PORTABLE DIGITAL AUDIO PROCESSOR FOR USE WITH HOME VCR

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Abstract

Described is the development of a new digital audio processor weighing only 3.1kg and designed for use with a home VCR for digital audio recording. While performance is the equal of conventional digital audio processors, this unit is highly portable and easy to operate. Technical features of this unit are simplified signal flow, a newly developed AD/DA converter using a new concept in 14-bit linear quantization, and a newly developed multi-function, high density N-MOS LSI.

1. Developmental Background

For both professional and home applications, we can see a markedly increased demand for high performance audio recording equipment. While this is in large part responsible for the trend towards digital (or PCM) recording, the high cost of present stationary head 16-bit multi-channel systems and rotary head type 16 and 14-bit 2-channel systems has stood in the way of digital recording's popularization. With the introduction of the digital audio disc (Compact Disc), we can expect even greater demand for the digital equipment required for master recording, editing, cutting, and duplicating.
It is against this background that we began development of the SV-100 digital audio processor designed for use with a VCR to offer the recordist high quality results at a modest price. Main specifications were established as follows.

A. To meet home use requirements and be suitable for professional applications, quantization is linear 14-bit.

To assure frequency response of up to 20kHz, sampling frequency is 44.056kHz. An additional benefit of this sampling frequency is that it is very close to the 44.1kHz used by the Compact Disc.

B. Signal format (including the above) is the EIAJ standard (STC-007). This format has been proposed to the AES, IEC, and other organizations and it offers compatibility with much other equipment already in use.
A VCR such as VHS or ß (beta) is used for recording; these machines are popular and widely available at a modest price.

C. For in-the-field recording, the unit must be truly portable with such characteristics as light weight, compact size, and battery operation.

2. Description of Technical Developments

Since conventional design methods and circuit devices were not suitable for our purposes, we divided our efforts between overall planning, circuit design engineering, and development of semiconductor devices.

2-1. Signal Flow Simplification

A digital audio processor has three basic functions. First, the recording function which converts the audio signal into digital form and then into the standard video signal format supplied to the VCR. Second, the playback function which converts the VCR's video signal into digital form and then into an audio signal. And third, the monitoring function which permits recording level setting and monitoring. These are shown in the figure-1 signal flow diagram.
We may note that the recording section and reproducing section do not operate at the same time and that low pass filters are used for both recording (to suppress unwanted signals in sampling) and playback. The monitoring section is required during both recording and playback. Through systematic combination of similar functions, we were able to simplify the signal path as shown in the figure-2 signal flow diagram. Here a single low pass filter is used for recording and playback, the AD and DA converters are combined in an integrated AD/DA converter, and the monitoring section is designed to operate

throughout the recording and reproduction process as well as during digital copying (for monitoring of master tape playback, as shown in the diagram). All this contributes to considerable size and weight reduction, not to mention reduction in the number of parts.

2-2. Combined AD/DA Converter

AD and DA converters represent some of today's most advanced electronic technology. Even for an experienced manufacturer, it is a difficult job to develop an AD converter having a conversion time of 10μsec or less and 14-bit resolution of the kind required for this digital audio processor. The variety of available converters is limited and they are surprisingly expensive. Furthermore it seems a waste to have to use separate AD and DA converters where only one or the other is working at a time.

Fortunately, we have experience in the development of these devices and by switching some of the circuitry, we were able to develop a new "combined AD/DA converter" (MA6196) capable of performing both tasks.

Figure 3 shows a block diagram of this combined AD/DA converter with its high speed 14-bit DA converter, high speed comparator, and successive approximation register logic (SAR). These are, of course, the essential components of an AD converter and figure 3 shows signal flow during AD conversion.

For DA conversion, switching takes place at two points. The digital input for DA is input to the SAR; it becomes 14-bit parallel data and is supplied to the DA converter, then is output as an analog signal. The comparator is not used in this process, but other components are employed for optimum efficiency.

This combined AD/DA converter achieves the outstanding performance specifications as shown in table 1, yet it is contained in a very compact 24-pin package, thereby contributing to this unit's compact dimensions.

2-3. LSIs for Digital Signal Processing

Three new LSIs which we developed are used in this unit. All are of the N channel MOS process variety. Details are listed in table 2.

EIAJ standard recording and playback signal processing is performed using these LSIs along with high speed random access

memory, and a number of TTL logic ICs. If we did not have these LSIs and had to do everything with conventional TTL logic ICs, it would take about 700 ICs to achieve the same functions and performance. Needless to say, this would mean so much space, weight, and power consumption that the unit would be disqualified as a portable recording system.
Figure 4 shows these LSIs. These three chips are equivalent to approximately 700 conventional TTL ICs.

2-4. Waveform Equalization and Microprocessor for Automatic Adjustment of Data Slice Level

During reproduction, digital data must be derived from the VCR's playback video signal. This is the job of the video demodulator in figure 1 and figure 2. This is an extremely important part of a digital audio processor since the demodulator's performance greatly affects playback stability and noise suppression. Here we come to the problem of the condition of the VCR and cassette used for recording and playback. Quality, type, and maintenance will all contribute to big differences in video playback signal quality. Specifically, we are faced with variations in waveform reproduction, signal-to-noise ratio, voltage reproduction, and dropout noise, to list the most obvious.
To deal with this reality, we have employed waveform equalization and a dedicated microprocessor in a circuit which automatically adjusts slice level to the optimum.
Waveform equalization depends on a delay line, the basic technology of which is employed in a number of fields. However, to the best of our knowledge, this is the first case of application in a digital audio processor. This greatly reduces the kind of waveform distortion called "intersymbol interference" encountered in data transfer, so data retrieval is dramatically improved.
In the EIAJ format, the digital information is contained in the picture signal component of the VCR playback video signal, as shown in figure 5. In this example, we have a typical VCR playback video signal. Due to waveform distortion and noise, the "eye" gets smaller at points A and B. If the data slice level is as shown in this example, then PCM data can be retrieved without error. However, if the data slice level is beyond the upper limit or lower limit, then errors will appear at point A and point B, thereby degrading reproduced sound quality.
To maintain data slice level at the optimum, we developed a new

microprocessor (MN1400PCA) especially for this purpose. This is shown in figure 6. By using the CRC to count PCM data error rate, this microprocessor finely adjusts slice level thirty times per second to minimize errors.

2-5. Recording Level Indication

Recent digital audio equipment sometimes suffers from the lack of proper level indication; there is nothing more than an "over" indication when input level exceeds the digital clipping level. In digital audio, the maximum signal level that can be handled is determined by the number of bits; clipping occurs when this level is exceeded. This can be compared to the situation with an amplifier using negative feedback. If clipping is low and for a brief enough time, then there will be little noticeable effect on sound quality. In other words, clipping is allowable in practice, but one must be cautious about it. The problem with having no more than an "over" indication is that one cannot determine the degree of clipping. Even if the situation is such that one can make another recording, there is no way of knowing how much to reduce the recording level.

To avoid these dual drawbacks, it is obviously necessary to have an indication of input signal level that extends above the digital clipping level. This permits correction during recording and provides a reference for setting recording level at a more suitable value if recording is to be performed again. How far such an indication should extend depends on the unit's performance and application, but in the case of the SV-100 we decided on +6dB. The +6dB value is equivalent to one bit more than 14-bits, in other words, 15-bit full swing level.

2-6. Recording Level Adjustment

Recording level adjustment must be performed accurately, and in many situations it must also be performed quickly. For this purpose, the SV-100 has a master level control and a stereo balance control. This rather unusual system was chosen because it is the most practical for most purposes, those being recording FM broadcasts or (in the near future) from Compact Discs. In both cases it is easiest to adjust the left and right channels at the same time; little if any stereo balance adjustment is needed. The same goes for live recording where the left and right channel microphones are normally matched in

the first place; after adjusting balance, one can simply adjust both channels to the optimum level at the same time. If using more than two microphones, then the mixer's faders are normally used to achieve the proper left and right channel balance.
It is obvious that having separate level and balance controls is the most rational approach; and it is in fact the standard for stereo amplifiers. Nevertheless, most tape decks use friction coupled left and right channel controls. These are convenient if no balance adjustment is required; but when it is, then both hands must be used and it frequently occurs that balance actually changes when level is adjusted afterward. But the biggest problem is when one tries to fade out at the end of a recording; one channel is completely attenuated before the other.
The second most common system is to have completely separate left and right channel controls. This makes it hard to achieve proper stereo balance in the first place and requires both hands to operate. This is suitable for data recorders which handle completely different signals on each channel but not for audio recorders, especially for home use or portable applications.

3. Portability

For use in the field, this unit operates on a rechargeable lead battery pack. This 12V 2AH battery is the same as that used in the matching portable VCR. A switching regulator supplies four voltages, +12V, +5V, -5V, and -15V, to the various sections of the unit. To avoid interference from the switching regulator, its oscillator frequency is set to match the PCM sampling frequency.
Since all parts are very small and lightweight, the total weight of the unit is only 3.1kg, on a par with today's lightest portable VCRs which weigh around 3kg. In combination, this unit and a portable VCR form a more compact, lightweight, and portable digital recording system than has previously been available.

4. Other Major Specifications

PCM Standard ----------------- EIAJ STC-007
Quantization ----------------- Linear 14-bit
Decoding -------------------- Linear 14-bit
Frequency response ----------- 2Hz ~ 20kHz (±0.5dB)

Total harmonic distortion ---- 0.01% or less (1kHz, 0dB)
Dynamic range ---------------- Better than 86dB

5. Conclusions

The field of digital audio is progressing at a rapid pace. With the introduction of the Compact Disc, digital audio will become commonplace in the home. In turn, this exciting new music source will most likely stimulate the demand for high quality recording equipment. Since it is designed for use with the already popular home VCR, a digital recording system of the type described here meets such demands.

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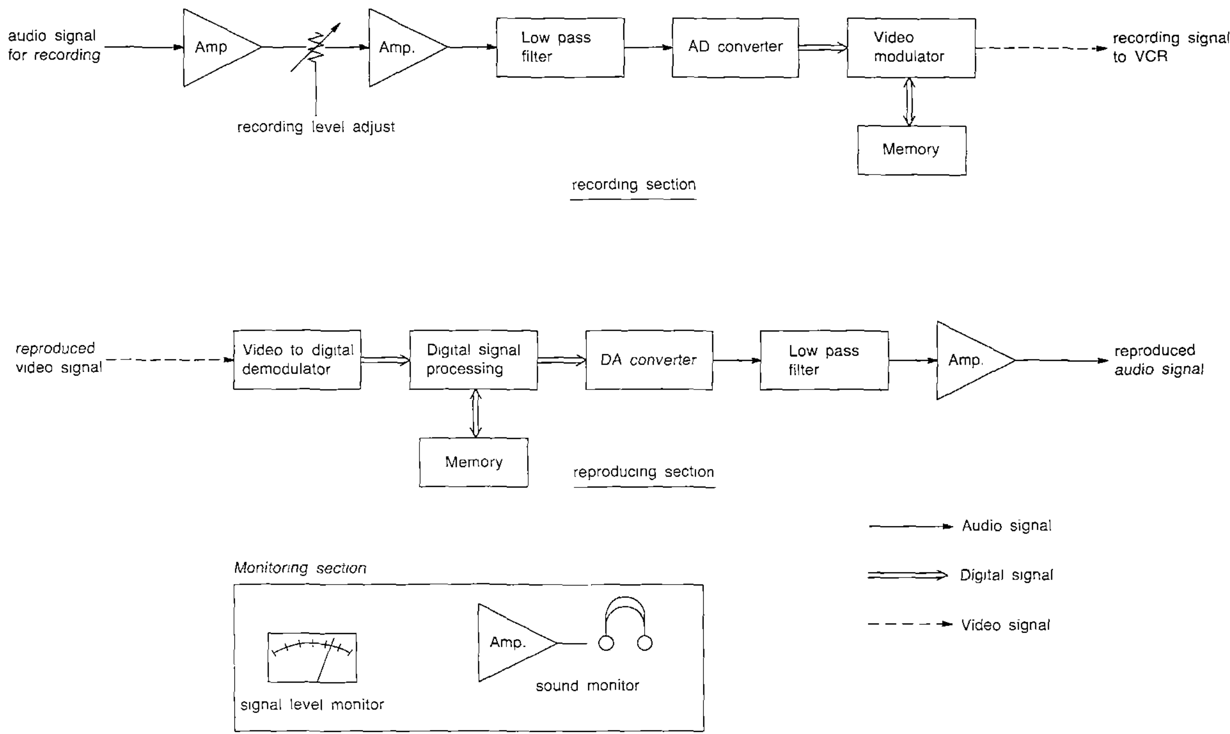

Fig. 1 Signal flow diagram of general recording processor

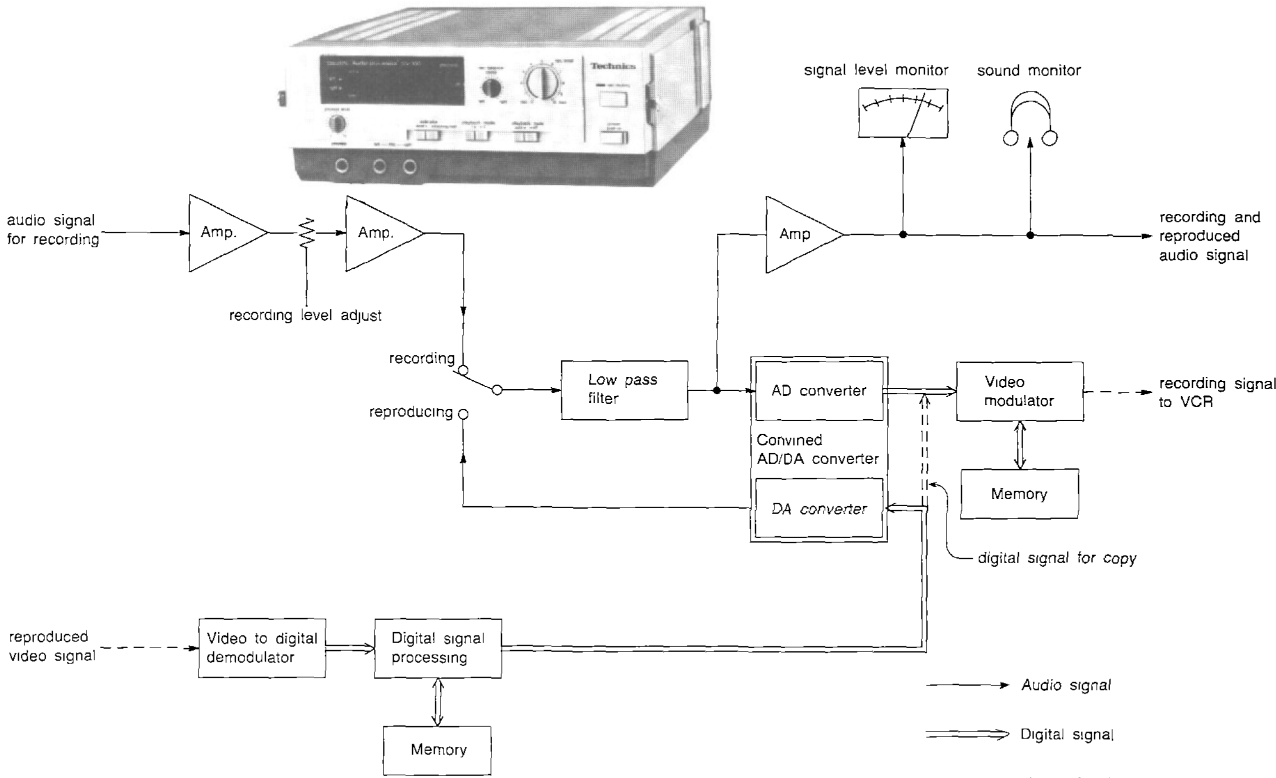

Fig. 2 New signal flow diagram of SV-100

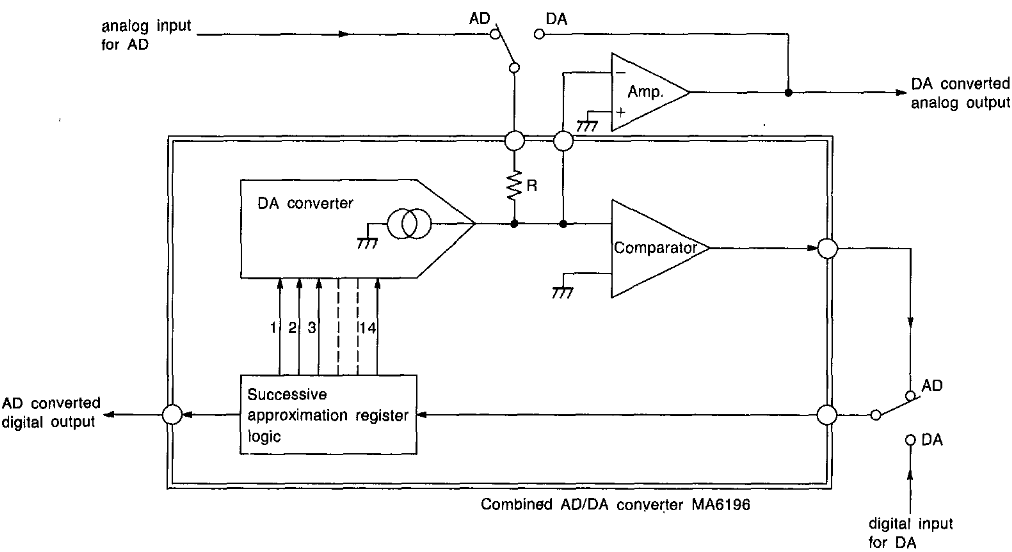

Fig. 3 Combined AD/DA converter and the application

MN6601 MN6602 MN6603

Fig. 4 LSIs for digital signal processing

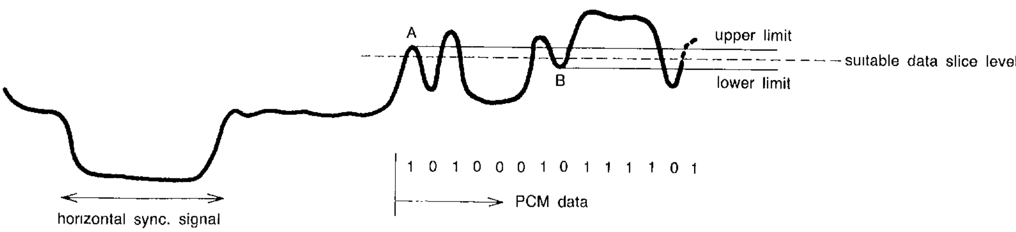

Fig. 5 Reproduced video signal wave form from VCR and data slice level

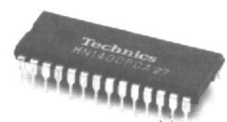

Fig. 6 Micro processor MN1400PCA
for automatic data slice level control

	AD operation	DA operation
resolution	14 bits	14 bits
error	1/2 LSB	1/2 LSB
conversion time	8.5 μsec	5.5 μsec
analog signal	±5 volts	±2 mA

Table 1 Basic Specification of MA6196

name	transistors	pins	purpose
MN6601	about 10000	64	recording processing data interleave CRC generation master timing generation
MN6602	about 15000	64	reproducing prossing data deinterleave error correction timing generation
MN6603	about 6300	28	recording and peproducing control AD/DA conversion control clock signal generation recording or reproducing selection self test signal generation

Table 2 Basic characteristics of LSIs