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The Digital Audio Signal Processor: Current Status and Future Trends

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Technical innovations in semiconductor devices and processing techniques have greatly improved the performance and functions of automobile audio equipment.
Not only does digital audio (CD and DAT) bring increased fidelity to sound reproduction, the digital signal enables the use of digital processing techniques to realize highly cost-effective audio systems.
Fujitsu TEN has developed a digital signal processing algorithm and a high-performance digital signal processor (DSP), the FT8800, for audio equipment.
Once analog sections of peripheral ICs have been implemented in digital circuits, they can be simplified and digital processing for high-frequency circuits made possible. A variety of new applications will thus be realized.

1. Introduction

Ten years have passed since the first digital signal processor (DSP) was developed. Early DSPs were used in communications systems having comparatively low sampling frequencies.

Rapid progress in CMOS VLSI technology has enabled high-speed processing, with instruction cycles of 100 ns or less. Presently, audio stereo signals are processed on a realtime basis.

Recent years have seen new features and functions added to audio equipment to improve functionality, convenience, and sales appeal. To provide these enhancements using analog circuits, a separate, dedicated, or custom IC is often required for each function, and subsequent modification is difficult and expensive.

One of the major advantages of the microprocessor is the ease and economy with which functions can be added and modified. With a microprocessor-based design, enhancements require a minimum of hardware change; the software is modified, instead. The DSP brings this advantage to audio equipment design.

The sections that follow report the development of the DSP and related techniques at Fujitsu TEN. They also describe the digital audio signal processor (DASP), an application-specific integrated circuit (ASIC) for audio application.

2. DASP development

2.1 Background

Table 1 outlines the progress in the use of digital techniques at Fujitsu TEN since 1975. Our purpose in the early stages was to replace mechanisms with wired logic or a microprocessor, i.e., to enable electronic control. The microprocessor proved to be flexible in meeting market requirements.

The microprocessor has a great impact on all industrial fields.

Fujitsu TEN first adopted microprocessor techniques for PLL synthesizer tuners in 1978. The microprocessor is now used in virtually all of Fujitsu TEN's products.

The microprocessor provides the following advantages when applied to audio equipment:

① Improved control accuracy (simplified mechanism)
② Increased design flexibility
③ Improved operating ease
④ Extended functions

Microprocessing techniques are also applied to audio signal processing, which now involves such varied techniques as equalizers which tailor the system frequency response to the automobile's interior, and sound quality control to produce the sound "flavor" appropriate to the music genre.

High-quality audio signal processing programs has also been developed.

Table 1. Introduction and progress of digital techniques

	Stage 1	Stage 2	Stage 3
	1975 to 1980	1981 to 1985	1986 to 1990
Purpose	Electronic control Operation improvement and indicator visibility	Size and weight reduction, performance improvement, interface circuit simplification	Function versatility and systemization, Digital processing for voice
Technique	Wired logic ① ② ③ ④	Microprocessor ⑤ ⑥ ⑦	DSP ⑧ ⑨ ⑩ ⑪
Application (product)	① AFC-synthesizer electronic tuner ② PLL-synthesizer electronic tuner ③ Microprocessor-controlled tape decks ④ Infrared emitting remote controllers	⑤ Single-chip integrated microprocessor-PLL ⑥ Single-chip microprocessor, A/D, LCD driver, and PLL (PLL *f*max: 150 MHz) ⑦ CD player	⑧ CD-ROM decoder ⑨ DAT ⑩ DASP ⑪ Sound field control with DASP

2.2 DASP development goals

The recorded music medium has evolved from analog to digital – the LP record and analog tape are rapidly being replaced by CD and DAT. This digital source data can be modified and manipulated by a microprocessor before D/A conversion. This makes efficient and cost-effective audio systems possible.

Figure 1 shows a DASP-based audio system in which the following development goals have been realized:

(1) Versatile functions and high performance (software control)
(2) Simple, compact hardware
(3) Digital user interface for individual devices

Table 2 shows the basic conditions for the DASP which realizes the above goals. Basic hardware development concepts are listed following the table.

Our basic hardware concepts involve:

① Simplifying the signal processing interface and control interface.
② Enabling versatile high-speed processing.
③ Providing high performance using floating-point operation.*1
④ Multichip configuration to enable versatile functions.
⑤ Enabling multichannel output (maximum 8 channels) (4 or 8 speakers).

We developed support tools in parallel with DASP development because DASP functions are performed by software (algorithm). (See "Digital Audio Signal Processor (DASP)" in this issue.)

*1 Floating-point operation: In operation in floating-point format, the length of the significant bit of the mantissa is maintained within the range in which the exponent can change, thus making operation highly precise.

Table 2. Basic conditions for DASP

1. Interface	① Analog input: Via A/D converter ② Digital input: CD, DAT, satellite broadcast ③ Digital output: Number of channels: Maximum of 8 (8 speakers) ④ User interface: Microprocessor controlled
2. Processing cycle time (realtime processing)	① Synchronized with external data cycle ② Digital input (*fs*): 32 kHz, 44.1 kHz, 48 kHz
3. Operation mode (assurance of repeat operation precision)	① Floating-point mode (18E6) ② Prevention of overflow/underflow
4. System configuration	① Multichip configuration ② Interface: EIAJ CP-340

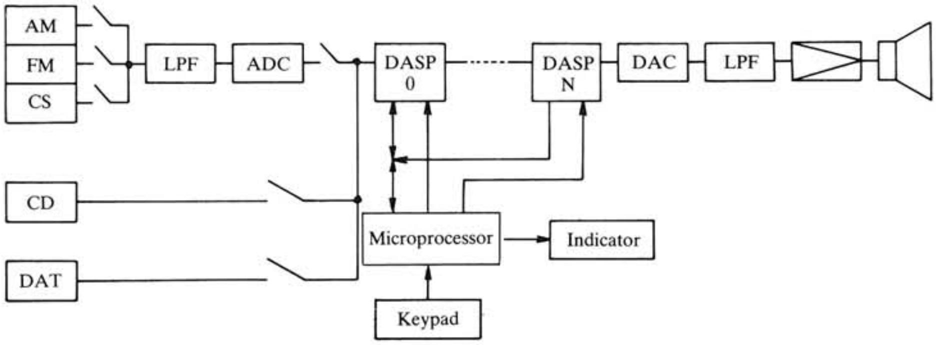

Figure 1. Audio system

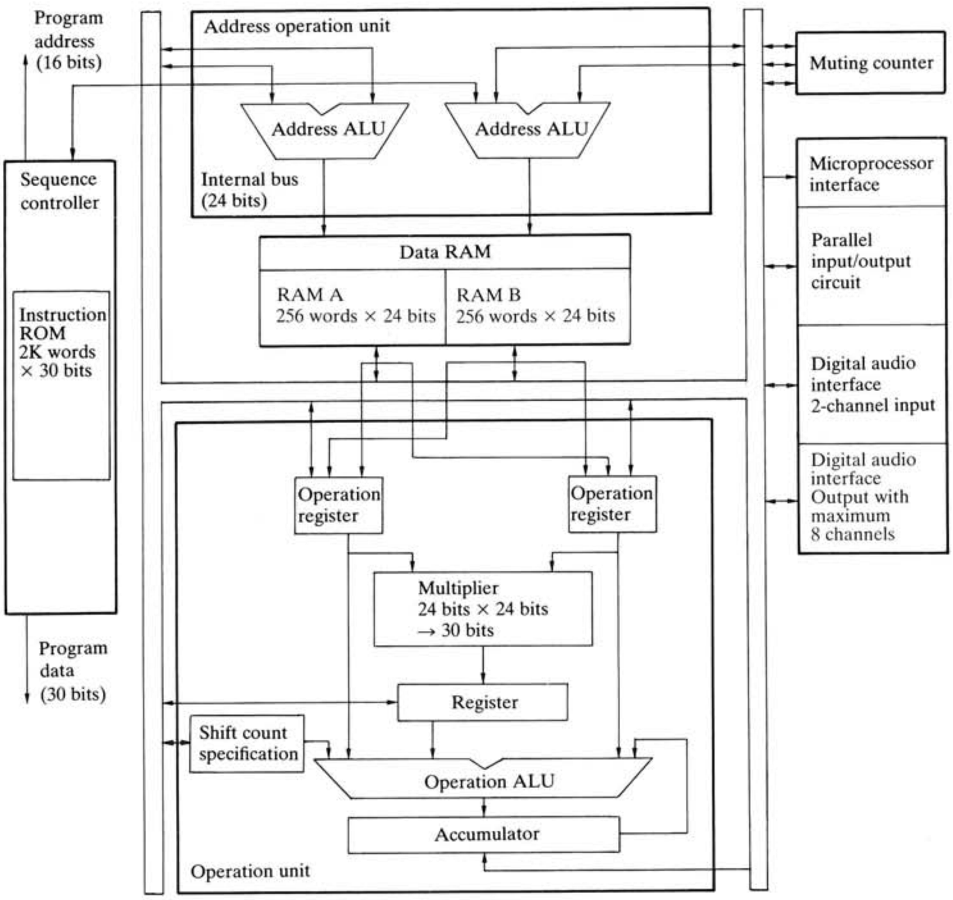

Figure 2. DASP

3. DASP application

Figure 2 shows the basic DASP configuration.[1)]

The DASP is a single-chip microprocessor which enables digital signal processing under software control. It contains memory for program storage and for data, and arithmetic operation and control circuits, An important feature is its multiplier. High-speed operation is possible with a pipeline architecture *2 using the multiplier and accumulator (adder).

*2 Pipeline architecture: Operations are divided into several steps, each of which is executed by different hardware.

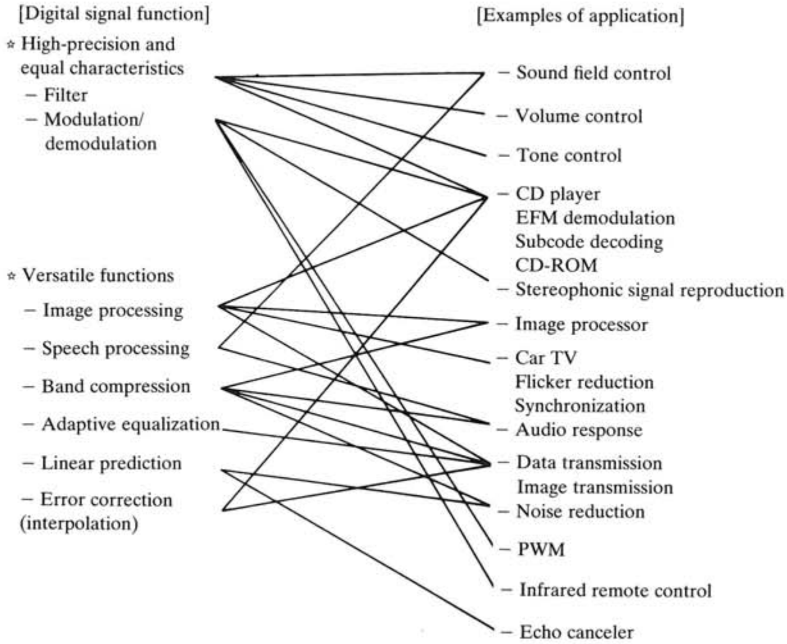

Figure 3. DSP functions and applications

The DASP enables the functions of following analog signal processing devices to be replaced by software:

- Modem consisting of resistors (R), capacitors (C), coils (L), and amplifiers.
- Filter
- Waveform shaper

High-precision circuits, e.g., adaptive filters, which are difficult to implement using analog techniques can also be implemented. Figure 3 shows some of the functions provided by the DSP, together with examples of application.

In conventional audio equipment, the most important object of audio signal processing was high fidelity. Implementing digital circuit in audio equipment has made it easier to obtain high-quality sources.

Next, to improve listening conditions within the confines of the passenger compartment (3 or 4 m^3), sound field control was used to create the illusion of being in a wider space, such as a large room or concert hall.[2]

When we listen to sound, we do not hear just the original sound, but echo, reverberation, and reflection in the space where the sound is transmitted. That is, the overall sound is created by the sound field (space) in which it is transmitted.

By analysis of the sound field of the performance space and the use of appropriate equalization, a sound field equivalent to a concert hall can be created in the passenger compartment. (See "Automobile Passenger Compartment Sound Field Control Analysis" in this issue.)

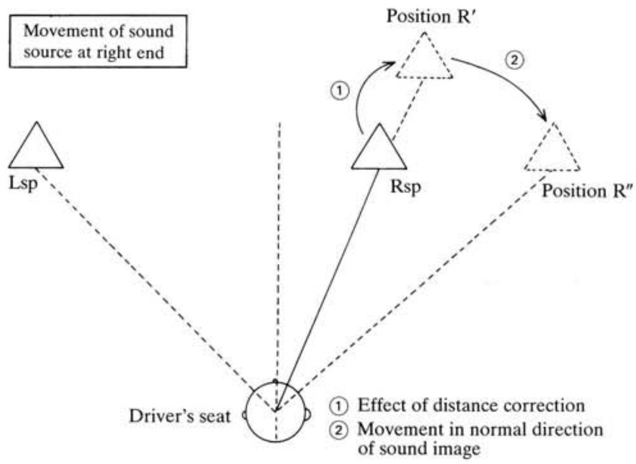

Figure 4. Sound image control

We developed the EQS-1000 sound field controller to reproduce presence without affecting tone quality. (See "Sound Processor (DASP Applications)" in this issue.)

In the passenger compartment, the positional relationship between the listening point and speakers is asymmetrical. The amount of delay, phase angle, and sound pressure level are controlled by the DASP, so that the sound from position R is heard from R" (Figure 4.) This is called sound image control.

Table 3. Major functions provided by DASP

Function \ Mode	A M	F M	TV (Sound)	Cassette tape	CD	DAT
Tone control	○	○	○	○	○	○
Equalization	○	○	○	○	○	○
Simulated stereo	○		○			
Sound field control	○	○	○	○		○
Sound image control		○		○	○	○
EFM demodulation, error correction					○	○
Noise reduction	○	○	○	○	○	

With sound image control, the acoustics of the passenger compartment are modified to simulate a listening room.

We will apply DSPs to the audio products as listed in Table 3. (circle indicates application)

4. Projected Improvements

If the analog sections of peripheral ICs have been implemented in digital circuits as mentioned above, analog circuits can be simplified and digital processing for high-frequency circuits becomes possible. For example, superheterodyne AM/FM tuner performance can be achieved by direct detection.[3]

The DSP is used for error correction for CD, DAT, as well as audio signal processing. CD use is expanding beyond music, e.g., CD-V video, digital road map data, and CD-ROM (Figure 5). DSP techniques are used to demodulate data signals, e.g., expanded adaptive differential pulse code modulation (ADPCM) and decoding of image coding.

The hands-free mobile telephone is desirable for driving safety. Howl generates by acoustic feedback from the speaker to the microphone in this system can be a problem. The adaptive digital filter changes the coefficient of the filter according to the acoustic environment and removes unnecessary signals. It

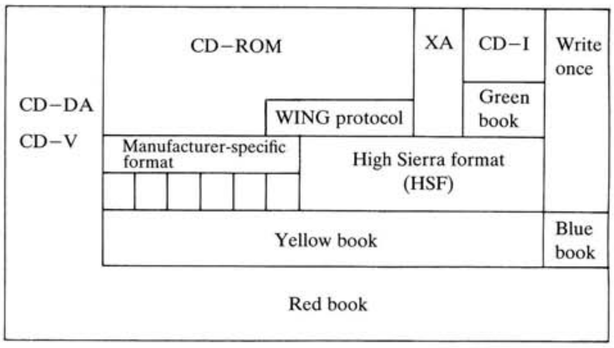

Figure 5. Expansion of compact disk applications

can be used for a howl or echo canceler.

One new application is in a millimeter-wave FM radar system used to sense the distances (forward and backward) from a car. It receives signals reflected from a target (obstacle) and calculates the distance from the target. The digital filter or fast fourier transform (FFT) can be used to detect the objective wave from reflected waves.[4]

As automobile equipment becomes increasingly complex and compact, it will come to require built-in self-diagnostics to enable practical maintenance. Such functions could be provided by the DSP's high-speed operation unit, and could conceivably involve neuroprocessors.

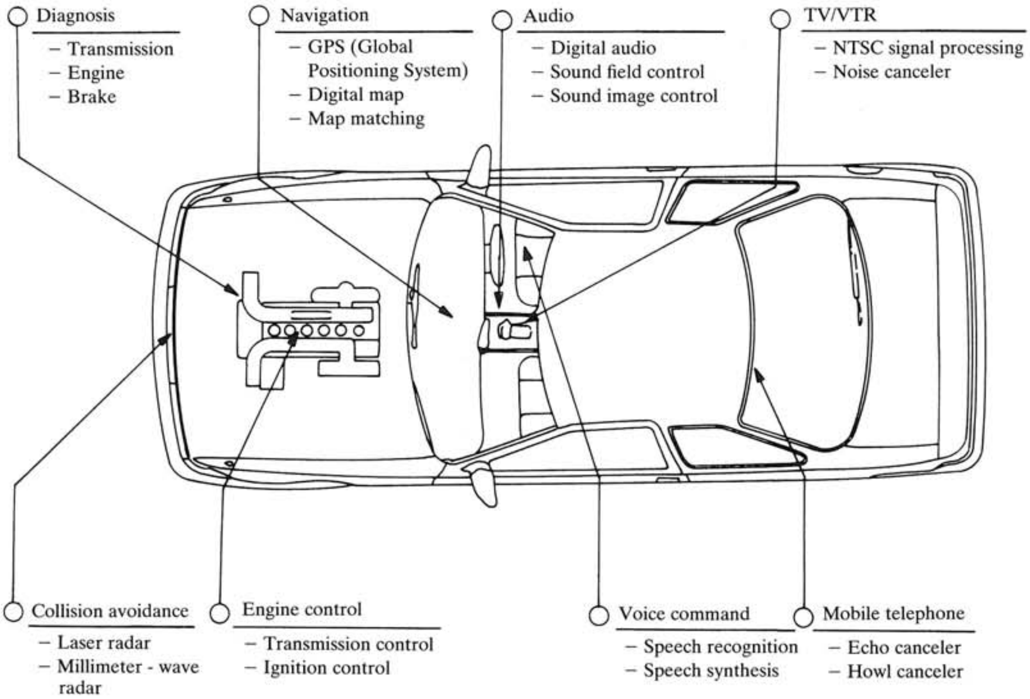

Figure 6. DSP applications in automobiles

Figure 6 shows some DSP applications in automobiles.

Because input/output conditions differ with the equipment, fixed circuits in the DASP will have to be converted to ASCICs to enable hardware and software easier to develop.

References

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