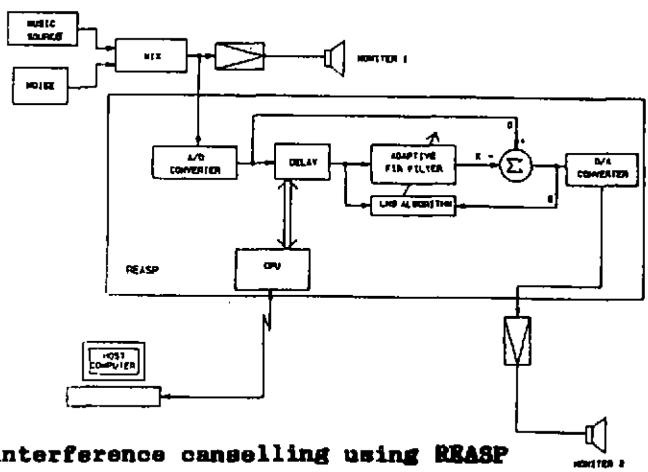

Fig.5-b Adaptive interference cancelling using REASP