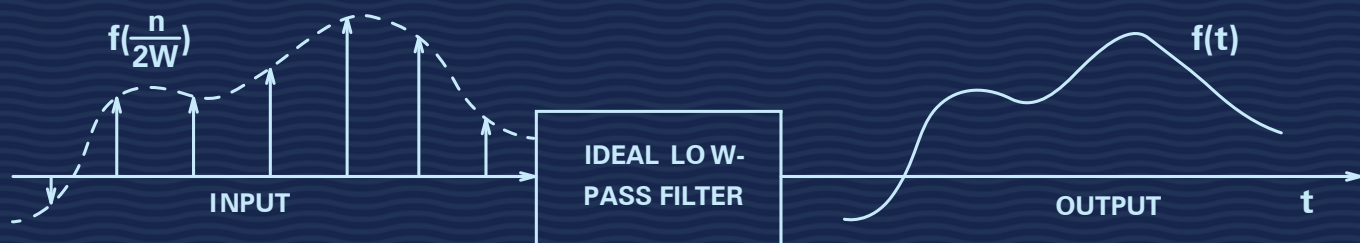International Telecommunication Union

Handbook

DIGITAL TELEVISION SIGNALS
CODING AND INTERFACING
WITHIN STUDIOS

Fundamentals and practical aspects

Edition 2010
(revised Edition of 1995)

$$f\left(\frac{n}{2W}\right) \qquad f(t)$$

INPUT   IDEAL LOW-PASS FILTER   OUTPUT   t

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Handbook

DIGITAL TELEVISION SIGNALS CODING AND INTERFACING WITHIN STUDIOS

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SUMMARY

This Handbook provides information concerning the current status of the digital technologies as it applies to digital television signals coding and interfaces within studios. Some historical data has been included to give some insight into the background of many of the decisions reached by the CCIR and later the ITU-R. The revision of the Handbook includes new text to cover current interfaces.

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1   Introduction

The pulse code modulation (PCM) system on which the coding of digital television signals depends was invented by Alec Reeves as long ago as 1937 but it was not until the early 1970s that the development of digital techniques had reached the stage where it was possible to pursue their application to television in earnest. International discussions on digital television began around 1972 and by 1974 the CCIR[1] had adopted a Question on the subject of "Standards for television signals using digital modulation" together with a Study Programme on the "Encoding of Colour Television Signals". The results of this work were gathered together in CCIR Report 629 and subsequently in Report 962, with more to follow. But it was not until 1982 that the CCIR finally adopted Recommendation ITU-R BT.601 which defines the form of component digital 525- and 625- line signals within studios. Four years later Recommendation ITU-R BT.656 for an interface standard based on Recommendation ITU-R BT.601 was adopted, and today they stand at the heart of any description of a studio coding system. Many items of equipment based on these Recommendations have now appeared.

The digital interface based upon Recommendation ITU-R BT.656 is now in wide spread use in all broadcasting and post production facilities, in addition a further ITU-R interface Recommendation ITU-R BT.1120 has been defined at a bit rate of 1.5 Gbit/s for HDTV applications. Definitions of the HDTV interface are available for both coaxial cable and fibre implementations.

While the original interface parameters were developed around basic image pixel arrays, there has been a continued expansion of the interface implementations that includes multiple "links[2]" of the original interface in order to achieve wider bandwidths, greater pixel depths, and extended pixel arrays. The interfaces are seen as data carriers rather than carrying just uncompressed image formats.

The transition to digital technology has introduced an entirely new set of digital switchers, routing switchers and an extensive range of support devices. More recently as the migration to HDTV is nearing its completion, switching devices have emerged that can handle both SDTV and HDTV signals including aspect ratio conversion, colorimetry conversion and pixel array rescaling.

This Handbook attempts to provide valuable information to readers concerning the current status of the digital technology. In some cases historic data has been included to give the reader some insight into the background of many of the decisions reached by the CCIR and later the ITU-R.

Sections of this Handbook have therefore been provided as a brief overview of the development of digital television coding and interfacing, to enable the material presented in §§ 3, 4, and 7 to be put into context. In general, the quotations from Reports are presented verbatim, but where subsequent developments have modified what are nevertheless important excerpts, some explanatory text has been added.

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[1] It should be noted that in the recent reorganization of the ITU the CCIR has become the Radiocommunication Sector of the ITU. For historical accuracy the abbreviation "CCIR" is used in this Handbook in referring to work undertaken and publications issued before March 1993. Following the reorganization, "CCIR" is generally replaced by the abbreviation "ITU-R". Similarly the CCITT has become "ITU-T".

[2] Links is a term in common use to cover implementations using multiple interfaces, usually requiring some form of signal/data multiplexing.

2

2 Coding

2.1 Fundamentals

2.1.1 Pulse code modulation

One of the most important contributions to an understanding of the fundamental principles of communication was made by Shannon, who developed equations enabling the theoretical capabilities of a transmission channel to be evaluated in terms of its information carrying capacity, and thereby defined the minimum requirements for a transmission circuit capable of conveying a given flow of information [Shannon, 1948].

One of Shannon's equations states that the information contained within a signal is approximately given by the bandwidth which it occupies multiplied by the signal/noise ratio expressed in logarithmic terms. (This statement assumes that the signal is appreciably greater than the noise, that the signal and the noise have sufficiently random amplitude distributions and similar spectral distributions, and that the signal/noise ratio is calculated in terms of mean signal and noise powers.) It follows that if the signal is encoded so as to occupy M times its original bandwidth, it can, by suitable coding, be transmitted through a channel whose signal/noise ratio in decibels is $1/M$ times that required at the output of the decoder. Therefore, if a circuit has a bandwidth which exceeds that of the analogue information it is required to convey, then the effective signal/noise ratio of that circuit can be greatly improved by dispersing the information over the broader spectrum.

Another contributor in this field was Nyquist who showed that analogue information which is inherently represented by a continuous function can be correctly represented by a number of discrete samples. If the number of samples per second is equal to or greater than twice the bandwidth in cycles per second occupied by the analogue signal, then the original information is preserved in the sampled signal [Nyquist, 1928]. Nyquist's theorem had been stated by several authors, the first of whom appears to have been Cauchy [Cauchy, 1841].

In 1938, Reeves devised a practical transmission system which made use of the theoretical concepts described by Nyquist. He conceived the idea of representing each Nyquist sample by a binary number, and transmitting these binary numbers in the form of groups of pulses, the presence or absence of a pulse indicating the state of each binary digit. Reeves called his system "pulse code modulation" [Reeves, 1938]. The minimum bandwidth required by PCM is equal to the original analogue bandwidth multiplied by the number of pulses used to describe each sample.

Although at first sight it seems extravagant to use many pulses of identical height instead of one pulse of variable height, this apparently extravagant use of bandwidth is, in fact, the strength of PCM. By forcing the signal to occupy a larger bandwidth Reeves was, in effect, making use of Shannon's theory to provide a form of signal coding which can tolerate very high noise levels. At the receiving end of a transmission system using PCM the message can be correctly decoded so long as the presence or absence of each pulse can be correctly established; only a peak noise level greater than half the magnitude of the transmitted pulses can cause errors.

The groups of pulses forming a PCM signal can be stored and delayed by simple devices and it is thus possible to convert a wideband PCM signal carried by a single channel into a number of lower bandwidth signals each carried by a separate channel. This flexibility allows available circuits to be used efficiently and is frequently used in tape recording digital data.

In addition to the virtue of resistance to impairment a PCM signal can easily be repaired or regenerated en route. A simple circuit containing a threshold detector and a pulse generator can reconstruct a "clean" version of the original signal which is in all practical respects as good as new. Thus the use of pulse code modulation with its inherent ruggedness and easy regeneration enables the quality of the decoded signal to be made independent of the characteristics of the transmission channel. It is then only necessary to ensure that the original coding process describes the given analogue signal in sufficient detail to satisfy the requirements of the recipient. In many modern installations there is no A/D conversion, like wise with the introduction of Flat Panel Displays the D/A process may also be by passed with pixels being displayed as a direct digital signals.

2.1.2 Sampling and filtering

The parameters of a PCM system can be determined from the required bandwidth and signal to noise ratio.

In § 2.1.1 it was pointed out that a minimum sampling frequency of twice the bandwidth of the analogue signal is essential if the received signal is to be free of distortion. This is a theoretical figure and in practice a somewhat higher ratio of sampling frequency to bandwidth is necessary in order to allow for the limitations of practical filters. The act of sampling effectively multiplies the input signal by a regular chain of unit impulses, and the resulting spectrum is defined by the low-pass filter characteristic with which the input signal is limited, repeated at multiples of the sampling frequency.

The original signal is obtained by submitting reconstructed samples to a further low-pass filtering operation. If the pass band of the input filter is not confined entirely to frequencies below half the sampling frequencies, "alias" components are introduced whose amplitude in the output signal will depend on the characteristics of the output filter.

Input and output filters are normally designed to have similar roll-off characteristics. In many applications such as computer graphics, character generator etc. the filtering process is often ignored resulting in reproduced signal distortion on edges or other types of alias effects.

Filtering may be carried out entirely in the analogue domain or more commonly by an initial use of higher-frequency sampling with associated digital filtering. In deciding the required filter characteristic one has also to bear in mind that many such filters may be used in cascade, and therefore particular attention has to be paid to the need for a flat response within the pass band. Taking all of these factors into account, it is usual to allow for a sampling rate of about 2.3 times the required bandwidth, i.e. the "super-Nyquist margin" is about 15%.

2.1.3 Quantizing

The effective signal/noise ratio is determined by the number of signal levels that can be described. Since each level is described by a number, it is necessary to ensure that a sufficient choice of discrete numbers, each identifying one signal level, is available in the coding equipment. If n binary digits are used, the number of levels that can be described is given by 2^n.

In general, the signal samples will not have magnitudes exactly equal to one of the levels that can be described. The nearest possible level is therefore selected. If the waveform of the input signal is sufficiently complicated, the errors thus introduced change in an arbitrary way from sample to sample, and in effect, add to the original signal a spurious signal which is very similar to white noise. The magnitude of this so-called "quantizing noise" is decreased when the number of available levels is increased.

Thus the required output signal/noise ratio specifies the minimum number of signal levels which must be described by a PCM system. It may be shown that with ideal low-pass filters and with full use of the coding range, the ratio (unweighted) of the peak to peak signal to r.m.s quantizing noise at the output of a PCM system is:

$$6.02\, n + -0.8 \qquad \text{dB}$$

where n is the number of bits per sample. This is modified in practice by the coding range and super-Nyquist margins.

The noise introduced by cascaded uncompressed PCM systems is in general uncorrelated and therefore subject to r.m.s. addition.

Taking all of these factors into account, it is generally accepted that to preserve adequate quality the television signal should be described using at least 8 and preferably 10 bits per sample. In some cases 12 bits are required to ensure that signal integrity is preserved.

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2.2   Application to TV signals

2.2.1   The signals to be coded

The early stages of work on digital television were mainly devoted to the problems of international programme exchange. It was natural, therefore, that the signals coded should be the same as those which had hitherto been sent in analogue form, namely the composite PAL, SECAM and NTSC signals as specified in Recommendation ITU-R BT.1700.

Interest then began to arise in the use of digital techniques within studio centres, and here too, attention was first paid to the composite signal. The reasons given for this were straightforward. It was seen that digital signals would be introduced over a period of many years and during the transition period the signal would be going into and out of the digital format many times. Digital timing correctors were beginning to be used with analogue video recorders, and this meant that there would be as many codecs in tandem as there were generations of tape copying.

Previous experience had led to the belief that once the signal had been safely encoded in composite form it was far better to leave it like that than to attempt to take it to pieces again. Composite decoding, it was realized, could never be perfect and some of the imperfections it introduces would be likely to accumulate with several decoding and re-encoding processes in tandem.

It therefore seemed obvious that digital processing within the studios should be carried out on the composite PAL signal, and that the signal should only be decoded where necessary in a side-chain and not in the main signal path.

Also by this time a considerable amount of work had been done on methods of low bit-rate coding for the economic use of transmission links and perhaps of digital recorders. In the United Kingdom, and in the United States of America, this work was principally based upon a process of sub-Nyquist sampling which had been shown to be surprisingly efficient and capable of high quality.

The method is to use a sampling frequency which is lower than twice the signal bandwidth and offset from a multiple of the line frequency. It had long been realized that this would be a good technique to use for monochrome signals, since the spectral components of the television signal tend to be clustered about multiples of the line frequency.

What had not been realized until the 1970s was that the same process can be applied to a PAL signal. This is because twice the colour sub-carrier frequency (2_{fsc}) is almost exactly at an odd multiple of half the line frequency, thus satisfying the sub-Nyquist sampling criterion.

Furthermore, it turns out that if the PAL signal is sampled at 2_{fsc} with the sampling appropriately phased with respect to the sub-carrier the chrominance alias components are reflected about the subcarrier frequency in such a way as to reinforce rather than interfere with the wanted components. Moreover, the line delay comb filters used to suppress luminance aliasing also restore the signal to its original PAL form at the output of the sub-Nyquist codec.

Therefore, those who had already been feeling that composite PAL constituted an efficient signal package were further encouraged in the belief that it was well suited to digital encoding too. It was further realized that if digital comb filters were used to tailor the signal appropriately for sub-Nyquist operation they could themselves operate at a frequency of 4_{fsc}, and a 4_{fsc} sampling structure is not only picture repetitive but almost exactly orthogonal. One can use a relatively simple method of sample re-timing as a bridge between a 4_{fsc} structure, and one based on exactly 1135 samples per line.

It seemed, therefore, that composite PAL, 4_{fsc}, 2_{fsc} and 1135 times line formed an attractive package on which to base a digital coding strategy. Similar thinking was applied to the NTSC signal in the United States of America [Rossi, 1976] and subsequently in the United Kingdom [Devereux, 1976] despite the fact that, with its half-line offset colour subcarrier, NTSC is not so amenable to sub-Nyquist sampling.

Meanwhile, however, a feeling was beginning to be expressed widely within the European Broadcasting Union (EBU) that the advent of digital techniques might perhaps provide an opportunity to rectify the present situation in which Europe was divided between PAL and SECAM, by establishing a method of describing signals which could be the same in all 625-line countries.

Luminance and colour-difference signals are, of course, a common factor to both types of composite coding, and it was natural, therefore, that the EBU should take the view that separate component signals would form the best basis for a unified approach to digital coding.

But another important development was to encourage a move away from composite coding. It had been realized for some time that the PAL signal was somewhat inconvenient for certain processes, in particular video-tape editing, because of its eight-field cycle. It was towards the end of the 1970s, however, that increasingly complex post-production processing started to appear, and this demanded access, not to the composite signal but to its components.

It also demanded a sampling structure that repeats exactly from one picture to another. It was fortunate in this respect that work carried out during that period established that it is possible to interface a component coding system with a composite PAL environment using a digital PAL decoder whose sampling rate was completely unlocked from the PAL sub-carrier.

In practice the use of digital composite coding never turned out to be a popular choice. Digitalizing a composite signal resulted in video being coded at about 7½ bits. Maintaining the color subcarrier sequence of 4 or 8 fields was not an easy task.

It was thus becoming clear that if the emphasis in international discussions was to be centred on the establishment of digital television standards based on component coding, the choice would not after all have to be restricted by the need for a composite/components bridge somewhere in the vicinity of a multiple of the subcarrier frequency.

The stage was thus set for an intense period of international discussion in the search for acceptable sampling frequencies. With the concentration now on component signals, and consequent meetings between the EBU and the Society of Motion Picture and Television Engineers (SMPTE) of the United States of America, the discussions took place within a worldwide context, with a new emphasis now on the achievement of a maximum degree of commonality between 525-line and 625-line applications.

The result was the now well known Recommendation ITU-R BT.601 which defines the main parameters of a coding system which has 525-line/625-line commonality to the extent of using the same sampling frequencies (13.5 MHz and 6.75 MHz) and the same numbers of samples per digital active line (720 and 360 for Y and U/V respectively) in the two applications. There are different numbers of samples per total line, but these are both integer (864 and 858) since the chosen sampling frequency is a multiple of 2.25 MHz.

3 Recommendation ITU-R BT.601

3.1 Introduction

The CCIR had brought together, during the study period 1982-1986 and previously, a large amount of information concerning the coding, processing, transmission and recording of picture signals in digital component form in the studio. This work has resulted in Recommendation ITU-R BT.601 on the encoding of the digital video signal for studio production, and Recommendations ITU-R BT.656 and ITU-R BT.799 on studio interfaces for the digital video signal.

Work on the component digital standard would not have been complete without a recording format capable of recording the 8 bit 4:2:2 signal. For a number of years from 1986-1990 the D-1 recording format was the only means of recoding the uncompressed signal. In recent years a number of recording formats have been introduced that use varying levels of compression. And that are capable of recoding the full 10 bits.

Recommendation ITU-R BT.601 has undergone a number of revisions since 1986:

– 10 bits was added to the definition.

– 18 MHz sampling was added to address wide screen issues- subsequently the 18 MHz sampling was removed, as there was little, or no commercial implementations.

– Informative notes were added that allowed SDTV signals to be encoded with the same colorimetry as HDTV signals.

3.2 Use of component coding

Two different approaches to the coding problem have been proposed:

– *component coding*: in this approach the luminance and colour difference signal components are digitally coded separately and transmitted as separate bit streams, time-multiplexed together; and

– *composite coding*: in this method the composite colour signal is coded in its composite form as a single bit stream. Composite digital coding was not adopted for international programme exchange. Commercial implementations were constrained to regional adoption.

Component coding is now accepted as the only digital coding method, as analogue transmissions are coming to an end there is no future for any digital composite implementations. Digital components are the basis for almost all compression schemes, specifically ITU-T Recommendations H.262 and H.264 which are the default compression technologies for transmission. Within Studio applications there are many compression schemes in use designed to with stand the production requirements of multiple generations.

Digital coding in the form of Y, C_R, C_B signals can represent a substantially greater range of signal values than can be supported by the corresponding ranges of R, G, B signals. Because of this it is possible, as a result of electronic picture generation or signal processing, to produce Y, C_R, C_B signals which, although valid individually, would result in out of range values when converted to R, G, B. [Devereux, 1987] explains that it is both more convenient and more effective to prevent this by applying limiting to the Y, C_R, C_B signals than to wait until the signals are in R, G, B form. Also, techniques are described by which limiting can be applied in a way that maintains the luminance and hue values, minimizing the subjective impairment by sacrificing only saturation.

3.3 Family of coding standards

In the formulative years of the development of digital coding standards for broadcasting there were a number of proposals that have evolved into a set of Recommendations defining a 4:2:2 and 4:4:4 family. In the case of some compression schemes, there exists a 4:2:0 member of the family. It is generally accepted that the 4:2:0 member be used for emission, or in applications not requiring high levels of image processing. In other industries the coding standards do not rely upon R,G,B primaries, yet they are still referred to as 4:4:4.

At one time there was interest expressed in other coding families such as 2:1:1, interest in investigation these lower order members of the families was over taken by the progress that was made in bit rate (compression) schemes of which there are many.

Rossi describes a set of digital filters adapted for a binary family, that accomplishes a two-to-one sampling rate down conversion and a one-to-two sampling rate up conversion, by comb filter interpolation [Rossi, 1981]. This work is reported here just for the historic record, implementation was not wide spread.

3.4 Bits per sample

The original choice of 8 bits per sample was influenced by a number of factors, processing distortion, bandwidth requirements, and speed of processing. Today, none of these perceived issues are a problem, handling 12 bits with image arrays of 4k × 2k at 60 frames is well within the range of commercial possibility. To deal with the limitations of 8 bits and the quantizing visibility a number "band Aids" have been used, rounding, dither etc. These were all stop gap measures and did not enjoy popular adoption.

Experience suggests that, within PCM signal processing equipment, more than 8 bits per sample should be retained to avoid a rapid accumulation of quantizing distortion from repeated rounding after each arithmetic process. However, 8 bits per sample has been found satisfactory for interconnections between equipment using digital Y, C_R, C_B signals coded according to Recommendation 601, provided that effective rounding is applied when converting from a higher number of bits to 8 bits at the equipment output. In [Croll *et al.*, 1987] a suitable form of rounding, called error feedback is described, in which those lower significance bits truncated from one sample are added to the following sample before truncation thus accumulating, rather than discarding, lower significance residues. With 8 bits per sample, rounding signals in this way causes higher frequency quantizing distortions which are not visible, whereas simple truncation can cause visible contouring. These historic views while valid at the time are no longer considered acceptable. The base line for most professional interfaces is 10 bits.

Thus, digital representation was initially defined in Recommendation ITU-R BT.601 for an 8-bit scale only. But many limitations appeared particularly in post processing. To overcome this problem it was proposed to use a 10-bit scale and to extend transmission interfaces (parallel and serial) for this 10-bit representation.

Therefore, Recommendations ITU-R BT.601, ITU-R BT.656 and ITU-R BT.799 from 1992 include the 10-bit scale as the preferred implementation value. It should be noted that a high degree of compatibility between the 8-bit and 10-bit parallel interfaces was maintained and that any 10-bit interface conforming to the present specification should operate when connected to an 8-bit interface, albeit with the loss of the two additional bits of signal data. Conversely, when an 8-bit interface is connected to a 10-bit interface the additional two bits are set to zero. The bit-serial interface operates with 10-bit words only: the two least-significant bits are zero when the interface conveys 8-bit source signals.

Unfortunately these same rules do not apply to ancillary data which may be added to the interface. Ancillary data has additional constraints that limit the compatibility between 8- and 10-bit implementations to a 6-bit word. Over time this limitation is expected to be removed as more and more implementation specify 10-bit operation as a minimum.

In the specification, the contents of digital words are expressed in both decimal and hexadecimal forms, denoted by the suffixes "d" and "h" respectively.

To avoid confusion between 8-bit and 10-bit representations, the eight most-significant bits are considered to be an integer part while the two additional bits, if present, are considered to be fractional parts. This bit representation has been adopted by the ITU-R for many of its Recommendations; other SDOs have used other means.
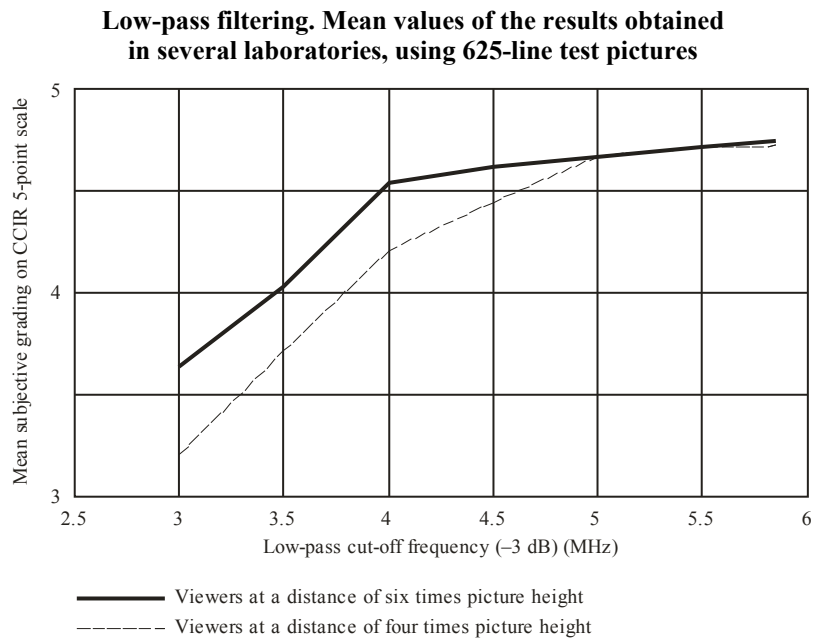
3.5 Filtering requirements

In choosing the sampling parameters to be used for component signals, the sampling frequencies are, clearly, closely related to the bandwidths of the component signals, but other parameters are concerned with the picture sampling structure or structures (see § 3.6).

3.5.1 Luminance signal bandwidth

In [24, 25] the results of tests by members of the EBU, to determine the relationship between subjective quality and the bandwidth of the luminance component, using a 625-line system and monochrome display are described. The tests were carried out using a method described in Report 405, using the impairment scale and pictures slightly more critical than average, as required by Recommendation ITU-R BT.500.

The main conclusion indicates that bandwidth limiting to 4.5 MHz (at –3 dB) using sharp cut-off low-pass filtering introduces an impairment which is imperceptible to 50% of the observers at a viewing distance of four times the picture height (see Fig. 1). The results indicate that the effects of filtering on electronically-generated captions are less critical than those on the pictures. These studies did not show the expected advantage for the use of a comb filter (with sub-Nyquist sampling) when compared with a low-pass filter cutting off at half the sampling frequency.

FIGURE 1

**Low-pass filtering. Mean values of the results obtained
in several laboratories, using 625-line test pictures**

Low-pass cut-off frequency (–3 dB) (MHz)

——————— Viewers at a distance of six times picture height

– – – – – – Viewers at a distance of four times picture height

digit_tv signals-01

The subjective test results obtained in these EBU tests have been analyzed in accordance with the method described in Annex III of Report 405 and this work is described in [26]. The result of the analysis is presented in Fig. 2.

FIGURE 2

**Impairment characteristics for bandwidth restriction
from EBU experiments: spread of results from different laboratories.
Viewing distance is 4 times picture height**

Curves X: Laboratory A (the least critical)
Y: Average of results from 7 laboratories
Z: Laboratory B (the most critical)

digit_tv signals-02

Digital television signals coding and interfacing within studios
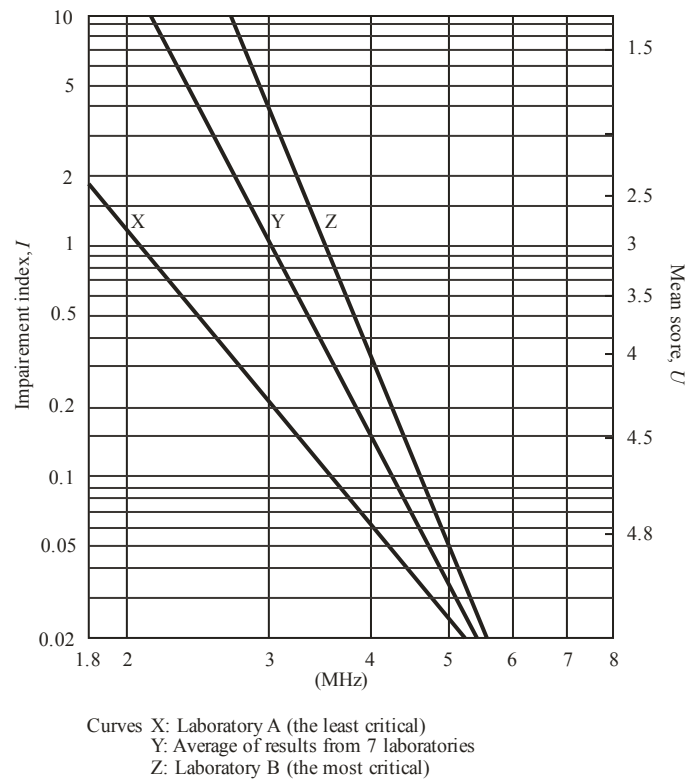
The same text discusses the substantial effect of the choice of picture material and the large spread of the results which can occur when Recommendation ITU-R BT.500 is not followed. It also expresses the view that an impairment criterion of $I = 0.05$ (corresponding to grade 4.8 in a five grade scale) is appropriate and that consequently the minimum bandwidth of the luminance signal should be 5.8 MHz.

Tests carried out in Poland and described in [27] also indicate that the luminance signal bandwidth should not be less than 4.5 MHz for 625-line systems. Work by the former Union of Soviet Socialist Republics [28] concludes that the appropriate bandwidth for the luminance signal is 6.0 MHz.

The relationship between picture quality and luminance signal bandwidth, for 525-line systems, has been investigated in Japan [29]. Separate tests were carried out, using colour pictures, for two different values of colour-difference signal bandwidths; the filters used in the luminance and colour-difference channels were of the Thomson Type (i.e. relatively slow cut-off). The results of the tests indicate that a luminance signal bandwidth of 5.6 MHz is suitable.

3.5.2 Colour-difference signal bandwidth

Tests in Poland [27] indicate that, for 625-line systems, the bandwidth of each colour-difference signal should not be less than 1.5 MHz, a result which agrees with that given in [28] from the USSR.

The work described in [29] also included tests to determine the colour-difference signal bandwidth appropriate to 525-line systems. The relationship between picture quality and colour-difference signal bandwidth was determined by separate tests, using colour pictures, in which two different values of luminance signal bandwidth were involved. From the results it is concluded that the colour-difference signal bandwidth should have a value of approximately 2.8 MHz.

Experiments conducted in France [Sabatier and Sallio, 1981] [Sabatier and Chatel, 1981] show very similar results for 625-line systems.

Investigations to establish the optimum characteristic for limiting the bandwidth of colour-difference signals, carried out in the Federal Republic of Germany, were made assuming that the maximum bandwidth available (the Nyquist limit) was 2.0 MHz [30]. Studies involving subjective tests have been made to determine the optimum characteristics for a filter, also with a bandwidth of 2.0 MHz [31].

Subjective assessments of the visibility threshold for colour signal ringing were carried out in Japan. The results show that the visibility threshold is reached with 2.5% or less, for electronically generated patterns and characters, and with 5% or more, for general pictures. Based on the results of the assessments, a colour signal shaping-filter was designed, for which the total disturbance due to ringing and aliasing was less than 2.5%. This characteristic can be realized by a digital transversal filter [32].

The specifications given in Recommendation ITU-R BT.601 were devised to preserve as far as possible the spectral content of the Y, C_R, C_B signals throughout the component signal chain. It is recognized, however, that the colour-difference spectral characteristic must be shaped by a slow roll-off filter inserted at picture monitors, or at the end of the component signal chain. In practice this assumed requirement may be ignored.

3.5.3 Bandwidth requirements for studio signal-processing applications

It must be borne in mind that the values of the luminance-signal bandwidth arrived at in §§ 3.5.1 and 3.5.2 are those regarded as desirable when considering the overall signal chain prior to the broadcast transmitter. When considering signal processing in the studio it should be pointed out, first, that the luminance bandwidth must be adequate to permit a reasonable amount of picture recomposition, without causing a perceptible impairment of the output picture. Secondly, the bandwidths of the colour-difference signals must be sufficient to enable good results to be obtained with chroma-key (colour-matte).

A colour-difference signal sampling frequency of 6 to 7 MHz (half that used for the luminance signal) appears to be satisfactory with regard to both picture quality and chroma-key (colour-matte) requirements, provided that the filtering characteristic shows 12 dB attenuation at half the sampling frequency; further, the frequencies at which the attenuation values are 12 dB and 3 dB should be related by the ratio 1.25:1. Over time, these assumptions have been born out in real world practical implementations.

The general problem of the design of the band-limiting filters used in the digital coding of television signals has been studied by several organizations. Computer simulations were also used to investigate the influence of several design parameters on the amplitude of overshoot and the amount of aliasing.

Based on the results of the studies mentioned above, and an examination of the problems of the practical realization of efficient filters to meet the requirements of all administrations, the characteristics given in Recommendation ITU-R BT.601 were drawn up.

Studies were continued to determine specifications for filters for the colour-difference signals for the 4:2:2 coding standard of Recommendation ITU-R BT.601. In these studies special attention has been given to:

a)   the need to maximize the usable bandwidths of both luminance and colour-difference signals;

b)   the need to ensure negligible impairments due to passband tolerances when a number of filter pairs are cascaded in a transmission chain;

c)   complementing b), the need to avoid unnecessarily stringent values and tolerances;

d)   the need for the specifications of both analogue and digital filters to be capable of being met in production at reasonable cost.

Some guidance on the practical implementation of the filters recommended in Recommendation ITU-R BT.601 is given in the following paragraphs.

In the proposals for the filters used in the encoding and decoding processes, it has been assumed that, in the post-filters which follow digital-to-analogue conversion, correction for the (sin x/x) characteristic is provided. The pass band tolerances of the filter plus (sin x/x) characteristic should be the same as given for the filters alone. This is most easily achieved if, in the design process, the filter, (sin x/x) corrector and delay equalizer are treated as a single unit.

The total delays due to filtering and encoding the luminance and colour-difference components should be the same. The delay in the colour-difference filter (see Fig. 5 of Annex 2 to Recommendation ITU-R BT.601) is double that of the luminance filter (see Fig. 4 of Annex 2 to Recommendation ITU-R BT.601).

As it is difficult to equalize these delays using analogue delay networks without exceeding the pass band tolerances, it is recommended that the bulk of the delay differences (in integral multiples of the sampling period) should be equalized in the digital domain.

In correcting for any remainder, it should be noted that the sample-and-hold circuit in the decoder introduces a flat delay of one half a sampling period. In modern day implementations digital filters with very accurate timing delays are employed, little if any analogue retiming is used.

The passband tolerances for amplitude ripple and group delay are recognized to be very tight. Present studies indicate that it is necessary so that a significant number of coding and decoding operations in cascade may be carried out without sacrifice of the potentially high quality of the 4:2:2 coding standard. Due to limitations in the performance of currently available measuring equipment, manufacturers may have difficulty in economically verifying compliance with the tolerances of individual filters on a production basis. Nevertheless, it is possible to design filters so that the specified characteristics are met in practice, and manufacturers are required to make every effort in the production environment to align each filter to meet the given templates.

Subjective assessments of the visibility threshold for colour signal ringing show that the visibility threshold is reached with 2.5% or less, for electronically generated patterns and characters, and with 5% or more, for general pictures.

The specifications given in Recommendation ITU-R BT.601 were devised to preserve as far as possible the spectral content of the Y, C_R, C_B signals throughout the component signal chain. It is recognized, however, that the colour difference spectral characteristic must be shaped by a slow roll-off filter inserted at picture monitors, or at the end of the component signal chain. It has been observed in Flat Panel Displays that presentation of digital signals may well contain alias components not seen on cathode ray tubes (CRTs) where the MTF of the spot beam was itself a form of LPF.

3.5.4 Bandwidth and filtering for conversion between different sampling rates

In [33] the digital filtering and interpolation characteristics necessary to interface between the 4:2:2 signals of Recommendation ITU-R BT.601 and signals conveyed using lower sampling frequencies are considered, and examples are given of filters of reasonable complexity, designed to preserve the information content. Only horizontal filtering is considered.

Subjective tests are described in [34] carried out using the above characteristic implemented by computer simulation. It was found that for natural pictures, the effects of sub-sampling are more visible in coloured areas, but some impairment was introduced by sub-sampling the luminance component of electronically generated pictures. The former conclusion led to consideration of quincunx sub-sampling for chrominance signals.

3.6 Sampling parameters

The sampling process is determined by three basic factors:
–   he sampling structure, i.e. the relative position of the samples in space and time;
–   the number of samples per line; and
–   the filtering process, which may be one-, two-, or three-dimensional.

The sampling structure mentioned above may, or may not, be repetitive with respect to the picture. Likewise, the number of samples per line may, or may not, be constant from line to line. In all the examples given below the sampling patterns were picture-locked.

A general theoretical survey is given in [Kretz and Sabatier, 1981], and some comparisons between orthogonal and quincunx sampling patterns are given in [35]. These indicate that the orthogonal pattern has some advantages.

3.6.1 Sampling rates

The EBU has studied the problem of defining a standard set of essential coding parameters for television studio equipment. The work was carried out with four main objectives:
–   to eliminate, in the production area, the differences between the existing 625-line systems;
–   to ensure that the picture quality obtained is as high as, or higher than, that obtainable with good modern practice using analogue techniques;
–   to ensure that the standard is suited to technology that is available at present, or is likely to be available in the near future; and
–   to arrive at parameter values which take into account the needs of picture processing in the studio.

In a first series of studies [36] orthogonal sampling was used with sampling rates of $768\,f_H$, (line frequency) for the luminance signal and $256\,f_H$ for the colour-difference signals. The colour-difference samples were co-sited and each pair of colour-difference samples was co-sited with a luminance sample. These studies indicated that further work was necessary, with particular regard to the influence of picture processing requirements on the choice of sampling rates.

The sampling parameters discussed in [36] are considered in [37] with regard to their suitability for digital signal transmission at a rate of 140 Mbit/s. The view is expressed that signals based upon these sampling parameters would be of higher quality than those now provided by analogue transmissions.

The EBU has carried out a further series of studies relating to sampling rates. This work included experiments to asses the picture quality available using a number of parameter-set values within the range 12:4:4 to 14.3:7.15:7.15 where the numbers in these ratios correspond to the sampling frequencies (MHz) used for the luminance signal and for the two colour-difference signals, respectively. Apart from the 12:4:4 set, all the parameter sets assessed had a ratio of 2:1 between the luminance and colour-difference sampling frequencies. In [38] and [EBU, 1981] these investigations are outlined and the results obtained with each of the parameter-set values, with regard to a number of attributes, are discussed.

First, concerning the picture quality obtained after one analogue RGB to digital YUV conversion, the results indicate, in broad terms, that the 12:6:6 parameter set gives a perceptibly better performance than the 12:4:4 set. However, they also indicate that the performance of the 14.3:7.15:7.15 set, in this regard, is not significantly better than that obtained using the 12:6:6 set.

Secondly, with regard to the quality of chroma-key (colour-matte) obtained, the results indicate that, while the 12:6:6 set clearly provides a better performance than the 12:4:4 set, a relatively steady improvement in performance can be noted as the luminance and colour-difference sampling frequencies are further increased to their maximum values, i.e. conforming to the 14.3:7.15:7:15 set.

Thirdly, it was found that, in tests involving a moderate amount of horizontal picture expansion, no aliasing could be observed, using natural (non-electronically generated) test pictures, for all luminance-signal sampling frequencies within the range 12 to 14.3 MHz. However, using an electronically generated horizontal frequency sweep, aliasing decreased with increasing sampling frequency: the decrease was most noticeable in the range 12 to 13 MHz.

Finally, with regard to bit-rate reduction, the studies show that it is possible to reduce the bit rate of a signal conforming to the 14.3:7.15:7.15 set so as not to exceed 140 Mbit/s, without affecting the picture quality or the potential capability for chroma-key (colour-matte) processing.

Some of the above-mentioned results were derived from work undertaken in the United Kingdom [39]. This work included tests on two parameter sub-sets, in order to investigate the properties of systems possibly qualifying as lower members of the family of compatible digital coding standards. In one of the sub-sets the luminance and colour-difference signal sampling frequencies were related by the ratio 4:1.

The subjective tests involved in this work were carried out using the double-stimulus method described in Recommendation ITU-R BT.500 and reference pictures derived by digitally coding the input signals according to the 14.3:14.3:14.3 parameter set.

The detailed subjective test results obtained during the above-mentioned tests have been analyzed by the method described in Annex III of Report 405 and the results of the analysis, described in [40], are given in terms of the impairment index I and the mean score $\overline{U}$.

From the analysis it can be concluded that with regard to basic picture quality the 13.5:6.75:6.75 parameter set is characterized by an impairment index I of 0.03 at 4 H (i.e. a mean score greater than 4.8) and that the corresponding results related to chroma-key (colour-matte) performance are $I = 0.3$ and $\overline{U} = 4.0$.

In the former Union of Soviet Socialist Republics (USSR) consideration has been given to the sampling parameters suitable for standards D and K. Early work included the study of a system in which the luminance signal was sampled orthogonally at a rate of 800 f_H (12.5 MHz) and each of the colour-difference signals were similarly sampled at 200 f_H (3.125 MHz); the investigation is outlined in [41].

In studies described in [42] the sampling parameters for digital TV studios were revised as it was found desirable to increase the sampling frequencies to about 13 to 13.5 MHz and 6.5 to 6.75 MHz for the luminance and colour-difference signals respectively.

In [43] the results of comprehensive subjective tests are reported which were carried out with the objective of determining the relationship between the reproduced colour picture quality and the sampling frequencies and sampling structures used for the luminance and colour-difference signals.

A standard designed to be a member of the family of digital coding standards below that recommended as the main studio standard is described in [44]. In this standard the luminance signal is sampled at 10.125 MHz and the useful bandwidth extends to 5 MHz. The colour-difference signals are sampled at a frequency equal to 3.375 MHz, and have a bandwidth of 1.5 MHz. Further studies are necessary to select the best sampling structure for this system. The system is known as a 3:1 system, owing to, first, the particular ratios used to calculate the sampling frequencies from the frequencies recommended for the main studio standard and, secondly, the use of line-sequential coding of the colour-difference signals. It is claimed that the picture quality obtained from this system is at least equal to that obtained using a conventional analogue PAL codec. This proposal was never commercially implemented.

A system for coding a television signal at a bit rate of 70 Mbit/s using the DPCM method (folded quantization with 5 bits/sample) is described in [45] and by [Wengenroth, 1982]. The sampling frequencies for luminance and colour-difference signals are derived from the studio standard by means of a digital filter which gives a conversion of the sampling frequency in the ratio 6:5. Line-sequential transmission is foreseen for the two colour-difference signals.

Extensive subjective tests, using a method described in Report 405, Annex IV, were carried out in the United States of America using digitally coded component signals conforming to the 525-line standard; these are outlined in [46] and [SMPTE, 1981]. The tests covered luminance-signal sampling rates corresponding to 768, 864 and 912 samples per total line and ratios of luminance-signal to colour-difference signal bandwidth of 4:4:4, 4:2:2, 4:1:1 and 2:1:1. Tests were carried out with regard to basic picture quality and on the properties of the various parameter sets with regard to production processes such as picture expansion, chroma-key (colour-matte), multi-generation digital recording and digital decoding from, and encoding into, analogue M/NTSC composite colour signals. The tests confirm the selection of the 4:2:2 parameter set as the one preferred as a studio standard, and showed that a small but rising picture quality was obtained with an increase of the sampling rate.

Subjective tests with similar sampling parameters were carried out in Japan, and these are described in [47]. The tests included digital chroma-key (colour-matte) processing for parameter sets with luminance signal sampling frequencies of 12.1, 13.6 and 14.3 MHz. The two main results can be summed up in the following way. Firstly, the picture quality decreases gradually as the sampling frequency of the colour-difference signals decreases from a value equal to that used for the luminance signal to a quarter of that value. Secondly, the picture quality obtained using the chroma-key (colour-matte) process decreases significantly as the sampling frequency used for the colour-difference signals is reduced; this decrease is more marked when the sampling frequency is reduced from half that used for the luminance signal to a quarter of that value.

It should be noted that all these tests were conducted using interlace capture, had progressive capture been employed the results may well have been different.

A theoretical study of picture distortions in line-sequential transmission of colour-difference signals with a two field cycle is contained in [48] and [Khleborodov, 1983].

In [49] results are described of subjective assessments of picture quality obtained with 4:1:1, 4:2:0, 2:1:1 and 3:1:0 coding, including the effects of line- and field-offset sub-sampling, and of line-sequential processing of the colour-difference signals. These tests used stationary test pictures. The results showed that the picture quality for 2:1:1 with field-offset sub-sampling was superior to that of the other members with a similar bit rate for most of the pictures tested.

Subjective tests are reported [50] on a 2:1:1 system employing field-offset sub-sampling which suggests that satisfactory portrayal of movement can be achieved.

3.6.2 Samples per line

There are equipment advantages for members of a family to have compatible numbers of samples per line. The members of the digital family should be chosen having regard to other possible television signal representations, in order to limit the number and complexity of transcoding operations in the signal chain. Some of the representations that have been proposed are listed in [51].

The nominal duration of the active lines for 525/60 and 625/50 systems are slightly different. To bring the two systems together in the digital component domain, a digital active line (DAL) is defined. This has sufficient digital samples to cover both the 525/60 and 625/50 active lines. There are obvious advantages in that both systems now need precisely the same amount of digital line storage, and the DAL can be processed in exactly the same way for either the 525-line or 625-line systems. The number of samples associated with analogue blanking is different, but this need not be carried through into the digital component domain. .

The 525 active line is the longer of the two, and with the tolerances currently in practical use, more than 710 samples are needed to cover a line. The 720 samples per digital active line given in Recommendation ITU-R BT.601 was chosen because it conveniently meets this requirement.

Over the years a number of debates have taken place related to the equivalent number of samples of the active line that corresponds to an analogue line. In all cases due to longer analogue blanking the full 720 samples are not used in analogue systems. It should be noted that in some implementations the number of samples truncated resulted in a centre of the picture shift. With the passage of time and the conversion to wholly digital technology these anecdotal analogue stories will diminish.

3.6.3 Sampling frequency tolerance

The sampling frequencies used should comply with the requirements of associated television systems. In particular, the tolerance for component signal sampling frequencies should be equal to that for line frequency in the relevant colour television standard [52, 53].

3.6.4 Changing the sampling rate

Sampling rate changing is a process required in many picture processing operations. One example is that involved in converting the signals conforming to one member of the family of compatible coding standards to another. A filtering process is described [54], based on comb filtering of the upper part of the signal spectrum, which enables signals to be converted easily between various members of a binary-related family (4:4:4; 4:2:2; 2:1:1). In [55] and [Nishizawa, 1981] a very sophisticated interpolation low-pass filter intended for the same purpose is described.

Change of sampling rate is also required when the family of compatible digital coding standards is not based upon binary ratios. In [56] it is indicated that the design of the interpolating filter need not be unduly complicated, provided that the change of sample rate involves a ratio described by a rational number.

In cases where the sampling frequencies are generated from the reference synchronization signal arriving from a distant main synchronization generator, special centralized synchronization signals containing reference frequency packets may prove useful for the purpose of increasing the phase stability of the generated frequencies [57].

3.7 Relevance to composite signals

The sub-Nyquist technique can be applied by sampling the PAL composite signal at a rate of twice the sub-carrier frequency [Devereux and Phillips, 1974]. The digital codec employing such sampling uses filtering which introduces minor losses of diagonal luminance resolution and vertical chrominance resolution. However, further filtering of this nature in any subsequent sub Nyquist sampling process should not cause any further resolution impairment [Stott and Phillips, 1977].

In [58] the EBU draws attention to the need for compatibility between the proposed methods for composite and component coding of 625-line signals.

EBU experiments [59] have shown that good quality analogue PAL to digital YUV coding and decoding, when using line-locked sampling, can be achieved provided a sophisticated codec is used.

In [Clarke, 1986], details of digital techniques are given for generating, from a line-locked sampling frequency, quadrature sub-carrier signals for use in composite colour encoders and for locking these signals to the incoming reference burst in colour decoders. This allows the advantages of more stable, better defined performance produced by digital implementation to be obtained in coders and decoders without the need for sample rate conversion. The description also includes the modifications necessary to provide operation with NTSC signals.

Further, in [Clarke, 1988], a number of filtering methods for achieving high-quality separation of the luminance and chrominance components of a PAL signal are described, primarily for obtaining digital Y, C_R, C_B signals from PAL in a digital studio context. These range from filters using relatively simple line-delay combs to three-dimensional comb filters using multiple line, field and picture delays. The results of subjective tests comparing the different methods indicate that, while a modest degree of temporal filtering can improve performance significantly, greater use of this technique can lead to impairments to moving objects.

While at the time when the conversion from the analogue world to the digital world was taking place, a great deal of attention and concern was expressed over the various conversions that would be required. In practice, while the concerns were real, the implementation for the most part resulted in very few conversion problems.

Implementation of digital composite systems was largely confined to NTSC countries where limited installations were employed. The complexities of handling digital composite signals plus the flexibility offered by the component systems were driving forces that resulted in digital composite systems having a short-lived life. Moreover, most of the popular compression schemes used by broadcasters were component based, which had the effect of accelerating the implementation of digital component systems.

4 Studio digital encoding parameters for HDTV

Recommendation ITU-R BT.709

Recommendation ITU-R BT.709 is one of the famous ITU-R Recommendations that lead to the word wide adoption of HDTV image parameters. There has been worldwide adoption of ITU-R BT.709 as a HDTV standard. Since the early 1980 demonstrations of HDTV the Recommendation has undergone a number of changes including the adoption of 24-frame/s.

Recommendation ITU-R BT.709 offered the industry a number of improvements over SDTV, in particular colorimetry encompassing a range which includes the majority of real-surface colours and a high contrast range. In addition the ability of interfaces to pass progressive HDTV images has become a reality with interfaces operating at 3 Gbit/s with either copper or fibre optic transports. Recommendation ITU-R BT.1367 defines an optical interface for both SDTV and HDTV.

In the beginning the only implementations were Analogue, Y, C_R, C_B, along with a modified analogue Format C VTR with a head to tape speed of 32 M/s. In most respects the signal parameter values and the suggested anti alias filters are scaled versions of Recommendation ITU-R BT.601. While Recommendation ITU-R BT.709 has two image formats only the 1 920 × 1 080 format has seen widespread implementation.

Since the mid 1990s, only digital implementations of Recommendation ITU-R BT.709 have been commercially available. With the inclusion of 24-frame/s as a part of the Recommendation many popular TV series ("Soaps", documentaries and drama) have become available on a worldwide basis without the need for so called standards conversion.

5 Interfaces

Recommendations ITU-R BT.656 and ITU-R BT.799

Once the basic parameters of the original digital component coding standard had been established in Recommendation ITU-R BT.601 that is, sampling at 13.5 MHz with 8 bits/sample – work could begin on devising interfaces to permit the interconnection of equipment operating to this standard, known as the 4:2:2 level. The aim was to enable the signal to remain in the digital domain and so avoid frequent conversions between analogue and digital formats and the impairments introduced in the analogue domain. Groups within EBU, SMPTE and elsewhere undertook this work. Before it was complete, interest in the use of 10-bit data-words had grown to the point that they too were accommodated in the interfaces.

The basic problem was how to convey the three component signals, luminance and two colour-difference signals, in a reliable and economical fashion. Given the sampling rate and sample resolution, the data to be conveyed amounts to a total of 27 Mword/s, or 270 Mbit/s at 10 bits/sample. (Actual video data rate was approximately 210 Mbit/s, the 270 Mbit/s number includes the blanking intervals.)

At an early stage it was recognized that a single cable interconnection is desirable, to avoid the inconvenience of multiple cables and complex routing arrangements. Thus work concentrated on creating a multiplex of the three components on a single cable.

The task facing those working on digital interfaces could be considered as having three parts: a common signal format definition, including multiplex structure and synchronization, a bit-parallel interface and a bit-serial interface.

The resultant interface specifications are incorporated in Recommendations ITU-R BT.656 and ITU-R BT.799. The following notes of this Handbook give background information.

5.1 Multiplex control

If the three signals are to be combined in a single multiplex, a means must be provided to allow them to be identified and separated from the multiplex. For this, some form of synchronization is required. There are other synchronization requirements: it must be possible to measure the timing offset between different data streams (e.g., for mixing) and the correct relationship between the sync waveform and the picture waveform has to be established when the digital signal is converted back to analogue form.

These requirements are met by the inclusion of a synchronization code, known as the timing reference code, in the data-stream. This has to be unique so that video or other data are not mistaken for it, it must be easily detected and it must be robust to permit detection even when corrupted by bit errors. To create unique codes, the data values FFF_H and 000_H (hexadecimal notation), corresponding to "all ones" and "all zeros" were designated as reserved values for housekeeping purposes and excluded from the video data coding range. The timing code comprises a fixed three-word preamble followed by a fourth word which indicates the states of line and field blanking and the first or second field. A further timing reference code is inserted immediately preceding and immediately following the 1440 words of video data in each line and are referred to as Start of Active Video (SAV) and End of Active Video (EAV) respectively.

Thus the video data can be regarded as being in packets, each of which begins and ends with a timing reference code. In the interests of simple signal processing, it was decided that all lines should be the same length; thus there are no half-lines in the signal format. The packets contain the same amount of data in the 525-line and 625-line systems.

5.2 Serial and parallel interfaces

Initially it was believed that serial interfaces would require the use of expensive, complex high-speed circuitry and that the majority of interconnections would be by bit-parallel bearers within a studio, with bit-serial connections reserved for long, inter-area links where the cost of the sending and receiving circuitry could be justified by the use of low-cost coaxial cable. Consequently, early work on implementing interfaces centred on the development on a bit-parallel interface, once it had been demonstrated that useful interconnection distances could be achieved. Connections up to 200 m were demonstrated, using specially designed multi-pair cables.

The bit-parallel interface used the widely available 25-way D type connector for its ten data pairs and clock pair. However, the size of the connector and the cost of assembling it, together with the difficulty of constructing large routing matrices with eleven cross points for each signal, led to interest in the alternative serial interface. Few if any large systems were implemented using the bit-parallel interface.

A serial interface specification, supported by the EBU had been incorporated in Recommendation ITU-R BT.656. This was based on mapping 8-bit data words to 9-bit transmission words (8B9B), the 9-bit words being selected for their desirable transmission characteristics (large number of transitions to ease clock recovery and low DC component to simplify equalisation). A ROM-based look-up table was used for the mapping, special words being used for the FF_H and 00_H code words in the video data stream to enable the de-serializer to be correctly synchronized.

This system proved complex to implement and unsuitable for use with the 10-bit and greater interfaces gaining in popularity and was therefore overtaken by a serial interface based on scrambling a 10-bit signal proposed in 1985. The latter was subsequently incorporated into Recommendation ITU-R BT.656.

The initial Serial Digital Interfaces use coaxial cable as the means of studio interconnections. It is now defined as an optical connection as well.

5.3   Key signal

In addition to interfaces for component video signals, interfaces are needed for other signals encountered in production studios, the most common being key signals often associated with a video signal. The various groups working on digital video interfaces agreed to treat key signals as luminance signals and to use the 4:2:2 interface for this purpose. Initially, the data carried in the "colour-difference" channels of the key interface was not specified but its use was reserved until such time as an agreed specification could be issued. In later years with the use of multiple channels of the interface, the key channel was carried as an "A" channel, commonly known as an Alpha channel.

5.4   Synchronization

Recommendation ITU-R BT.711

Recommendation ITU-R BT.711 deals with synchronising reference signals for the component digital studio. Report ITU-R BT.1219 provides background information on the considerations leading to the preparation of the Recommendation.

The move to digital operation in studios has not removed the need for synchronisation. On the contrary, several levels of synchronization need to be considered:

–   synchronization of an interface demultiplexer with the multiplexer so as to assemble and disassemble the multiplex correctly;

–   synchronization of video data streams being combined in digital mixing and keying operations;

–   synchronization of the resulting analogue video data to the analogue synchronization waveform at the point where digital-to-analogue conversion takes place;

–   synchronization of signals from outside the studio with those generated within; and,

–   synchronization of the outputs of devices such as cameras, caption generators, digital tape recorders and disk stores to the local studio reference.

The first four of these can make use of the timing reference codes embedded in the digital video data streams, but the last requires the distribution of a studio synchronization reference signal.

The signal that is used to synchronize digital equipment is the analogue color black signal defined in Recommendation ITU-R BT.711. From time to time efforts were made to replace the analogue signal with some form of digital signal. The complexities and costs related to the implementation of two reference signals has not warranted any further investigation.

5.5   4:4:4 interfaces

Whilst the 4:2:2 interface is adequate for all the normal studio operations, there are some specialized applications where the use of full bandwidth RGB or colour-difference signals can give worthwhile improvements, examples being chroma-key and computer graphics. For these applications, the 4:4:4 level of Recommendation ITU-R BT.601 is appropriate.

At this level, either the red, green and blue primary signals or the luminance and full-bandwidth colour difference signals can be conveyed from one equipment to another, in each case accompanied by a key or A signal if required.

The interface for this level is based on the use of two 4:2:2 interfaces operating in parallel. A 4:2:2 interface can be considered as having two 13.5 Mword/s channels, one of which carries the luminance signal while the other carries the two half-bandwidth colour-difference signals. By operating two such interfaces in parallel, the resulting four channels can be used for full-bandwidth RGB plus Key, or luminance, two colour-difference signals and Key.

The 4:4:4 interface (also referred to as the 4x4 or 4:4:4:4 interface, because of the inclusion of the key channel) is the subject of Recommendation ITU-R BT.799 .

6 Testing

Recommendation ITU-R BT.801

In digital systems, just as in analogue systems, testing is used to give type approval to new designs of equipment and to verify their correct operation in service. However, impairments in digital systems are likely to result from errors in the conversions to and from the digital domain (including filtering, sampling and quantization) and by degradation of the signal itself (such as bit-errors, excessive timing jitter or loss of synchronization). The nature of the errors occurring in digital systems are such that the use of conventional analogue techniques such as insertion test signals (e.g. pulse and bar waveform) are not appropriate.

Sets of test signals have been specified in Recommendation ITU-R BT.801 to facilitate testing digital component equipment. These enable the analogue-to-digital conversion process to be examined, the presence of reserved codes (000_H and FFF_H) to be detected, the electrical performance of the parallel interface to be verified, and the operation of the blanking circuits to be observed. These signals are useful when examining new designs of equipment.

For the in-service testing of equipment, Recommendation ITU-R BT.1304 provides for a monitoring system based on the generation of a checksum from the video data at one point in the signal chain which is carried with the video data and compared with checksums generated at subsequent points down the chain. Any discrepancy between the locally generated checksum and that carried with the video data would indicate the occurrence of a fault. The check sum scheme was implemented for SDTV only interfaces. The HDTV interface defined by Recommendation ITU-R BT.1120 contains an ECC scheme on every line. While there is no error correction this is an error indicator. The typical error rate of an HDTV SDI connection is order of 10E14.

In addition the error handling schemes employed by Recommendations ITU-R BT.656, ITU-R BT.799, and ITU-R BT.1120, specific test signals have been developed to stress test the ability of SDI receivers to function correctly.

7 Types of ancillary signal

It was recognized from the outset of the work on digital video interfaces that there is a substantial amount of capacity in the data stream not used for carrying useful video data but carrying line- and field-blanking data, commonly referred to has HANC and VANC. Since each digital active line corresponds to 1440 data words in the interface and in the 525-line (625-line) system there are 39 (49) blanked lines the capacity available due to field blanking alone corresponds to an unformatted data channel of 1.4 (1.76) Mbit/s. In addition there are data words available in the line blanking of each line.

The horizontal and vertical blanking periods are capable of carrying ancillary data such as audio and other data formatted according to Recommendation ITU-R BT.1364. Readers should be aware that not all digital processing equipment such as video switchers, VTRs, file servers may transparently pass the blanking intervals; HANC and or VANC data may be corrupted.

For example, existing digital-component video tape recorders do not record the line blanking data or all lines of the field blanking.

8 HDTV interfaces

HDTV Serial Digital Interfaces (HD-SDI) use a component format, usually Y, C_R, C_B similar to that adopted in Recommendation ITU-R BT.601 and multiplexed in a similar way as set out in Recommendation ITU-R BT.656. These 4:2:2 signals are carried by HD-SDI at an interface bit rate of 1.5 Gbit/s.

The synchronization arrangements are similar, including extensions to handle 4:4:4:4 signals at 3 Gbit/s. The 3 Gbit implementations can be either two channels of 1.5 Gbit/s or a single 3 Gbit/s link.

For the serial HDTV interface, the same scrambling algorithm that has been adopted in Recommendation ITU-R BT.656 has been applied to the HD-SDI.

For HDTV, the use of optical fibre for serial interconnections becomes necessary at distances over about 150 m and specifications have been developed for fibre interfaces Recommendation ITU-R BT.1367.

All current system implementations of HDTV use either the coaxial cable SDI or optical SDI.

8.1 Progressive segmented frame

The television systems in current use have typically used interlace capture (acquisition) and transmission. The frame/field rates of these systems have been 50/60 Hz, a rate that when presented on CRT display devices did not require any associated picture flicker correction. Television systems of the future will support both interlace and progressive capture and display technology.

In addition to the support of interlace and progressive capture and display, there will be extended frame rates to be supported, along with new display technology. For a number of years there will be a mix of the old and new technologies.

Specifically, the progressive segmented frame (PsF) technology is intended to be implemented only when frames rates of 30 Hz and lower are being used.

A large percentage of television programming is produced on film that has a frame rate of 24-frames/s and sometimes 30-frames/s. Past practice was to perform post production by editing the film to produce a complete programme on film. The final film could be transferred to 60 Hz video by employing the 3:2 pull down technique. For 25 Hz release the film could be transferred by running the 24-frame film at 25-frames/s.

8.1.1 24-frame/s production

Using the CIF of $1\,920 \times 1\,080$, $\times$ 24Fp/s material may be transferred using progressive capture. This transfer will provide the highest resolution capture, with no 3:2 pull-down artifacts, moreover both 30 Hz frame rate and 25 Hz frame rate versions may be created from a single master with no quality loss.

The 30 Hz frame rate copy may be created by playing the 24-frame/s original and inserting the 3:2 pull-down during the replay process. This process also has the advantage of maintaining the 3:2 pull-down sequence during the replay process such that any downstream picture processing, such as an MPEG encoder, will not be affected by any 3:2 discontinuities.

The 25 Hz frame rate copy may be created by simply playing back the 24 Hz rate original at the slightly faster 25 Hz rate; there is no picture quality loss.

8.1.2 Progressive/interlace compatibility

The post-production world has a need to cater for both progressive and interlace television signal formats for the foreseeable future. Therefore, any new signal format such as 24P, the original frame rate, will need to coexist with interlace formats of 25 Hz and 30 Hz systems. One of the constraints in monitoring the 24-frame/s systems is the picture flicker that is present when displaying a 24-frame/s signal on a CRT display. Interlace systems minimize this flicker by refreshing the CRT phosphors every 60th/50th of a second. There are at least two solutions to the flicker created by the 24-frame/s systems, install a frame store in every monitor, or provide to the monitor a signal that emulates the interlace refresh rate.

24PsF/25PsF/30PsF are transmission formats that will provide monitoring devices with signal refresh rates hat will permit direct monitoring of the original frame rate of the material.

It should be noted that in some cases users may want to monitor 24-frame/30-frame material at other than the original frame rates.

The use of 24PsF/25PsF/30PsF does not in any way limit the monitoring of the signal by the newer flat panel displays.

A second potential use of the 24PsF/25PsF/30PsF transmission format is in the area of digital post production switchers. A common switcher design handling both interlace and progressive signals is economically possible, and addresses the requirements of end users who have a requirement to work in interlace and progressive formats with common equipment. The digital interface of an interlace signal and a PsF signal are common, only the signal content is different.

8.1.3   Signal mapping

The 24PsF/25PsF/30PsF transmission format maps a progressive image onto the interlace digital serial interface.

The odd lines of the progressive image are mapped to one field of the interlaced interface while the even lines are mapped to the other field of the interlaced interface. The progressive image when reconstructed does not lose any of the progressive capture characteristics, there is no vertical filtering applied as is the case with interlaced images.

The PsF format is not related to any interlace format characteristics. It is a way to convey a progressive image that has been captured at a 24/25/30 Hz rate. Capture at these low frequencies may require special monitoring considerations. The PsF transmission format is intended to provide an economical solution while still retaining the compatibility with interlace systems.

8.2   HDTV SDI testing

Stressing of the automatic equalizer is accomplished by using a signal with the maximum number of one's or zeros, with infrequent single clock period pulses to the opposite level. Stressing of the PLL is accomplished by using a signal with a maximum low-frequency content; that is, with a maximum time between level transitions.

Channel coding of the HDTV serial digital signal utilizes scrambling and encoding into NRZI accomplished by a concatenation of the two following functions:

$$G1\,(x) \,=\, x^9 \,+\, x^4 \,+\, 1 \qquad\qquad G2\,(x)\, =x+\,1$$

As a result of the channel coding, long runs of zeros in the $G2\,(x)$ output data can be obtained when the scrambler, $G1\,(x)$, is in a certain state at the time when the specific words arrive. That certain state will be present on a regular basis; therefore, continuous application of the specific data words will regularly produce the low-frequency effects.

Although the longest run of parallel data zeros (40 consecutive zeros) will occur during the EAV/SAV timing reference sequence (TRS) words, the frequency with which the scrambling of the TRS words coincide with the required scrambler state to permit either stressing condition is low. In the instances where this coincidence occurs, the generation of the stressing condition is so time limited that equalizers and PLLs are not maximally stressed.

In the data portions of digital video signals (excluding TRS words in EAVs or SAVs, and ANC data flag words), the sample values are restricted to exclude data levels 0.00 to 0.75 and 255.00 to 255.75 (000_h to 003_h and $3FC_h$ to $3FF_h$ in 10-bit hexadecimal representation and 00.0_h to $00.C_h$ and $FF.0_h$ to $FF.C_h$, in 8.2 hexadecimal notation) (see Note 1). The result of this restriction is that the longest run of zeros, at the scrambler input, is 16 (bits), occurring when a sample value of 128.00 (200_h or 80.0_h) is followed by a value between 1.00 (004_h or 01.0_h) and 1.75 (007_h or $01.C_h$). This situation can produce up to 26 consecutive zeros at the NRZI output, which is (also) not a maximally stressed case.

Other specific data words in combination with specific scrambler states can produce a repetitive low-frequency serial output signal until the next EAV or SAV affects the scrambler state. It is these combinations of data words that form the basis of these test signals.

Because of the *Y/C* interleaved nature of the component digital signal, it is possible to obtain nearly any permutation of word pair data values over the entire active picture area by defining a particular flat colour field in a noise-free environment. Certain of these permutations of word pair data values will produce the desired low-frequency effects.

Receiver equalizer testing is accomplished by producing a serial digital signal with maximum d.c. content. Applying the sequence 192.00 (300_h or $C0.0_h$), 102.00 (198_h or 66.0_h) continuously to the *C* and *Y* samples (respectively) during the active line will produce a signal of 19 consecutive high (low) states followed by one low (high) state in a repetitive manner, once the scrambler attains the required starting condition. Either polarity of the signal can be realized, indicated by the level of the 19 consecutive states. By producing approximately half of a field of continuous lines containing this sequence, the required scrambler starting

condition will be realized on several lines, and this will result in the generation of the desired equalizer testing condition.

Receiver PLL testing is accomplished by producing a serial digital signal with maximum low-frequency content and minimum high-frequency content (i.e., lowest frequency of level transitions). Applying the sequence 128.00 (200_h or 80.0_h), 68.00 (110_h or 44.0_h) continuously to the C and Y samples (respectively) during the active line will produce a signal of 20 consecutive high (low) states followed by 20 low (high) states in a repetitive manner, once the scrambler attains the required starting condition. By producing approximately half of a field of continuous lines containing this sequence, the required scrambler starting condition will be realized on several lines, and this will result in the generation of the desired PLL testing condition.

Because the equalizer test works by producing a serial digital signal with a bias, steps must be taken to ensure that both polarities of bias are realized. To change the polarity of the bias from one frame to the next, the sum total of all the bits in all the data words in all the lines in a video field must be odd.

To ensure that the polarity of the bias can change often, a single Y sample data word in the signal is changed from 120.00 (198_h or 66.0_h) to 100.00 (190_h or 64.0_h) (a net change of 1 data bit), once every other frame. This causes the bias polarity to alternate at a frame rate regardless of whether the original frame bit sum is even or odd. The data word in which the value substitution is made is the first Y sample in the first active picture line of every other frame. The specific word and line for each signal format is listed in Table 34 as the polarity control word.

The sequence 192.00 (300_h or $C0.0_h$), 102.00 (198_h or 66.0_h) and 128.00 (200_h or 80.0_h), 68.00 (110_h or 44.0_h) applied to C and Y samples results in shades of purple and gray, respectively. Reversing the C and Y ordering for each of these two sequences results in lighter and darker shades of green, respectively.

If the ordering is reversed, then the polarity control word is changed to 128.00 (200_h or 80.0_h). The polarity control word in either case is located at the first Y sample in the first active picture line in the field(s).

8.3   HD-SDI as a data channel

In addition to the HD-SDI being used for transmission of uncompressed video and audio signals, it is possible to use the payload of the HD-SDI to carry almost any data in the defined active video area. Care needs to be taken to avoid the words used by the HD-SDI (or SDTV SDI) for synchronization essentially reducing the useable payload by 20%. The term HDSDTI is used to describe the interface when carrying formatted data. (Serial Digital Transport Interface.) Recommendation ITU-R BT.1577 defines the mapping and coding rules for the HDTV-SDTI interface. Recommendation ITU-R BT.1381 covers the SDTV- SDTI.

Recommendation ITU-R BT.1577 specifies a data stream used to transport packetized data within a studio/production centre environment. The data packets and synchronizing signals are compatible with Recommendation ITU-R BT.1120 (see Fig. 3). The Recommendation describes the assembly of two channels of 10-bit words multiplexed onto one HD-SDI line for the purpose of transporting the data streams in a structured framework. The HD-SDTI data blocks and synchronizing signals provide a data transport protocol that can readily be added to the infrastructure described in Recommendation ITU-R BT.1120.

FIGURE 3

SDI-SDTI Compatible Formatting

Arrangement of HD-SDTI wrapped around Recommendation ITU-R BT.1120

digit_tv signals-03
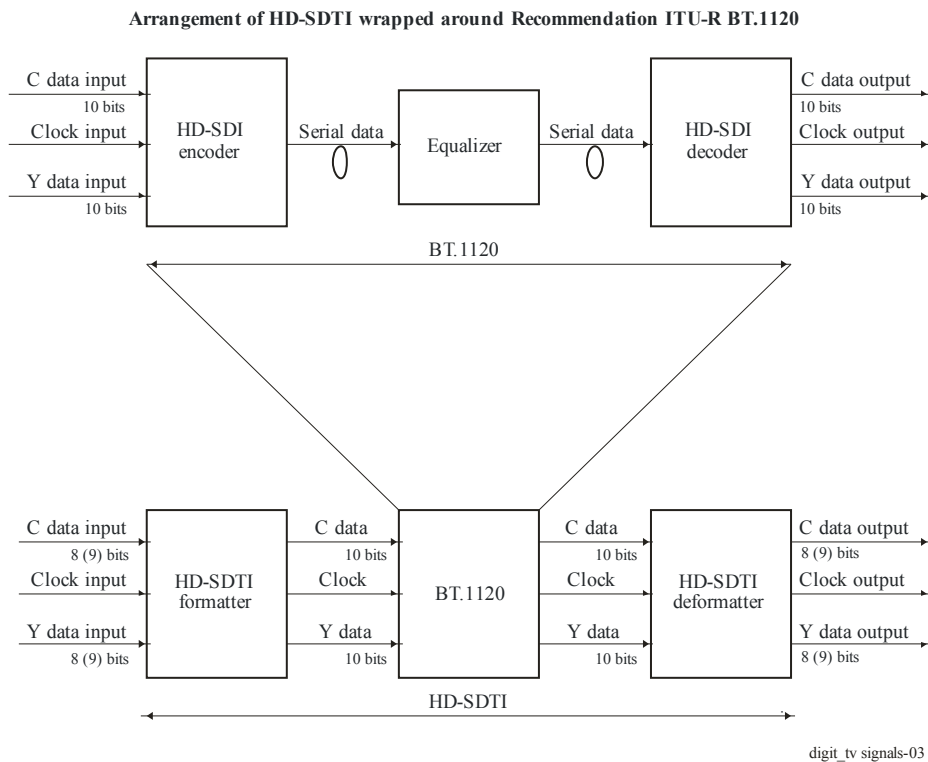
The HD-SDTI data shall be serialized, scrambled, coded, and interfaced according to Recommendation ITU-R BT.1120 and the associated source format standard. The signal specifications and connector types shall be as described in Recommendation ITU-R BT.1120.

The data word length shall be 10 bits defined as bits B0 through to B9. B0 is the least significant bit (LSB) and B9 is the most significant bit (MSB). The order of bit-transmission shall be LSB first as defined in Recommendation ITU-R BT.1120.

As described above the HD-SDI may be used as a general transporter of data, many Recommendations describe the packetized data that is carried by the interface including payload identifiers (see Recommendation ITU-R BT.1614).

Further extensions to the use of the HDTV-SDI are expected for the mapping of UHDTV images onto a multiplex of 1.5 Gbit/s into a 10 Gbit/s optical stream.

9 Present and future routing strategies

This Handbook has, until now, been concerned with the interfaces through which the equipment within a studio area may be interconnected. It has not addressed the expanding use of file transfers and their related interfaces, or how the internet an IP services are handled.

Although the tendency is towards the setting up of largely self-contained studio production areas where possible, the requirements of efficient usage of specialised equipment call for a complex network of interconnections between studio areas and areas remote to a central facility. Also some programmes, e.g. news and sport, require the extensive use of signals originated remotely.

The result is that, at present, every large studio centre relies on an infrastructure consisting of numerous interconnecting cables carrying sound, video, talkback and production control, and a miscellany of ancillary

data signals either embedded in serial interfaces or carried separately. Connection is made through one or more multi-level switching matrices. The expense results partly from the necessary complexity and partly from the stringent technical specifications, Not surprisingly, such systems, once installed, are expected to last for many years. Spare capacity is usually incorporated in an attempt to anticipate increased demand, but a change in the format of the signals will generally present a major problem.

Two different approaches to large systems designs is taking place. One utilizes the Serial digital interface in much the same manner as analogue implementations, the second approach is to use file transfers with both real time capability and non-real time capability. The later technology is still in the development phase along with the use of many competing file formats. This Handbook does not deal with the systems issues or system design of such systems.

10   Interfaces with point-to-point transmission networks

Recommendations ITU-R BT.656 and ITU-R BT.1120 are now well established as a means of access to and from transmission networks. The serial interfaces have become the standard interconnection to almost all television related equipment including telephone company connections. Currently there are extensive investigations being conducted into the use of WAN technology and technology that would lock/synchronize TV stations and their synchronizing systems to GPS.

11   Conclusions

Recommendations ITU-R BT.601 and ITU-R BT.709 are now well recognized image formats along with serial digital interfaces Recommendations ITU-R BT.656 and ITU-R BT.1120, these formats forming the basis for SDTV and HDTV systems designs.

The TV plant of the future is likely to be a mix of serial interfaces for streaming applications and the use of file transfers for production and post-production applications.

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