Digital Television Transmission Using Bandwidth Compression Techniques

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DIGITAL transmission and signal processing techniques have long been viewed as promising and powerful means to achieve efficient television transmission by combining bandwidth compression with digital transmission systems. Recent progress in LSI and digital technologies have made complicated signal processing a technically feasible reality and have led to reasonable hardware size and cost. High-capacity and low-cost MOS memories have made it possible to store an entire television picture frame. This in turn has led to progress in digital television encoding through the development of interframe coding, by which the transmission bit rate can be greatly reduced.

At the same time, progress in digital transmission systems using multiphase modulation has allowed the use of digital transmission formats in microwave links and satellites. These trends have generated enthusiastic efforts to develop and use digital approaches in actual television networks.

In addition, the transmission of video teleconferencing services is growing rapidly. Teleconferencing has long been anticipated as an alternative to travel and many trials and evaluation tests have and are being conducted. Visual communications is a key to teleconferencing. Full motion video transmission is thought to be very helpful and useful in accomplishing interactive communications. However, transmitting a full television signal to accomplish video teleconferencing would be very expensive, requiring a thousand times wider bandwidth than a voice telephone channel. Bandwidth compression is, therefore, required to provide economical teleconferencing systems with compression ratios of 1:10 or even less.

Thus, the growth of broadcast television as well as teleconferencing services will rely more and more on sophisticated video processing techniques. These have already produced reasonable cost/performance results and have made digital television transmission a present reality in actual field applications.

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WHY DIGITAL TELEVISION TRANSMISSION

Digital transmission is considered to be advantageous from the channel capacity point of view, particularly when it employs an efficient encoding scheme [6],[7]. Existing television transmission links are generally analog FM. They carry a single network quality television signal per radio channel. Although one might double the channel capacity by restricting the modulation bandwidth [8] or by field interpolation multiplex [9], the resultant transmission quality would not meet network television standards. Furthermore, with analog techniques, high-compression ratios such as 1:10 could never be achieved.

Fig. 1 shows the relationship between transmitter power and RF bandwidth of a radio link. For analog FM, the transmitter power increases significantly as RF bandwidth is reduced (at an SNR of 54 dB weighted). On the other hand, digital transmission using quadraphase PSK modulation, requires smaller RF bandwidth and smaller transmitter power as the transmission bit rate reduces. At equivalent picture quality, digital technology offers obvious advantages over analog methods at the bit rate below 60 Mbits/s.

Furthermore, it carries with it various advantageous features of digital transmission. Picture quality is almost solely determined by the terminal equipment and can be made almost independent of transmission line impairments including those of digital terrestrial links. This implies that equal quality television service can be achieved uniformly over a wide service network, irrespective of distance. Sophisticated digital processings can be employed such that error control techniques improve the tolerance of the system to bit error rate, digital television signals can be easily encrypted to protect communication privacy, and adaptive bit sharing

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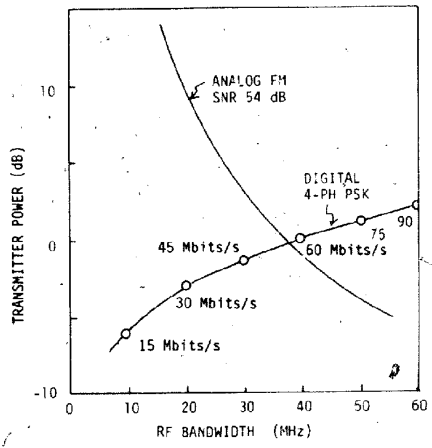

Fig. 1. A comparison of analog FM and digital transmission in terms of RF bandwidth and transmitter.

techniques can further reduce the required bit rate by sharing some of the total bit rate among plural channels.

WHAT IS INTERFRAME CODING?

Many investigations have been made to realize efficient digital coding [11],[12]. Digital television encoding schemes are generally categorized into three classes: 1) conventional PCM, 2) intraframe coding and 3) interframe coding, as shown in Fig. 2. For NTSC color television signals, conventional PCM or straight A-to-D conversion provides high-quality encoding with 7- or 8-bit encoding at about a 10 MHz sampling, resulting in a 75 through 86 Mbit/s transmission rate [10]. Intraframe coding is well known as a technique to reduce the transmission bit rate by intraframe processing of the signal such as higher order differential PCM (HO-DPCM) or orthogonal transform coding. By these intraframe coding methods, the transmission bit rate can be reduced to about 30 through 60 Mbits/s depending on quality requirements and technique employed. There are also certain tradeoffs between picture quality, bit rate, and hardware complexity.

Much greater reduction in transmission bit rate can be

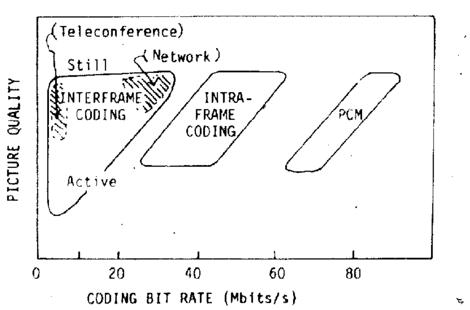

Fig. 2. Coding bit rate versus picture quality for typical coding methods.

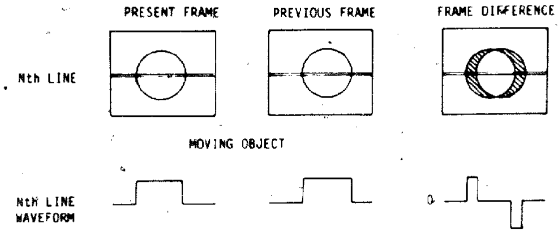

Fig. 3. Principle of basic interframe coding.

achievable by use of interframe coding, in which the difference signal between two successive frames is encoded and transmitted. The general concept of bit rate reduction by interframe coding can be better illustrated by a simplified model as shown in Fig. 3. Suppose that a soccer ball is crossing a television screen. On looking at two successive television frames, the soccer ball in the present frame is positioned slightly differently from that of the previous frame, and the difference corresponds to the movement of the soccer ball during the time period: one frame. Instead of transmitting the entire information, if the difference information of the two successive frames is transmitted, the amount of information to be transmitted can be greatly reduced. In fact, if the movement is zero, theoretically, no information need be transmitted. The information to be transmitted is dependent on picture object movement: the more active the movement, the greater the information to be transmitted becomes.

The basic configuration of the interframe coder is shown in Fig. 4. The digitized television signal, converted by an A-to-D converter, is encoded by an interframe coder, in which essentially the difference signal between the present and the previous frame is encoded. The previous frame signal is obtained from frame storage built in the interframe coder. The output of the encoder is again processed through a variable length coder to remove the redundancy contained in the frame differential signal. As mentioned earlier, the information rate is directly dependent on the movement of picture objects. It should be smoothed out to obtain a constant transmission bit rate. This function is accomplished through buffer memory and feedback control to the interframe coder to suppress the excessive generation of information. Since the control of the information generation rate is made by changing the quantizing step size, the signal-

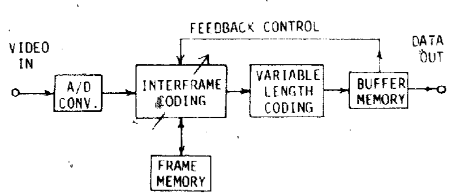

Fig. 4. A basic configuration of interframe encoder.

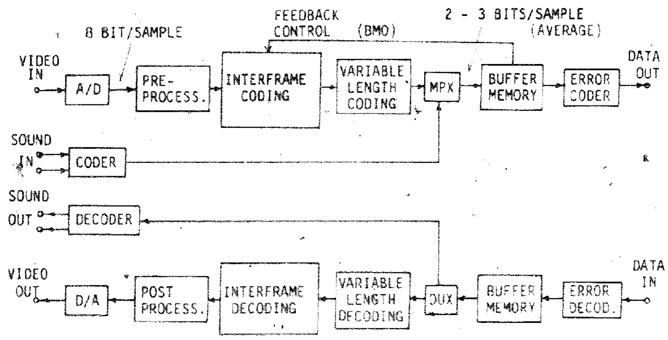

Fig. 5. An interframe encoder and decoder block diagram.

to-noise ratio is decreased as the amount of motion increases. When the information generation rate exceeds the transmission rate, picture quality starts to degrade. The encoded picture quality thus varies according to the video source materials.

As will be stated later, for network television signals where much more active motion is encountered, excellent picture quality can be transmitted at a bit rate around 20 through 30 Mbits/s [24],[29]. Relatively still pictures such as those encountered in conference room scenes can be transmitted at 6 Mbits/s, 3 Mbits/s, or even lower bit rates [13]-[22].

It should be noted here that the picture quality measure for interframe coding is different from the existing analog evaluation measure. Even at 3 Mbits/s, the signal-to-noise performance for a still picture is as good as that of 60 Mbit/s PCM. Picture quality impairments occur only when the picture moves actively, and the type of impairment is very much different from simple noise impairment. Presently, no objective evaluation standard has been established. Picture quality can only be subjectively evaluated.

INTERFRAME CODING FOR NETWORK QUALITY TELEVISION

The encoder/decoder arrangements for high-quality transmission of broadcast quality is shown in Fig. 5 [24],[29]. An input composite NTSC color television signal is first converted by an A-to-D converter into a 8 bit PCM signal. The PCM signal is then encoded into a reduced bit rate format of 2-3 bits/sample on average through digital signal processing. In the network television application, the transmission system is required to be transparent to the composite NTSC color video signal. Therefore, the codec is designed so that the composite signal is directly encoded and the waveform of the input signal is preserved except for quantization error.

In an NTSC color television signal, the color subcarrier phase is different by 180° between successive frames. This is undesirable because it generates a large frame difference even when the picture is perfectly still. To solve this, phase inversion is made by a preprocessing circuit in which the composite signal is separated into luminance and chrominance components by reversible

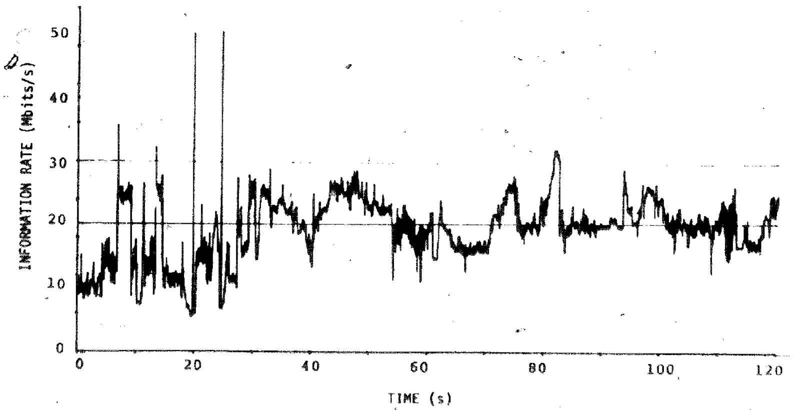

Fig. 6. Variation of encoded data rate with time. Signal source is "Superbowl '79," most active picture example.

transformation and the chrominance component polarity is inverted frame by frame. After phase correction the signal is encoded by an interframe predictive encoder. Adaptive intraframe/interframe coding is used by selecting either an intraframe prediction or an interframe prediction, whichever is appropriate. For quantization of the prediction error signal, eight kinds of quantization characteristics, Q0-Q7, are prepared and one of them is adaptively chosen through feedback control depending on buffer memory occupancy value. Quantization steps for the small amplitude region are, 1(Q0), 1.5(Q1), 2.5(Q3), 4.5(Q4), etc. respectively, where the magnitude 1 corresponds to one quantization step size of the 8 bit A/D conversion. The quantized prediction error is then coded with a variable length coder to reduce a bit rate. The compressed data are stored in a buffer memory with a capacity of about 1 Mbit, and is then transmitted to a line. Buffer memory occupancy value (BMO) is fed back to the adaptive quantizer to control the encoded data generation rate in order to prevent buffer overflow.

The performance of the interframe coder is better illustrated by way of example in Fig. 6. The example is taken from "Superbowl 79," a most violent picture example, for a period of 2 min. The waveform shows the information generated from the interframe coder without feedback control. The amount of information here indicates more or less original source information. As is seen, the amount of information varies tremendously according to the source scenes. Then, by adding feedback control, the system operates to allow coarse quantization for the portions of excessive source information, and as a result, keeps the output bit rate constant. This is the case for active motion, although generally television programs yield less source information. Fig. 7 shows the statistical data of source information for various source scenes. The dotted line shows the probability of source information rate for various source materials taken from broadcast television programs for 36 h [29]. It is seen that the average rate is around 15 Mbits/s, and as is seen on the solid line for the cumulative probability, 93 percent of the scenes are handled at 20 Mbits/s and 99 percent at 30 Mbits/s without appreciable noise impairments.

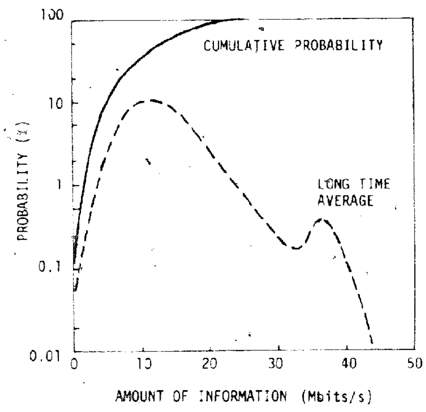

Fig. 7. Statistics of the encoded data rate for actual broadcast television signals for 36 h in total.

FURTHER IMPROVEMENT IS ACHIEVED BY ADAPTIVE BIT SHARING THROUGH PLURAL CHANNELS

From what we have observed through these statistics, excellent picture quality can be obtained by 20 Mbit/s coding most of the time. Buffer fill occurs with very small probability. According to actual measurements, buffer fill probability decreases very rapidly as the transmission bit rate is increased. The concept of sharing the transmission bit rate among plural television channels becomes quite effective in reducing the probabilities of buffer fill, and in improving picture quality for extremely active motion.

Adaptive bit sharing is a concept quite similar to that of TASI (time assignment speech interpolation) [27], using the advantage of statistical difference among plural channels. When a channel is transmitting a rapidly moving picture, other channels may be transmitting relatively quiet pictures, because the probability of occurrence of rapid picture motion is generally very small. Therefore, we can assign a larger bit rate to the rapidly moving channel, and a smaller bit rate to the other relatively quiet channels, keeping the total bit rate constant. In fact, for three channel transmission with total 60

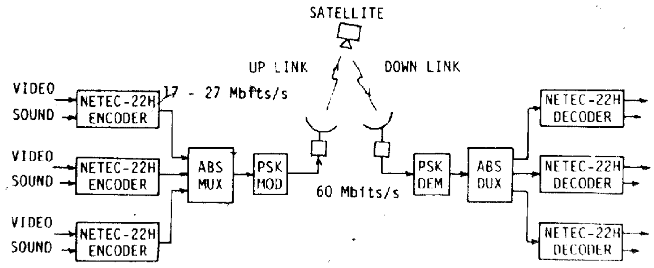

Fig. 8. A three channel television transmission system.

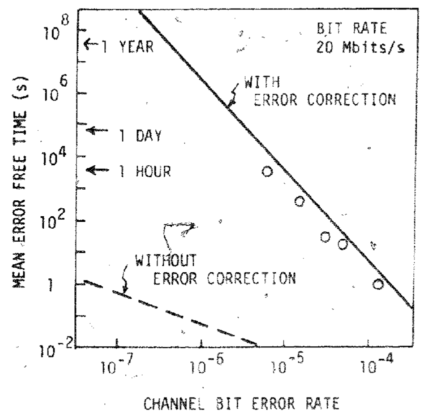

Fig. 9. Error correction performance of double error correction (DEC) BCH code. The solid line is theoretical and the circles are the field test results.

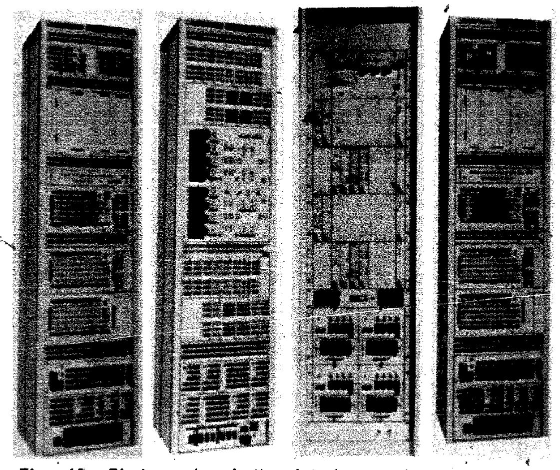

Fig. 10. Photograph of the interframe encoder/decoder NETED-22H and ABS-MUX system.

Mbit/s rate, the bit rate for each channel can be adaptively varied within the 17 to 27 Mbit/s range, with an average of 20 Mbits/s.

Interframe coding performance improvement by adaptive bit sharing is measured [29] by using a real hardware system shown in Fig. 8. Three different broadcast television program signals are supplied to the three encoders, and the adaptive bit rate assignment is performed by an ABS-MUX (adaptive bit sharing multiplexer) with the total bit rate kept constant at 60 Mbits/s.

ERROR CONTROL

This system has a forward acting error control circuit with 239/255 double error correcting BCH code. This error control has excellent error correction capability as shown in Fig. 9 [28]. The calculated mean error free time is longer than 1 h at a line bit error rate of 10^{-5}, and is about 5 s even at 10^{-4} which is generally considered the digital link threshold. The actual observed mean error free time, obtained from a satellite transmission experiment, is plotted in the figure. These data coincide very well with calculated error control performance. In normal satellite conditions, the bit error rate performance is generally better than 10^{-7}, and the mean error free time gets to be much longer than a year.

In addition to forward acting error control, in the decoder, an erroneous line is automatically replaced by the previous line, making the error less observable to the human eye.

A typical realization of the interframe encoder/decoder for network television is the NETEC-22H developed by Nippon Electric Company, Ltd., as shown in Fig. 10. This configuration consists of three encoders (from the left), an ABS-MUX/DUX bay, a PSK-MODEM, and two receiving decoders. Also each encoder/decoder contains two 15 kHz audio channels.

INTERFRAME CODING FOR VIDEO TELECONFERENCING

So much for the interframe encoding for high-quality network television. Now we will describe the interframe technology for teleconferencing applications where the advantage of interframe coding is more significant in reducing the transmission bit rate down to 6 Mbits/s, 3 Mbits/s, or less. Basic principles of interframe coding are the same as those for network television, but there exist differences in technologies related to achieve high-data compression under conditions of slower picture movement. For this application, transparency is not as important as in broadcast network use. Algorithm design is therefore directed to a best compromise of the transmission bit rate and picture quality.

A functional block diagram of the interframe encoder and decoder terminal is shown in Fig. 11 [17],[18]. The terminal encodes color video and audio signals into a digital stream and multiplexes them into a bit stream of 3 through 6 Mbits/s. Video signal input and output are NTSC color television signals which are widely used in conventional video equipment. An audio signal with 7 kHz bandwidth is encoded at a bit rate of 128 kbits/s.

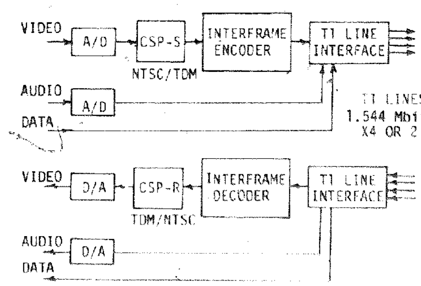

Fig. 11. A digital television transmission terminal, NETEC-6/3, for teleconferencing use.

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