Digital Television Transmission With 34 Mbit/s

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Digital modulation methods for TV satellite transmission can offer advantages in satellite transmission power and bandwidth if the bit rate can be reduced to 34 Mbit/s (to 34.368 Mbit/s, the bit rate for 480-channel PCM systems). A very efficient coding technique, differential pulse code modulation (DPCM), is described, and its application for TV transmission is shown. A test system for demonstrating the feasibility of digital TV transmission is described in detail. In this system the composite TV signal is split into its luminance and chrominance components. Tests via directional radio and satellite line have been carried out using a modem with 4-PSK (phase-shift-keying) as interface. After DPCM encoding these picture signals and accompanying sound signals are multiplexed and transmitted together with a code word for synchronization. On the receive side the signals are demultiplexed and after decoding are provided as analog signals. An example of digitally transmitted pictures is given to show that good picture quality has been achieved.

Introduction

Digital transmission of TV signals via satellites can offer advantages compared to analog transmission. These advantages come partly from the inherent properties of digital transmission and partly from the transmission link via satellite.

In satellite transmission it is highly desirable to keep the transmission power as low as possible; therefore, power-saving modulation methods which exchange power consumption and bandwidth have to be used. In the following, satellite transmission power requirements are compared between frequency modulation and digital phase-shift-keying (PSK) modulation.

Assuming a weighted signal/noise ratio of 52 dB and the parameters of a 12-GHz broadcast satellite link (Ref. 1), the satellite transmission power P has been calculated as a function of transmission bandwidth B (as described in Ref. 2). The result is shown in Fig. 1. The dashed curve shows the power requirements for FM transmission assuming a bandwidth of the modulating TV signal of 5 MHz (video bandwidth of PAL G).

The FM curve shows that the bandwidth can be reduced at the expense of the transmission power, or vice versa, while in the case of digital transmission, power can be saved if the bit rate can be reduced.

The solid curve shows the satellite transmission power for 4-phase shift keying (4-PSK) with differential encoding and coherent demodulation and the transmission bit rate as a parameter. This bit rate depends on the source encoding of the video signal. Applying straightforward PCM encoding will result in a bit rate of 80 Mbit/s for a color TV signal. The transmission of sound further increases the bit rate.

It is evident from Fig. 1 that, in this case, digital transmission is inferior to FM; however, if the bit rate can be reduced to less than 50 Mbit/s, digital transmission becomes superior. This can be achieved by applying more sophisticated source encoding techniques than PCM. These methods are described in the following section.

Before recommending the introduction of a digital TV transmission system, however, sufficient test results on picture quality, transmission errors, reliability, comparability with other systems, and indications for possible system simplifications must be available. For this purpose, we at Standard Elektrik Lorenz AG (SEL) have developed a system for testing the digital transmission of a color TV signal including two broadcast sound signals at a bit rate of 34.368 Mbit/s, in accordance with Refs. 2 and 5.

Bit-Rate Reduction

Television pictures contain in general a large amount of redundancy because of the statistical dependencies of the picture samples. This high redundancy stems from the fact that often a picture consists of areas of the same brightness level.

Encoding of the components of the TV signal offers a way of achieving bit rates as low as 34.368 Mbit/s with good picture quality. This bit rate is highly attractive because it would fit into the third-order European PCM hierarchy. Figure 1 shows that at 34 Mbit/s the necessary bandwidth is about 22 MHz. With this bandwidth, a reduction in transmission power by a factor of nearly five, compared with FM, is achieved.

For applying bit-rate reduction methods to the digitization of TV signals, there are basically two approaches. The TV signal can either be treated as an entirety (encoding of the composite signal) or split up into its components, luminance and chrominance, which are then encoded separately. At first glance, encoding of the composite signal appears more attractive since it preserves the standardized TV signal. However, bit-rate reduction methods for color TV signals are not very efficient, requiring minimum bit rates of at least 60 Mbit/s (see Refs. 3 and 4), and thus do not provide the power-saving advantages mentioned above. Moreover, these methods, based on linear prediction, can be applied only to PAL and NTSC standards. They are not applicable to SECAM because SECAM chrominance signals are transmitted by frequency modulation, which is a nonlinear operation. Therefore, bit-rate reduction methods based on elimination of linear statistical dependencies are due to fail.

Principle of DPCM

Today's most commonly applied redundancy reduction method is DPCM (differential pulse code modulation), which is especially well suited because of its high efficiency of achievable bit-rate reduction and acceptable system complexity. In DPCM the difference between the actual sample and an estimated value obtained from a predictor is transmitted.

Instead of transmitting the picture samples, after appropriate quantization and encoding in PCM, a difference between actual picture sample x and a predicted value $\hat{x}$ is transmitted. This difference is quantized. It is then fed into the DPCM feedback loop and also transmitted to the receiver. As can be seen in Fig. 2, the feedback loops in the transmitter and the receiver are identical. The task of the receiver or feedback loop is to add the quantized difference d_q to the predicted value $\hat{x}$. This will result in a reconstituted signal x_q which approximates the input signal $\hat{x}$ the closer, the smaller the quantization distortions are.

The reason that the PCM bit rate may be reduced by DPCM encoding is based on the fact that the transmitted difference — if the prediction is a good one — will be much smaller than the actual sampled value. Thus, the difference can be encoded with fewer bits per sample.

In a DPCM system, however, typical quantization errors can occur. If the smallest quantization step is chosen too large this will result in a coarse reproduction of picture areas with nearly constant brightness level or amplitude. This effect is called "granular noise" and is rather annoying to the human observer since the eye is sensitive to noise, especially in such picture areas. On the other hand, large quantization steps must be available for reproduction of large differences resulting from inaccurate prediction at edges or contours. If the available quantization steps are too small, this will result in an effect called "slope overload" which means that the DPCM signal cannot follow the actual sample quickly enough. The effect on picture quality is a loss of definition. These

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contradictory demands for quantization step size in picture areas and at contours can be satisfied partly by a nonlinear quantization characteristic. However, coarse quantization of edges is liable to introduce another effect called "edge busyness" resulting from nonidentical reproduction of edges in consecutive lines. This is due to unavoidable noise in the picture signal leading possibly to different quantization steps at edges.

In the test system these disadvantages have been largely avoided by using a controlled quantizer, which applies a chosen quantizing characteristic according to the picture contents. This will be discussed in detail in a section to follow on luminance signal processing.

Concept of the Test System

The block diagram of Fig. 3 shows how this DPCM method is applied separately to the processing of the luminance signal and the two chrominance signals (coding of the components).

On the transmitting side the video components Y (luminance), $R - Y$, $B - Y$ (chrominance) are digitized and the bit rates reduced in DPCM systems. A sync code word generator provides synchronization code words for identifying the frame start of the video signal. Together with two digital sound signals T_1, T_2, video and synchronization information is multiplexed. Because protection against transmission errors is of much greater importance for signals with reduced redundancy than for PCM encoded signals, error protection will generally be necessary. The error protection equipment has not been developed by SEL because several possibilities are already known[10] for reducing bit error rate (BER).

The digital TV signal can be transmitted by various transmission media, such as satellite or directional radio systems, coaxial lines, and optical fibers. An adaptation to the operating mode of the chosen transmission medium by an appropriate interface circuit — for instance, 4-PSK for satellite transmission — is necessary. With a bit rate of 34.368 Mbit/s, terrestrial links with the third-order European PCM hierarchy (PCM 480) may also be used. On the receiving side, the synchronization code word is extracted, and the necessary timing information for error decoding and demultiplexing is derived. After error correction and demultiplexing, the video signals are reconstituted in DPCM receivers to digital PCM signals. Digital/analog converters provide analog video and sound signals.

The test system has been designed to be as flexible as possible because the technology is new and there are many variants on the DPCM principle that could improve the picture quality. So far there is still not complete agreement as to *which* variant (method and parameters) is best suited for maximizing picture quality while minimizing transmission error insensitivity and system complexity.

After PCM coding, picture and sound signals are processed completely by digital

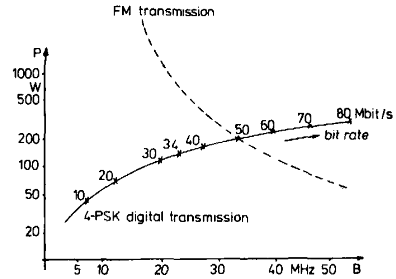


Fig. 1. Comparison of satellite transmission power (P) and bandwidth (B) for FM and for 4-PSK transmission.

means. This approach offers advantages of accuracy and reproducibility of the video signal as well as allowing for integration of functional units in MSI and LSI circuits. The exact reproducibility of digital circuits also simplifies the problem that in DPCM systems some functional units such as the adder and predictor — which are common to the transmitter and receiver unit — have to provide absolutely identical signals for a correct reconstruction of the transmitted signal. It is furthermore necessary for an exact prediction to synchronize the sampling frequencies and the picture processing clocks with the picture signal and to keep them so stable that the temporal sampling error in the picture signal is small compared to the sampling interval. This requirement is met in the test system by locking all clock frequencies in a phase locked loop with a voltage controlled crystal os-

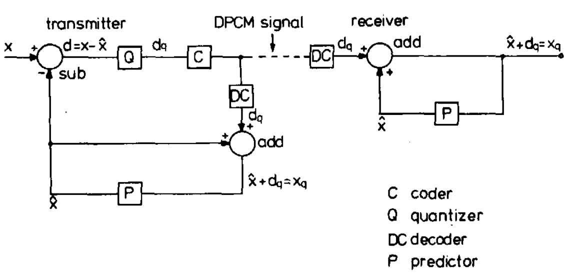
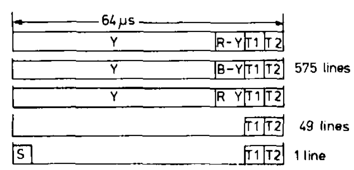


Fig. 2. Principle of differential pulse code modulation (DPCM).



signal	bandwidth	bits/sample	sample rate	bit rate in 52 μs	bit rate in 64 μs
Y	5 MHz	3	10.0 MHz	30.0 Mbit/s	24.375 Mbit/s
R-Y / B-Y	1 MHz	4	2.0 MHz	8.0 Mbit/s	6.500 Mbit/s
T1	0.015 MHz	14	31.25 kHz		0.438 Mbit/s
T2	0.015 MHz	14	31.25 kHz		0.438 Mbit/s
			resulting bit rate		31,751 Mbit/s
			error protection and additional information		2.617 Mbit/s
			transmission bit rate		34.368 Mbit/s

Fig. 4. Frame structure and bit rates.

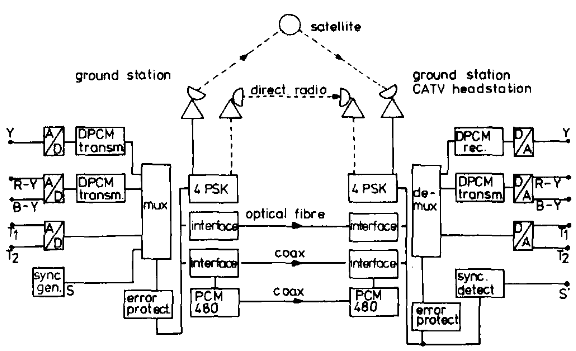


Fig. 3. Block diagram of the test system for satellite and terrestrial transmission.

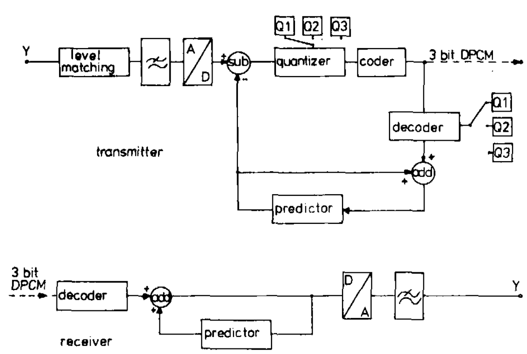


Fig. 5. Block diagram of luminance processing.

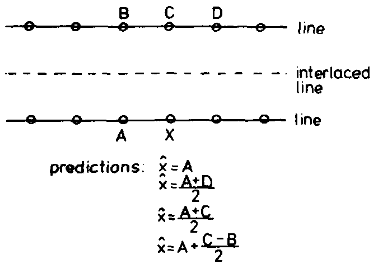


Fig. 6. Picture element configuration and predictions.

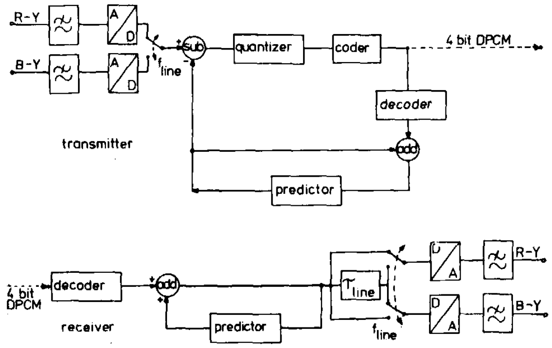


Fig. 7. Block diagram of chrominance processing.

cillator to the line frequency of the picture signal.

Figure 4 shows how the combination of the luminance, chrominance, sound, and sync signal in the lines of the digitized TV picture results in the bit rate of 31.7 Mbit/s. The two chrominance signals are transmitted line sequentially. This results in the halving of the bit rate for the chrominance coding but not in an intolerable loss in vertical resolution. Because the synchronization is carried out once per frame, providing the horizontal blanking interval for picture signal transmission, the bit rate for the picture encoding is reduced according to the ratio of the active to the total line period. The resulting bit rate for the encoding of luminance, chrominance, and two high-quality sound channels is 31.7 Mbit/s. The final bit rate for transmission is 34.368 Mbit/s thus allowing for an additional redundancy of 8.24% of the resulting bit rate which can be used for error protection. It will be necessary to add some useful redundancy in the form of an error correcting code, because the transmitted signal itself has largely lost its own redundancy by DPCM processing and is more sensitive to transmission errors.

To reduce the BER from 10^{-4} to 10^{-9} by error protection equipment, less than 3% of the bit rate is necessary for error correcting code information. This would enable the test system to transmit additional information, such as commentary channels, within the transmission bit rate of 34.368 Mbit/s.

Functional Units of the Test System

Processing of the Luminance Signal (Fig. 5)

The luminance signal is sampled with 10 MHz, converted to 8-bit PCM, and coded with a DPCM system using two-directional prediction and a controlled quantizer.[6] This means that the amount of the difference of two picture elements already processed yields a control signal which selects the optimal quantization characteristic for processing the actual picture element. Depending on the absolute difference between two samples, one of three quantization characteristics (Q_1, Q_2, Q_3) having coarser or finer steps is chosen. In order to keep the transmitted difference signal as small as possible, the predicted value of the next picture element must be very accurate. This is obtained by the two-directional predictor which uses previous picture elements of the same line as well as picture elements of the previous line.

The subtractor forms the difference ($x - \hat{x}$) between the signal of the actual picture sample x and the corresponding predicted value $\hat{x}$.

The quantizer subdivides the range of the difference signals (±256 are possible) into only eight subranges. These parts correspond to the steps of the quantization characteristic. These eight steps are encoded in three bits and transmitted to the receiver. If different quantization characteristics are used, the size of the subranges must be altered.

The demonstration model uses up to three different quantization characteristics. Each of these can be selected by switches, while in the dynamic mode they may be chosen under control of the predictor. The decoder assigns to each subrange transmitted by the quantizer a certain code word which gives the actual value of the quantized difference signal.

Depending on the sign bit, the adder unit adds or subtracts the 8-bit representative words of the decoder and the corresponding predicted value. In addition to these simple arithmetic operations, the adder provides a double-sided limitation to "black" and "white." The limitation is necessary because the output signal of the adder may be outside the possible 8-bit range of 256 quantization levels due to the inaccuracies introduced by the quantization.

The predictor can be switched to four different algorithms. In the simplest one, the predicted value of the picture sample $\hat{x}$ is given by the previous sample A (Fig. 6). In this way, the horizontal edges in the TV picture are predicted optimally but this prediction will fail for all other directions, especially vertical edges. This drawback can be overcome by a two-dimensional prediction using picture elements from the previous line. Computer simulation and subjective tests have shown that the prediction $\hat{x} = A + (C - B)/2$ in general yields the best results. If the previous line is stored some other predictions are available and realizable, for example $\hat{x} = (A + D)/2$ or $(A + C)/2$. For testing and comparing the different prediction algorithms they can be selected manually by switches.

Depending on the difference $|D - A|$ the quantizer characteristic is selected as described above, and the meaning of the transmitted code word is changed to the value corresponding to the related quantizer characteristic.

The receiver for the luminance signal is shown in the bottom part of Fig. 5. The input signal is a transmitted 3-bit DPCM word. Two bits are processed in the decoder. Depending on the sign bit the adder unit adds or subtracts. Adder and predictor are the same as on the transmitting side.

Chrominance and Sound Processing

Figure 7 is a block diagram of the transmitter and receiver sides of chrominance processing. After digitization into 8-bit samples at a 2-MHz sampling rate, the signals are processed alternately in a DPCM system. This system uses a simple previous sample predictor. Quality improvement compared to a conventional DPCM is obtained by using two's complement arithmetic and a special quantization characteristic.[7] This approach does not require transmission of the sign of the transmitted difference which is quantized to 4-bit code words. After the DPCM receiver, both components $R - Y$ and $B - Y$ are provided simultaneously, replacing one of the actual components by the line-delayed signal.

Digitization of the two sound signals is carried out with 14-bit linear PCM at twice the line frequency.

Multiplexing and Modulation

The multiplexed signal is a dibit stream consisting of the following parts (Fig. 8): 2 × 780 luminance bits are followed by 2 × 208 chrominance bits and 2 × 28

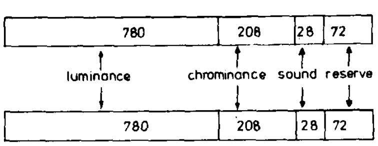


Fig. 8. Structure of the dibit multiplex signal.

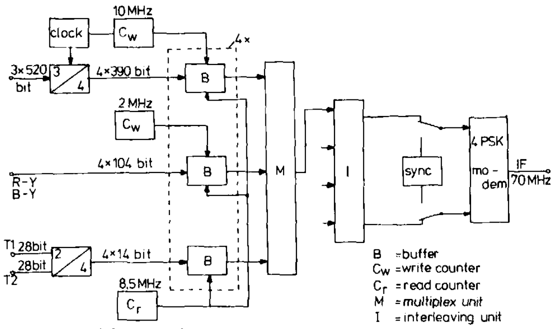


Fig. 9. Multiplex unit.

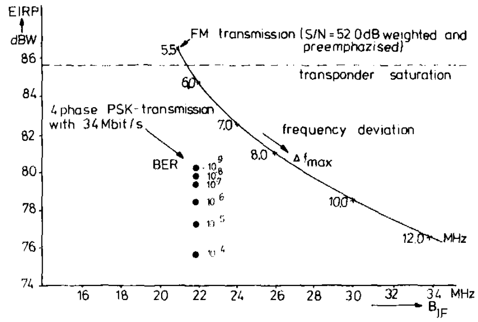


Fig. 10. Results of transmission test via Symphonie satellite.

sound bits. The remaining 2 × 72 bits are reserved for redundancy bits as a protection against transmission errors and for additional information. The total bit rate of the dibit stream is 17.184 Mbit/s; this results in a transmission bit rate of 34.368 Mbit/s.

Figure 9 is a block diagram of the multiplexer. In order to obtain a lower processing bit rate in the memories the total information is spread across four interleaving channels with 8.5 Mbit/s. The DPCM chrominance signal is already divided into four channels by the 4-bit encoding while the 3-bit encoded luminance information has to be converted from three channels with 520 bits/line to four channels with 390 bits/line. The reading counter distributes the address bits to the buffers and simultaneously, in its function as a central clock unit, provides several timing pulses necessary for the correct filing of the signals in the multiplex frame.

For transmission via satellite, the dibit stream at the output of the multiplexer (2 × 17.184 Mbit/s) modulates a 4-PSK modulator which operates with a 70-MHz carrier. The phase-shift is controlled by the difference of consecutive dibits.

Demodulation and Demultiplexing

In the receiver the 4-PSK signal is demodulated first to a dibit stream with the same bit rate as in the transmitter. Because of the differential encoding in the modulator, the phase ambiguity of $n\pi/2$ (for $n = 0, \ldots 3$) can be eliminated.

The following demultiplexer divides the incoming dibit stream into four channels which are first buffered. After converting the divided bit streams into parallel form for luminance, chrominance, and sound, the information is processed in DPCM decoders for luminance and chrominance. The sound signals are D/A-converted and, after low-pass filtering, become available at the output of the receiver.

Both chrominance signals $R - Y$ and $B - Y$ are processed in the same DPCM decoder because of their line sequential transmission. D/A conversion and low-pass filtering are necessary for monitoring the transmitted video signal with an analog monitor. The synchronization code word is transmitted once each TV frame. It controls, together with a line sync combination and a central clock, the processing of the signals in the receiver. Because the functions of the multiplexer are nearly inverse to those of the demultiplexer it is not necessary to show a block diagram of the latter.

Results and Future Work

A test system for the digital transmission of color TV signals and two broadcast sound signals was developed and realized in mid-1978. The picture quality attainable, after transmission over the complete system, scores between 4 and 5 on the internationally used 1 to 5 picture quality scale.

After conducting transmission tests over coaxial links as well as directional-radio and fiber-optic channels, measurements were made in June 1978 in a trial broadcast via the Symphonie satellite (German/French communications satellite for test purposes), from Raisting, the German ground terminal. The results achieved were well in agreement with the theoretical values shown in Fig. 1 for the relation between bandwidth and transmission power for digital (4-PSK)- and frequency-modulation.

Figure 10 shows the measured dependence of EIRP (equivalent isotropically radiated power) versus B_{if} (intermediate frequency bandwidth). Naturally, 4-PSK transmission could be made only with the transmission bit rate of the system (34 Mbit/s) which requires a bandwidth of about 22 MHz. The EIRP for several bit error rates is shown in the diagram. Additional tests via directional radio equipment with simulated link attenuation have been carried out in order to obtain an impression of the influence of bit errors on the attainable picture quality. All these tests and measurements were carried out without error protection equipment.

The next step will be to test the system with error protection equipment which is capable of reducing a bit error rate of 10^{-4} to a value of 10^{-9}. This value is necessary

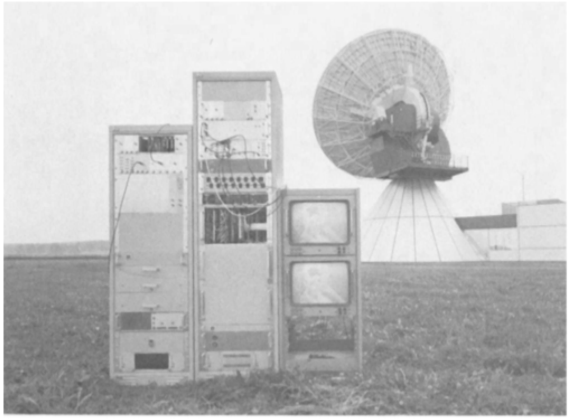


Fig. 11. The test system with the Symphonie ground terminal.


Fig. 12. Digitized picture.

for achieving a good picture quality. As shown in Fig. 10, power reduction by this procedure is 4.5 dB.

Figure 11 shows a photograph of the test system in front of the Symphonie ground terminal in Raisting. Because of the flexible design of the picture processing equipment, the system is considerably larger than would be necessary for a final version.

Figure 12 shows a transmitted digitized picture, with 3-bit DPCM, adaptive quantization, and prediction $\hat{x} = A + (C - B)/2$.

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Comments Solicited by the Board of Editors

Reviewers: It seems unlikely that the luminance TV signal can be DPCM encoded with 3 bits and still give good quality pictures. The authors use a fairly unsophisticated intrafield, two-dimensional prediction. According to V. G. Devereux (in his paper "Differential Coding of PAL Video Signals Using Intrafield Prediction," in *Proceedings IEE*, Dec. 1977), 3-bit encoding will cause the picture impairment to be somewhat objectionable, and 5 bits per sample are needed for acceptable picture quality. One opinion is that 4 bits is the smallest number that can be tolerated.

There is also question about using a 10-MHz sampling rate for a 5-MHz video signal. Due to the finite response of the analog filters used with a PCM codec, it is impractical to encode at the Nyquist sampling limit. The authors may be getting only about 4 MHz of Y response.

The Authors: Our objective has been a coding method for color TV, including transmission, an objective sought by only a few other researchers.

We agree that when using pictures for testing the resolution, 3 bits per sample for the luminance signal is not enough compared with analog transmission. But the CCIR Recommendation requests that test patterns not be used for subjective tests of picture quality. Our subjective tests have been carried out in accord with CCIR Rec. 500, the slides being the three commonly used test slides titled Playboy, Strawhat Girl (Fig. 12), and Beach Scene, and two others (Cityscape and Puppet). The 25 test persons included five experts. In all cases we got a quality score of more than 4 on the CCIR scale, when the bit error rate was 10^{-9} or less.

The use of interframe coding would improve picture quality, but more sophisticated equipment would be required; we do not see a disadvantage in using an unsophisticated method of encoding.

We suggest that a 5-bit DPCM encoding for the luminance signal will not be standardized because of the final bit rate for transmission. The international trend seems to be for a standard of 140-Mbit/s PCM encoding in the studio area and of 34-Mbit/s DPCM encoding for transmission. In this case, a maximum of 4 bits for each luminance sample may be used for transmission purposes. In Webster's "Digital Picture Coding" in *Wireless World* for October 1978, we note advice that "Excellent picture quality can be achieved by means of a 3-bit difference signal for each picture element."