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Digital Audio Signal Processor (DASP)

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Fujitsu TEN has developed a digital audio signal processor which consists of a basic processing unit employing 24-bit floating-point operation (conforming to IEEE. 754), two types of serial data transfer circuits, and a muting coefficient register.
This configuration enables the direct transfer of two types of PCM data and external control required for the user interface.
This LSI makes it possible to design flexible and high-performance digital audio systems with simple configurations.
This paper reports the outline, characteristics, and applications of this LSI.

1. Introduction

In 1979, the first digital signal processor was developed by Intel. This was soon followed by devices from other semiconductor manufacturers.

The development of these devices and the progress of LSI technology expanded the area of realtime processing of digital signals.

In the car audio industry, digital audio sources are quickly replacing analog sources, and high-quality speech processing (Hi-Fi processing) is required to make the most of the quality of these sources.

To meet this requirement, Fujitsu TEN developed a processor dedicated to car audio signal processing. This digital audio signal processor (DASP) employs floating-point operation.

Our DASP has the following characteristics:

(1) Basic instruction cycle: 75 ns (13.3 MIPS)
(2) 24-bit floating-point operation
(3) Audio interface
(4) The signal processing system is completely isolated from the control interface.
(5) Large-capacity external data memory (64 kW × 24 bits × 32)
(6) Audio muting counter
(7) 28-bit parallel I/O port
(8) Multi-channel data transfer capability:
- 16-bit fixed-point data × 8 channels
- 24-bit floating-point data × 4 channels

2. Basic specifications

Figure 1 is a photograph of the DASP chip; Table 1 lists its basic specifications.

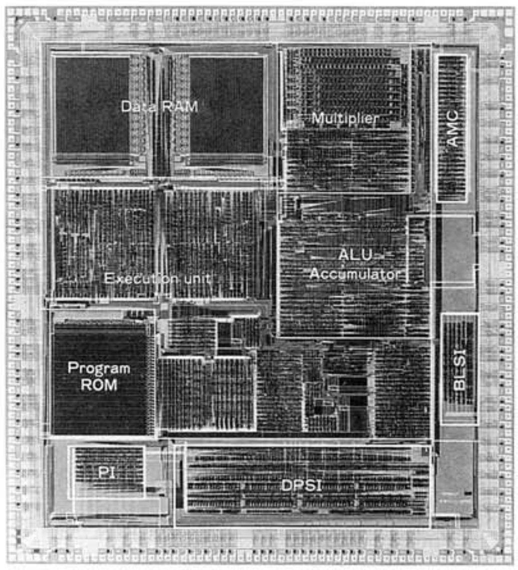

Figure 1. DASP chip layout

Table 1. DASP specifications

	Specification
Data format	24 bits (18-bit mantissa + 6-bit exponent → 24 bits)
Operation format	24 bits × 24 bits → 30 bits (result)
Bus	24 bits (both internal and external)
Instruction cycle	75 ns × 1 (basic instruction execution) 75 ns × 2 (floating-point operation)
Program memory	Instruction ROM: 2K × 30 bits (usable as data ROM)
Data memory	Internal RAM: 512 × 24 bits External: 64K × 24 bits × 2
Interface	Dual-port serial interface Binary low-speed serial interface Audio muting counter Parallel interface

The DASP consists of a core for operation execution and a peripheral data transfer circuits.

For sufficient precision for audio equipment, the core adopts a data format of 24 bits (18E6) (18-bit mantissa and 6-bit exponent). To increase operation precision, the length of the accumulator which holds the operation result is 30 bits (24E6). In general, the quality of digital audio processing is determined by the bit-length of the mantissa and the accumulator length. It is desirable that these lengths should be larger. However, the above word lengths were adopted considering the hardware scale and processing quality.

The instruction execution cycle is 75 ns. Most basic instructions are executed in one cycle; floating-point operations (except for division) are executed in two cycles.

To simplify the system configuration, the DASP incorporates the following functions:

- Two types of serial interface
- Parallel interface which enables separate input/output
- Audio muting counter (AMC) which prevents "popping noise" during algorithm switching

3. Numeric representation

The DASP uses two types of numeric representation: floating-point representation (18E6) conforming to IEEE, and fixed-point representation obtained by floating-point conversion.[1)]

For arithmetic operations, floating-point representation ensures operation precision. Figure 2 shows the numeric representation formats.

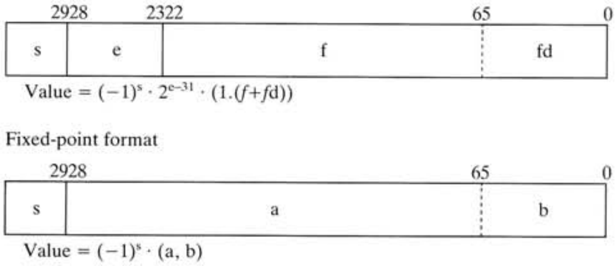

Figure 2. Numeric representation

The above figure shows a value in the 30-bit accumulator. When the value is transferred to memory, the 24 most significant bits are used.

Floating-point data representation consists of a 1-bit sign (MSB), a 6-bit exponent, and a 23-bit mantissa (fraction).

The exponent part uses excess-31 notation to make exponent's range non-negative.

4. Merit of floating-point operation

The DASP uses floating-point data representation for precise audio processing and a wide dynamic range.

Floating-point operation enables the precision of mantissa (fraction) to be maximized as long as the value is within the variation range of the exponent. (See Figure 3.)

As compared to fixed-point operation for the same data length, the precision of floating-point operation is superior except in the neighborhood of the maximum value.

Most audio signals are of small amplitude; the coefficient of the digital filter to process the signals is usually 1 or less.

For ordinary applications, therefore, most intermediate operation results are small values. To process small-amplitude signals or the rare large-amplitude signals, scaling may be used in fixed-point operation to ensure precision. With this method, the input value is multiplied by the coefficient at operation and is restored at output.

For floating-point operation, the algorithm can be simplified and signal quality ensured without complicated processing. In actual applications, the DASP obtains better results (characteristics, acoustic evaluation, etc.) than the DSP using fixed-point operation.

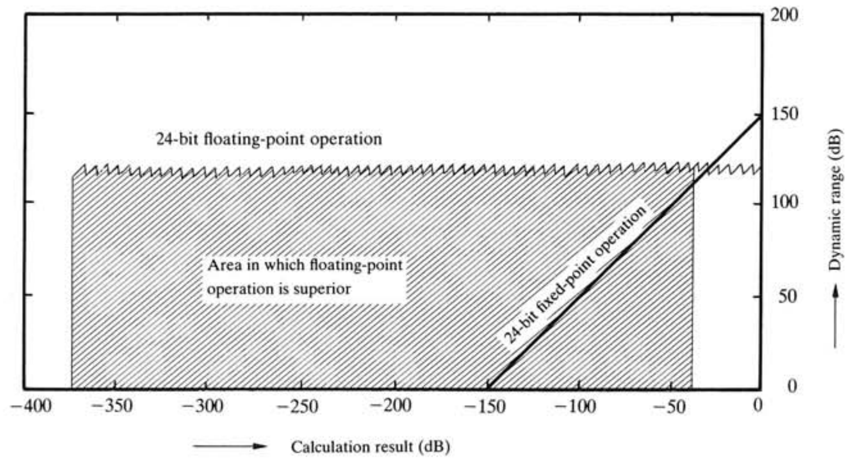

Figure 3. Comparison of floating-point operation and fixed-point operation

5. Dual port serial interface (DPSI)

To perform more high-speed and complicated signal processing, a multiprocessor configuration, in which two or more DASPs are connected in series or parallel, is used.

Such a system requires data transfer as follows:

- Transfer of BTC (fixed-point) data between A/D or D/A converters and DASPs
- Transfer of floating-point data between DASPs

This is because the high-precision operation result in floating-point format must be transferred to the next DASP without reducing the precision. This process aims at securing the same precision as that obtained when processing is performed by one chip.

Simultaneous transfer of multichannel data is also required to divide the signal processing smoothly and to effectively use the multiple speakers of the car audio system.

The DPSI contains two serial ports and uses the two data transfer modes and two clock modes shown in Table 2.

The following two clock modes are used:

- Single clock mode, in which data is output at the trailing edge of the clock and it is input at the leading edge
- Double clock mode, in which data is input and output at the leading and trailing edges

  In the double clock mode, twice as much data can be transferred with the same clock.

Table 2. Data transfer modes

Transfer mode \ Clock mode	× 1	× 2
Floating-point data transfer	24 bits × 2 channels	24 bits × 4 channels
Fixed-point data transfer	16 bits × 4 channels	16 bits × 8 channels

The DPSI enables multichannel input/output to allow a multiprocessor system to be constructed easily.

Figure 4 shows data transfer using the DPSI.

6. Binary low-speed serial interface (BLSI)

To construct an efficient audio system, it is desirable that the complicated processing required by the user interface should be separated from signal processing. Processing for each should be allotted to the microprocessor and DSP separately.[2)]

For this purpose, the DASP incorporates a binary low-speed serial interface (BLSI) for data transfer between the microprocessor and DASP.

The BLSI is a two-way interface which consists of commands, an address reception register (RXR), a data transmission/reception register (TXR), and three control lines. It is used to access the data memory in the DASP.

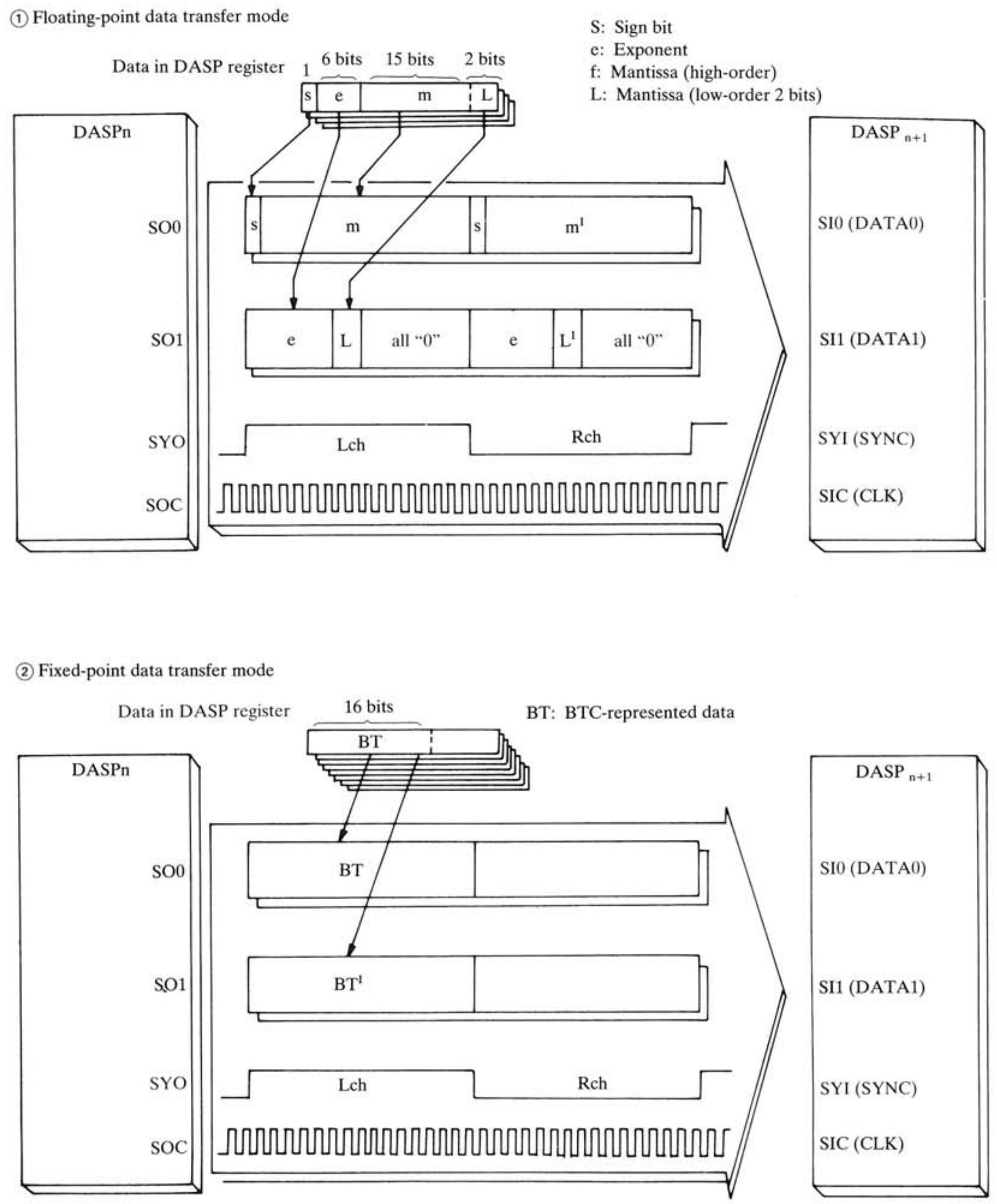

Figure 4. Data transfer using DPSI

Figure 5 shows the configuration of BLSI.

An example of a user interface using the BLSI is given below.

To change the sound volume or tone, the system operator transfers data from the microprocessor to the DASP and changes the data in the DASP.

The signal processing program is repeatedly executed in the DASP. The characteristics desired by the operator can be obtained because the data in the DASP changes the coefficients or processing algorithm used for processing.

Conversely, a spectrum analyzer or level indicator can be provided by using the audio data in the DASP.

As explained above, the internal program of the DASP provides the ideal user interface and efficient signal processing without complicated processing.

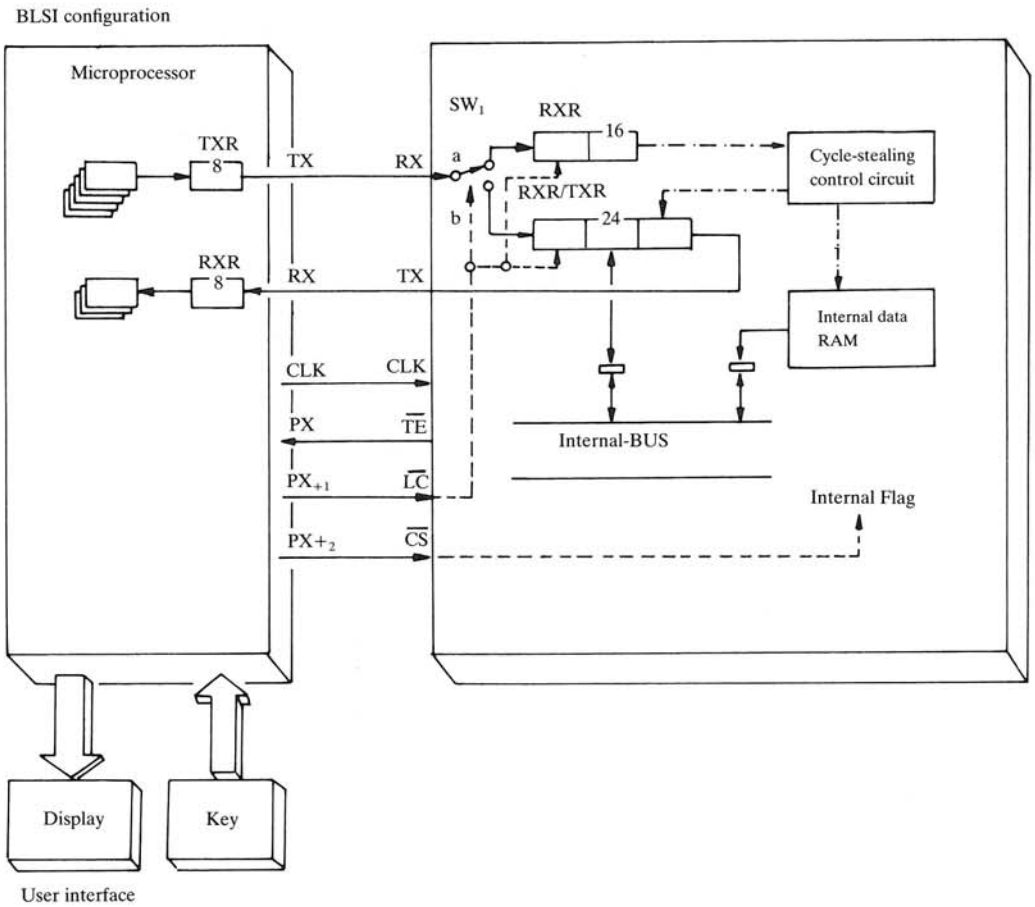

Figure 5. BLSI configuration

7. Audio muting counter (AMC)

If the algorithm is switched or a large amount of coefficients are rewritten during audio signal processing, discontinuous points occur in the output data. This is heard as a "pop."

To prevent this, the DASP is equipped with an audio muting counter (AMC). The AMC is initialized by setting three parameters (coefficient variation range, variation quantity, and increase/decrease direction) and the count mode.

Figure 6 shows an example of output signals when input signals (sine wave) are processed with the coefficients read from the AMC.

The AMC is automatically updated each time it is accessed, and its variation range need not be considered. Therefore, calculation of muting coefficients, which is performed by software in conventional systems, becomes unnecessary, thus simplifying the algorithm.

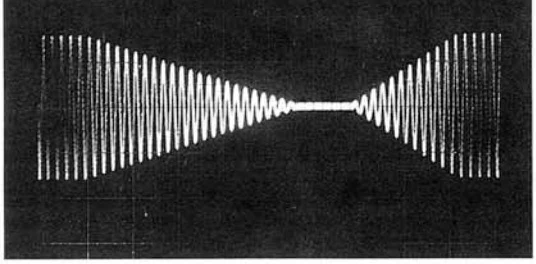

Figure 6. Example of audio muting counter operations (muting operation and release operation with sine-wave input)

8. Examples of system configuration

Figure 7 shows an example of connecting a DASP to an A/D converter and CD or DAT. Analog signals are converted to PCM and directly input by the DPSI. For digital audio sources such as CD or DAT, the PCM signals output from the source are also directly input.

Figure 8 shows an example of output to several D/A converters. Up to eight channels of data can be output by using both clock edges (leading edge and trailing edge). This function satisfies the requirement for multichannel output.

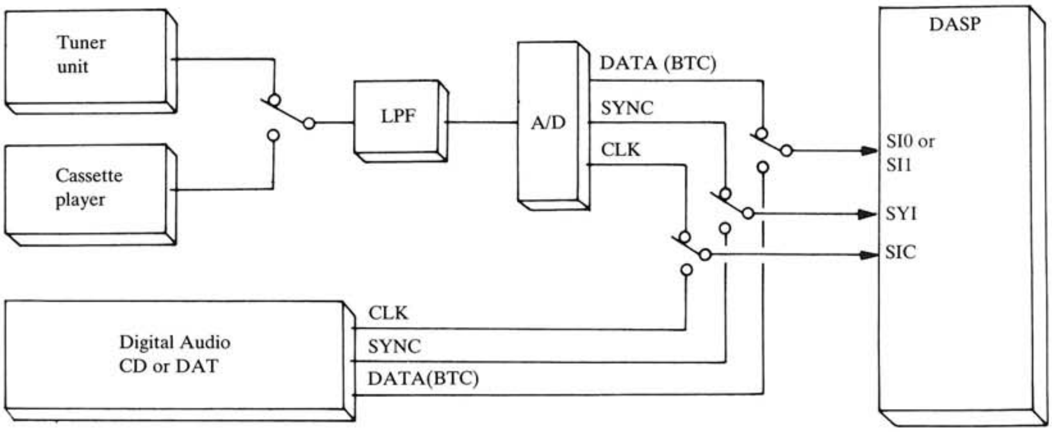

Figure 7. Example of connecting a DASP to A/D converter and CD or DAT

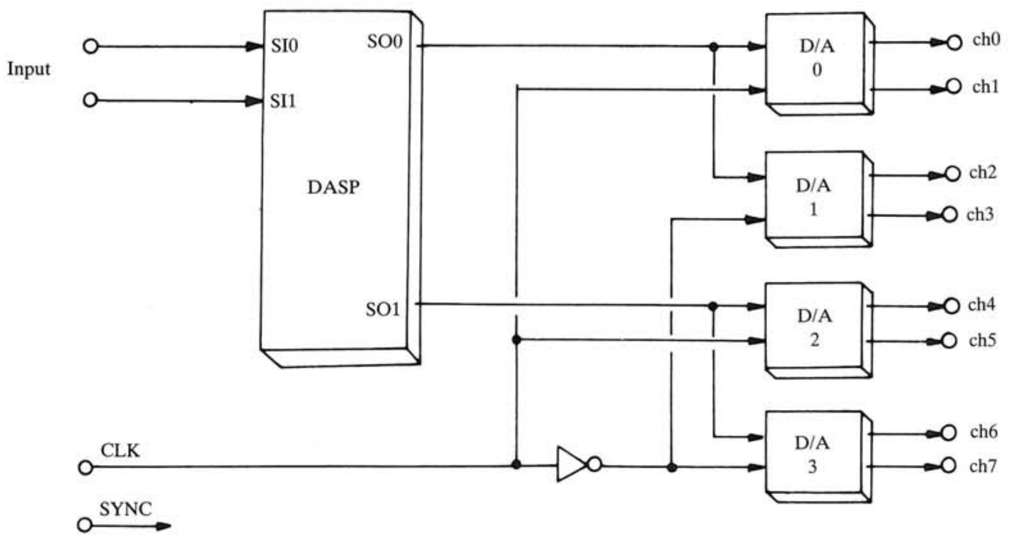

Figure 8. Example of connection to D/A converter

9. DASP chips

Figure 10 shows the DASP chip. The chip at the left is the prototype used for algorithm development. The chip at the right is the production model.

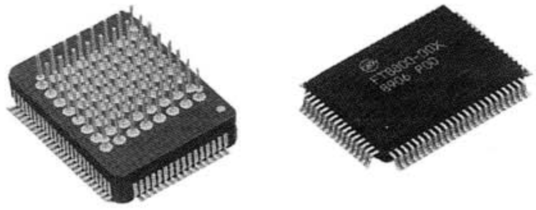

Figure 9. DASP

10. Development support system

Figure 10 shows the configuration of the DASP development support system; Table 3 lists the components of the system. The DASP development support system is classified into the following categories:

① Filter design program and simulation tool
② Cross assembler for development of signal processing program and emulation tool
③ Evaluation board for product development and ROM board for program
④ In-circuit emulator (ICE) to enable program evaluation

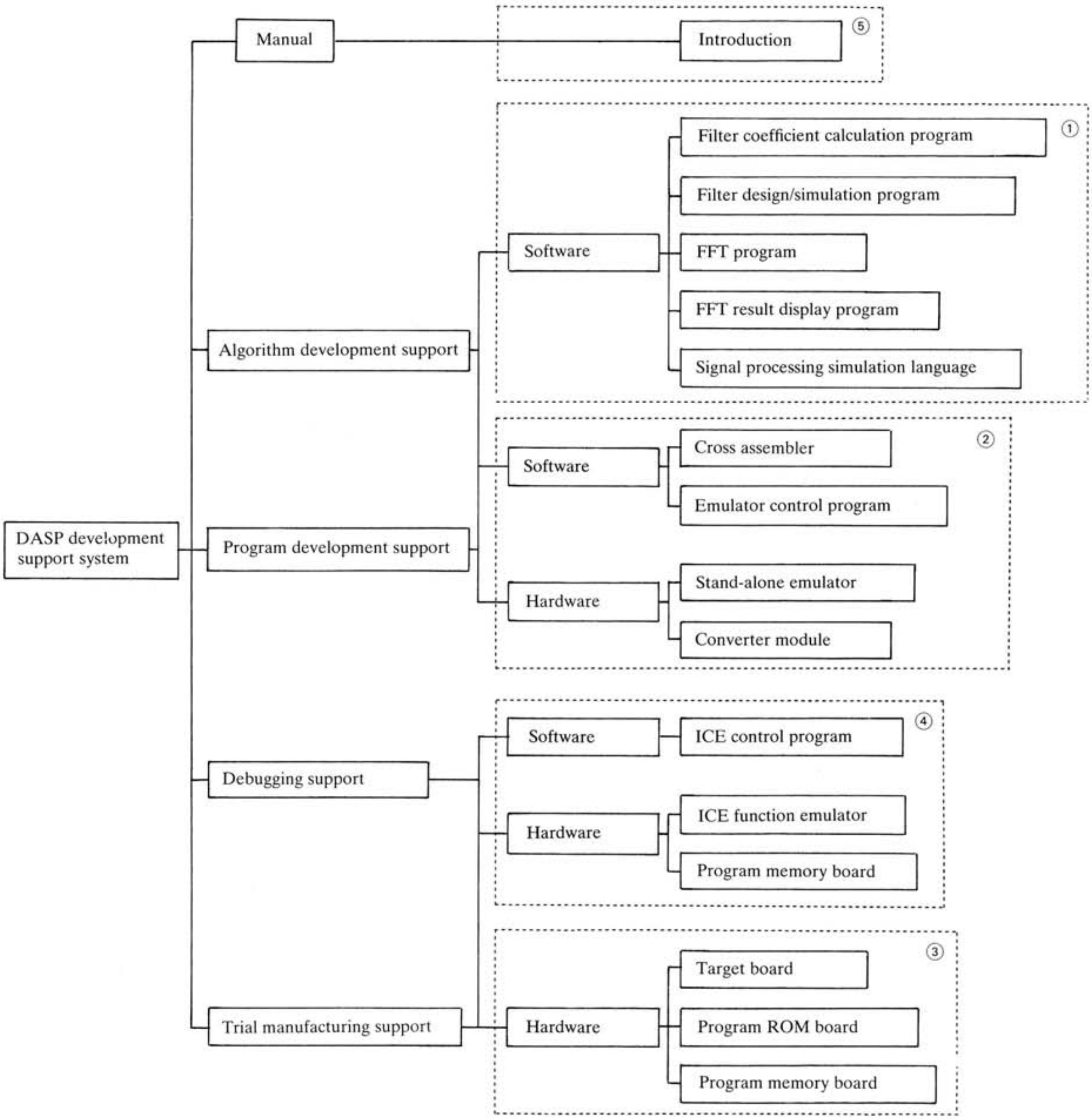

Figure 10. DASP development support system

Table 3. Components of development support system

Name	Description
Introduction	Manual which describes the functions and use of DASP
Filter coefficient calculation program	Prgram which calculates the coefficients of elliptic IIR filter and linear-phase FIR filter
Filter design/simulation program	Design and simulation program for Butterworth/Chebyshev IIR filter
FFT program	General-purpose FFT program for numeric data
Cross assembler	Cross assembler for DASP
Stand-alone emulator	System which includes two or more DASPs and their peripheral circuits (extended memory, program memory, ADC, DAC, etc.). It is mainly used for audio application development.
Emulator control program	Program which controls the stand-alone emulator and supports DASP program execution, debugging, and user application development.
ICE function emulator	Controls only the debugging terminal of DASP chip. Peripheral circuits prepared by the user are used. Program memory board is required. This emulator can be used for various applications.
Program memory board	Connected to debugging terminal of DASP chip. PROM, EPROM, and RAM can be mounted.
ICE control program	Control program which controls ICE
Target board	Small board for DASP evaluation on which masked ROM and program ROM board are used
Program ROM board	Program memory board for PROM

⑤ Manuals for DASP and tools

DSP software can be developed efficiently by using this development support system.

The emulator consists of the following components:

- Main unit in which two DASP chips are mounted
- Converter module which incorporates A/D (ADC) and D/A converters (DAC)
- Emulator control program

Figure 11 shows the emulator. Using the emulator, the signal processing program for DASP can be run and audio signals processed and evaluated.

Figure 11. Emulator

(1) Configuration

The emulator includes DASP chips and all peripheral circuits required for operating the chips. The emulator enables software development in parallel with equipment design.

For general use, the two DASPs are cascade-connected. Application development with one-chip or two-chip configuration is possible.

Figure 12 shows the block diagram of the emulator.

The emulator contains the following circuits:

- Clock circuit which generates DASP master clock
- DASP control circuit
- External data memories, external program memories, and their peripheral circuits
- ADC
- DAC

The emulator is externally controlled by a personal computer.

(2) Functions

Table 4 lists the emulator functions. Various audio functions can be developed by using these functions.

(3) Control software

The emulator software controls the emulator and enables efficient software development.

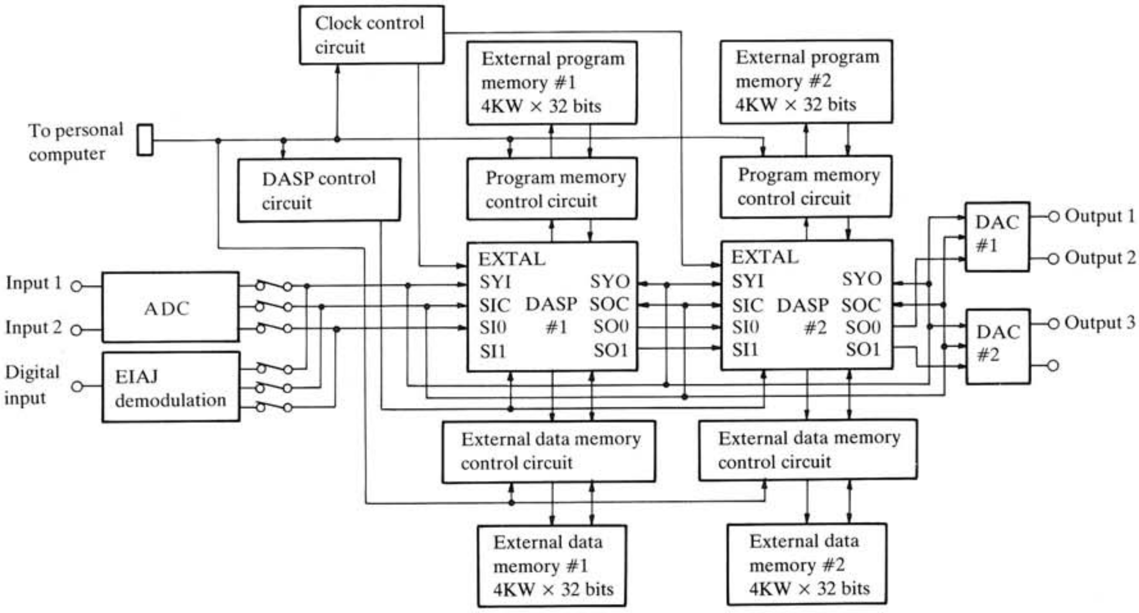

Figure 12. Block diagram of emulator hardware

Table 4. Emulator hardware functions

Function	Description
DASP control	Halt, step execution, continuous execution interrupt control, master clock control
DASP internal register access	Operation registers (A, B, D, P, SFT, SP), address operation registers (B0, B1, X0, X1, PGI, VSM), special registers (MOD, ST, MASK), peripheral registers (BLSI, DPSI, AMC, PI) Read/write operations for the above registers
Memory access	Read/write operations for external program memories, DASP internal memories, and DASP external data memories
Input	Analog input: 2 channels Digital input: 1 system (coaxial)
Output	Analog output: 4 channels
Sampling frequency	Optional for 48 kHz or 44.1 kHz

The DASP software is developed interactively by the designer and the development support system (control software) using a personal computer. Therefore, it can be said that this interface determines the system quality.

To provide an easy-to-use system for software designers, priority was given to the user interface.

For example, multiwindows and pull-down menus were adopted for comprehensible and easy operation. Figure 13 outlines the operation of the emulator software.

This system enables basic design and simulation of DASP programs, emulation using the chip, and characteristic evaluation. The powerful user interface increases the efficiency of software development.

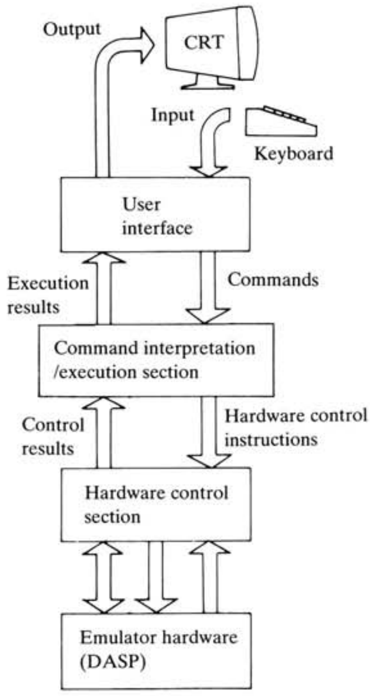

Figure 13. Program support function

11. Applications

As explained before, the DASP can be applied to general audio processing, sound field control, active control for noise, and so forth.[3),4)]

For use in a narrow processing band (e.g., mobile telephone), many operations can be performed with a single chip.

Since the DASP provides parallel input/output (also used for external data memory ports), it can be applied to signal analysis, such as diagnosis (diagnostic system) and control.

12. Conclusion

We discussed the outline, characteristics, and applications of the DASP and the DASP development support system.

Using the DASP simplifies system configuration because converters and the interface circuits between DASPs are not required.

Since floating-point operation is employed, high-precision processing is possible and characteristic deterioration with signal processing is small.

Stability with temperature change and aging can be ensured; the same hardware can be used for various purposes depending on the software.

For the above reasons, the DASP is suitable for car audio equipment, which must be compact, lightweight, reliable, and environment resistant.

Although the DASP has many possibilities, no systems can be realized without algorithm development.

We are developing generic control algorithms for audio use and sound field control and sound image control algorithms which enable stereophonic and natural sound to be reproduced in the car. We are confident that these algorithms will improve the quality of car audio equipment.

References

1) C. Ashton: "Designer's Guide to Floating-point Processing," *EDN*, 1986, 1/9, Vol. 31, No. 1, 115-121.
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Joined the company in 1978. He is currently engaged in the development of car audio products and related electronic equipment in the Research and Development Department.