ABSTRACT

In this article we will describe the architecture, performance and possible improvements to a digital sound synthesizer developed recently at Padua University.

INTRODUCTION

This work was being developed at C.S.I. (Centro di Studio e Investigazione) at the University of Padua, which is one of the most important centers in Italy concerned with scientific research in the field of Computer Music.

In the field of hardware research, one of the most important results obtained at C.S.I. in the "Mus 180" project and implementation of an "Elagatore sequencer" (ES) for real-time sound synthesis. This type of computer is designed to function as a slave computer of any other master computer, which must supply the E8 with data and instructions which constitute a sub-generation program.

From this research in the area of digital signal processing and synthesis, we have developed an architecture for a digital sound processor (DSP) with interesting characteristics. Within the past year we have assembled and debugged a prototype of this DSP. It consists of four standard boards with almost 280 independent circuits - TTL, CRT and high-speed CMOS. We have worked intensively on this prototype during the last few months and have obtained some interesting results, which will be illustrated later. Let us now explain something about the structure of this DSP.

DSP SYNTAX AND PERFORMANCE

Some of the characteristics of the DSP are:

1) eight voices polyphonic output, i.e. one can play up to 8 different notes simultaneously;
2) individually assignable timbres for each voice and dynamic allocation of the voices, this means that one can arbitrarily define a different timbre for each voice, and that any number of assignments can be set dynamically during synthesis;
3) stereo/dual stereo digital output to D/A converters;
4) 12 bit output wordlength for each voice; total output wordlength up to 15 bits for 8 voices. This wordlength can be extended to 10 bits in cases of cascading more than one DSP;
5) the synthesis network implemented at this time is one of the known techniques, we have implemented various versions of:
   - table look up;
   - waveform filtering, as used in analog synthesizers,
   - amplitude modulation,
   - ring modulation,
   - frequency modulation,
   - waveshaping;
6) internal pseudo-random noise generator;
7) 8 bit input bus with eight data lines, for easy control from any microcomputer;
8) possibility of tremolo, vibrato and portamento effects, independently assignable to each voice;
9) possibility of dynamic keyboard control velocity and aftertouch pressure;
10) possibility of multiple cascading of processors, without additional hardware.

On observing these characteristics, a large variety of applications is possible for the DSP. For example, it can be used as a simple sound generator (in musical organ-like keyboards), or as a base for complex programmable synthesizers, because it is possible to define each synthesis parameter. The DSP can also be utilized to build equipment such as sound interface devices for a peripheral of a personal or mini computer, or as a powerful sound channel in home computers.

The most noticeable difference between this processor and its predecessors is in its architecture. As can be seen in Fig. 1, the DSP contains...
tive independent units, performing different functions: a control unit and a synthesis unit. With this device it has been possible to minimize the external processors work load in the communication with the DSP.

In fact, this has been one of the greatest problems up to now for digital sound processors - i.e. the very high data rate required to perform synthesis.

With this DSP, the external processor only needs to send a short stream of data bytes each time a note is set on or off, or another command is required. The length of the stream is variable according to the type of command that is sent, and ranges from 5 to 32 bytes. In the latter case, all the synthesizer parameters are sent to the synthesis unit. Data leading from the master computer can be very fast, up to 300 nsec per byte, allowing direct access to a microcomputer data bus.

The function of the control unit is to unpack and, before input data. This work is held under the control of an integral microprocessor in the DSP. The sound parameters obtained are communicated to the synthesis unit. This choice allows microprocessor changes, for example if one wants to define other commands for sound synthesis or modification, only changing ROM contents.

Obviously there is also a synchronization logic between the control unit microprocessor and the external processor, as well as with the DSP's synthesis unit, in this way, synthesis parameters relative to one voice cannot be updated if this voice is in progress by the synthesis unit, easing row operation.

The synthesis unit performs synthesis algorithms under microprocessor control. The microprocessor has developed over the last revision according algorithm (or, in other words, has been made to include future procedures. Hence, it is possible to easily implement other algorithms like phase distortion, etc.

As an example of obtained results in sound synthesis, Fig. 4 shows waveforms and their spectra generated by means of a feedback FM algorithm, corresponding to various modulation indexes.

The synthesis microprocessor length is fixed, which allows for the generation of a fixed spacing clock dividing the master clock of the DSP. In the prototype, the sampling frequency is set to 20 kHz, with a master clock of 20 MHz.

A data RAM and a ROM, with a total capability of up to 12 kwords, for data tables has been provided in the prototype.

Because of the very high speed of the DSP, an internal interpolation logic has been implemented, so that waveform or distortion function tables can be very short. This choice allows sound quality without a large amount of memory. In Fig. 10 and 11, it is possible to see the resulting spectra of two 100 Hz and 1 kHz tones. In Fig. 12, the waveform is obtained by means of a waveshaping algorithm with a decade-like distortion function.

Tables for envelopes i.e. amplitude or modulation index envelopes - have also been provided. These are 16 point tables, with linear interpolation between two adjacent points.

The microprocessor also includes time-varying digital filters. At present, we have implemented the low pole IIR filter profile. Filter parameters are dynamically controllable by the master processor or by the DSP itself by controlling several tables. These digital filters ensure enhanced sound performance and are seen as a synthesis tool to avoid the wrap-around effect is overflow. We have also concluded and solved the zero sum limit cycles problem by injecting an appropriate digital noise.

In Fig. 13 to 15 it is possible to see the saturation effect on a sinusoid with an increasing filter gain. Some filtering characteristics are shown in Fig. 16 to 20. Here data are obtained by filtering themicrowave noise source (white spectrum is in Fig. 21).

The synthesis unit is the number coefficient capable of over 1 million of arithmetic operations per second at 20 MHz clock rate. It is adapted for working with sound synthesis and generating algorithms, and includes an A/D, multiplier, shifter, specialized registers etc. The internal arithmetic format allows the minimization of computing components.

Features such as requiring instead of truncating are also used. In order to obtain the highest level of performance, parallel processing, pipelining and specialized hardware structures are used intensively.

The final target of this work is to integrate the DSP in a line output module, which has been supported by the G.E.S. MicroAudioSemi-

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other controllers. Furthermore, the master processor usually needs RAM and ROM to store data and programs.

Each chip receives the stream for data input and the data stream from the master processor.

The clock is the same for all the chips. Every chip needs its own ROM with waveform tables. The chips are connected together in a daisy chain, so that the output of the last chip is the sum of the samples calculated by all the chips.

A digital post-processing is obviously possible before A/D conversion. After the post-processing (typically filtering or other effects such as equalization) the sound arrives at the output amplifiers.

**NEW DEVELOPMENTS**

During the development of this prototype, we have found some improvements to the DSP. For example, it is possible to increase the internal data format to 16 bits per voice, and the sampling frequency to 44,100 Hz. Furthermore, by providing the DSP with an I/O interface, it will be possible to process any compact disc standard digital signal.

In more general terms, the present and future research is concerned with the following topics:

1) Speech and singing synthesis:
   - Investigation of vowel synthesis methods
   - Evaluation of the results which one can obtain (i.e., speech and singing quality, number of contemporary voices)

2) Musical applications:
   - Modification and optimization of already implemented algorithms
   - Research on new synthesis methods

3) Real-time audio processing:
   - Investigation of the possibility to carry out audio signal processing such as line filtering for equalization and tone control, digital reverb, signal distribution among more channels, dynamic range signal compression and expansion, distortion etc.
   - Evaluation of the possibility of the use of an external RAM to obtain delays for the realization of reverberation, echo or other special effects (e.g., chorus, flanger)

4) Other applications:
   - Research on the use in the field of real-time process control
   - Applications in the telecommunication field

**CONCLUSIONS**

We have shown the characteristics and the architecture of a new digital sound processor. This DSP has been designed to be used in future musical instruments for the consumer market, as well as for computer music use. It is envisaged that in a short time the development of a silicon prototype will be completed.
Fig. 4 to 9 waveforms and their spectra obtained by means of FM feedback algorithms, corresponding to different modulation schemes.
Fig. 10 100 Hz sinusoid.

Fig. 11 2 kHz sinusoid.

Fig. 12 Diode characteristic obtained from a non linear transfer function stored in table.

Fig. 13

Fig. 14

Fig. 15

Fig. 16 10 kHz saturation effect in two pole JFET mixer.
Fig. 16 to 21 White noise filtered by a two pole IIR filter. Low pass, high pass, bandpass and all pass characteristics shown.