A DIGITAL SIGNAL PROCESSING SOLUTION FOR SOUND SIMULATION

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ABSTRACT

The design of sound simulators for aircraft and other vehicles has often presented a variety of problems in areas of integration, flexibility, maintenance and life cycle cost. Recent developments in digital signal processing (DSP) technology have provided a powerful and cost effective solution to these problems by way of a special device known as a single-chip digital signal processor. This technology allows fixed hardware to be highly flexible by using software algorithms to perform functions that would normally require analog oscillators, noise generators, filters and amplifiers. This approach eliminates recurring hardware design, simplifies integration, increases system reliability and provides better quality and control of sound parameters. This paper describes the features and advantages of a DSP-based sound simulator prototype that is capable of generating complex tone scenarios such as those found in avionic systems and other sounds such as those developed by a vehicle and its surrounding environment.

INTRODUCTION

In many training systems a sound simulator plays an important role in providing a trainee with a convincing and effective degree of realism. Over the years manufacturers have used a variety of methods for meeting sound simulation requirements including the use of actual recordings, design of unique sound generating circuitry, or, in more recent times, use of dynamically modified digital recordings. Interest in designing digital-based sound simulators has stemmed from demands for greater performance and reduction of life-cycle cost. The traditional analog-based approach has been very limiting with regard to meeting these cost/performance objectives because each type of training system has usually required a different sound simulator design. Furthermore, such systems have often required costly circuit changes to implement modifications of simulated sounds.

Digital signal processing (DSP) technology has grown significantly over the past few years and has forced many analog circuit designers to re-evaluate their options. Several companies have produced devices known as single-chip digital signal processors which can replace numerous analog circuits and are well suited to performing audio synthesis and processing. To demonstrate the capabilities of a single-chip DSP processor consider the case of simulating a missile launch from a fighter aircraft. The sound is typically simulated by generating random noise, filtering the noise to obtain the required center frequency and bandwidth and then amplitude modulating the signal with the proper attack and decay times. A single DSP processor has been programmed to perform this entire synthesis process. Similarly, a variety of other complex waveforms and tone sequences can be numerically defined and executed on a single-chip DSP processor. Sounds that are not required simultaneously can be grouped and executed by a single processor.

Using the approach outlined above, a DSP-based sound simulator prototype system has been developed which uses a software library to program fixed generic hardware for a specific training requirement. This paper discusses the following features and benefits of this prototype system:

- A single type of circuit board is used as the system building block. The system can easily accommodate expansion and allows different types of trainers to be realized using the same hardware design.
- Reliability is enhanced and periodic calibration requirements are eliminated by using digital components and implementing a built-in self-test.
- 16-Bit digital-to-analog converters are used to supply wide dynamic range.
- System can supply multiple analog output nodes for both monophonic and stereophonic sounds.
- Numerical processing is used to create sounds. Enhances controllability and accuracy of sounds and allows quick turn-around time for system realization and modification.
- Host integration is simplified by using a short command set and communicating via shared memory.

SYSTEM HARDWARE

The hardware architecture of this digital sound simulator is based on the use of multiple DSP processors to create a network of complex waveform generators. The network is formed using several copies of a single type of circuit board that allows the system to be expanded to meet the sound simulation requirements of a particular training system.

The DSP Circuit Board

The DSP circuit board is functionally and physically divided into the following three sections: A four-processor DSP array that executes sound synthesis programs, a communications interface that allows a host computer to communicate with the DSP processors through shared memory, and an analog section that provides two channels of digital-to-analog conversion and signal smoothing. The board's input/output ports allow two channels of synthesized audio to be digitally summed with signals on other boards, or the analog outputs can be summed using an audio mixer. A simplified block diagram is shown in figure 1.

Proper operation of digital and analog circuitry is verified by a built-in self-test that executes at system reset or at the request of a host computer. Information supplied by the self-test allows a host computer to locate a faulty circuit board or identify specific failures on a given board. To minimize the impact of a component failure the circuit board makes no attempt to terminate its operation upon detection of a failure.

The hardware features identified above provide several advantages that have a direct impact on performance and life-cycle cost. Maximizing the use of digital components and implementing a self-test greatly increases reliability. The use of DSP processors provides extensive flexibility and eliminates recurring hardware design. Using shared memory simplifies system integration. A multi-board network comprised of a single type of circuit board minimizes the overhead required for support of the hardware.

The DSP Network

A network of signal processors is formed using several copies of the DSP circuit board in a single chassis. The number of boards required for a particular training requirement depends on the number of monophonic sounds, the number of stereophonic sounds, the number of sounds that are likely to occur simultaneously, and the number of stereophonic sounds that can be cross-coupled between processor pairs. A typical fighter aircraft may require between 8 and 10 of these DSP boards. The present addressing format allows for a maximum of 16 DSP boards per single chassis.

Once the required size of the network is determined, analog output nodes can be established based on the summing methods defined within the DSP software. The resulting analog signals may be connected to power amplifiers and other hardware such as an intercom system. Figure 2 shows how this DSP-based sound simulator can be implemented in a training system using

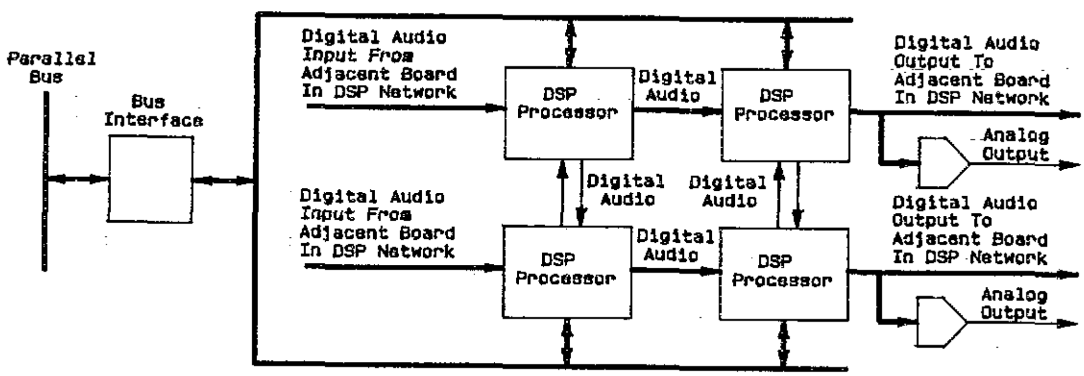

Figure 1. Simplified Block Diagram Of The DSP Circuit Board

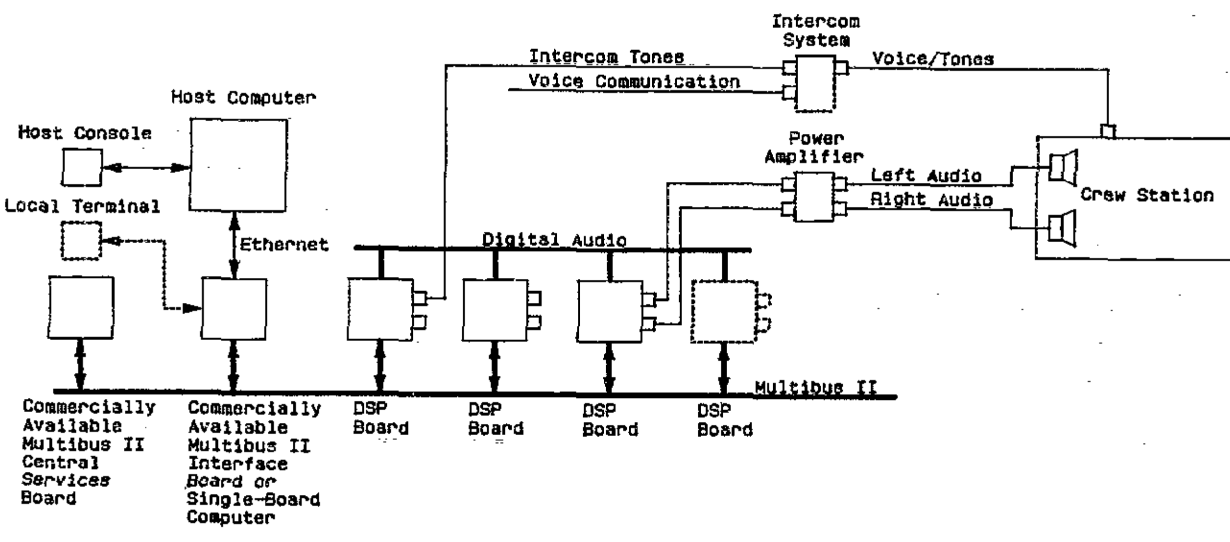

Figure 2. Implementation Of The DSP-Based Sound Simulator Using Ethernet and Multibus II

Ethernet® and Multibus® II.

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SYSTEM SOFTWARE

Developing application-specific software for this system involves a five-part process of data acquisition, data analysis, sound simulator programming, host programming and system evaluation. This section describes the methodology used to perform these tasks.

Data Acquisition and Data Analysis

Obtaining real-world data typically involves using a recorder to capture the sounds on magnetic tape. The recordings are analyzed, and the unique features of each sound are identified. These procedures are common to most sound simulator designs and are not detailed in this paper.

Programming the DSP-Based Sound Simulator

Software support for this sound simulator includes a library of special purpose and general purpose algorithms that can be used to create sound synthesis programs. Waveform generating routines, filter routines and modulation algorithms are linked to create programs that can synthesize the time and frequency domain parameters that have been identified during analysis of audio recordings. Constants, coefficients and data tables are assigned by the programmer to specify frequencies, amplitudes, phase relationships, sweep rates, bandwidths, and the shapes of periodic signals and modulation waveforms. Using the available software support, programs can be developed with considerable savings in time and effort.

After the sound generating programs have been created, they are grouped as a single library of sounds that represent the requirements of a particular training system. Files within this library can be linked with the firmware used by the digital signal processors, or they can be transferred to a host computer and downloaded to the sound simulator's random-access memory.

Programming the Host Computer

Software for the host computer is developed using information obtained from analysis of real-world data. The host is responsible for providing control data such as on/off commands, specific data such as engine RPM and intensity, or one-time requests for sounds of finite duration. Although data may be sent at regular intervals, this sound simulator does not require refreshing unless the status of a sound or group of sounds actually needs to be changed.

The host is also responsible for downloading the sound generating programs to the sound simulator's random-access memory if the programs are not executed in the sound simulator's firmware. This can be done as part of system initialization, or, if the number of DSP boards is to be minimized, the host can download programs on-the-fly to instantly change the sounds emanating from the DSP processors. As an example, consider an aircraft training system which, among other sounds, requires simulation of runway rumble and gun fire. Since these sounds are not required simultaneously, they can be downloaded to the same DSP processor as required by the mission. If a user would prefer not to download programs from the host computer, the shared processor concept is still available by linking multiple programs into the firmware of a single DSP processor.

System Evaluation

The performance of each sound synthesis program can be compared with information obtained from analysis of real-word data. If discrepancies occur, the particular sound synthesis program(s) can be edited to adjust signal parameters. This evaluation task can be performed with an alternate host such as a single-board computer. The Multibus II interface board shown in figure 2 can be substituted with a single-board computer to allow local interaction with the sound simulator system. If desired, this single-board computer can remain as a permanent intermediate host that can serve both as a bus interface and as a local evaluation/diagnostic tool. Using support software, the sound generating programs can be exercised, evaluated and modified as required with rapid turn around time.

CONCLUSION

Digital signal processors have been used to develop a sound simulator that can be programmed to numerically synthesize the sounds required by a particular training system. Adopting a DSP approach has eliminated the need for recurring hardware design while preserving system flexibility and growth potential. Using numerical methods to create sounds allows a programmer to control virtually any sound parameter in a predictable and accurate manner. Life cycle cost is reduced by increasing system reliability and reducing the time and effort required for system implementation.

ABOUT THE AUTHOR

Mr. Brian Leger obtained his B.S. degree in electrical engineering from Western Michigan University in 1985. He is a laboratory Engineer with McDonnell Douglas Corporation and has been working in the Flight Simulation Department of McDonnell Aircraft Company for two years. Mr. Leger is a member of the Acoustics, Speech and Signal Processing Society of IEEE.