DIGITAL AUDIO AND DATA TRANSMISSION SYSTEM FOR CATV LINE

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ABSTRACT

There is an increasing need for digital data transmission system using CATV line today. With the above in mind, we have developed a system which is able to transmit digital data of approximately 7.4 MBPS using a frequency bandwidth of 6 MHz (equivalent to one arbitrary TV channel), and which can also be connected to currently used CATV system without any alteration.

2-level VSB transmission method is employed for this system because of its suitability for the various characteristics of CATV system and simplicity of instrumentation in paticular at the receiving side. The system is also provided with a very flexible data format, enabling a wide application in designing the system.

The system enables to simultaneously transmit four ultra-high-fidelity stereo audio programs, as well as computer and game software, facsimile data, still picture, etc. to all or specified subscribers.

1. INTRODUCTION

A system that enables to offer a wide menu of new services by use of current CATV line is in demand.

Time division multiplex digital data transmission can be considered as one of the method well-responding to the above. The method is advantageous because of its flexibility in multiplexing various kinds of signals. It is also advantageous because the signals are seldom degraded by noise or distortion. This method is therefore well-suited for the multi-channel-broadcast of ultra-high-fidelity digital audio programs when a large amount of transmission capacity can be obtained.
Some of the data transmission method for CATV system has already been considered. However, most of them, for example, the method which uses blanking period of TV signal, are difficult to get large capacity and have only a restricted application.

Taking the above conditions into consideration, we have developed a wholly new style of data transmission system for the current CATV.
In this system a frequency bandwidth of 6 MHz which is equivalent to one arbitrary TV channel is used for data transmission, and digital data of approximately 7.4 MBPS can be transmitted by use of a flexible data format which can effectively process both audio and non-audio data and realize addressable function. The system has been prudently considered about the suitability for various characteristics of CATV, and is able to use in currently used CATV without any alteration. The receiver unit can be reasonably implimented concerning both the cost and the hardware size.

2. SYSTEM STRUCTURE

The captioned system is composed of a transmitter with a system control computer and home receiver-converters.
The transmitter, which are connected to Head End of the current CATV system, also includes, a modulator, a frequency converter, and A/D converters if necessary.
Although the level of the system control computer will depend on the contents of application, micro-computer can be generally used, giving sufficient performance.
The receiver is connected to drop-line from a tap-off of CATV line at each subscriber-sites.

Figure 1 and 2 show a photograph of the experimental system and an illustration of the system concept, respectively.

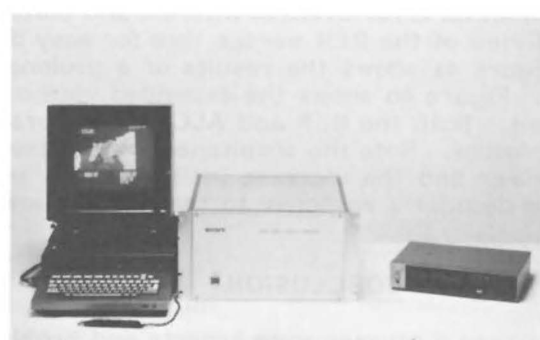

Fig. 1 : EXPERIMENTAL SYSTEM

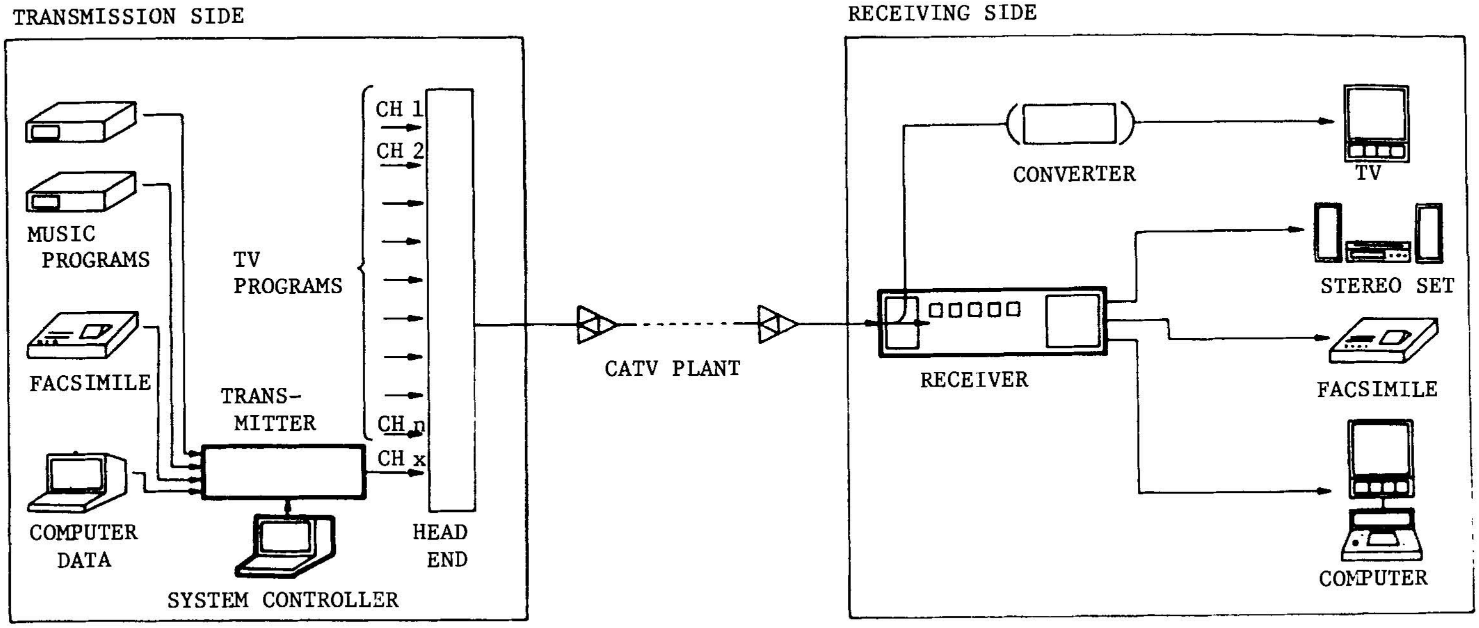

Fig. 2 : SYSTEM CONCEPT

3. TRANSMISSION METHOD

Some of the modulation methods used for digital data transmission are FSK(Frequency Shift Keying), PSK(Phase Shift Keying), and SSB(Single Side-Band AM) and VSB (Vestigial Side-Band AM).

FSK and PSK, however, require a more complicated demodulator compared to AM in order to get a large capacity using a restricted bandwidth of 6 MHz, which results in the cost increase of the receiver.
SSB and VSB are possible method for AM.
SSB has an advantage of getting a large transmission capacity using a restricted band. However, it is not a practical method for data transmission because its filter requires an extremely strict accuracy in order to accomplish distortionless transmission. Consequently, considerations like the above lead us to select VSB for the captioned system.

In our system, the implementation of the above method becomes as follows:
The modulation depth is max. 50%. Envelope detection is applied at the receiver, taking advantage of the good C/N of the line; the current CATV line in use maintains C/N of min. 36 dB, providing sufficiently large carrier. Consequently, this method, in which outstanding waveform distortion seldom occurs, shows sufficient noise performance for digital data transmission.
Figure 3 shows the spectrum of this method.
This method requires no alteration of current equipment, such as Head End, line, repeater, etc. because a position of the carrier and the bandwidth can be adjusted to those of current TV signals.
This method is also advantageous in economical implementation of the receiver because mass-produced TV parts, such as VIF IC, SWF, tuner, etc., can be used as hardwares of the receiver.

In order to suppress intersymbol interference, an accurate sinusoidal roll-off bandwidth restriction must be done at the baseband. The roll-off factor concerns both transmission capacity and amount of intersymbol interference.
In this system 21% roll-off is applied by LPF and BTF(Bynary Transversal Filter).
In this case the Nyquist frequency is, as shown in Figure 4, approximately 3.7 MHz, and the transmission capacity becomes approximately 7.4 MBPS.
Figure 5 shows the eye-pattern of the detected baseband signals.

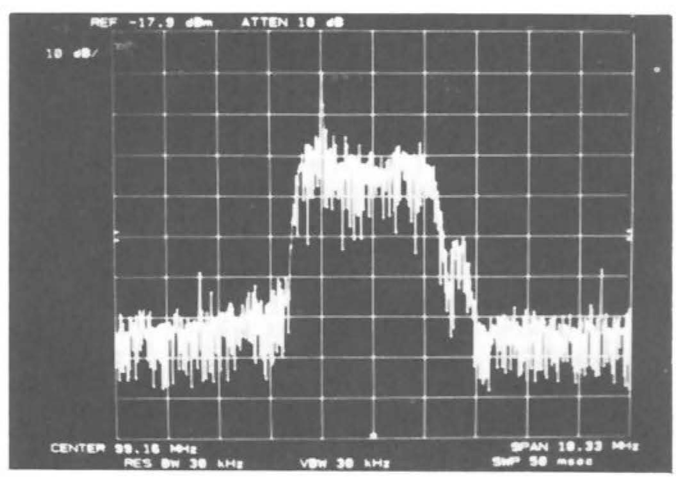

Fig. 3 : VSB SPECTRUM OF THIS SYSTEM

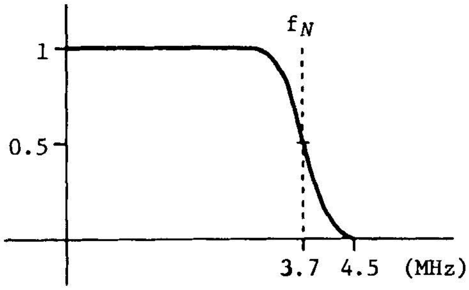

Fig. 4 : BANDWIDTH RESTRICTION

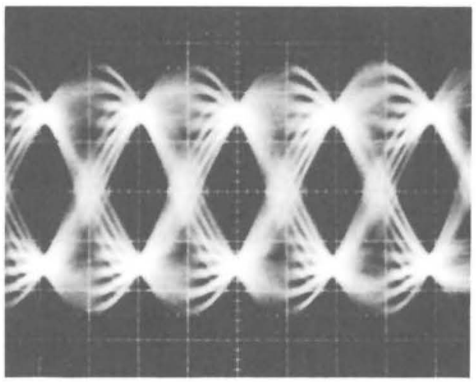

Fig. 5 : EYE PATTERN

4. DATA FORMAT

Various kinds of data must be processed in this system, responding to the respective demands of CATV operators. Therefore, it is indispensable for this system to provide a flexible data format, enabling a wide application in designing the system.

One of the main purposes of this system is, as mentioned before, to transmit data of ultra-high-fidelity audio programs, and the audio signal must be real-time-processed in general.
Taking the above into consideration, the minimum unit of this format, which is called q-unit, has been settled as shown in Figure 6. The q-unit consists of 32 information bits and 7 check bits, and is processed in approximately 22.7 µsec (= 1/44.1 kHz), i.e. one ultra-high-fidelity stereo audio program (16-bit quantization, 44.1 kHz sampling frequency) can be processed using a q-unit.
The format provides four independent q-unit, enabling simultaneous transmission of four ultra-high-fidelity audio programs.

Transmission format which is called "frame" is shown in Figure 7.

4 bits of service bit which are used to realize addressable function and 8 bits of synchronization data are attached to 4 q-units of data which are time division multiplexed for each and every bit, forming a frame consisting of 168 bits.
Therefore, the bit rate in the above process becomes;

$$168(\text{bits}) \times 44.1(\text{kHz}) = 7.4(\text{MBPS})$$

In order to facilitate the management of the service bits, a larger unit named "super frame" has been defined. One super frame consists of 256 frames and one super frame sync is attached to every 256 frames.

In order to answer to the various demands, the q-unit has four different modes as shown in Figure 8.

Mode A is used for transmitting the aforementioned ultra-high-fidelity stereo audio program. Mode B can transmit two stereo audio programs (8-bit quantization, 44.1 kHz sampling frequency) using one q-unit. In this case, high quality, which is better than current FM, is obtained when the noise reduction is applied. Mode C enables to transmit 8 monaural audio programs (8-bit quantization, 22.05 kHz sampling frequency) for BGM and announcement. Mode D is a combination of Modes B and C. Consequently, three quality-levels of audio programs can be transmitted by use of the above format. As for the non-audio data, each mode may be used depending on the data rate, using a certain interface if necessary.

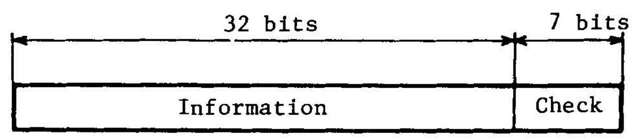

Fig. 6 : Q-UNIT

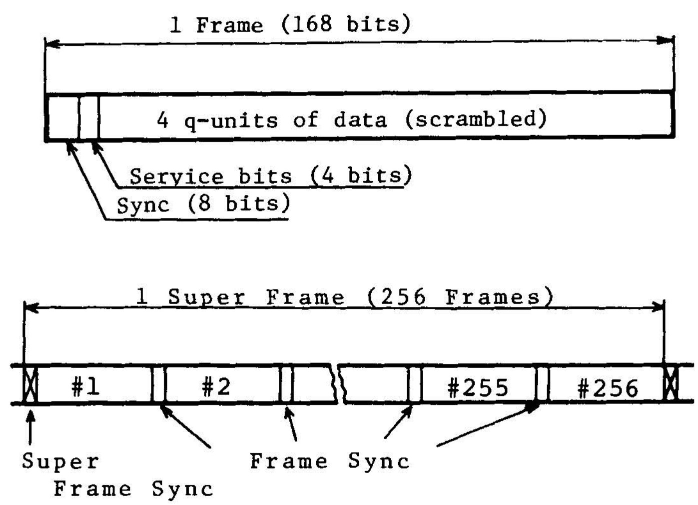

Fig. 7 : FRAME STRUCTURE

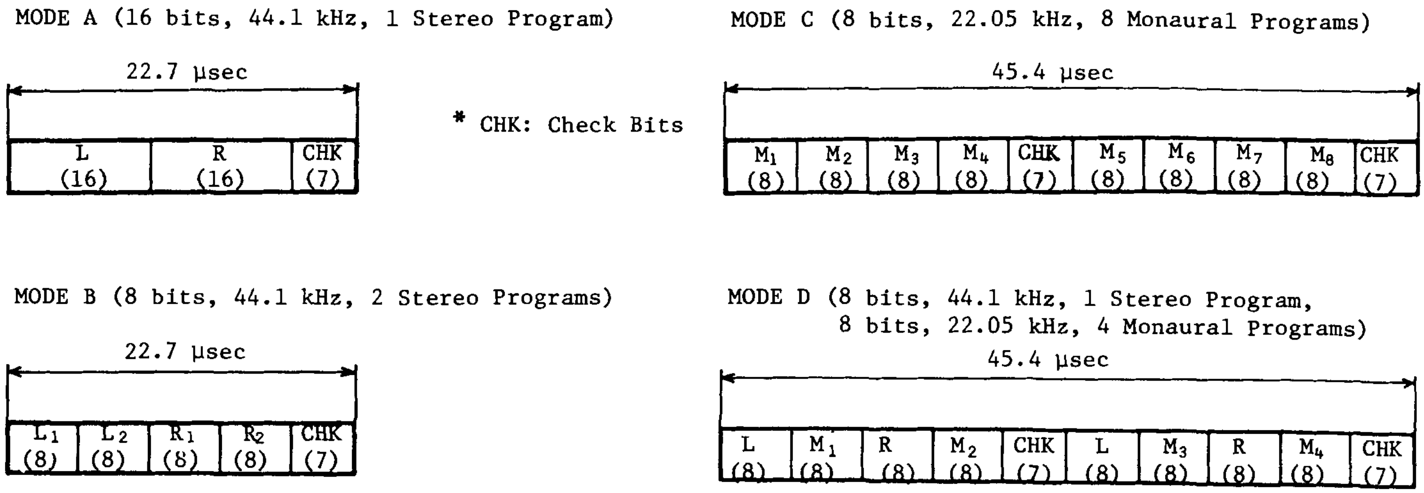

Fig. 8 : Q-UNIT MODE

5. SIGNAL PROCESSING

Figures 9 and 10 show the block diagrams of the transmission side and the receiving side of the system, respectively.

At the transmission side, input digital data are first converted into a serial data, and then sent through an encoding circuit where check bits are generated and attached. [63, 56] Extended Hamming Code is used in a shortened form, as shown in Figure 11. This code is capable of correcting single error in any digit, and detecting all patterns of double errors and some of the triple errors, owing to the shortening effect. Figure 12 indicates the theoretical performance of this code. The encoded data is then time division multiplexed and scrambled by use of an M-sequence in order to ease the bit-clock-recovery, while sync and service bits which are sent from the system control computer are attached to them, forming one data stream. Afterwards, the data stream is sent through a BTF and a LPF, at which 21% roll-off bandwidth restriction are accomplished. Output signal of the LPF is AM modulated, passed through a VSB filter, frequency-converted, and sent to the Head End.

At the receiving side, signal which is fed from a tap-off of CATV line is detected after going through the Front End and VIF circuit which are currently used in TV sets. Digital data and clock are recovered at the level-comparator and clock recovery circuit, respectively, and then sent through digital signal processing circuits. In these circuits synchronization data and service bits are separated from the data stream and sent to the built-in micro computer. The computer processes these data according to the data format and the instruction given by the subscriber, and generates some control signals. The data is descrambled by use of an M-sequence, and error controlled, and finally sent out in various forms according to the control signals.

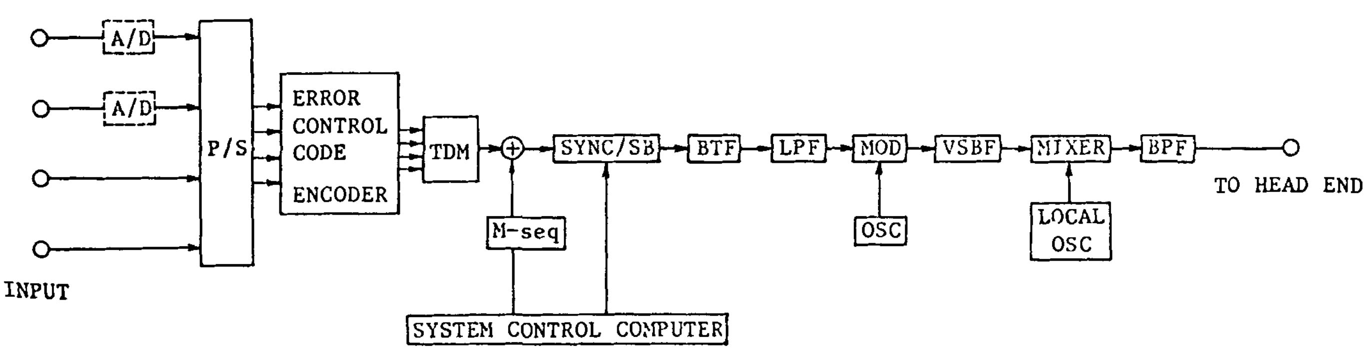

Fig. 9 : BLOCK DIAGRAM OF TRANSMISSION SIDE

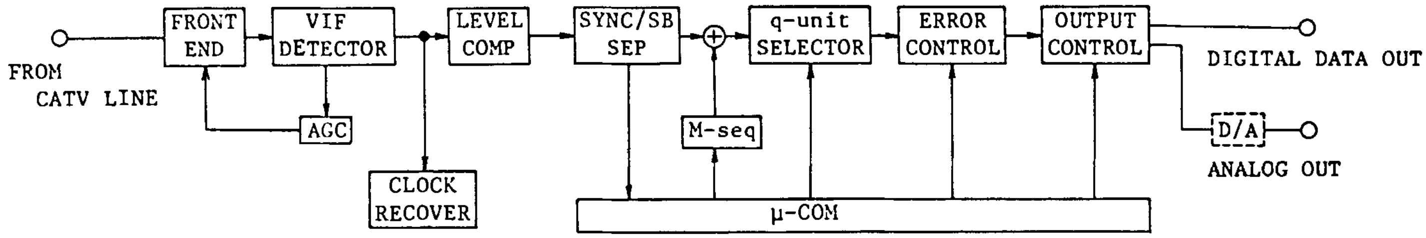

Fig. 10 : BLOCK DIAGRAM OF RECEIVING SIDE

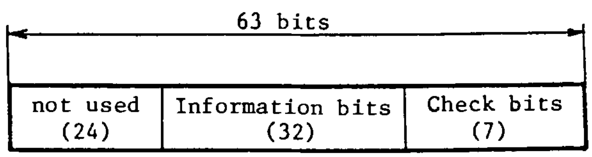

Fig. 11 : ERROR CONTROL CODE

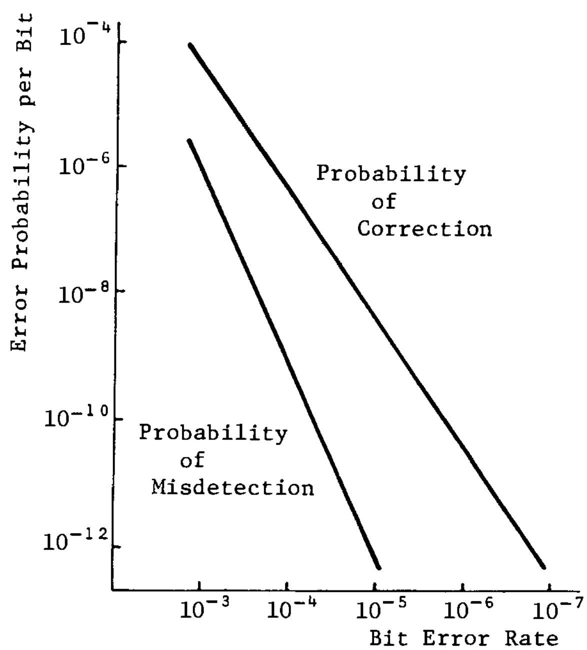

Fig. 12 : PERFORMANCE OF ERROR CONTROL

6. APPLICATION

As is already stated at the beginning of this paper, many applications are possible in designing this system depending on the various mode-selection of q-unit and handling of service bits.

The system is able to transmit three quality-levels of audio programs, as well as computer and game software, and facimile data, etc. Still pictures can be transmitted as well, provided that frame memory is applied to the transmitter and receiver. A certain application of the service bit leads to addressable function which enables, for example, specified announcement and facimile transmission. Furthermore, various pay service system can be designed when addressable function is effectively combined to the technique of scrambling and encription.

7. CONCLUSION

The system reported in this paper has proven to be effective in our North American and Japanese CATV system transmission experiments.

We believe the system is totally suitable to transmit various digital data using CATV, and will be used in the near future to offer various new services.

ACKNOWLEDGEMENT

The authors would like to express their deep appreciation to Dr. H.Nakajima, former Director, Dr. S.Miyaoka, Director, and T.Waku, Manager, Audio/Video Technology Center, Sony Corporation, for their kind support and valuable advice in developing this system.