

Refresher Topics –Television Technology

by Rudolf Mäusl



ROHDE & SCHWARZ

Introduction

Rudolf Mäusl, Professor at the University of applied Sciences Munich, gave a detailed overview of state-of-the-art television technology to the readers of "News from Rohde & Schwarz" in a refresher serial.

The first seven parts of the serial were published between 1977 and 1979 and dealt with fundamentals of image conversion, transmission and reproduction, including a detailed description of the PAL method for colour TV signals. Further chapters on HDTV, MAC and HD-MAC methods, satellite TV signal distribution and PALplus were added in two reprints.

In 1998, these topics were no longer of interest or in a state of change to digital signal transmission. This background has been fully taken into account in the current edition of this brochure which also presents a detailed description of digital video signal processing in the studio, data compression methods, MPEG2 standard and methods for carrier-frequency transmission of MPEG2 multiplex signals to DVB standard.

An even more detailed discussion of the subject matter as well as of state-of-the-art technology and systems is given in the second edition of the book by Rudolf Mäusl "Fernsehtechnik - Übertragungsverfahren für Bild, Ton und Daten" published by Hüthig Buch Verlag, Heidelberg 1995 (only in German).

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1 Transmission method

The principle of TV transmission with a view to reproducing black-and-white pictures can be summarized as follows: the optical image of the scene to be transmitted is divided into small picture elements (pixels).

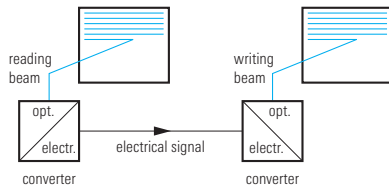


Fig 1
Principle of TV transmission.

An opto-electrical converter, usually a camera tube, consecutively translates the individual elements into electrical information depending on their brightness. This signal is then transmitted at its actual frequency or after modulation onto an RF carrier. After appropriate processing at the receiving end, the information is applied to an electro-optical converter and reproduced in accordance with the brightness distribution of the pattern. Continuous transmission is ensured by producing a defined number of frames as in cinema films.

1.1 Scanning

The pattern is divided into a number of lines which are scanned from left to right and from top to bottom (Fig 1). The scanning beam is deflected horizontally and vertically, writing a line raster. Synchronizing pulses are transmitted to ensure that the reading and the writing beams stay in step, covering the correct, corresponding picture elements.

Scanning converts the individual picture elements from the geometrical into the time domain. Fig 2 gives a simplified representation assuming that the scanning beam returns to the lefthand picture margin within a negligible period of time. In general, the signal current obtained is a train of multishape pulses of varying mean value, corresponding to the mean

brightness of the pattern. This signal current, which may contain components of very high frequency due to fine picture details, must be applied to the receiver without distortion. This requirement determines the essential characteristics of the transmission system.

1.2 Number of lines

The quality of the reproduced picture is determined by the resolution, which is the better the higher the number of lines, a minimum number being required to ensure that the raster is not disturbing to the viewer. In this context, the distance of the viewer from the screen and the acuity of the human eye have to be considered.

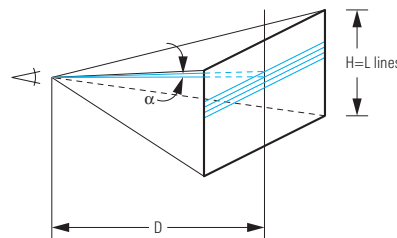


Fig 3
Angle of sight when viewing TV picture.

The optimum viewing distance is found to be about five times the picture height, i.e. $D/H = 5$ (Fig 3). At this distance, the line structure should just be no longer visible, i.e. the limit of the resolving power of the eye should be reached.

Under normal conditions the limit angle is α about $\alpha_0 = 1.5'$. From the equation:

$$\tan \alpha = \frac{H/L}{D} \approx \alpha \quad (1)$$

where $\alpha = \alpha_0 = 1.5'$ and $\tan \alpha_0 = 4 \times 10^{-4}$ the following approximation formula for calculating the minimum line number is obtained:

$$L = \frac{2500}{D/H} \quad (2)$$

For $D/H = 5$, this means a number of $L = 500$ visible lines [1]. In accordance with CCIR, the complete raster area has been divided into 625 lines, 575 of which are in the visible picture area due to the vertical flyback of the beam (525 lines in North America and Japan with about 475 active picture lines).

1.3 Picture repetition frequency

When determining the picture repetition frequency the physiological characteristics of the eye have to be considered. To reproduce a continuous rapid motion, a certain minimum frame frequency is required so that no annoying discontinuities occur. 16 to 18 frames per second, as are used for instance in amateur films, are the lower limit for this frequency. 24 frames per second are used for the cinema. This number could also be adopted for television; however, considering the linkage to the AC supply frequency, a picture repetition frequency (f_r) of 25 Hz for an AC supply of 50 Hz has been selected (30 Hz for a 60 Hz AC supply in North America and Japan).

However, the picture repetition frequency of 25 Hz is not sufficient for flicker-free reproduction of the picture. The same problem had to be solved for the cinema where the projection of each individual picture is interrupted once by a flicker shutter, thus producing the impression that the repetition frequency had been doubled.

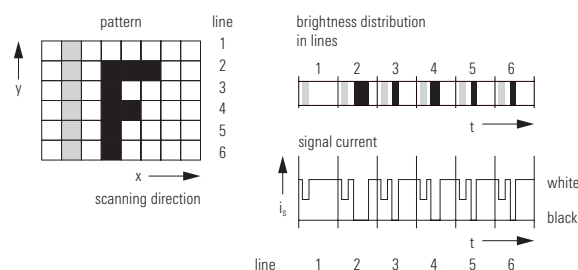


Fig 2
Waveform of signal current in case of line-by-line scanning of pattern.

This method cannot be used for television; here the solution found has been interlaced scanning. The lines of the complete raster are divided into two fields, which are interlaced and transmitted consecutively. Each field contains $L/2$ lines and is swept within the interval $T_V/2$. This means that lines 1, 3, 5 etc are in the first field and lines 2, 4, 6 etc in the second field (geometrical line counting) (Fig 4).

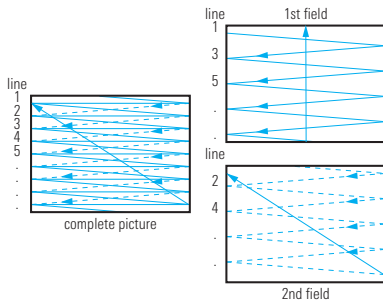


Fig 4 Division of complete raster for interlaced scanning.

When reproducing the two fields it is essential that they be accurately interlaced since otherwise pairing of lines may cause the field raster to appear in a very annoying way. In a system using an odd line number, for instance 625, the transition from the first to the second field takes place after the first half of the last line in the first field. Thus no special auxiliary signal is required to ensure periodic offset of the two fields. This detail will be discussed in section 2.2.

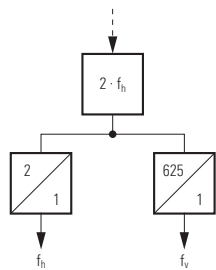


Fig 5 Coupling of horizontal and vertical deflection frequencies in case of interlaced scanning according to CCIR.

Thus 50 fields of $312\frac{1}{2}$ lines each are transmitted instead of 25 pictures of 625 lines, the field repetition or vertical frequency being $f_v = 50$ Hz.

The resulting line or horizontal frequency is

$$f_h = 25 \times 625 = 50 \times 312\frac{1}{2} = 15\,625 \text{ Hz.}$$

The period of the horizontal deflection is $T_h = 64 \mu\text{s}$, that of the vertical deflection $T_v = 20$ ms. The horizontal and vertical frequencies must be synchronous and phase-locked. This is ensured by deriving the two frequencies from double the line frequency (Fig 5).

1.4 Bandwidth of picture signal

The resolution of the picture to be transmitted is determined by the number of lines. With the same resolution in the horizontal and vertical directions, the width of the picture element b is equal to the line spacing a (Fig 6).

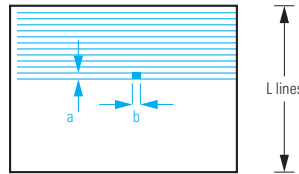


Fig 6 Resolution of pattern by line raster.

At the end of a line, the scanning beam is returned to the left. After sweeping a field, it is returned to the top of the raster.

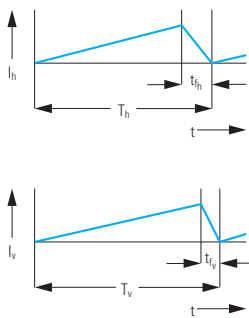


Fig 7 Periods of horizontal and vertical deflection with flyback intervals.

During flyback both the reading and the writing beams are blanked. The required flyback intervals, referred to the period T_h of the horizontal deflection and T_v of the vertical deflection, are given in Fig 7.

In accordance with CCIR, the flyback intervals are defined as follows:

$$T_{fh} = 0.18 \times T_h = 11.52 \mu\text{s}$$

$$t_{fv} = 0.08 \times T_v = 1.6 \text{ ms}$$

Thus, for transmitting the picture information, only the line interval $T_h \times (1 - 0.18) = 52.48 \mu\text{s}$ of the total line period T_h and the portion $L \times (1 - 0.08) = 575$ lines of the L-line raster ($= 2 T_v$) can be used, the raster area available for the visible picture being reduced (Fig 8).

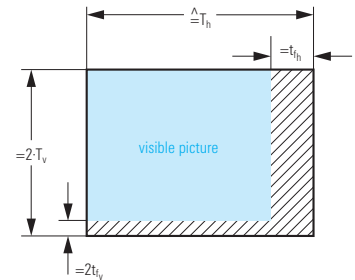


Fig 8 Raster area reduced due to flyback intervals.

For optical and aesthetic reasons a rectangular format with an aspect ratio of 4:3 is chosen for the visible picture.

With the same horizontal and vertical resolution, the number of picture elements per line is:

$$\frac{4}{3} \times 625(1 - 0.08) = 767$$

and the total number of picture elements in the complete picture:

$$\frac{4}{3} \times 625 \times (1 - 0.08) \times 625 \times (1 - 0.08) = 440\,833$$

This number of picture elements is transmitted during the time interval:

$$64 \mu\text{s} \times (1 - 0.18) \times 625 \times (1 - 0.08) = 30.176 \text{ ms}$$

Thus the time T_{PE} available for scanning one element is:

$$T_{PE} = \frac{30.176 \text{ ms}}{440\,833} = 0.0684 \mu\text{s}$$

The highest picture signal frequency is obtained if black and white picture elements alternate (Fig 9). In this case, the period of the picture signal is:

$$T_P = 2 \times T_{PE} = 0.137 \mu s$$

Due to the finite diameter of the scanning beam the white-to-black transition is rounded so that it is sufficient to transmit the fundamental of the squarewave signal. This yields a maximum picture signal frequency of:

$$f_{P_{max}} = \frac{1}{T_P} = 7.3 \text{ MHz}$$

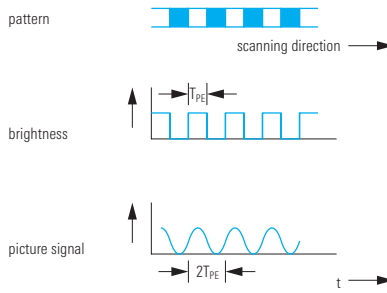


Fig 9
Rounding of picture signal due to finite beam diameter.

Considering the finite beam diameter, the vertical resolution is reduced compared with the above calculation. This is expressed by the Kell factor K. With a value of $K = 2/3$, the bandwidth of the picture or video signal, and the value laid down in the CCIR standards, results as:

$$BW = 5 \text{ MHz.}$$

2 Composite video signal

The composite video signal (CVS) is the complete television signal consisting of the scanned image (SI), blanking (B) and sync (S) components. The scanned image signal was dealt with in section 1.

2.1 Blanking signal

During the horizontal and vertical beam return, the scanned image signal is interrupted, i.e. blanked. The signal is maintained at a defined blanking level which is equal to the black level of the video signal or differs only slightly from it. In most cases the setup interval formerly used to distinguish between blanking level and black level is nowadays omitted for the benefit of making better use of the whole level range. The signal used for blanking consists of horizontal blanking pulses with the width:

$$t_{bh} = 0.18 \times T_h$$

and vertical blanking pulses with the width:

$$t_{bv} = 0.08 \times T_v$$

Thus the signal coming from the video source is completed to form the picture signal (Fig 10).

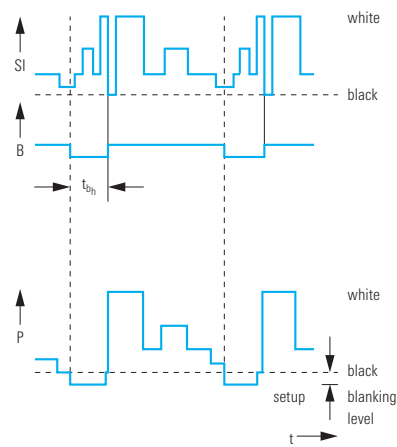


Fig 10
Horizontal blanking signal and generation of picture signal.

2.2 Sync signal

Synchronization pulses are required so that line and field of the picture at the receiver stay in step with the scanning at the transmitter. These sync signals drive the deflection systems at the transmitter and receiver ends. The sync pulse level is lower than the blanking level, thus corresponding to a "blacker-than-black" region (Fig 11).

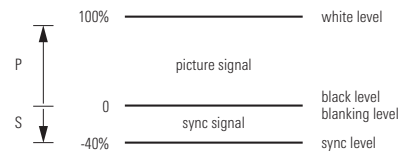


Fig 11
Level range of composite video signal.

In this level range, the horizontal and vertical sync signals must be transmitted in a distinctive way. This is why differing pulse widths are used.

This and the different repetition frequency permit easy separation into horizontal or vertical sync pulses at the receiver end.

The horizontal sync pulse is separated from the sync signal mixture via a differentiating network.

Thus the leading edge of the pulse, whose duration is $4.5 \mu\text{s}$ to $5 \mu\text{s}$, determines the beginning of synchronization, i.e. at the beam return. The front porch ensures that the beam returns to the lefthand picture margin within the blanking interval t_{bh} (Fig 12). The back porch is the reference level. But it is also used for transmitting additional signals, such as the colour synchronization signal.

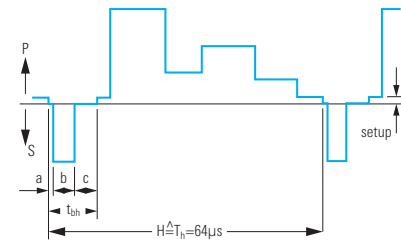


Fig 12
Horizontal sync signal.

The vertical sync pulse is transmitted during the field blanking interval. Its duration of $2.5 H$ periods ($2.5 \times 64 \mu\text{s}$) is considerably longer than that of the horizontal sync pulse (about $0.07 H$ periods). To obtain regular repetition of the horizontal

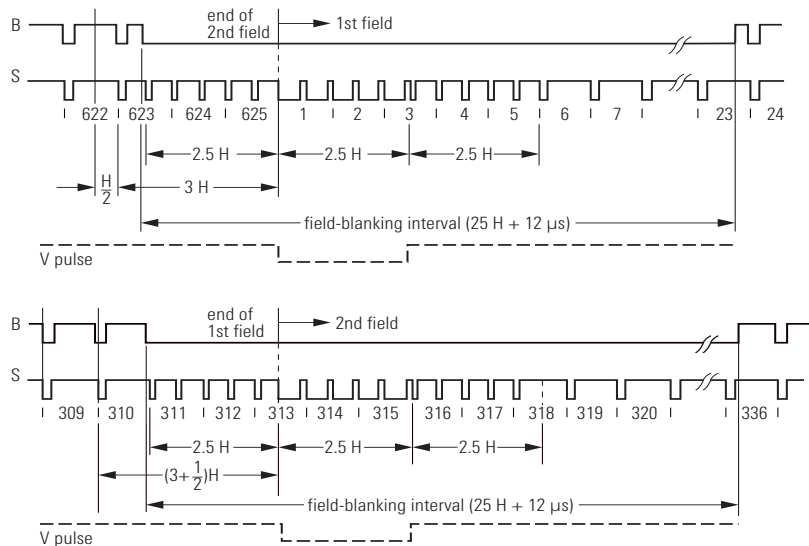


Fig 13
Vertical sync signal with pre- and postequalizing pulses.

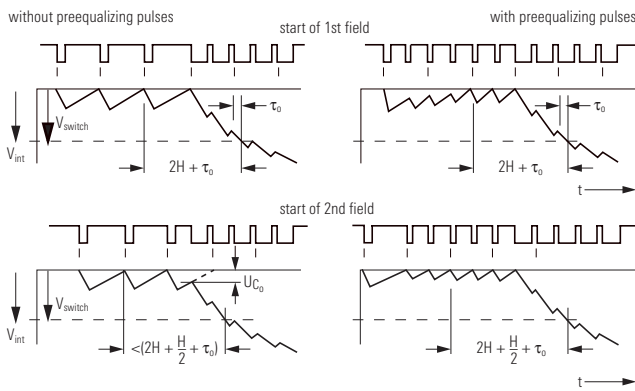


Fig 14

Effect of preequalizing pulses:

left: integration of vertical sync pulse without preequalizing pulses;

right: integration of vertical sync pulse with preequalizing pulses.

sync pulse, the vertical sync pulse is briefly interrupted at intervals of $H/2$. At the points marked in Fig 13, the pulse edges required for horizontal synchronization are produced. Due to the half-line offset of the two rasters, the interruption takes place at intervals of $H/2$. Interlaced scanning also causes the vertical sync pulse to be shifted by $H/2$ relative to the

horizontal sync pulse from one field to the next.

Since the vertical sync pulse is obtained by integration from the sync signal mixture, different conditions for starting the integration (Fig 14, left) would result for

the two fields due to the half-line offset. This in turn might cause pairing of the raster lines. Therefore five narrow equalizing pulses (preequalizing pulses) are added to advance, at $H/2$, the actual vertical sync pulse so that the same initial conditions exist in each field (Fig 14, right). In a similar way, five postequalizing pulses ensure a uniform trailing edge of the integrated vertical part pulses.

The following explanation of the line numbering of Fig 13 is necessary. In television engineering, the sequentially transmitted lines are numbered consecutively. The first field starts with the leading edge of the vertical sync pulse and contains $312\frac{1}{2}$ lines. The first $22\frac{1}{2}$ lines are included in the field blanking interval. After $312\frac{1}{2}$ lines the second field begins in the middle of line 313, also with the leading edge of the vertical sync pulse, and it ends with line 625.

After the complete sync signal with the correct level has been added to the picture signal in a signal mixer, the composite video signal (CVS) is obtained.

3 RF transmission of vision and sound signals

For radio transmission of the television signal and for some special applications, an RF carrier is modulated with the composite video signal. For TV broadcasting and systems including conventional TV receivers, amplitude modulation is used, whereas frequency modulation is employed for TV transmission via microwave links because of the higher transmission quality.

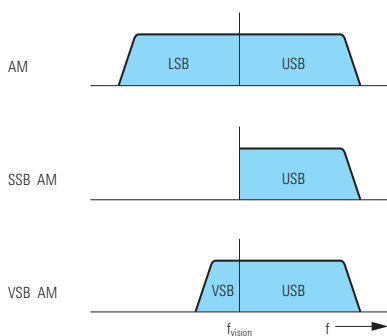


Fig 15
RF transmission of CVS by modulation of vision carrier:
top: amplitude modulation, carrier with two sidebands;
center: single sideband amplitude modulation;
bottom: vestigial sideband amplitude modulation.

3.1 Vestigial sideband amplitude modulation

The advantage of amplitude modulation is the narrower bandwidth of the modulation product. With conventional AM the modulating CVS of $BW = 5 \text{ MHz}$ requires an RF transmission bandwidth of $BW_{RF} = 10 \text{ MHz}$ (Fig 15, top). In principle, one sideband could be suppressed since the two sidebands have the same signal content. This would lead to single sideband amplitude modulation (SSB AM) (Fig 15, center).

Due to the fact that the modulation signals reach very low frequencies, sharp cutoff filters are required; however, the group-delay distortion introduced by these filters at the limits of the passband causes certain difficulties.

The problem is eluded by using vestigial sideband amplitude modulation (VSB AM) instead of SSB AM. In this case, one complete sideband and part of the other are transmitted (Fig 15, bottom).

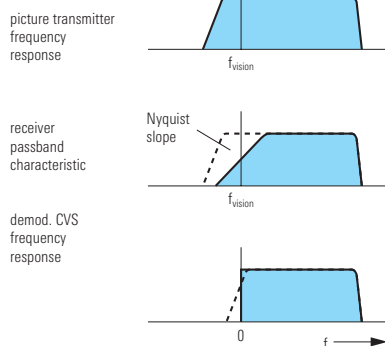


Fig 16
Correction of frequency response in vestigial sideband transmission by Nyquist filter.

At the receiver end it is necessary to ensure that the signal frequencies in the region of the vestigial sideband do not appear with double amplitude after demodulation. This is obtained by the Nyquist slope, the selectivity curve of the receiver rising or falling linearly about the vision carrier frequency (Fig 16).

In accordance with CCIR, 7 MHz bands are available in the VHF range and 8 MHz bands in the UHF range for TV broadcasting. The picture transmitter frequency response and the receiver passband characteristic are also determined by CCIR standards (Fig 17). In most cases, both modulation and demodulation take place at the IF, the vision IF being 38.9 MHz and the sound IF 33.4 MHz.

The modulation of the RF carrier by the CVS is in the form of negative AM, where bright picture points correspond to a low

carrier amplitude and the sync pulse to maximum carrier amplitude (Fig 18).

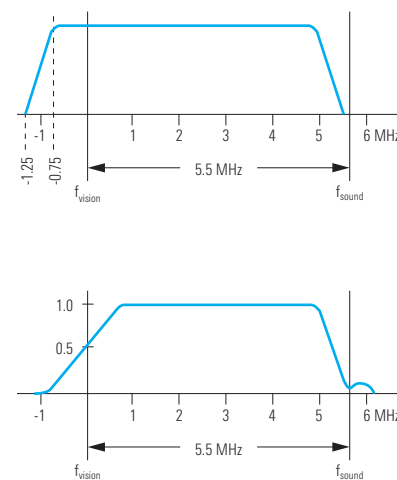


Fig 17
CCIR standard curves for picture transmitter frequency response (top) and receiver passband characteristic (bottom).

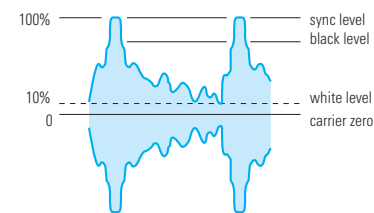


Fig 18
Negative amplitude modulation of RF vision carrier by CVS.

A residual carrier (white level) of 10% is required because of the intercarrier sound method used in the receiver. One advantage of negative modulation is optimum utilization of the transmitter, since maximum power is necessary only briefly for the duration of the sync peaks and at the maximum amplitude occurring periodically during the sync pulses to serve as a reference for automatic gain control in the receiver.

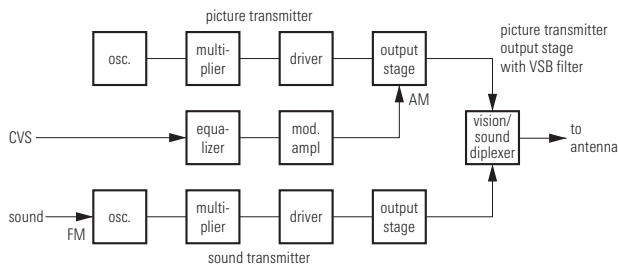


Fig 19
Block diagram of TV transmitter using output stage modulation in picture transmitter.

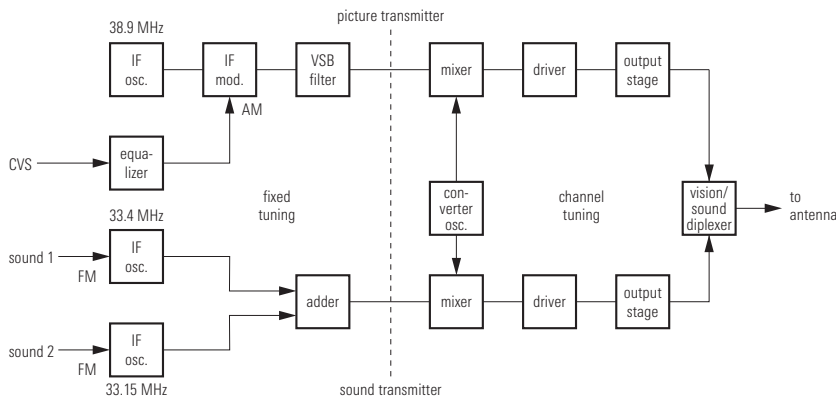


Fig 20
Block diagram of TV transmitter using IF modulation in picture and sound transmitters.

3.2 Sound signal transmission

In TV broadcasting the sound signal is transmitted by frequency-modulating the RF sound carrier. In accordance with the relevant CCIR standard, the sound carrier is 5.5 MHz above the associated vision carrier.

The maximum frequency deviation is 50 kHz. Due to certain disturbances in colour transmission, the original sound/vision carrier power ratio of 1:5 was reduced to 1:10 or 1:20 [2]. Even in the latter case no deterioration of the sound quality was apparent if the signal was sufficient for a satisfactory picture.

As mentioned above, the intercarrier sound method is used in most TV receivers. The difference frequency of 5.5 MHz is obtained from the sound and vision carrier frequencies. This signal is frequency-

modulated with the sound information. The frequency of the intercarrier sound is constant and is not influenced by tuning errors or variations of the local oscillator.

More recent studies have shown further possibilities of TV sound transmission, in particular as to transmitting several sound signals at the same time. A second sound channel permits, for instance, multilingual transmission or stereo operation.

With the dual-sound carrier method, an additional sound carrier 250 kHz above the actual sound carrier is frequency-modulated, its power level being 6 dB lower than that of the first sound carrier. A multiplex method offers further possibilities by modulating an auxiliary carrier at twice the line frequency or using the horizontal or vertical blanking intervals for pulse code modulation.

3.3 TV transmitter and monochrome receiver

The RF television signal can be produced by two different methods.

If the modulation takes place in the output stage of the picture transmitter (Fig 19), the RF vision carrier is first brought to the required driving power and then, with simultaneous amplitude modulation, amplified in the output stage to the nominal vision carrier output power of the transmitter. The modulation amplifier boosts the wideband CVS to the level required for amplitude modulation in the output stage. The sound carrier is frequency-modulated with a small deviation at a relatively low frequency. The final frequency and the actual frequency deviation are produced via multiplier stages. The picture and sound transmitter output stages are fed to the common antenna via the vision/sound diplexer.

When using IF modulation (Fig 20), first the IF vision carrier of 38.9 MHz is amplitude-modulated. The subsequent filter produces vestigial sideband AM. One or two sound carriers are also frequency-modulated at the IF. Next, mixing with a common carrier takes place both in the vision and in the sound channel so that the vision/sound carrier spacing of 5.5 MHz is maintained at the RF. Linear amplifier stages boost the vision and sound carrier powers to the required level.

The advantage of the second method is that the actual processing of the RF television signal is carried out at the IF, thus at a lower frequency, and band- and channel-independent. However, for further amplification, stages of high linearity are required, at least in the picture transmitter.

Table 1 Frequency ranges, vision/sound carrier spacing, channel width, sound modulation

Standard	CCIR	OIRT	French VHF	FCC (USA)
	B, G	D	E	M
VHF, band I / MHz	47 to 68	48.5 to 100	50 to 70	54 to 88
VHF, band III / MHz	174 to 230	174 to 230	160 to 215	174 to 216
UHF, band IV/V / MHz	470 to 853			470 to 890
Vision/sound carrier spacing	5.5 MHz	6.5 MHz	11.15 MHz	4.5 MHz
Channel width	7 MHz (B) 8 MHz (G)	8 MHz	13.15 MHz	6 MHz
Sound modulation, FM deviation	FM, 50 kHz	FM, 50 kHz	AM	FM, 25 kHz

Table 2 Composite video signal

Standard	CCIR	OIRT	French VHF	FCC (USA)
	B, G	D	E	M
Number of lines	625	625	819	525
Field-repetition frequency	50 Hz	50 Hz	50 Hz	60 Hz
Line frequency	15 625 Hz	15 625 Hz	20 475 Hz	15 750 Hz
Video bandwidth	5 MHz	6 MHz	10.6 MHz	4.2 MHz
Line duration H	64 μ s	64 μ s	48.84 μ s	63.5 μ s
Field duration	20 ms	20 ms	20 ms	16.667 ms

Reproduction of the image on the receiver screen is based on proper amplification of the RF signal arriving at the antenna (Fig 21). To this effect, the incoming signal is converted into the IF in the VHF/UHF tuner, where also standard selection by the Nyquist filter and the required amplification are provided. The subsequent demodulator generates the CVS and the 5.5 MHz sound IF. The latter is limited in amplitude, amplified and frequency-demodulated. The CVS is applied to the video amplifier; after separation of the sync component, the signal is taken to the control section of the CRT via the video output stage. The sync signal brought out from the video amplifier by way of a sync separator is fed to the horizontal deflection system via a differentiating network and to the vertical deflection system via an integrating network.

The line deflection frequency is produced in the horizontal oscillator and compared to the incoming horizontal sync pulses in a phase discriminator. A control circuit ensures that the correct frequency and phase relation to the transmitter sync signal is maintained. In the horizontal output stage, the required deflection power is produced and the high voltage for the CRT is obtained from the line retrace pulses. The vertical oscillator is synchronized directly by the vertical sync signal. The blanking pulses required for the beam retrace are derived from the horizontal and vertical output stages.

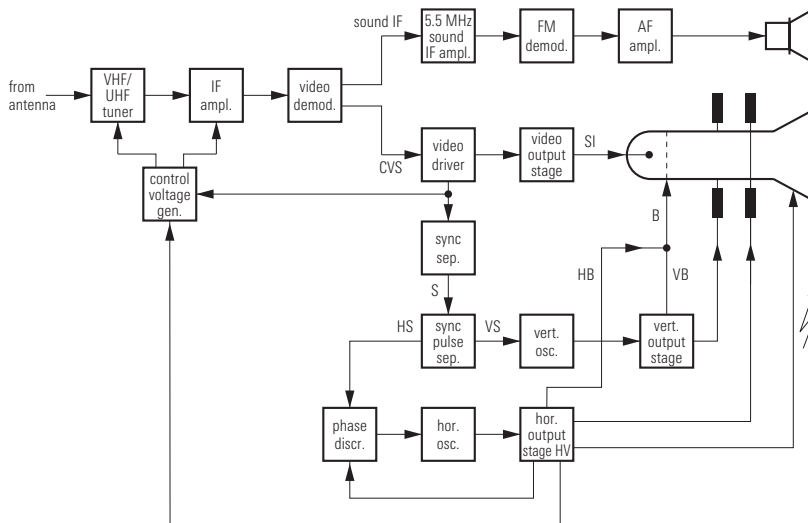


Fig 21 Block diagram of monochrome TV receiver.

3.4 TV standards

The characteristics of the television signals mentioned in sections 1, 2 and 3 refer to the CCIR standard. Various other standards are in use; for the differences in the standard specifications see Tables 1 and 2.

4 Adding the colour information

To reproduce a colour image of the pattern, additional information on the colour content, i.e. the "chromaticity" of the individual picture elements, must be transmitted together with the brightness or luminance distribution. This requires first the extraction of the colour information and then a possibility of reproducing the colour image.

4.1 Problem

The problem of colour transmission consists in maintaining the transmission method of black-and-white television and broadcasting the additional colour information as far as possible within the available frequency band of the CVS. This means for any colour TV system that a colour broadcast is reproduced as a perfect black-and-white picture on a monochrome receiver (compatibility) and that a colour receiver can pick up a monochrome broadcast to reproduce a perfect black-and-white picture (reverse compatibility).

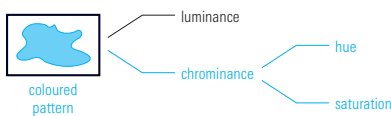


Fig 22 Representation of coloured pattern by luminance and chrominance components.

These requirements can be met only if

- information on the luminance distribution and
- information on the colour content

are obtained from the coloured pattern and then transmitted.

The chromaticity is characterized by the hue – determined by the dominant wavelength of light, for instance for distinct colours such as blue, green, yellow, red – and by the saturation as a measure of spectral purity, i.e. of colour intensity with respect to the colourless (white)

(Fig 22). The chrominance signal cannot be obtained directly from the pattern. Instead the three primaries (red, green, blue) are used in accordance with the three-colour theory (Helmholtz). The red, green and blue signals are also required for reproducing the colour picture. Thus the scheme of compatible colour transmission is established by the luminance signal Y and the chrominance signal F (Fig 23).

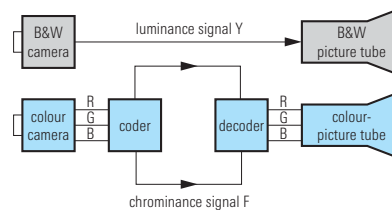


Fig 23 Principle of compatible colour transmission.

4.2 Chromatics and colorimetry

Light is that part of the electromagnetic radiation which is perceived by the human eye. It covers the wavelengths from about 400 nm (violet) to 700 nm (red). The light emitted by the sun consists of a multitude of spectral colours merging into each other. Spectral colours are saturated colours. Mixing with white light produces desaturated colours.

Coloured (chromatic) light can be characterized by its spectral energy distribution. The radiation of the wavelength λ causes the sensations of brightness and colour in the eye. The sensitivity to brightness of the human eye as a function of the wavelength is expressed by the sensitivity characteristic or luminosity curve (Fig 24).

This characteristic indicates how bright the individual spectral colours appear to the eye when all of them have the same energy level. It can be seen from this characteristic that certain colours appear dark (e.g. blue) and others bright (e.g. green).

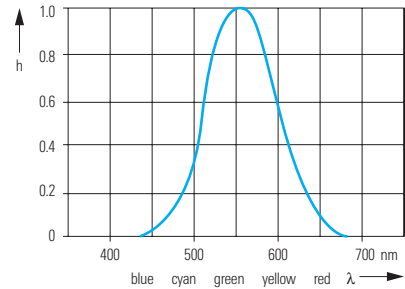


Fig 24 Brightness sensitivity characteristic of human eye.

In monochrome television, where only the luminance distribution of a coloured pattern is transmitted, this sensitivity characteristic of the eye has to be taken into account. This is done by using the spectral sensitivity of the camera tube and, if required, correction filters in connection with the colour temperature of the lighting.

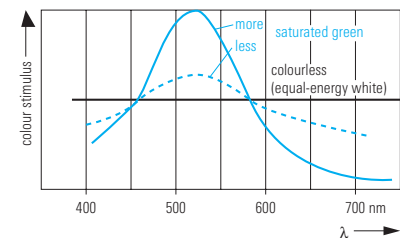


Fig 25 Colour stimulus at different degrees of saturation.

The colours of objects are those colours that are reflected from the light to which the object is exposed. The colour stimulus curve shows the associated spectral distribution (Fig 25). In most cases, the object colours are not spectral colours but rather mixtures consisting of a number of closely spaced spectral colours or of several groups of spectral colours.

This is an additive process. White (colourless) can also be produced by mixing. Fig 26 shows typical examples of additive colour mixing.

Investigations into the colour stimulus sensitivity of the human eye have shown that a colour sensation is produced by mixing the part sensations caused in the primaries red, green and blue. This leads to the conclusion that any colour appearing in nature can be obtained by combining the corresponding portions of the primaries red, green and blue. In accordance with the Helmholtz three-colour theory, Grassmann (1854) found the following law:

$$F = R(R) + G(G) + B(B) \quad (3)$$

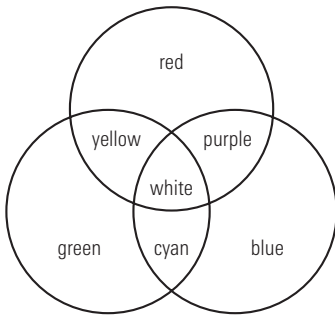


Fig 26 Additive colour mixing using three primaries R, G, B.

This means that a distinct colour stimulus F can be matched by R units of the spectral colour red (R), G units of the spectral colour green (G) and B units of the spectral colour blue (B).

Monochromatic radiations of the wavelengths

$$\lambda_R = 700 \text{ nm}, \lambda_G = 546.1 \text{ nm}, \text{ and } \lambda_B = 435.8 \text{ nm}$$

have been determined as the standard spectral colours, called primary colours or stimuli. None of the three primaries must be obtainable from the other two by mixing.

Based on the equation (3), colour mixture curves were plotted, showing the portion of each primary stimulus required for the different spectral colours (Fig 27). The ordinate scale refers to equal-energy white. As can be seen from the curves, negative amounts or tristimulus values are associated with some components. This means that for matching certain spectral colours a specific amount of a

primary stimulus must be added to the colour stimulus.

In a colorimetric representation of a colour stimulus, the three primaries yield a space vector. The direction of the colour vector in space determines the chromaticity, its length being a measure of the brightness.

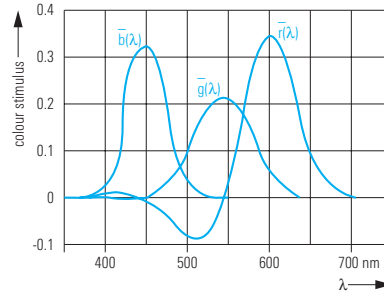


Fig 27 Colour mixture curves $\bar{b}(\lambda)$, $\bar{g}(\lambda)$, $\bar{r}(\lambda)$, referred to R, G, B.

However, a three-dimensional coordinate system is not convenient for graphic representation. But since brightness and chromaticity are independent of each other, the tristimulus values can be standardized to the luminance component:

$$\frac{F}{R + G + B} = \gamma$$

$$\gamma = \frac{R(R)}{R + G + B} + \frac{G(G)}{R + G + B} + \frac{B(B)}{R + G + B}$$

$$= 1 \quad (4)$$

or, using the chromaticity coordinates:

$$r + g + b = 1. \quad (5)$$

These reduced values no longer contain any luminosity information but merely the chromaticity.

Since, however, the sum of r, g and b is always unity, one of the three coordinates can be omitted when specifying the chromaticity so that a two-dimensional system, the colour surface, is obtained. When entering the chromaticity coordinates found from the colour-mixture

curves in an r-g diagram, the locus of all spectral colours is plotted (Fig 28).

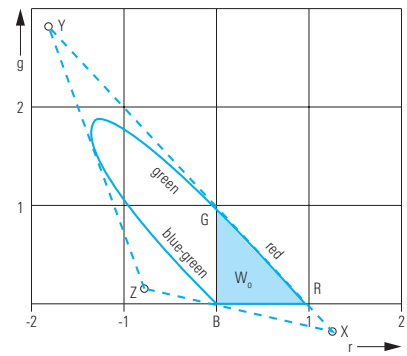


Fig 28 Colour surface in r-g diagram.

Due to the negative portion of the $\bar{r}(\lambda)$ colour mixture curve, here again negative values are obtained. Coordinate transformation referring to a new set of fictive, i.e. non-physical primaries X, Y and Z, yields a curve which comprises only positive colour values [3]. When using the fictive primaries (standard reference stimuli X, Y, Z), the relationship according to the equation (4) still holds, expressed by the standard reference summands x, y, z:

$$x + y + z = 1. \quad (6)$$

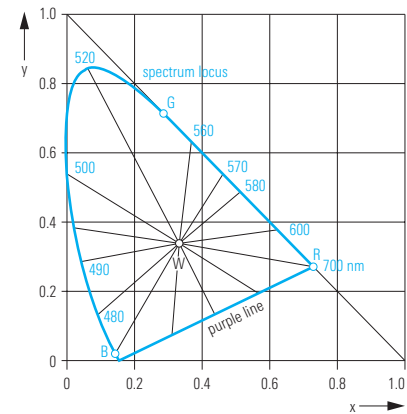


Fig 29 Standard colour diagram; colour surface in x-y diagram.

The two-dimensional representation of chromaticity in the x-y coordinate system is called the standard colour diagram in accordance with CIE (Commission Internationale de l'Eclairage) or, briefly colour

triangle (Fig 29). The area of colour stimuli that can be realized by additive mixing is enclosed by the spectrum locus and the purple line. The line connecting the white point W (equal-energy white) of $x = 0.33$ and $y = 0.33$ to the position of any colour F yields the dominant wavelength, i.e. the hue, when extended to its point of intersection with the spectrum locus.

The ratio of the distance between the hue F and the white point W to the distance between the spectrum locus and the white point W on the connecting line passing through the colour position gives the colour saturation. The closer the colour point to white, the weaker the colour saturation. The position of a mixture colour is situated on the line joining the loci of the two colours mixed or, if three colours are added, within the triangle formed by the connecting lines.

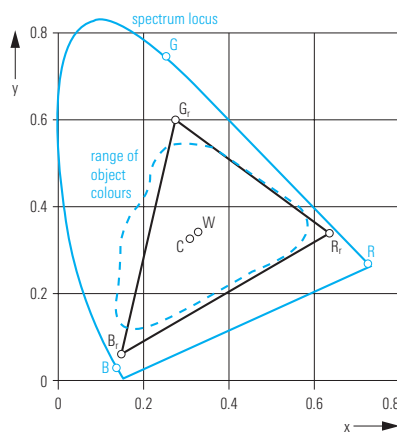


Fig 30 Colour coordinates of receiver primaries and representable colour range.

When defining the tristimulus values in a colour TV system, it is essential to consider the question of how the primary stimuli can be realized at the receiver end. On the one hand, the requirements regarding the receiver primaries are determined by the necessity of providing as wide a range of representable mixture colours as possible, i.e. the chromaticity coordinates of the receiver primaries should be located on the spectrum locus. On the other hand, primaries of especially

high luminance—which can be produced economically—are required. In accordance with recent EBU (European Broadcasting Union) specifications, the receiver primaries R_r , G_r and B_r are used; their colour coordinates are given in Fig 30.

The colour mixture curves plotted with the aid of the primary stimuli R, G and B are based on equal-energy white W. In colour TV technology, standard illuminant C is used as reference white, corresponding to medium daylight with a colour temperature of about 6500 K, the standard tristimulus values being:

$$x_C = 0.310, y_C = 0.316, z_C = 0.374.$$

If now the tristimulus values of the receiver primaries R_r , G_r and B_r are found for all spectral colour stimuli of equal radiation energy and plotted as standardized tristimulus values as a function of the wavelength λ (maximum of the curve referred to 1), the colour mixture curves used in TV engineering are obtained (Fig 31).

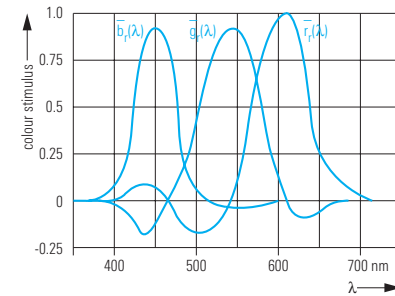


Fig 31 Colour mixture curves $\bar{b}_r(\lambda)$, $\bar{g}_r(\lambda)$, $\bar{r}_r(\lambda)$, referred to receiver primaries R_r , G_r , B_r .

Here again negative colour values appear due to the colour stimuli located outside of the triangle formed by R_r , G_r and B_r .

Therefore, the colour mixture curves are slightly modified for practical purposes (dashed line). The signals produced by the camera tubes in the red, green and blue channels of the colour camera must be matched with these colour mixture

curves using their spectral sensitivity and additional colour filters (see Fig 23). The output voltage of the colour camera in the three channels must have the same relationship as the tristimulus values. With standard illuminant C, the three output signals must be equal, even at different luminance values.

4.3 Luminance and chrominance signals, colour difference signals

For reasons of compatibility, the colour camera has to deliver the same signal from a coloured pattern, i.e. the luminance signal to the monochrome receiver, as the black-and-white camera. The spectral sensitivity of a black-and-white camera corresponds to the brightness sensitivity curve of the human eye to ensure that the black-and-white picture tube reproduces the different colour stimuli as grey levels of the same brightness as perceived by the eye.

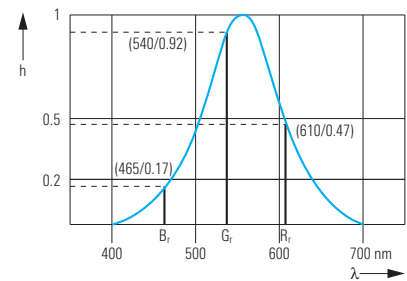


Fig 32 Brightness sensitivity of human eye to receiver primaries.

The colour camera, however, delivers three signals with spectral functions matching the colour mixture curves. To obtain one signal whose signal spectral function corresponds to the sensitivity curve of the eye, coding is required. To this end, the three colour values represented by the functions $\bar{r}_r(\lambda)$, $\bar{g}_r(\lambda)$ and $\bar{b}_r(\lambda)$ are multiplied by the relative luminosity coefficients h_r , h_g and h_b and then added up. Except for the proportionality constant k , the result must be identical to the sensitivity function of the human eye $h(\lambda)$.

$$h(\lambda) = k[h_r \times \bar{r}_r(\lambda) + h_g \times \bar{g}_r(\lambda) + h_b \times \bar{b}_r(\lambda)]. \quad (7)$$

The relative luminosity coefficients h_r , h_g and h_b are obtained by normalization from the corresponding values $h_{(R_r)}$, $h_{(G_r)}$ and $h_{(B_r)}$ of the eye (Fig 32):

$$\begin{aligned} h_{(R_r)} &= 0.47 \\ h_{(G_r)} &= 0.92 \\ h_{(B_r)} &= 0.17 \end{aligned}$$

$$\Sigma h = 1.56$$

The following figures are calculated for the relative luminosity coefficients:

$$h_r = \frac{h_{(R_r)}}{\Sigma h} = \frac{0.47}{1.56} = 0.30$$

$$h_g = \frac{h_{(G_r)}}{\Sigma h} = \frac{0.92}{1.56} = 0.59$$

$$h_b = \frac{h_{(B_r)}}{\Sigma h} = \frac{0.17}{1.56} = 0.11$$

Due to normalization:

$$h_r + h_g + h_b = 1. \quad (8)$$

Thus for equation (7)

$$h(\lambda) = k [0.30 \times \bar{r}_r(\lambda) + 0.59 \times \bar{g}_r(\lambda) + 0.11 \times \bar{b}_r(\lambda)]. \quad (7')$$

is obtained or, written in a simplified way, for the luminance signal Y corresponding to the sensitivity curve of the eye

$$Y = 0.30 \times R + 0.59 \times G + 0.11 \times B. \quad (9)$$

This equation is one of the most important relations of colour TV engineering.

Technically the luminance signal V_Y is obtained from the tristimulus signals V_R , V_G and V_B via a matrix (Fig 33). The signals listed in Table 3 below result for a pattern consisting of eight colour bars (standard colour bar sequence) – the three primaries plus the associated complementary colours and the colourless stripes white and black.

Table 3 Signals of standard colour bar sequence

Pattern	R	G	B	Y
White	1	1	1	1.00
Yellow (R + G)	1	1	0	0.89
Cyan (G + B)	0	1	1	0.70
Green	0	1	0	0.59
Purple (R + B)	1	0	1	0.41
Red	1	0	0	0.30
Blue	0	0	1	0.11
Black	0	0	0	0

To reproduce a colour pattern, the three tristimulus signals R, G and B are required. For compatibility, however, colour transmission uses the luminance signal Y and the chrominance signal F. The latter cannot be obtained directly from the tristimulus values but only by way of the colour difference signals:

$$R - Y, \quad G - Y, \quad B - Y.$$

These are the colour values minus the luminance component.

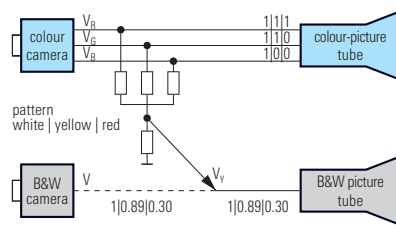


Fig 33 Producing luminance signal V_Y from tristimulus voltages V_R , V_G , V_B and compatibility relationship.

The chrominance signal carries information on hue and saturation. Therefore two colour difference signals are sufficient to describe the chrominance component. For this purpose, the quantities $R - Y$ and $B - Y$ were selected [4].

referring to the voltage of the luminance signal derived from the coder yields these two colour difference signals as:

The equation

$$V_Y = 0.30 \times V_R + 0.59 \times V_G + 0.11 \times V_B \quad (10)$$

$$V_R - V_Y = 0.70 \times V_R - 0.59 \times V_G - 0.11 \times V_B \quad (11)$$

and

$$V_B - V_Y = -0.30 \times V_R - 0.59 \times V_G + 0.89 \times V_B \quad (12)$$

The colour difference signals only carry information on the colour content. For a monochrome scene ($V_R = B_G = V_B$) they are equal to zero.

The amplitude of the colour difference signals shows the departure of the hue from the colourless, which is a measure of colour saturation.

The hue is determined by the amplitude ratio and by the polarity sign of the colour difference signals. Transformation of the orthogonal coordinate system ($B - Y$ and $R - Y$) into polar coordinates (Fig 34) yields the colour saturation from the vector length A:

$$A = \sqrt{(B - Y)^2 + (R - Y)^2} \quad (13)$$

and the hue from the angle α :

$$\alpha = \arctan \left(\frac{R - Y}{B - Y} \right) \quad (14)$$

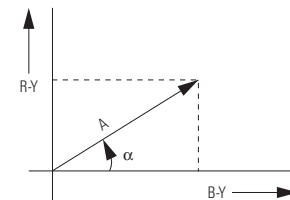


Fig 34 Representation of chromaticity as a function of colour difference signal.

Investigations have shown that the resolution of the eye is lower for coloured pattern details than for brightness variations. Therefore it is sufficient to transmit only the luminance signal with the full bandwidth of 5 MHz. The bandwidth of the chrominance signal can be reduced to about 1.5 MHz by taking the two colour difference signals via lowpass filters.

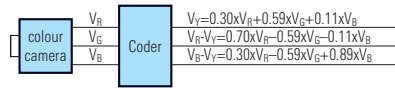


Fig 35
Producing luminance signal V_Y plus colour difference signals $V_R - V_Y$ and $V_B - V_Y$.

In the coder, the signals V_R , V_G and V_B produced by the colour camera are converted into the luminance component V_Y and the colour difference signals $V_R - V_Y$ and $V_B - V_Y$ (Fig 35) and, in this form, applied to the reproducing system. However, the tristimulus values are required for unbalancing the red, green and blue beams. Two different methods are commonly used for restoring these colour values:

1. Driving colour picture tube with RGB voltages (Fig 36)

The tristimulus voltages V_R , V_G and V_B are produced from the luminance component V_Y and the two colour difference signals $V_R - V_Y$ and $V_B - V_Y$ via matrices and applied directly to the control grids of the colour picture tube, the cathodes being at fixed potential.

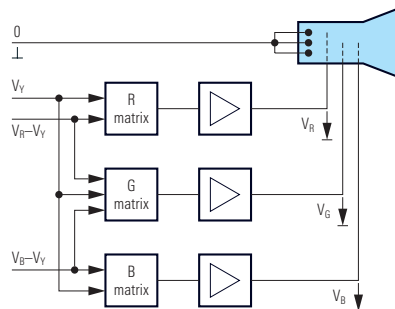


Fig 36
Restoring tristimulus values when driving colour picture tube with RGB.

2. Driving colour picture tube with colour difference signals (Fig 37)

The third colour difference signal $V_G - V_Y$ is obtained in a matrix from the two quantities $V_R - V_Y$ and $V_B - V_Y$, based on the following equations

$$\begin{aligned} V_Y &= 0.30 \times V_R + 0.59 \times V_G + 0.11 \times V_B \\ V_Y &= 0.30 \times V_Y + 0.59 \times V_Y + 0.11 \times V_Y \\ V_Y - V_Y &= 0.30 \times (V_R - V_Y) \\ &\quad + 0.59 \times (V_G - V_Y) \\ &\quad + 0.11 \times (V_B - V_Y) = 0 \end{aligned} \quad (15)$$

or, after rewriting,

$$\begin{aligned} V_G - V_Y &= -0.51 \times (V_R - V_Y) \\ &\quad - 0.19 \times (V_B - V_Y) \end{aligned} \quad (16).$$

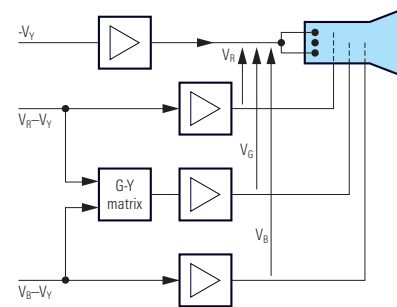


Fig 37
Restoring tristimulus values when driving colour picture tube with colour difference signals.

The colour difference signals are taken to the control grids of the deflection systems; the negative luminance signal is applied to the cathodes so that the tristimulus signals are obtained as control voltages at the three systems, for instance:

$$U_{\text{contr R}} = (U_R - U_Y) - (-U_Y) = U_R. \quad (17)$$

The advantage of this method is that the bandwidth of the final amplifier stages for the colour difference signals is smaller than that of the stages for the tristimulus signals and that a black-and-white picture appears if the colour difference signals fail. The disadvantage, is that higher voltages must be produced in the final stages associated with the colour difference signals. Fig 38 shows signals for the standard colour bar pattern.

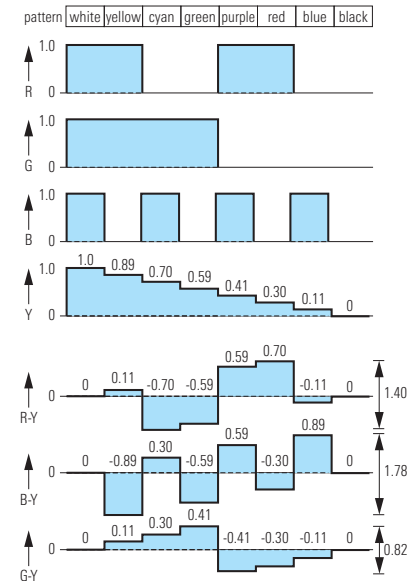


Fig 38
Tristimulus values, luminance component and colour difference signals for standard colour bar pattern.

5 Transmission of chrominance signal with colour subcarrier

As explained in the preceding chapter, the colour TV signal is transmitted in the form of the luminance signal Y and the two colour difference signals R-Y and B-Y for reasons of compatibility. To transmit the complete picture information—luminance plus chrominance—a triple transmission channel would be required. Here one might think of making multiple use of the TV transmission channel either by frequency or time multiplex. However, neither method is compatible with the existing black-and-white transmission method.

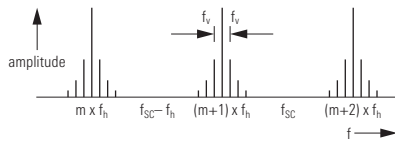


Fig 39
Detail of CVS spectrum.

A decisive thought for the colour transmission method actually selected is derived from spectral analysis of the luminance signal or CVS. It becomes apparent that only certain frequency components occur in the CVS spectrum, these components being mainly multiples of the line frequency due to the periodic scanning procedure.

The varying picture content amplitude-modulates the line-repetitive pulse sequence producing sidebands spaced at multiples of the field frequency from the spectral components of the line pulse. Fig 39 shows a detail of the CVS spectrum. Essentially the spectrum is occupied only at multiples of the line frequency in their vicinity. Between these frequency and groups the spectrum exhibits significant energy gaps.

Since the colour information is also line-repetitive, the spectrum of the chrominance signal consists only of multiples of the line frequency and the corresponding sidebands. Therefore it is appropriate to insert the additional colour information

into the gaps of the CVS frequency spectrum. This is done by modulating the chrominance signal onto a colour subcarrier whose frequency f_{SC} , and thus also the spectrum of the line-repetitive modulation products, is located between the frequency components of the CVS (Fig 40).

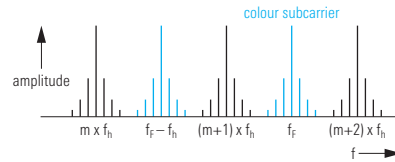


Fig 40
Spectrum of CVS and modulated colour subcarrier.

5.1 Determining the colour subcarrier frequency

One condition for determining the colour subcarrier frequency results from the symmetrical interleaving of the CVS and chrominance signal spectra: the frequency f_{SC} should be an odd multiple of half the line frequency f_h :

$$f_{SC} = (2n + 1) \times \frac{f_h}{2} \quad (18)$$

In this way the half-line offset is obtained.

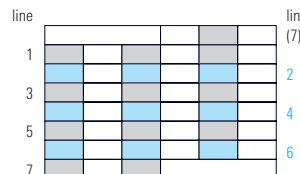
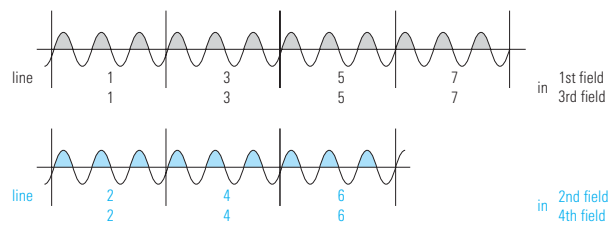
Generally, the interference which a colour subcarrier produces on the black-and-white picture can be explained as follows, if it is assumed that a sinewave in

the frequency region of the CVS produces a bright-dark interference pattern on the screen.

If the colour subcarrier frequency is an integer multiple of the line frequency, i.e. if there is no offset with respect to the line frequency, an interference pattern of bright and dark vertical stripes appears, their number corresponding to the factor n (Fig 41). As a result of the half-line offset the phase of the colour subcarrier alternates by 180° from line to line of a field. However, because of the odd number of lines, bright and dark dots coincide after two fields. The interference pattern occurring in a rhythm of $f_v/4 = 12.5 \text{ Hz}$ would thus be compensated over four fields (Fig 42). Nevertheless, the compensation of the interference pattern on the screen is not perfect due to the nonlinearity of the picture tube characteristic and the inadequate integrating capability of the human eye.

If a half-line offset is assumed between the colour subcarrier frequency and the line frequency, the subjective annoyance can be further reduced by selecting the colour subcarrier frequency as high as possible. In this way, the interference pattern takes on a very fine structure. However, the colour difference signals modulate the colour subcarrier so that for transmitting the upper sideband a certain minimum spacing of the colour subcarrier frequency from the upper frequency limit of the CVS has to be maintained.

Fig 41
Interference pattern caused by colour subcarrier with integer relationship between colour subcarrier frequency and line frequency.



$$f_{SC} = n \times f_h$$

e.g. n=3

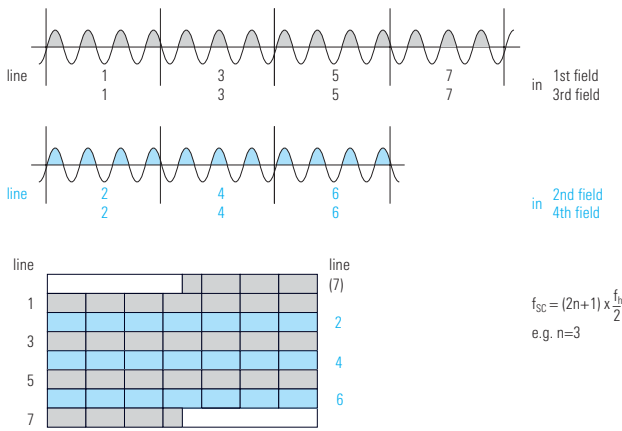


Fig 42
Compensation of interference pattern with half-line offset of colour subcarrier frequency.

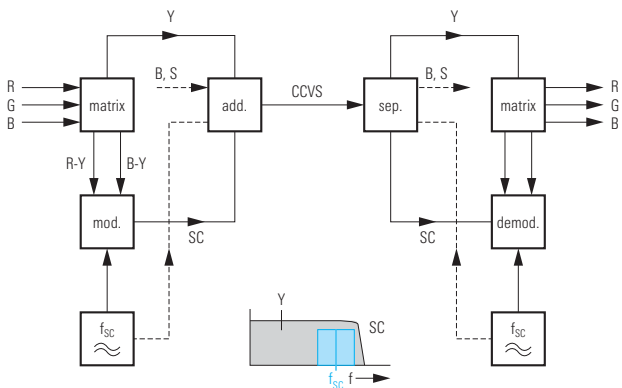


Fig 43
Principle of compatible colour TV transmission using luminance and chrominance signals.

The best compromise has proved to be a colour subcarrier frequency of about 4.4 MHz. Thus the principle of compatible colour TV transmission is found using the luminance signal and the chrominance signal modulated onto the subcarrier; this is the basis of the NTSC system and its variants (Fig 43). For the CCIR-modified NTSC method, which will be discussed in detail later, a colour subcarrier frequency of

$$f_{sc} = 567 \times \frac{f_h}{2} = 283.5 \times f_h = 4.4296875 \text{ MHz} \quad (19)$$

has been fixed.

A further development of the NTSC method is the PAL method, which is being widely used today. In the PAL system, one component of the subcarrier is switched by 180° from line to line. However, this cancels the offset for this subcarrier component so that a pronounced interference pattern would appear in the compatible black-and-white picture. This is avoided by introducing a quarter-line offset of the subcarrier plus an additional offset of $f_v/2 = 25 \text{ Hz}$. Thus the CCIR

standard colour subcarrier frequency in the PAL system is

$$f_{sc} = 283.75 \times f_h + 25 \text{ Hz} = 4.43361875 \text{ MHz} \quad (20)$$

The rigid coupling of the subcarrier frequency to the line frequency is ensured by deriving the line frequency f_h or twice the line frequency $2f_h$ from the frequency f_{sc} of the colour subcarrier (Fig 44).

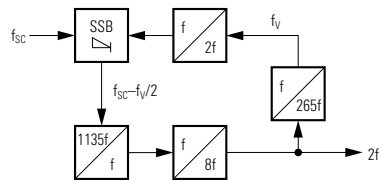


Fig 44
Coupling of colour subcarrier frequency and line frequency.

5.2 Modulation of colour subcarrier

The chrominance signal is transmitted by modulating the colour subcarrier with the two colour difference signals. The modulation method must permit the colour difference signals to be extracted separately at the receiver end.

In the NTSC and PAL methods, double amplitude modulation is used. The 0° component of the subcarrier is amplitude-modulated by the (B-Y) signal and its 90° component by the (R-Y) signal, the subcarrier being suppressed at the same time. This results in quadrature modulation (Fig 45). The modulation product is a subcarrier frequency whose amplitude and phase are determined by the two colour difference signals. Amplitude and phase modulation take place at the same time. Compared to the chrominance signal represented in Fig 34, the colour saturation now corresponds to the amplitude SC and the hue to the phase angle ϕ_{SC} of the modulated colour subcarrier. For this reason, the modulated colour subcarrier is also called the chrominance signal.

$$SC = \sqrt{(B-Y)^2 + (R-Y)^2} \quad (21)$$

$$\phi_{SC} = \arctan \frac{R-Y}{B-Y} \quad (22)$$

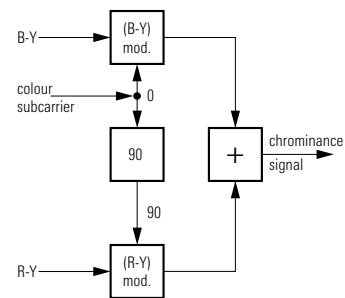


Fig 45
Producing chrominance signal by quadrature modulation of colour subcarrier with two colour difference signals.

A vector diagram of the chrominance signal shows the position of the different colours on the colour circle (Fig 46). Similar to the colour triangle in Fig 29, the complementary colours are located on opposite sides of the coordinate zero (colourless). When transmitting a colourless picture element, the colour difference signals and the amplitude of the colour subcarrier equal zero. In this case, the subcarrier does not cause any interference in the black-and-white picture.

To demodulate the chrominance signal, the unmodulated carrier of correct phase is required.

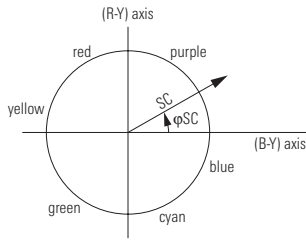


Fig 46
Vector representation of chrominance signal: colour circle.

Synchronous detection is used, evaluating only the chrominance component which is in phase with the reference carrier. Since the actual subcarrier is not transmitted, it must be produced as a reference carrier at the receiver end. For synchronization with the subcarrier at the transmitter end, a reference signal is inserted into each line in the H blanking interval, i.e. the colour sync signal or burst. This signal consists of about ten oscillations of the subcarrier at the transmitter end and is transmitted on the back porch (Fig 47).

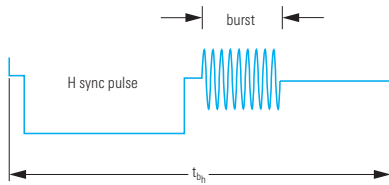


Fig 47
Colour sync signal (burst).

In the NTSC system, the burst phase is at 180° compared to the 0° reference phase of the colour subcarrier.

In the receiver the burst is separated from the chrominance signal by blanking. A phase comparator produces a control voltage from the departure of the reference carrier phase from the burst phase; this voltage controls the frequency and phase of the reference carrier oscillator via an integrating network. The control voltage becomes zero if the phase difference is 90° . Coming from the reference carrier oscillator, the 90° component is taken directly to the (R-Y) synchronous detector and, after a negative 90° phase shift, as the 0° component to the (B-Y) synchronous detector (Fig 48).

5.3 Composite colour video signal

The chrominance signal is combined with the composite video signal (CVS) to form the composite colour video signal (CCVS). The CCVS is amplitude-modulated onto the RF vision carrier. The full level of the colour difference signals would cause overmodulation of the RF vision carrier for certain coloured patterns. This is shown for the signals of the standard colour bar sequence (Table 4).

Overmodulation occurs in both directions (Fig 49). In particular, the periodic suppression of the RF carrier and its falling short of the 10% luminance level would cause heavy interference. For this reason, the chrominance signal amplitude has to be reduced. As a compromise between overmodulation on the one hand and degradation of signal-to-noise ratio on the other—an overmodulation of 33% in both directions with fully saturated colours has been permitted since, in practice, fully saturated colours hardly ever occur. This is ensured by using different

reduction factors for the two colour difference signals, multiplying them by

0.49 for the (B-Y) signal
and 0.88 for the (R-Y) signal

In this way the reduced colour difference signals U and V are obtained:

$$U = (B-Y)_{red} = 0.49 \times (B-Y) \\ = -0.15 \times R - 0.29 \times G + 0.44 \times B \quad (23)$$

$$V = (R-Y)_{red} = 0.88 \times (R-Y) \\ = 0.61 \times R - 0.52 \times G - 0.10 \times B \quad (24)$$

(The values are rounded off.)

The signal values listed in Table 5 are obtained for the colour bar pattern with 100% saturated colours. Fig 50 shows the line oscillogram of the CCVS for a 100% saturated colour bar sequence.

For measurements and adjustments on colour TV transmission systems the test signal used is the standard colour bar sequence for which all chrominance sig-

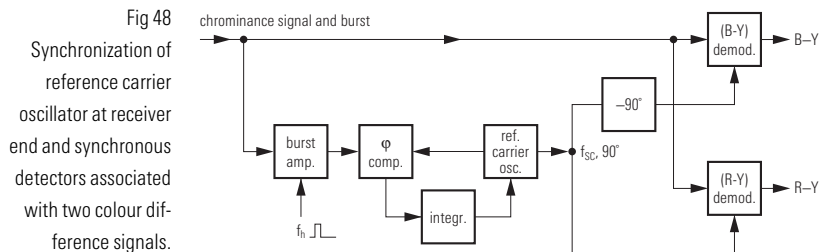


Fig 48
Synchronization of reference carrier oscillator at receiver end and synchronous detectors associated with two colour difference signals.

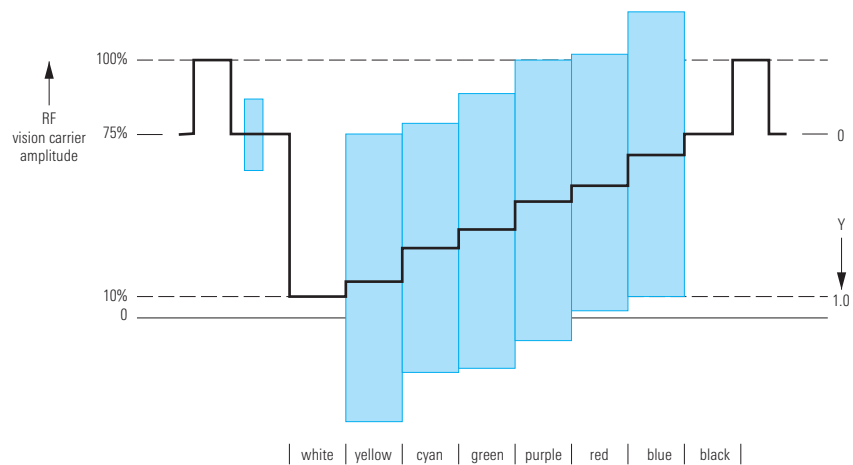


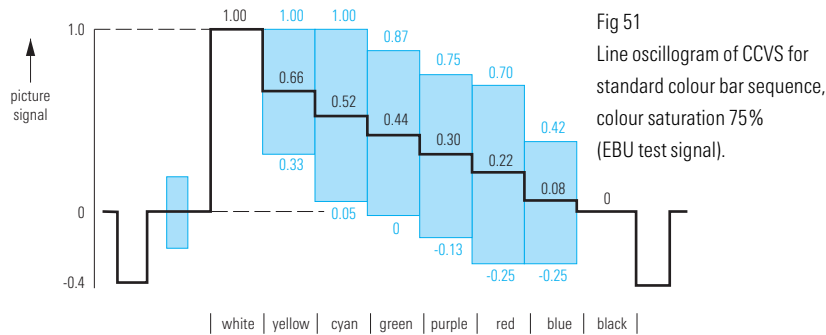
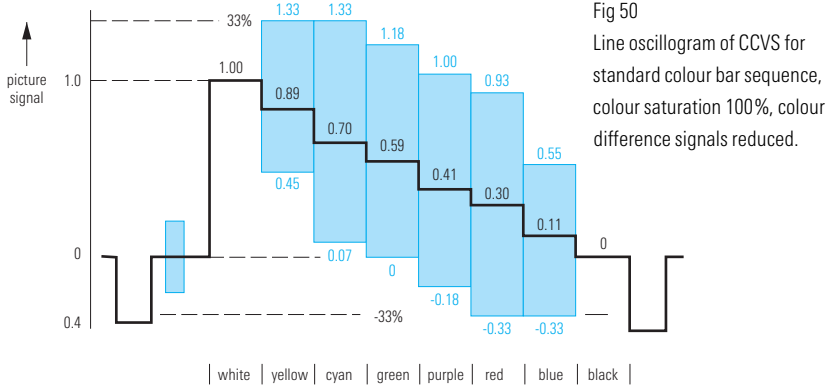
Fig 49
Amplitude modulation of RF vision carrier by CCVS without reducing colour difference signals.

Table 4 Overmodulation of RF vision carrier by standard colour bars.

Colour bar	Y	B -Y	R -Y	SC	Y + SC	Y -SC
White	1	0	0	0	1	1
Yellow	0.89	-0.89	+0.11	0.89	1.78	0
Cyan	0.70	+0.30	-0.70	0.76	1.46	-0.06
Green	0.59	-0.59	-0.59	0.83	1.42	-0.24
Purple	0.41	+0.59	+0.59	0.83	1.24	-0.42
Red	0.30	-0.30	+0.70	0.76	1.06	-0.46
Blue	0.11	+0.89	-0.11	0.89	1.00	-0.78
Black	0	0	0	0	0	0

Table 5 Modulation of RF vision carrier by standard colour bars with reduced colour difference signals

Colour bar	Y	U	V	SC _{red}	Y + SC _{red}	Y -SC _{red}
White	1	0	0	0	1	+1
Yellow	0.89	-0.44	+0.10	0.44	1.33	+0.45
Cyan	0.70	+0.15	-0.62	0.63	1.33	+0.07
Green	0.59	-0.29	-0.52	0.59	1.18	0
Purple	0.41	+0.29	+0.52	0.59	1.00	-0.18
Red	0.30	-0.15	+0.62	0.63	0.93	-0.33
Blue	0.11	+0.44	-0.10	0.44	0.55	-0.33
Black	0	0	0	0	0	0



nals, except in the white bar, are reduced to 75% in accordance with the EBU (European Broadcasting Union) standard. In this way, 33% overmodulation by the colour subcarrier is avoided (Fig 51).

To determine the colour subcarrier phase or the colour points in the (B -Y) (R -Y) plane, a vectorscope is used. This is an oscilloscope calibrated in polar coordinates including two synchronous detectors for the U and V components. The detected colour difference signals are taken to the X and Y inputs of the oscilloscope. Fig 52 shows the vector oscillogram of the standard colour bar sequence.

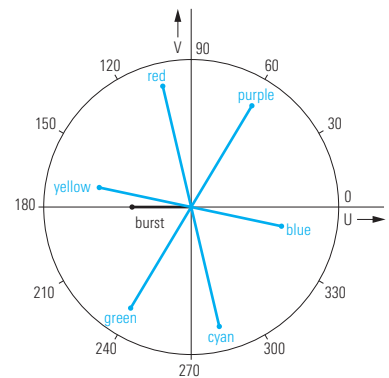


Fig 52
Vector oscillogram of standard colour bar sequence.

5.4 NTSC method

The NTSC, PAL and SECAM methods, which are mainly used for colour TV transmission, differ only with respect to the modulation of the colour subcarrier. The NTSC method, named after the National Television System Committee, constitutes the basis for the improved variants PAL and SECAM.

The principle of the NTSC method has basically been described in the chapters on modulation of the colour subcarrier and on the composite colour video signal. However, the original NTSC system (US standard) does not transmit the reduced colour difference signals U and V but instead the I and Q components, which are referred to a coordinate system rotated counter-clockwise by 33° (Fig 53).

In the colour triangle, the I axis corresponds to the axis for which the eye has maximum colour resolution (Fig 54). This ensures better transmission of colour transitions.

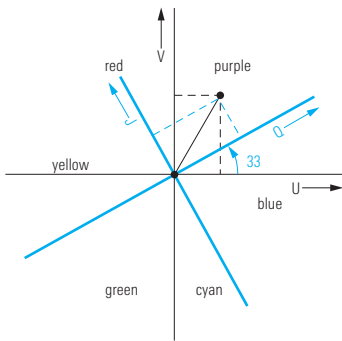


Fig 53
I and Q components of reduced colour difference signals in original NTSC system.

The modulation signals are thus

$$I = V \times \cos 33^\circ - U \times \sin 33^\circ \quad (25)$$

$$Q = V \times \sin 33^\circ + U \times \cos 33^\circ \quad (26)$$

or, using the tristimulus matrix equations,

$$I = 0.60 \times R - 0.28 \times G - 0.32 \times B \quad (27)$$

$$Q = 0.21 \times R - 0.52 \times G + 0.31 \times B \quad (28)$$

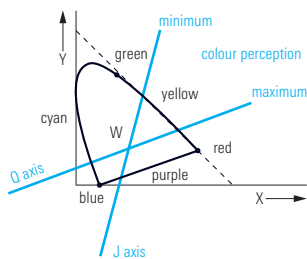


Fig 54
I-Q axes in colour triangle.

The two signals I and Q are transmitted with different bandwidth, i.e.

- the I signal with 1.3 MHz
- and the Q signal with 0.5 MHz.

Fig 55 shows the complete block diagram of an NTSC coder. Except for the 33° phase shift, the functioning of the corresponding decoder is essentially explained in Fig 48.

The human eye reacts very strongly to incorrect hue. The hue of the colour image reproduced by the picture tube is determined by the phase angle of the chrominance signal referred to the phase of the burst. When producing the CCVS in the studio, it may happen that the chrominance signal from different sources has different delays and thus different phases with respect to the burst. To correct wrong hue resulting from static phase errors in the transmission path, the NTSC colour TV receiver is provided with a control permitting the phase of the reference carrier to be adjusted. In most cases this is done by referring to the hue of a well-known picture detail, such as the flesh tone.

However, this hue control does not allow correction of differential phase distortion. In accordance with DIN 45 061 differential phase means the difference of phase shifts through a four-terminal network at two different points of the transfer characteristic at the subcarrier frequency. The differential gain is defined in a similar way.

Due to the shift of the operating point on the transmission characteristic as a function of the Y components of the CCVS, the chrominance signal suffers a gain change (because of the change in slope of the characteristic) and a phase change (because the transistor input capacitance is dependent on the emitter current and thus on the operating point) when passing, for instance, through an amplifier stage with preceding tuned circuit. While the differential gain can be eliminated to

a large extent by negative feedback, the differential phase can be reduced only by limiting the driving level. Fig 56 shows this influence on the CCVS and the effect in the vectorscope representation.

5.5 PAL method

The effects of static and differential phase errors are considerably reduced with the PAL method, the occurring phase errors being corrected with relatively little extra outlay. The PAL system is based on the following concept: an existing phase error can be compensated by a phase error of opposite polarity. This is realized technically by alternating the phase of one of the two chrominance signal components, for instance the SC_V component, by 180° from line to line. PAL stands for phase alternation line.

If a phase error exists in the transmission path, alternately positive and negative departures of the chrominance signal phase from nominal are produced in the receiver after elimination of the line-to-line polarity reversal of the SC_V component generated at the transmitter end. Delaying the chrominance signal for the duration of a line (64 μs) and subsequent addition of the delayed and the undelayed signals cause two phase errors of opposite polarity to coincide and thus to cancel each other. It should be mentioned, however, that this method is based on the assumption that the chromaticity does not change within two consecutively transmitted lines. If horizontal colour edges exist, the eye hardly perceives a falsification of the colour transition even in this case.

Fig 55
Block diagram of NTSC coder.

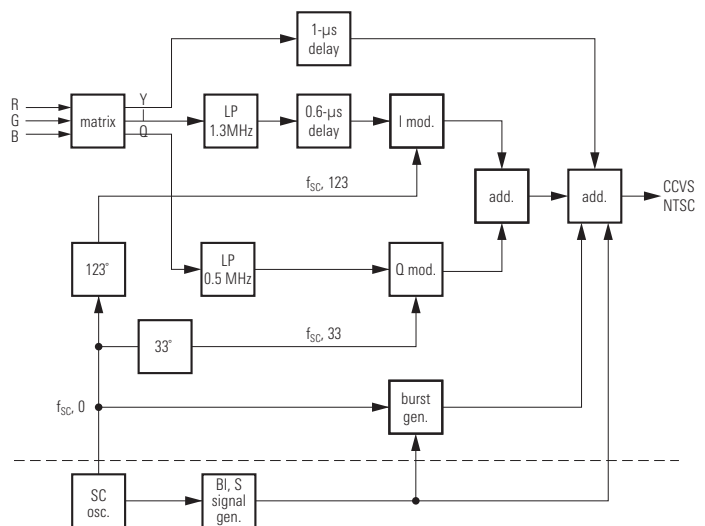


Fig 57 shows the compensation of a phase error with the PAL method. The assumed phase error α affects the chrominance signal with respect to the burst on the transmission link.

After elimination of the SC_V polarity reversal (PAL switchover) and addition of the chrominance signals in two successive lines, the phase angle of the resulting signal SC_{res} is equal to that of the transmitted chrominance signal, and the original hue is thus maintained. After reducing the resulting signal to half its amplitude, this signal exhibits only slight desaturation.

An additional identification is transmitted with the burst to ensure correct phase reversal to the SC_V component in the receiver or of the reference carrier for the

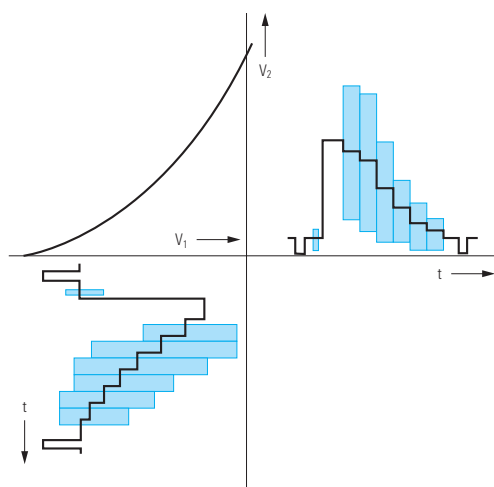


Fig 56
Generation of differential gain and phase distortion.

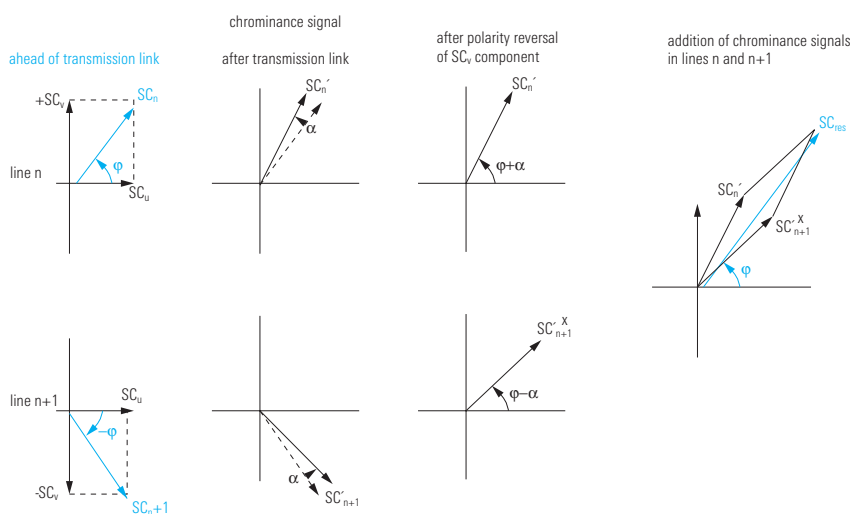


Fig 57
Compensation of phase error with PAL.

(R - Y) sync detector. To this effect, the burst is split into two components, one being transmitted at 180° and the other at $\pm 90^\circ$ alternating from line to line in phase with the SC_V reversal. This yields a swinging burst of $180^\circ \pm 45^\circ$ (Fig 58). The actual burst reference phase (180°) is obtained by averaging.

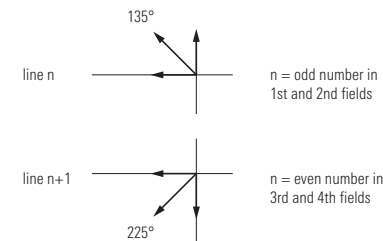
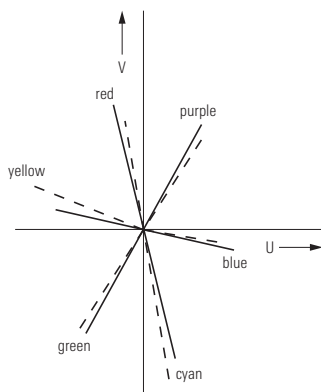


Fig 58
Swinging burst with PAL method.



The reference carrier oscillator in the receiver adjusts, via the phase control, to 90° with respect to the mean burst phase. The identification signal (burst flag) for synchronizing the PAL switch is derived from the burst phase discriminator (see Fig 62). With PAL the reduced colour difference signals U and V are directly transmitted, the bandwidth being 1.3 MHz. Limiting the sidebands of the modulated colour subcarrier to different widths no longer has an annoying effect thanks to the phase error compensation. Fig 59 is a block diagram of a PAL coder. Compared with the NTSC coder, the 33° phase shift of the colour subcarrier components is omitted, but reversal of the subcarrier component for the (B - Y) modulator and generation of the swinging burst are added.

The technical realization of the PAL error compensation requires special explanation as against Fig 57. For this purpose it is best to start with the group delay decoder included in the PAL decoder. In contrast to the NTSC decoder, the chrominance signal is not simultaneously applied to the two sync detectors in the PAL decoder but is first split into the SC_U and SC_V components.

This is performed in the group delay decoder (Fig 60). At its output, the incoming chrominance signal is divided into three components. It is taken to the two outputs via a $64 \mu s$ delay network (line duration) directly and after a 180° phase shift. At the outputs, signal addition takes place. The chrominance signal of the preceding line (SC_n) and that of the ongoing line (SC_{n+1}) are added at the SC_U output. Successive lines contain the SC_V component with a 180° phase alternation so that the SC_V component is cancelled every two lines. Thus the SC_U component of the chrominance signal is constantly available at this output.

The input signal is taken to the SC_V output with a 180° phase shift. Addition of the delayed chrominance signal cancels the SC_U component, and the SC_V component appears at this output, although with a 180° phase alternation from line to line. Based on the vector diagrams in Fig 61, the functioning of the group delay decoder can be explained very easily.

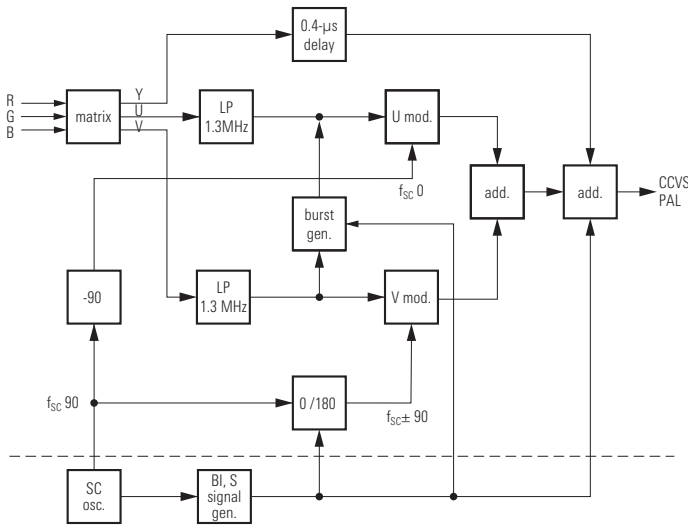


Fig 59
Block diagram of NTSCcoder.

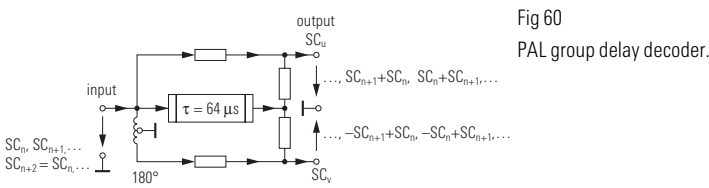


Fig 60
PAL group delay decoder.

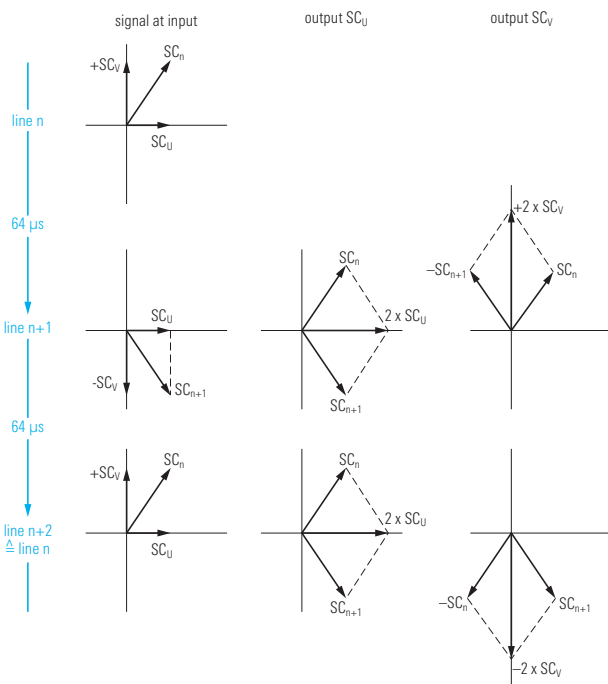


Fig 61
Division of chrominance signal into SC_U and SC_V components in PAL group delay decoder.

The line-to-line phase alternation of the SC_V component can be disabled by a controlled switchover. However, it is easier to provide for line-to-line phase reversal of the reference carrier in the (R - Y) sync detector. Within the complete PAL decoder this task is performed by the PAL switch, which is synchronized by the swinging burst (Fig 62).

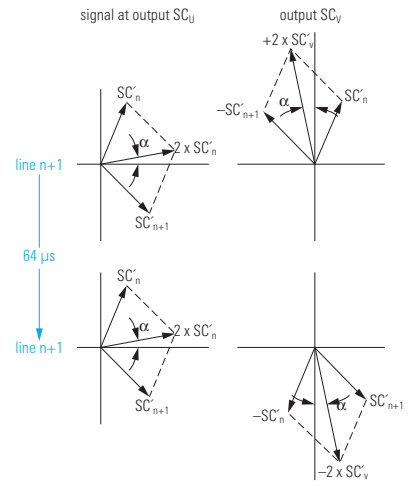


Fig 63
Effect of phase error on signals with PAL group delay decoder.

A phase error in the transmission path affects both the SC_U and the SC_V components in the same sense (Fig 63). Since, however, only the component in phase with the reference carrier is weighted in the sync demodulators, the (B - Y) demodulator delivers the signal $U' = |SC_U| \times \cos \alpha$ and the (R - Y) demodulator the signal $V' = |SC_V| \times \cos \alpha$. Both colour difference signals are reduced by the same factor so that the ratio V/U or $(R - Y)/(B - Y)$ remains constant and the hue of the reproduced image is not affected. Desaturation, which corresponds to the factor $\cos \alpha$, becomes significant only with large phase errors.

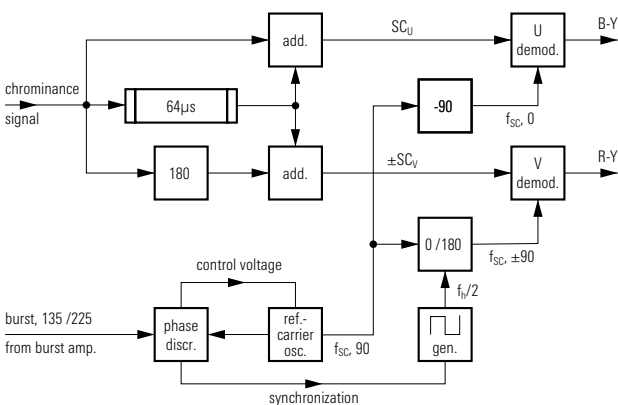


Fig 62
PAL decoder with reference carrier generation.

5.6 SECAM method

As against the NTSC method, the SECAM method too brings an improvement with respect to wrong hues caused by phase errors in the transmission path. Like the PAL method it is based on the assumption that the colour information does not essentially vary from line to line or that the human eye does not perceive any annoyance if the vertical colour resolution is reduced to a certain extent.

Therefore, the colour difference signals (B-Y) and (R-Y) characterizing the colour information need not be transmitted simultaneously. They can be sent separately in successive lines. In the receiver, the signal content of one line is stored for 64 μs via a delay line and processed together with the signal of the next line. The short form SECAM derived from "séquentiel à mémoire" indicates that this is a sequential colour system with memory.

As the two colour difference signals are transmitted separately, the type of modulation can be freely selected. SECAM uses frequency modulation, which is not very interference-prone. However, the reference frequency of the FM demodulator must be kept very stable so that the demodulated colour difference signals are not falsified.

Fig 64
Simplified block diagram
of SECAM coder (above)
and decoder (below).

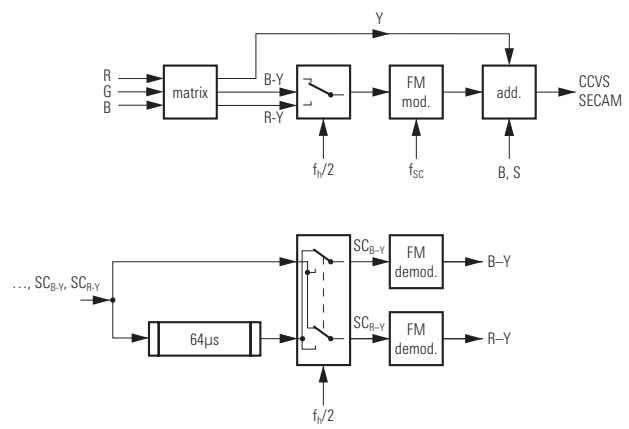


Fig 64 is a simplified block diagram of a SECAM coder and decoder. In the coder, the (B-Y) and the (R-Y) signals are applied to the frequency modulator in alternate lines. To ensure that in the decoder the demodulated colour difference signals are in synchronism with the transmitter end, identification pulses in the form of the modulated colour subcarrier are transmitted during nine lines of the field blanking interval.

When frequency-modulating the colour subcarrier, the latter is not suppressed. In particular with colours of low saturation, this would produce an interference pattern on a black-and-white receiver in spite of the colour subcarrier offset. Therefore the colour subcarrier is attenu-

ated by preemphasis at the transmitter end and boosted by deemphasis at the receiver end. The effect of noise is reduced by video frequency preemphasis and deemphasis.

SECAM has gone through several phases of development. Its latest variant, SECAM III b or SECAM III opt., is based on slightly different colour subcarrier frequencies for the (B-Y) and (R-Y) signals, further reducing the interference pattern caused by the colour subcarrier.

As against PAL, SECAM features some system-dependent weak points since frequency modulation is utilized at its physical limits [3].

6 Colour picture pickup and reproduction

Previous explanation was based on an electrical picture signal obtained from the pattern to be transmitted by optoelectrical conversion. Below, the converters used at the TV transmitter and receiver ends are briefly looked at and finally the reproduction of the colour picture by the TV receiver is explained.

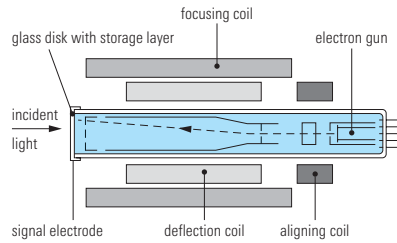


Fig 65 Design of Vidicon camera tube.

6.1 Principle of colour picture pickup

An optoelectrical converter is used to translate brightness variations into an electrical picture signal. Different converter systems are available, but only the pickup tubes with a photosensitive semiconductor layer are really important for TV technology.

In Vidicon tubes a semiconductor layer is used as the storage plate or target, its blocking resistance varying with the intensity of the light falling upon it. The characteristics of the converter differ depending on the composition of the semiconductor target. Frequently a Plum-bicon is used; this has a target of lead monoxide and, compared to the Vidicon using an antimony-trisulphide layer, features higher sensitivity and less inertia.

Fig 65 shows a Vidicon pickup tube with its deflection and focusing coils. The Vidicon works as follows: the electron beam, caused to emanate from the cathode by an electric field, negatively charges the side of the target facing the beam-producing system. Positive charge carriers are bound on the picture side of the target with the aid of the positive target-plate voltage. At the points upon which light falls, the incident photons release electrons in the semiconductor layer

causing a charge compensation at the corresponding picture elements due to the resulting lower blocking resistance. During the next charging process, electrons are again bound at these places on one side of the target plate and on the other side the same number of electrons are set free. These electrons flow across the external circuit resistance causing a signal voltage to be produced. Fig 66 shows the equivalent circuit of a picture point on the target, represented by a capacitor shunted by an exposure-dependent resistor..

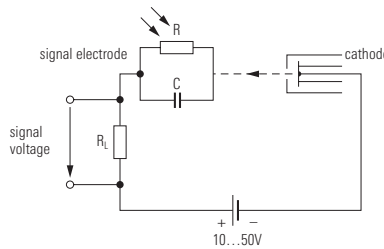


Fig 66 Equivalent circuit for picture element on storage plate of Vidicon camera tube.

In line with the basic principle of colour transmission, three pickup tubes are required; the image to be televised is projected, using the primaries red, blue and green, onto three photosensitive semiconductor targets via an optical beam splitter, called colour splitter, and via correcting filters for matching the signals to the spectral sensitivity of the semiconductor layers (Fig 67). To make the three partial images coincide accurately with their rasters, high mechanical and electron-optical precision is required. Coincidence errors of the colour rasters would cause a loss in definition for the luminance signal. For this reason, colour television cameras with a separate pickup tube for the luminance signal are also used. Progressive developments, already partly implemented in portable colour TV cameras, point to a single-tube colour TV camera producing the tristimulus signals in the red, green and blue channels by a multiplex method.

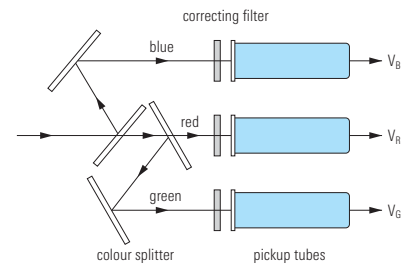


Fig 67 Splitting of incident light into three primaries in colour TV camera.

6.2 Colour picture reproduction using hole or slot mask tube

For reproducing the brightness pattern, television uses picture tubes with a phosphor screen which lights up in accordance with the intensity of the electron beam falling upon it. The electron beam is deflected within the raster with the aid of magnetic fields produced by the deflection currents in the horizontal and vertical deflection coils (Fig 68). The intensity of the electron beam is influenced by the voltage across the control electrode.

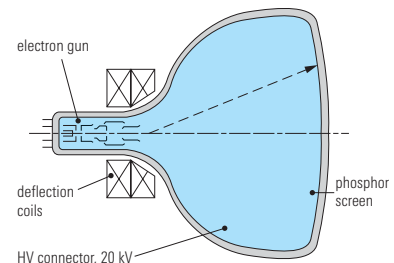


Fig 68 Black-and-white picture tube with deflection system.

Whereas a homogeneous, whitish-blue phosphor screen is used in picture tubes for monochrome reproduction, the screen of the colour picture tube must emit the primaries red, green and blue.

However, colour detail resolution should go as far as the individual picture elements. For this purpose each picture element is represented on the screen by

three luminescent dots in the colours red, green and blue, called colour triad (Fig 69). About three times 400 000 luminescent dots are accommodated on the screen area.

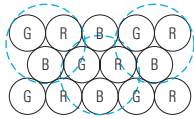


Fig 69
Arrangement of luminescent dots in colour picture tube.

The colour picture tube includes three beam-producing systems. In the **delta colour picture tube**, which for many years was practically the only colour picture tube in use, the beam-producing systems are arranged at angles of 120° to one another. The intensity of the emitted electron beams is controlled by the tri-stimulus signals that are applied.

To obtain –with common deflection of the three electron beams –a clear assignment to the luminescent dots, a hole mask is placed about 15 mm from the phosphor screen (Fig 70).

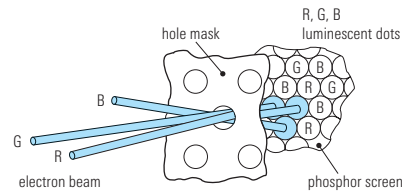


Fig 70
Part of hole mask and phosphor screen in delta colour picture tube.

More recent colour picture tubes are fitted with a slot mask. Accordingly, the luminescent areas on the screen are either oblong or in the form of stripes. With this **in-line colour picture tube** the three beam-producing systems are arranged in one plane (Fig 71). This configuration yields high colour purity, i.e.

the electron beams only strike luminescent stripes of the correct colour associated with the corresponding beam producing system.

Moreover, in conjunction with a special deflection field, excellent convergence is achieved, i.e. the correct luminescent areas associated with the picture elements are excited. With the delta colour tube, greater complexity of circuitry was required for this purpose.

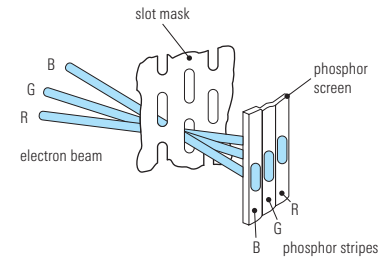


Fig 71
Part of slot mask and phosphor screen with in-line colour picture tube.

7 Block diagram of PAL colour TV receiver

Detailed signal tracing is often no longer possible in the circuit diagram of a modern TV receiver. The reason is the frequent use of integrated circuits combining several functional units. Even though this entails a large degree of standardization in circuit design, there is still a broad line of integrated circuits being offered for colour TV receivers due to the variety of possible combinations of the functional units.

For better explanation of the interworking of the functional units included in a PAL colour TV receiver, the individual units are shown separately in Fig 72. The dash-dotted lines indicate the possibilities of combination that are feasible and have been realized in practice by way of including several functional units in integrated circuits.

The RF signal coming from the antenna is converted into the IF in the VHF-UHF tuner; next it is taken via the Nyquist filter to compensate for the vestigial sideband component and then boosted in the IF amplifier to the level required for demodulation. To ensure that there is no intermodulation between the colour subcarrier and the intercarrier sound carrier, the IF sound carrier is isolated from the vision IF demodulator. In a separate diode circuit, the 5.5 MHz intercarrier sound IF signal is produced from the sound IF and the vision IF. More recent developments tend towards the split-carrier method, i.e. the 33.4 MHz sound IF signal is directly amplified and detected.

The detected CCVS is derived from the vision IF demodulator. Next the sync signal is separated and the chrominance signal is brought out selectively via a 4.43 MHz bandpass filter. The remaining Y component is delayed by about $1 \mu\text{s}$ to match the longer signal delay in the chrominance amplifier; after further amplification, the signal is taken to the matrix circuit.

After amplification the chrominance signal is applied to the group delay decoder, where it is split into the SC_U and SC_V components. As already explained, the colour difference signals are retrieved in the two synchronous demodulators. The tristimulus signals for driving the colour picture tube are obtained from the matrix circuit.

Parallel to its application to the chrominance amplifier, the burst is taken to the phase discriminator via an amplifier triggered by the line sync pulse. Here the signal is compared with the reference car-

rier. The control voltages for the reference carrier oscillator and the sync signal for the PAL switch are derived from the phase discriminator. Furthermore, the phase discriminator drives the colour killer, which inhibits the chrominance amplifier if the burst is absent so that no coloured noise occurs when reproducing a black-and-white picture.

Following the sync separator, the sync signals are fed separately to the horizontal and vertical deflection circuits as with a black-and-white receiver. Additional correction signals, which are applied to

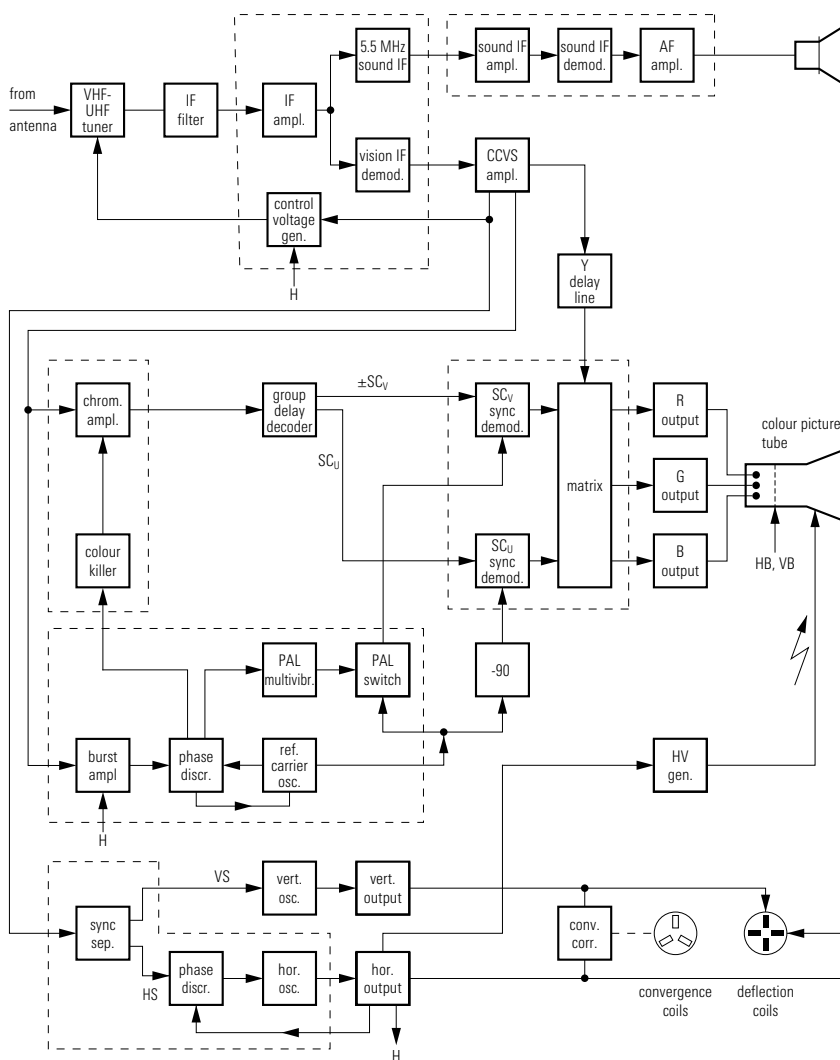


Fig 72
Block diagram of PAL standard colour TV receiver.

the deflection coils and/or the convergence correction coils of the deflection system of the colour picture tube, are derived from the horizontal and the vertical deflection signals.

Modular technology and the use of integrated circuits lead in part to subassemblies whose functions extend to other units. If required, further subassemblies can be added for user controls, for instance, multi-standard decoding, time insertion, picture-in-picture insertion or viewdata and teletext

The chapters added to the reprints of the refresher serial have been omitted in the present edition for the following reasons:

There is no point now in examining the analog HDTV signal as the source signal from the TV camera or the film scanner is immediately subjected to an A/D conversion and transmitted in the form of a digital HDTV signal. This is described in the chapter "Digital video studio signal" of the present revised edition.

The D2-MAC method introduced in 1983 for use in the whole of Europe has been superseded by the rapid development of digital TV. Only about 10% of the program signals presently transmitted via satellite are to D2-MAC standard and the method has never been used for terrestrial transmissions. For this reason there is no room for this method in a compendium on TV technology and a description of the HD-MAC method also becomes superfluous.

Broadcasting of TV signals via satellite using analog FM will remain important for some time. However, the satellite channels have become so numerous since the last edition of this brochure that it is not possible to deal in detail with the coverage of the ASTRA and EUTELSAT satellites, which are only relevant for Europe. Furthermore, the direct-reception satellite TV-SAT 2 has been assigned a special application. The literature on satellite reception and distribution techniques has become so extensive by now that this subject will not be dealt with either in the following.

So, the PALplus method is the only topic remaining of the extensions to the first edition, but even this enhanced PAL method with a wide aspect ratio will in the near future be replaced by the digital MPEG2 standard. Because of its compatibility to PAL and therefore to the great number of PAL colour TV sets in use, this analog procedure can also be employed in parallel with digital TV.

A digital video source signal is required for both the PALplus and the MPEG2 encoder. So a detailed description of the processing of digital video studio signals has been added after the revised chapter on PALplus.

8 PALplus system

In cooperation with renowned industrial companies, scientific institutes, European broadcasting companies and the Institute for Broadcasting Technology (IRT), an enhanced PAL system called PALplus was developed permitting transmission with a wide aspect ratio.

Characteristic features of the PALplus system are

- suppression of cross colour and cross luminance effects for full utilization of the 5 MHz luminance bandwidth
- 16:9 aspect ratio

and planned

- digital sound transmission
- correction of echoing.

Since PALplus is fully compatible with the standard PAL system, PALplus signals can be transmitted in existing 7 MHz or 8 MHz channels or in the PAL satellite channel. Pictures with a 16:9 aspect ratio are displayed on a standard 4:3 screen with the black bands at the top and bottom known from widescreen film reproductions.

In addition to the planned digital audio signals, analog FM audio signals will also be transmitted. The correction of echoing is another technical feature of the PALplus system.

8.1 Spectrum of PAL CCVS

In the frequency range between approx. 3 MHz and 5 MHz, the PAL CCVS contains components of the luminance signal Y and the chrominance signal F (Fig 73, top). A detailed analysis of the PAL CCVS spectrum yields interleaved spectra of the Y signal, i.e. the CVS, and of the chrominance signal components F_U and F_V (Fig 73, bottom). The signal comes as a frequency spacing multiplex with the components completely separated in the undisturbed transmission channel [5].

Comb filters are used in the receiver for separating the spectral components. In a standard PAL colour TV receiver, the comb filter function for separating the F_U and F_V components of the chrominance signal is assumed by the group delay decoder (see Fig 60). The periodic transfer function is obtained by adding or subtracting the luminance signal delayed by the line period H, i.e. $284 \times T_{SC} = 64.056 \mu s$, for setting a phase shift of $k \times 2\pi$. Fig 74 illustrates the transfer function of the notch filter for the two outputs F_U and F_V .

With the comb filter transfer function superimposed on the PAL CCVS it can be seen that the chrominance signal is divided into F_U and F_V components but that high-frequency components of the luminance signal persist in the two signal spectra (Fig 75). As a result of these cross-colour or cross-chrominance effects, an interference colour pattern is displayed on the screen when the chrominance components are demodulated into the primaries R, G, B.

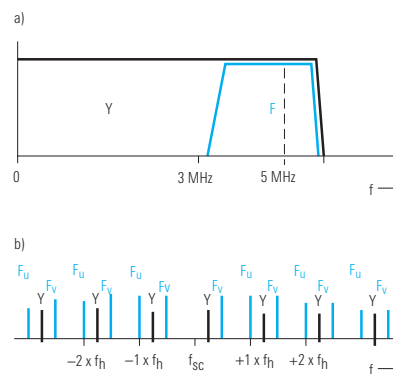


Fig 73 Spectrum of PAL CCVS
top: luminance signal Y and chrominance signal F
bottom: interleaved spectra of Y, F_U and F_V signals in colour subcarrier range (f_{sc}).

Chrominance signal components appear in the broadband Y signal between 3.1 MHz and 5 MHz, causing cross-lumi-

nance effects which are however less disturbing. In simple PAL TV receivers these effects are eliminated by limiting the frequency band of the luminance amplifier to approx. 3.5 MHz.

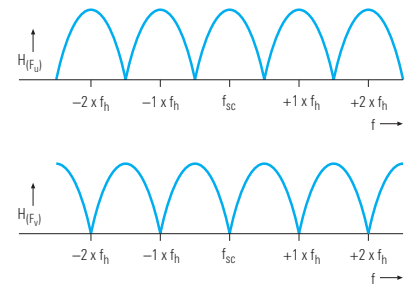


Fig 74 Transfer function of group delay decoder acting as comb filter for outputs F_U and F_V in colour subcarrier range (f_{sc}).

Different methods have been developed for the suppression of cross-colour effects, some of them being already implemented in PAL TV receivers. The suppression of luminance/chrominance crosstalk artefacts usually involves expensive memories and also adaptive measures for eliminating the side effects of the suppression.

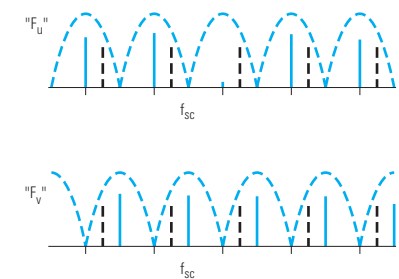


Fig 75 Separation of chrominance signal in group delay decoder into F_U and F_V components with crosstalk of Y signal.

8.2 Colour plus method

The suppression of crosstalk artefacts in the standard PAL system can be considerably improved through the use of the **colour plus method**. It is based on a compensation of interfering signal components by averaging the signals of two consecutive fields with respect to time, which is also referred to as **intraframe averaging** (IFA) [6].

After averaging, the luminance signal Y and the colour difference signals U and V in two successive fields are identical. The colour subcarrier and therefore the chrominance signal F are however shifted by 180° after one field, i.e. after exactly 312 lines. This is obtained from the colour subcarrier period as defined by

$$\begin{aligned} T_{sc} &= 1/f_{sc} \\ &= 1/4.43361875 \text{ MHz} \\ &= 0.2255 \mu\text{s} \end{aligned}$$

over $312 \times 64 \mu\text{s} = 19\,968 \mu\text{s}$ with 88 530.5 colour subcarrier periods.

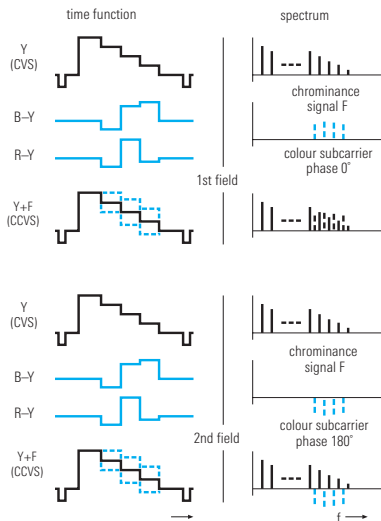


Fig 76 Spectral components of PAL CCVS with phase assignment in 1st and 2nd field.

The spectrum of the PAL CCVS with components of the (Y)CVS and the F signal can be shown in magnitude and phase in a simplified diagram (Fig 76). The crosstalk artefacts within the frequency range 3.1 MHz to 5 MHz are compensated for by adding or subtracting the two signals

spaced by 312 lines after storing the signal received first.

- Adding yields a cross-effect-free luminance signal, i.e. the high-frequency component Y_H ,
- subtracting yields the cross-effect-free chrominance signal F .

Fig 77 illustrates the principle used in a colour plus encoder and decoder. In practice, averaging in the decoder is performed with the demodulated colour difference signals using the 312H memory in the chrominance signal path.

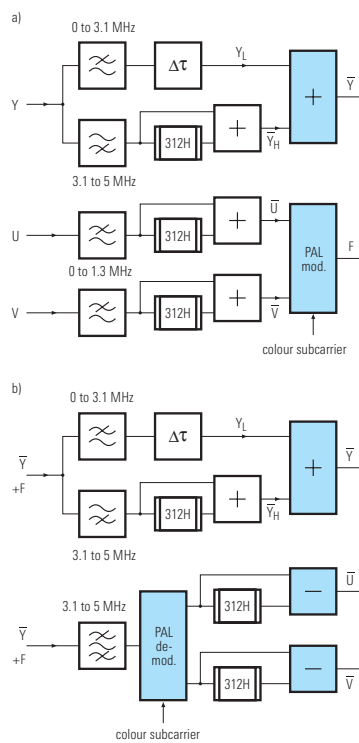


Fig 77 Principle of colour plus encoder (top) and colour plus decoder (bottom).

Averaging in the encoder reduces the vertical temporal resolution of the high-frequency luminance signal components and colour difference signals.

The vertical resolution is reduced from 576 to 288 active lines. This is hardly perceived by the human eye as the eye is less sensitive to the vertical resolution than to the horizontal one.

The time resolution is based on 25 picture motions per second. If the video signal is obtained by **film scanning**, the signal source does not allow for more than 25 movements/s anyway, i.e. the picture content of the two fields is referenced to the same motion phase.

If the video signal comes from an **electronic camera**, the motion phases in the two fields may differ with rapidly changing pictures. In this case, the crosstalk artefact is not compensated for and only the low-frequency component Y_L of the luminance signal down to about 3.1 MHz is transmitted to avoid the chrominance signal being disturbed by luminance signal components.

Another variant of the colour plus method matched to **camera** and **film mode** is the

motion adaptive colour plus (MACP) method.

Here a stepwise (4 steps for the luminance signal) or a continuous switchover (in the colour channel by means of a non-linear characteristic) takes place between

- transmission of the averaged high-frequency luminance signal component \bar{Y}_H and the averaged colour difference signals \bar{U} and \bar{V} and
- elimination of the averaged high-frequency luminance signal components and transmission of the original non-averaged colour difference signals as required by the picture motion [7].

The picture motion on the screen is determined by a pixel-by-pixel comparison. The averaged colour difference signals are applied to a motion detector providing the control signals for fading in and out in the luminance and chrominance channels. Fig 78 shows the principle of luminance signal and colour difference signal processing in the MACP coder. Digital signal processing is used for implementation.

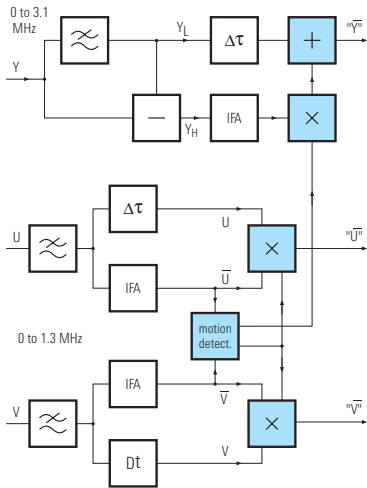


Fig 78
Principle of MACP coder.

A motion detector is also required in the MACP decoder as the signals in the receiver are not synchronized to the frame head and the picture contents are compared with the aid of corresponding fields [8].

8.3 Compatible transmission with 16:9 aspect ratio

The change to the widescreen 16:9 aspect ratio is an important innovation. The method used must however ensure full compatibility with the existing transmission method employing the 4:3 aspect ratio.

Two methods can be used for reproducing pictures with a 16:9 aspect ratio on a 4:3 standard receiver:

1. Side panel method

A 4:3 section of the 16:9 picture starting from the picture center or variable (pan and scan) is transmitted in line with the existing standard. In addition to the picture information of the side panels, which remain invisible on the screen of the conventional receiver, is transmitted using special coding for the 16:9 receiver.

2. Letterbox method

The 16:9 picture is reproduced in full width on the 4:3 receiver. The picture height has to be reduced however, which causes black bands to be displayed at the

upper and lower screen edges (letterbox format). The picture height is reduced through vertical filtering and by removing every fourth line from the total of 576 lines visible with PALplus. The remaining 432 lines are pushed together causing the picture height to be reduced to three quarters of its original size.

The luminance information of the removed 144 lines is transmitted in 72 lines each at the upper and lower screen edge in such a way that it remains invisible on the standard (4:3) receiver. The 16:9 receiver processes the additional information in the "vertical helpers" and, together with the information conventionally transmitted in the 432 lines for the 4:3 receiver, it restores the initial colour picture with 576 visible lines (Fig 79).

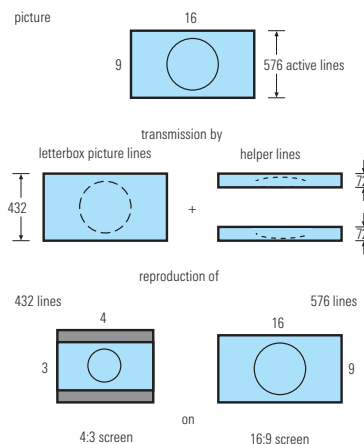


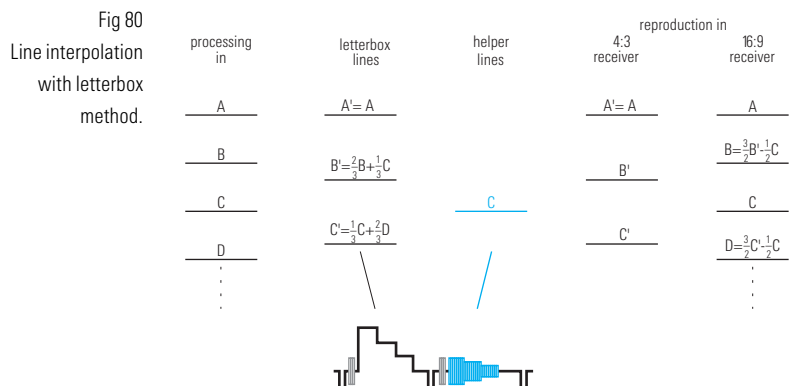
Fig 79
Transmission of 16:9 picture in letterbox format with PALplus.

Reversible line interpolation is performed as shown in the simplified diagram of Fig 80. The 16:9 receiver restores the original lines A, B, C, D from lines A', B', C' transmitted in the compatible 432-line picture and from lines C in the vertical helper.

Vertical filtering of the luminance signal is performed by means of a quadrature mirror filter (QMF). The frequency range to be divided into a lowpass and a high-pass band is filtered so that the bands do not overlap and a flat frequency response is obtained when the subbands are combined again in the receiver.

After interpolation (up sampling) by the factor of three and decimation (down sampling) by the factor of four, the vertical low-frequency component of the 432-line letterbox picture signal is obtained from the low-frequency component of the vertical spectrum (referred to 576 active lines). The high-frequency component of the vertical spectrum is transmitted in the vertical helper signal.

Simpler linear phase filters may be used for vertical filtering of the colour difference signals since no vertical high-frequency components are obtained in this case. As already pointed out, the helper signals only contain the luminance information.



With vertical filtering, a distinction is made between frame filtering (film mode) and field filtering (camera mode).

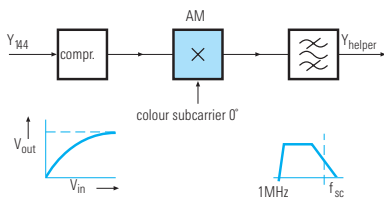


Fig 81 Conversion of luminance signal of helper lines.

For the vertical helpers to remain invisible, the ancillary information is modulated after filtering and nonlinear precorrection onto the colour subcarrier by vestigial sideband amplitude modulation with carrier suppression and is then transmitted with reduced amplitude about the black level in the 72 lines at the upper and 72 lines at the lower screen edge (Fig 81). Compatibility checks have shown that the visibility of the vertical helpers in the letterbox bands is sufficiently low.

Signal processing in the PALplus encoder is fully digital. This means that the component signals applied to the encoder input have to be in digital form. The interface to the input is the digital studio signal which will be dealt with in the next chapter.

The signal symbols of the PAL standard are used in the simplified block diagram of the PALplus encoder (Fig 82).

The lines in the luminance signal path are first separated into the 432 letterbox picture lines and 144 helper lines using the above-mentioned vertical filter (QMF).

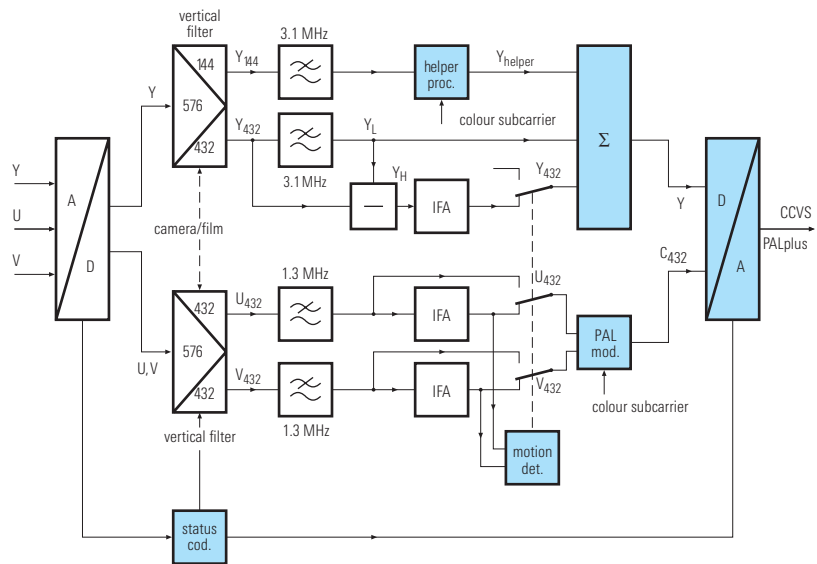


Fig 82 Simplified block diagram of PALplus encoder.

The low-frequency component Y_L is directly obtained from the letterbox picture signal. The high-frequency component Y_H is averaged over two fields (IFA), depending on the motion in the picture, or not transmitted at all after a stepwise reduction. A single switchover is shown in Fig 82 in a simplified form. As described above, the luminance signal from the helper lines is modulated onto the colour subcarrier in the frequency band 1 MHz to 5 MHz and added to the letterbox picture signal (Y_L and Y_H).

The colour difference signals U and V are processed in parallel. The information contained in the letterbox picture lines alone is transmitted –again as a function of the picture motion –as an average value of two consecutive lines (IFA) or temporarily in the unfiltered original signals.

The status information is also transmitted. Data signals inserted in the "active" part of line 23 by means of 14 bi-phase code elements of 1.2 μ s duration each (Fig 83), inform the receiver in a

- 1st group on the aspect ratio at (4:3, 16:9, or others),
- 2nd group on the signal source and image-improving measures (camera mode, film mode; standard PAL, MACP; helper),
- 3rd group on the type of subtitles (none or subtitles in the teletext, subtitles within or outside the active picture),
- 4th group on miscellaneous (e.g. surround sound).

Since the transmission of this information must be highly immune to interference, the bi-phase code is used, and with this aspect ratio the 3-bit code word is additionally protected by a parity bit. The 14 bi-phase code elements are preceded by a run-in code and a start code consisting of (29 +24) elements of the 5 MHz system clock.

A reference burst of the colour subcarrier with defined amplitude ($V_{pp} = 300 \text{ mV}$) is also transmitted in line 23. Together with the white reference bar (700 mV) in line 623 it guarantees correct voltage assignment in the helper signal for the letterbox picture signal (see Fig 83) [7].

Since lines 23 and 623 are occupied by data and reference signals, only 430 of the original 432 letterbox picture lines remain for image reproduction. This however has no visible effect on the picture displayed .

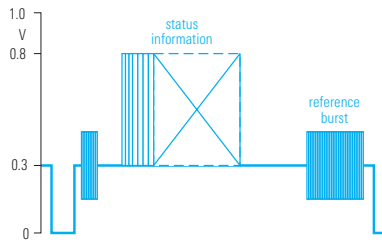


Fig 83
Status bit information and reference burst in line 23.

Signal processing in the receiver must be identical but in inverse form. In a standard receiver configuration, two signal processors (serial video processors) are used together with a field memory and a frame memory. Simpler receiver concepts do without MACP decoding as some interference suppression is already achieved during processing of the PALplus CCVS in the transmitter.

No decision has as yet been made regarding the introduction of digital audio signal transmission and the transmission of test signals for echo correction. Further comments on this subject will therefore not be made.

9 Digital video studio signal

The deployment of digital techniques for processing effects and characters and for storing video signals on digital tape recorders paved the way for digital studios. The introduction of PALplus made it necessary to apply a digital source signal to the coder to provide all the prerequisites required for digital TV in the studio. The standardized digital video studio signal DSC 270 Mbit/s can be applied to both the PALplus encoder and the MPEG2 encoder for further processing.

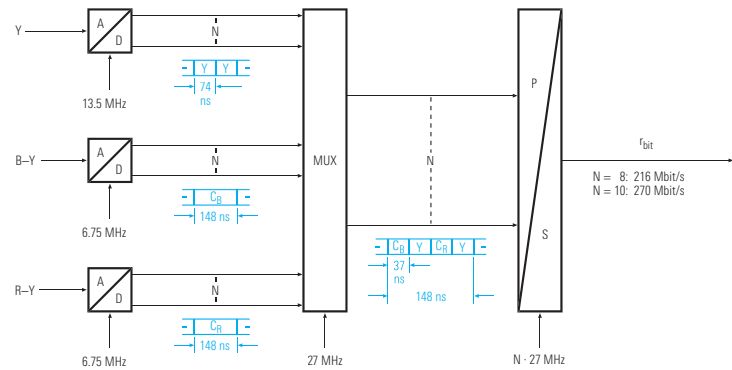
As early as in 1982, the digital studio standard CCIR 601 valid worldwide for 525-line and 625-line systems was issued in the form of a recommendation. At the end of 1992, CCIR (Comité Consultatif International des Radiocommunications) became the Radiocommunication Sector (RS) of the International Telecommunication Union (ITU). Since then the digital studio standard is available as Rec. ITU-R BT.601 under the title **Encoding Parameters of Digital Television for Studios**.

A distinction is made between the 4:2:2 and the 4:4:4 sampling schemes. The 4:2:2 scheme is used in the following description. The 4:2:2 and 4:2:0 schemes will be dealt with later on.

The analog source signals Y, B-Y and R-Y, which are limited in a lowpass filter to the 5.75 MHz (Y) and 2.75 MHz (B-Y, R-Y) band, undergo analog/digital conversion using the principle of pulse code modulation (PCM). In compliance with the sampling theorem by Shannon, the sampling frequency f_s must be at least twice the highest signal frequency $f_{sig\ max}$ in the sampled signal.

The sampling frequency for the Y signal is $f_{S,Y} = 13.5$ MHz with 720 samples in the active part of each line (53.33 μ s digital video signal), and $f_{S,B-Y} = f_{S,R-Y} = 6.75$ MHz for the colour difference signals with 360 samples in the active part of each line.

Fig 84
Processing
of digital
serial video
studio
signals.



The samples are encoded with 10 bits per sample at equally spaced quantization levels.

The ITU-R BT.601 standard also allows for 8 bit encoding which however causes visible quantization noise in critical images. The signals to be encoded are converted to a uniform voltage range of $V_{pp} = 1$ V which corresponds to the picture signal level for white in the Y signal for which the following holds

$$Y = 0.30 \times R + 0.59 \times G + 0.11 \times B \quad (29)$$

In the case of ± 0.5 V colour difference signals it corresponds to the maximum value of the standard colour bar with the new colour difference signals reduced to the following reference:

$$C_B = 0.56 \times (B-Y) - 0.17 \times R - 0.33 \times G + 0.50 \times B \quad (30)$$

$$C_R = 0.71 \times (R-Y) + 0.50 \times R - 0.42 \times G - 0.08 \times B \quad (31)$$

The quantization range of 0 to +1 V for the Y signal with 10 bit encoding and the quantization range -0.5 V to +0.5 V are assigned the quantization interval numbers 64 to 940. Thus a small working margin remains. The code words for the quantization interval numbers 0 and 1023 are excluded.

Analog-digital conversion is performed separately for the Y, B-Y and R-Y signals. Fast parallel converters are used

where the digital signal is available at $N = 10$ parallel outputs for the sampling period T_S . This period is $T_{S,Y} = 74$ ns for the luminance signal Y and $T_{S,B-Y} = T_{S,R-Y} = 148$ ns for the two colour difference signals.

A multiplexer controlled by the 27 MHz clock ($T = 37$ ns) combines the code words in the serial sequence $C_{B-Y}C_{R-Y}$ on $N = 10$ parallel lines. Subsequently a parallel-serial conversion of the multiplexed signal is performed with the 270 MHz bit clock, yielding a bit rate of $r_{bit} = 270$ Mbit/s for the **digital serial video components signal DSC 270** (Fig 84).

$$r_{bit} = 13.5 \times 10^6 \frac{1}{s} \times 10 \text{ bit} + 2 \times 6.75 \times 10^6 \frac{1}{s} \times 10 \text{ bit} = 270 \text{ Mbit/s} \quad (32)$$

The **active video signal** contained therein with 576 active lines and 720 pixels per active line has a bit rate of

$$r_{bit} = 720 \times 576 \times 10 \text{ bit} \times \frac{1}{40 \text{ ms}} + 2 \times 360 \times 576 \times 10 \text{ bit} \times \frac{1}{40 \text{ ms}} = 207.360 \text{ Mbit/s} \quad (33)$$

With 8 bit encoding a value of 165.888 Mbit/s would be obtained.

The bit rate for a **digital serial HDTV studio signal**, for which an international standard does not exist because no agreement could be reached between

the countries using 525 lines and those using 625 lines, is calculated with the 8 bit encoding employed in HDTV in the case of

- **progressive scanning** using 1152 active lines, 50 frames/s, 1920 samples per active line for the Y signal and 960 samples for each colour difference signal as

$$r_{\text{bit}} = 1920 \times 1152 \times 8 \text{ bit} \times \frac{1}{20 \text{ ms}} + 2 \times 960 \times 1152 \times 8 \text{ bit} \times \frac{1}{20 \text{ ms}} = 1769.472 \text{ Mbit/s} \quad (34)$$

- **interlaced scanning** using 1152 active lines, 50 fields/s, 1920 samples per active line for the Y signal and 960 samples for each colour difference signal as

$$r_{\text{bit}} = 1920 \times 576 \times 8 \text{ bit} \times \frac{1}{20 \text{ ms}} + 2 \times 960 \times 576 \times 8 \text{ bit} \times \frac{1}{20 \text{ ms}} = 884.736 \text{ Mbit/s.} \quad (35)$$

The **HDTV 1440 standard** to be used for transmission using

- 1152 active lines, 50 fields/s and 1140 samples per active line for the Y signal and 720 samples for each colour difference signal

yields a bit rate of

$$r_{\text{bit}} = 663.552 \text{ Mbit/s.} \quad (36)$$

for the active HDTV video signal.

The digital standard signal components are combined to form the **digital serial components signal DSC 270 Mbit/s** as defined in Recommendation **ITU-R BT.656** (Interfaces for Digital Components Video Signals in 525-Line and 625-Line Television Systems operating at the 4:2:2 Level of Recommendation ITU-R BT.601)

This recommendation also describes the digital sync signals **start of active video (SAV)** and **end of active video (EAV)** (timing reference signals) as well as the digital blanking signals, which are mostly replaced by **ancillary data signals** that may also come in the form of **digital accompanying audio signals**, e.g. for

specifying the signal source and the aspect ratio. In contrast to the analog CCVS, the digital serial components signal **DSC 270** also contains accompanying audio signals (Fig 85).

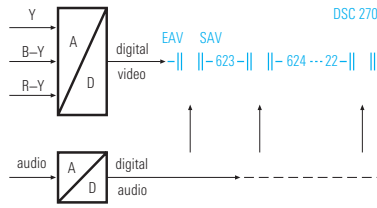


Fig 85 Sync signals (timing reference signals SAV, EAV) and digital audio signals in digital video studio signal.

The ITU-R Recommendations 601 and 656 refer to the **4:2:2 sampling scheme**. It should be pointed out in this context that the 4:4:4 scheme is used as a basis. The values given here define the sampling frequencies of the R-G-B or Y-C_B-C_R signals with 4 x 3.375 MHz = 13.5 MHz each. The sampling ratio is therefore 4:4:4.

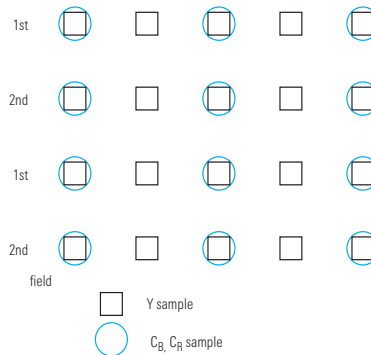


Fig 86 Constellation of Y, C_B and C_R sampling points with 4:2:2 sampling scheme.

Since the resolution of the colour difference signals may be lower than that of the luminance signal, the number of samples of the colour difference signals C_B and C_R and therefore the sampling frequency can be reduced to half the value of the luminance signal, i.e. to 6.75 MHz. In this case the sampling ratio is 4:2:2. This yields the 4:2:2 picture sampling scheme shown in Fig 86 with samples for the luminance signal Y and the colour difference signals C_B and C_R.

The logic of the sampling frequency ratios does not hold for the **4:2:0 scheme**. Starting with a reduction of the horizontal resolution of the colour difference signals in the 4:2:2 scheme, the vertical resolution is halved in a next step. This is performed by averaging the superimposed chrominance samples of two consecutive lines and assigning the averaged value to the line pair symmetrically to four luminance samples. Fig 87 shows the constellation of the Y and C_B-C_R samples.

Based on equation (33) and 8 bit encoding, the bit rate of the active video signal using the 4:2:0 sampling scheme and the vertically reduced C_B and C_R sampling points is

$$r_{\text{bit}} = 720 \times 576 \times 8 \text{ bit} \times \frac{1}{40 \text{ ms}} + 2 \times 360 \times 288 \times 8 \text{ bit} \times \frac{1}{40 \text{ ms}} = 124.416 \text{ Mbit/s}$$

in comparison with the 165.888 Mbit/s obtained with the 4:2:2 scheme.

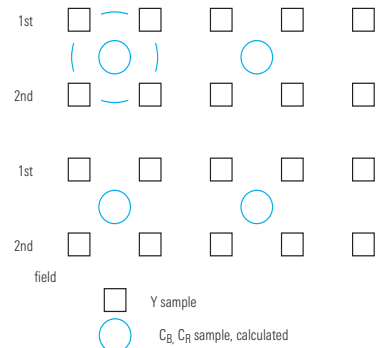


Fig 87 4:2:0 sampling scheme.

The 4:2:0 scheme is often used as a first step for the data compression described in the next chapter.

10 Data compression techniques

For the transmission of digital video studio signals in existing broadband channels to and between TV studios and particularly for the distribution to viewers, the amount of data to be sent must considerably be reduced. This data reduction may however not affect the picture quality.

The bit rate of the digital video source signal is reduced by removing redundancy and irrelevant information from the signal.

Redundant information is transmitted in

- blanking intervals together with the sync signal as well as in
- consecutive pixels and frames

and can be removed without affecting the picture quality as perceived by the human eye.

Redundancy reduction does not entail any loss in information and is therefore referred to as **lossless coding**.

Irrelevant information in

- details of the picture and in
- the form of high chrominance resolution

can be removed by quantization and by using different resolution for the luminance and chrominance signal components without any quality degradation noticeable to the human eye.

Irrelevancy reduction involves a loss of information and is therefore referred to as **lossy coding**.

The first step to redundancy reduction of the digital video studio signal is that only the active video signal is processed. Synchronization is transmitted by means of a few code words in the data stream. The main part of redundancy reduction is performed by utilizing the correlation between neighbouring pixels and particularly between successive frames.

10.1 Redundancy reduction for the video signal

Redundancy reduction can be effected for example by transmitting only the difference of the successive picture pixel values by way of **differential pulse code modulation (DPCM)**.

The signal value transmitted first serves as a predictive value which is then corrected with the difference to the actual value. The corrected value becomes the new predictive value. Fig 88 gives an example where the 8 bit PCM code words obtained with the sampling clock T_S are reduced to 4 bit DPCM code words for the difference transmitted. This halves the bit rate in the transmission channel.

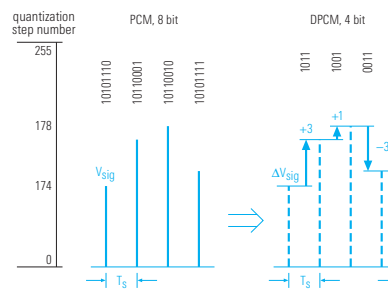


Fig 88 Transition from pulse code modulation (PCM) to differential pulse code modulation (DPCM).

In practice, a PCM signal is used for generating differential pulse code modulation (Fig 89). The difference between the current signal value and the predictive value obtained with the aid of the "predictor" is applied, after quantization with fewer steps than at the input signal, to

the decoder in the receiver for restoring the original PCM signals and to a decoder in the transmitter for creating the prediction signal. The difference value is of course smaller than the quantization range of the input signal because of the statistical distribution of signal values. In the transmit and receive signal paths the 4 bit DPCM signal is first reconverted to an 8 bit code word signal. To obtain the new prediction or to restore the new signal value in the receiver, which is identical to the predicted value in the transmitter, the difference is added to the signal value supplied last by the predictor.

The data quantity to be transmitted depends on the difference between the current signal value and the predicted value. The better the prediction the smaller the difference and the lower the amount of data to be transmitted. When the difference is small, the restored receive signal will correspond better to the original signal. The aim of data reduction is to obtain the best possible prediction.

Video signals of moving pictures contain a lot of redundancy as there is a significant degree of commonality between successive frames. It seems obvious therefore that only the difference between successive frames is transmitted. This is done by determining the difference between two frame sections, i.e. between $m \times n$ pixel blocks.

If the prediction obtained by shifting the pixels of a block of frame n in the horizontal and vertical direction optimally matches the corresponding block in the

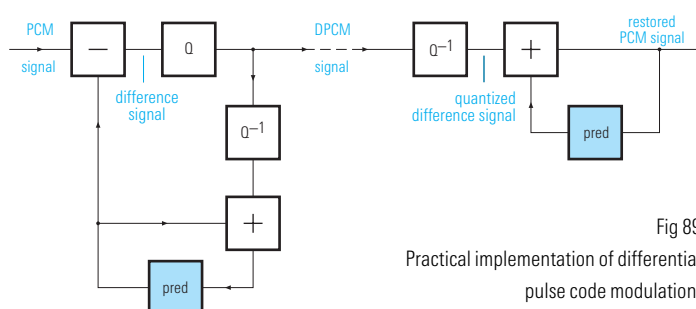


Fig 89 Practical implementation of differential pulse code modulation.

next frame $n + 1$, the difference between comparable blocks is very low or approaches zero. This is referred to as **motion compensation**.

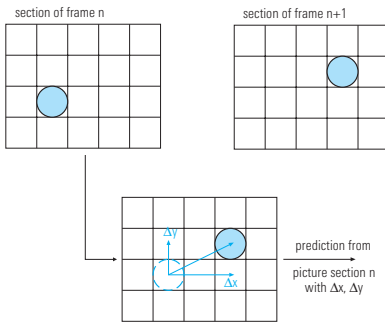


Fig 90
Example of motion compensation.

Fig 90 gives an example of this type of compensation. The prediction in the block is obtained by comparing (motion estimation) the picture content of a block of frame $n + 1$ with the corresponding block in frame n and successively shifting the viewed block of frame n by the values Δx and Δy , referred to as **motion vectors**, until the difference is minimized.

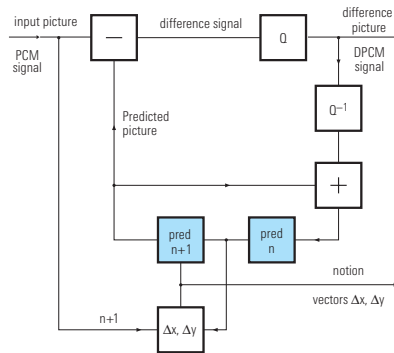


Fig 91
Generation of DPCM (difference picture) signals with motion compensation.

In addition to the picture difference from the $m \times n$ pixel block obtained by way of DPCM, the two motion vectors Δx and Δy for magnitude and direction are transmitted in short code words to the receiver (Fig 91), which uses the motion-compensated block for frame reconstruction. This technique considerably reduces the amount of data to be transmitted in the signal.

10.2 Irrelevancy reduction for the video signal

Irrelevant information is reduced by transformation coding where details of the picture are transmitted with reduced resolution as a result of frequency-responsive quantization. The video signal is subjected to a linear, reversible transformation. The coefficients of the transformed signal thus obtained are quantized and transmitted at a relatively low data rate. The favoured method of transformation coding of video signals is **discrete cosine transform (DCT)**.

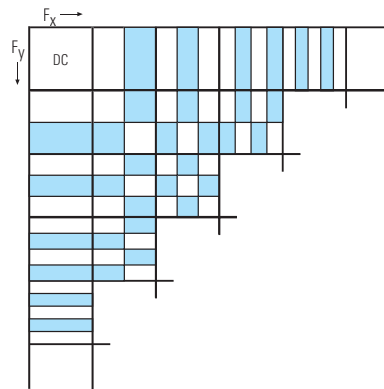


Fig 92
Section of DCT basic patterns.

The picture is divided into a number of blocks of $m \times n$ pixels. Each block is interpreted by means of a weighted superposition (summing) of basic pictures, which are transmitted as coefficients in coded form. Starting from the DC component, which corresponds to the mean brightness or luminance of the $m \times n$ pixel block, the structure of the basic pictures becomes finer the more the frequency increases in the vertical and horizontal direction. Fig 92 shows a section of the normally used 8×8 basic patterns with the limit value (black-white).

The two-dimensional DCT of a pixel block with subsequent quantization of transform coefficients for irrelevancy reduction is used as an example for describing the first stage of another reduction technique (Fig 93).

Here coding is effected in an 8×8 pixel block the luminance distribution $Y(x,y)$ of which is coded with 8 bits per pixel (value 0 to 255). The luminance distribution in the block is specified by a weighted superposition of the spectral coefficients $F(x,y)$ onto the 8×8 DCT basic function. In an intermediate step, the luminance distribution is referenced to the mean brightness (128). The DCT coefficients are

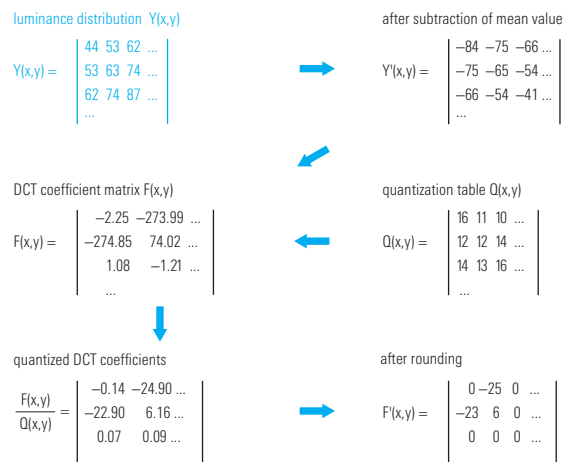
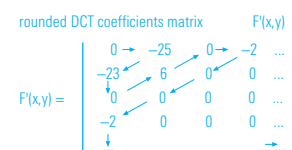


Fig 93
Transformation of luminance distribution $Y(x,y)$ of a block to the rounded DCT coefficient matrix $F(x,y)$ and zigzag readout of coefficients.



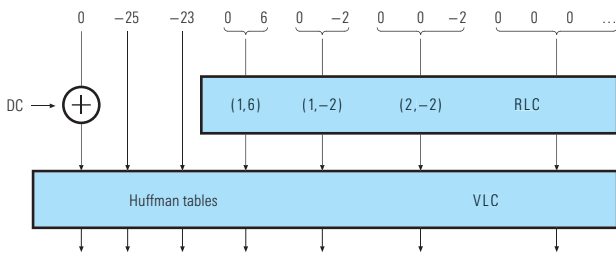


Fig 94
Redundancy reduction by run length coding (RLC) and variable length coding (VLC).

stored in a matrix. The coefficient in the top lefthand corner represents the DC component. The horizontal frequency coefficient F_x increases from left to right, the vertical frequency coefficient F_y from top to bottom. The higher the frequency, the lower the eye's sensitivity for fine structures.

For this reason a frequency-responsive quantization of DCT coefficients is performed using a quantization matrix $Q(x,y)$ with the result that the quantized coefficients become very small as the distance to the DC component increases in the horizontal, vertical and diagonal directions and are set to zero after rounding of the quantized coefficients. Thus matrix $F'(x,y)$ containing quantized and rounded DCT coefficients is obtained.

Starting from the DC component, the matrix $F'(x,y)$ is read out in zigzag fashion. The resulting sequence of coefficients may be used for further data reduction techniques.

The zigzag coefficient sequence is followed by two-digit **run length coding** (RLC), the first number indicating the length of the continuous string of zeroes preceding the value equalling the second number. Run length coding is completed when only zeroes are left in the string, which occurs soon with most of the pictures.

The value pairs thus obtained are then subjected to **variable length coding** (VLC) with the aid of Huffman tables, where frequently recurring value pairs are assigned a few bits only, and infrequent ones are transmitted with longer code words (Fig 94).

Discrete cosine transform with quantization and rounding, zigzag readout of coefficients as well as run length coding and variable length coding result in a data reduction of 10:1 without noticeable loss in picture quality. Differential pulse code modulation with motion compensation brings about a further reduction by a factor of 4:1.

The hybrid DCT coder normally used for data compression of video signals is a combination of DPCM with motion compensation using 16×16 pixel blocks and DCT using 8×8 pixel blocks. This is shown in a simplified block diagram in the next chapter.

10.3 MUSICAM for the audio signal

Although the source data rate of a stereo audio signal of CD quality is only about 1.4 Mbit/s and therefore of minor importance in comparison with the 100 times higher data rate of the video signal, the audio signal is also subjected to data reduction. The reason is that in sound broadcasting digital audio signals are transmitted in relatively narrowband transmission channels. An exception is **DAB** (digital audio broadcasting) where transmission is based on an OFDM signal (see chapter 12).

Digital audio signal transmission is widely used in the **ADR system** (ASTRA digital radio) where the digital audio signals are broadcast on subcarriers with 180 kHz

spacing above the spectrum of the analog video signal, or as a complete signal in the frequency band 0 to 8.5 MHz with frequency modulation in an analog satellite channel.

The idea behind source coding with **MUSICAM** (masking pattern universal subband integrated coding and multiplexing) is that certain components of the sound level cannot and need not be perceived by the human ear in order to identify the sound. Such components are not processed by the MUSICAM encoder.

This source coding method involves both redundancy and irrelevancy reduction. The result is a hardly perceivable loss in sound quality depending on the degree of data reduction and the sensitivity of the test person.

With MUSICAM, the digital source signal is generated with a data rate of 768 kbit/s using a 48 kHz sampling frequency and 16 bit coding and then spread to 32 subbands of 750 Hz width giving an audio frequency bandwidth of 24 kHz. The sampling frequency of each subband is only 1.5 kHz. Twelve successive sampling values of the subbands are combined to a block of 8 ms duration, for which the maximum signal level is measured and a scale factor with up to 64 steps is derived and coded with 6 bits. With volume classes of approx. 2 dB a dynamic range of about 120 dB is obtained. Since the scale factors rarely change for short periods of time, variable length coding is an appropriate technique to achieve further data reduction.

As can be seen from Fig 95, an exact analysis of the entire spectrum by means of FFT (fast Fourier transform) is performed in parallel to band splitting to define the

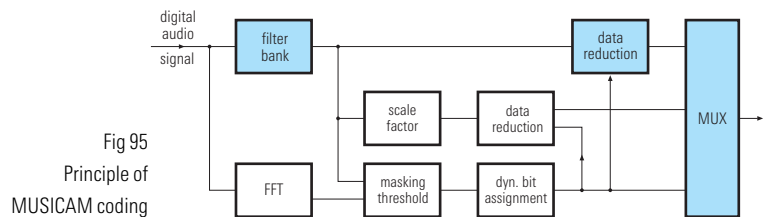


Fig 95
Principle of MUSICAM coding

masking threshold below which a tone is no longer perceived by the human ear. Quantization in the 32 subbands is controlled by a psycho-acoustic model simulating the masking characteristics of the human ear.

Through dynamic bit assignment, the constant output bit rate is split up into quantized signals in the subbands, scale factors, bit error correction and ancillary information, if any. The multiplexer com-

bines these substreams to generate an output data stream with a bit rate of 192 kbit/s, 128 kbit/s, 96 kbit/s or less per audio channel.

Other procedures will be described in the next chapter on audio coding to the MPEG2 standard.

The preceding chapters dealing with the digital video studio signal and data com-

pression techniques form the basis for the following description of **digital TV**.

Digital TV involves the emission and broadcasting of TV signals processed in digital form as a bit sequence in the studio and transmitted to the viewers with the aid of suitable carrier modulation methods. The next chapter therefore describes processing of the data stream applied to the transmission channel.

11 Video and audio coding to MPEG2 standard

In 1988, a working group was formed by the Joint Technical Committee (JTC) of the International Standards Organization (ISO) and the International Electrotechnical Commission (IEC) under the name

Moving Pictures Experts Group,
in short MPEG.

The aim of this group was to present a standard for coding methods and algorithms for video and audio signals. Standardization was to be implemented in several phases and has been concluded by now.

In the first phase, source coding for storing video and audio data on CD-ROM in multimedia systems was defined in the **MPEG1 standard**. The characteristic features of this standard are a maximum bit rate of 1.5 Mbit/s, frame processing and three quality classes for mono and stereo audio signals.

The MPEG2 standard is based on the MPEG1 standard but intended for standard TV signals with data rates from 2 Mbit/s to 15 Mbit/s and HDTV signals from 16 Mbit/s to 40 Mbit/s. In a special case an upper limit of 20 Mbit/s or even 100 Mbit/s is possible. The MPEG2 standard processes video signals in the field and frame format. Audio coding was extended to multitone with up to five channels.

The **MPEG3 standard** originally intended for HDTV signals does not exist because source coding of HDTV signals is covered by the MPEG2 standard.

The **MPEG4 standard** is being prepared at present with a view to defining very low bit rate coding in the kbit/s range for multimedia applications.

The main features of the MPEG2 standard relevant for digital TV are described below. In contrast to the digital studio standard, the MPEG1 and MPEG2 standards do not define parameter sets but tool boxes as defined by MPEG2 in

Systems	ISO/IEC 13818-1
Video	ISO/IEC 13818-2
Audio	ISO/IEC 13818-3

for creating optimized parameter sets for different applications. This ensures optimum compatibility between the different quality levels [10].

The **MPEG data stream** is characterized by two levels:

- The **system level** comprising bit organization, frame structure and other information required for demultiplexing the video and audio data stream and for synchronizing video and audio signals during reproduction.
- The **compression level** containing the compressed video and audio data stream.

11.1 Definition of profiles and levels in MPEG2 video

To allow a systematic classification of the great variety of video coding methods to MPEG2, criteria were defined for applications under **profiles**, and for quality under **levels**.

Definition of profiles:

- **Simple profile** (SP) uses coding to the 4:2:0 sampling scheme and simple prediction without motion compensation.
- **Main profile** (MP) also uses the 4:2:0 sampling scheme and is employed for most applications.
- **SNR scalable profile** (SNRP) uses scalability of quantization in the blocks. It permits the reproduction quality to be fixed depending on the bit error rate without a total failure of the picture.
- **Spatial scalable profile** (SSP) uses spatial scalability with the picture resolution being changed as a function of the transmission quality. SNR scalability is included.
- **High profile** (HP) uses coding to the 4:2:2 sampling scheme and complies with the most stringent demands but represents the most complex solution.

Levels	Maximum bit rate	Profiles				
		Simple	Main	SNR scalable	Spatially scalable	High
High	80 Mbit/s (100 Mbit/s)*		MP@HL			HP@HL (x)*
High 1440	60 Mbit/s (80 Mbit/s)*		MP@H14L		SSP@H14L	HP@H14L (x)*
Main	15 Mbit/s (20 Mbit/s)*	SP@ML	MP@ML	SNRP@ML		HP@ML (x)*
Low	4 Mbit/s		MP@LL	SNRP@LL		

Profile/level combinations for video; *): higher bit rates with double chrominance resolution.

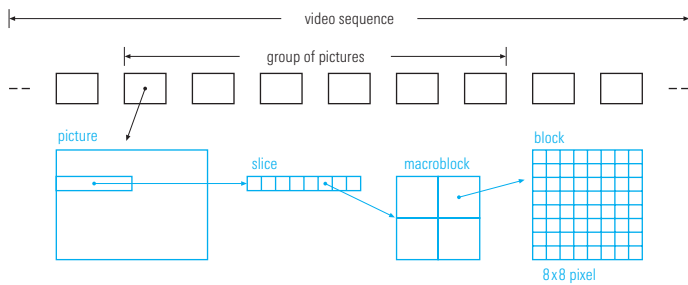


Fig 96
Layers of MPEG2 video data stream.

Levels define maximum parameters for picture resolution and refresh rate. The following levels are available:

- **Low level (LL)** for coding TV pictures with reduced resolution, similar to the MPEG1 SIF (standard image format) with 352×288 pixels and 25 frames/s or 352×240 pixels and 30 frames/s for luminance resolution.
- **Main level (ML)** for coding TV pictures with a standard resolution (SDTV) of 720×576 pixels and 25 frames/s or 720×480 pixels and 30 frames/s.
- **High-1440 level (H14L)** for coding HDTV pictures with 1440×1152 pixels and 25 frames/s or 1440×1080 pixels and 30 frames/s.
- **High level (HL)** especially for coding HDTV signals using a 16:9 aspect ratio with 1920×1152 pixels and 25 frames/s or 1920×1080 pixels and 30 frames/s.

The following useful profile/level combinations have been selected from 20 feasible combinations.

The high profile chrominance resolution can be doubled by way of the 4:2:2 sampling scheme (x). The values in brackets apply for the maximum bit rate.

Designations frequently used for specifying the quality levels of TV signals can be assigned to the following profile/level combinations:

- **LDTV** (low definition television): simple video and audio signal quality, with SP@ML, bit rate about 1.5 Mbit/s to 3 Mbit/s.

- **SDTV** (standard definition television): PAL quality of video signal, with MP@ML, bit rate about 3 Mbit/s to 6 Mbit/s.
- **EDTV** (enhanced definition television): approximately studio quality, with HP@ML, bit rate about 6 Mbit/s to 8 Mbit/s.
- **HDTV** (high definition television): high picture resolution, with HP@HL, bit rate about 20 Mbit/s to 30 Mbit/s [11].

11.2 Layers of video data stream

The main objective in the definition of coding algorithms is to keep the design of the decoder as simple as possible. In order to separate the basic elements of the coding procedure in the data stream and to allow suitable access mechanisms to become effective, a hierarchical structure in the form of six **layers** has been created for the **video data stream**. The assignment is shown in Fig 96.

- **Sequence layer** (sequence of video signals)
- **Group of pictures layer (GOP)**
- **Picture layer** (frame or field)
- **Slice layer** (slice)
- **Macroblock layer** (macroblock)
- **Block layer** (block)

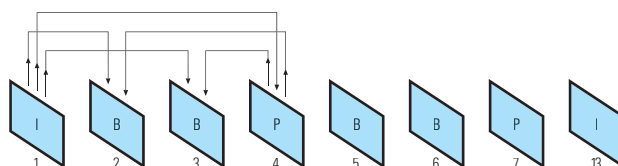


Fig 97
Sequence of intra-frame-coded pictures (I), forward predicted pictures (P) and bidirectional predicted pictures (B) in a group of pictures.

The **sequence layer** represents the highest level where the video signal sequence can be accessed, mostly for inserting programs. It contains information on the aspect ratio, the number of vertical and horizontal pixels (the number must be divisible by 16), the number of pictures displayed per second and the bit rate determining the size of the buffer memory required for decoding.

The **group of pictures layer** describes a group of frames taken from three different picture types:

- **Intra-frame coded pictures** (I frames) coded with DCT without prediction in the frame or field.
- **Forward predicted pictures** (P frames) predicted with reference to a previous intra-frame-coded picture with motion compensation.
- **Bidirectional predicted pictures** (B frames) transmitted with reference to a previous and subsequent frame.

To ensure that decoding starts after switch-on or after a transmission failure, the I frames have to be transmitted regularly approximately every 0.5 ms (after 12 frames).

P and B frames are sent in a defined sequence between the I frames (Fig 97). Compared to the I frame, the quantity of data transmitted is about one third in the P frame and about one ninth in the B frame.

The sequence of the frames differs from the constellation shown in Fig 97 because a P frame must be received after the I frame for the restoration of a B frame.

The **slice layer** is responsible for picture synchronization. Its header transmits a defined spot of the picture. A slice is a 16-line frame section containing a minimum of 16 pixels (i.e. a macroblock) or the full frame width when the main profile is used.

In the **macroblock layer** reference is made to the frame section used for motion compensation. A macroblock for the luminance signal contains four blocks and, depending on the sampling format used for the two chrominance signals C_B and C_R , two blocks in 4:2:2 format or one block in 4:2:0 format (Fig 98). This constellation has the advantage that motion estimation is only required for the luminance signal and that the motion vectors of the luminance components can be used for the chrominance signal.

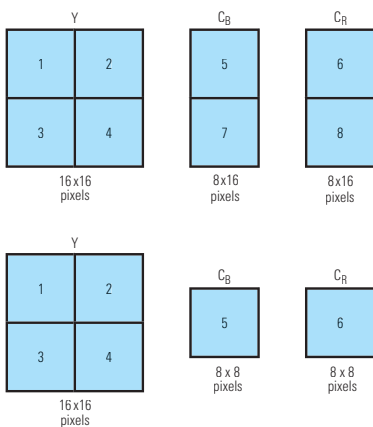


Fig 98 Block-to-macroblock assignment with 4:2:2 sampling format (top) and 4:2:0 sampling format (bottom).

In the **block layer**, the blocks (8×8 pixels) coded by means of DCT are transmitted. With I frames the information from the original picture is transmitted, with P and B frames the difference between frames only.

The operating principle of an MPEG2 video encoder is shown in Fig 99 [5]. The digital input signal (DSC 270 Mbit/s) is reduced to the coding format through filtering and decimation. Luminance and chrominance signals are processed in parallel although motion estimation is

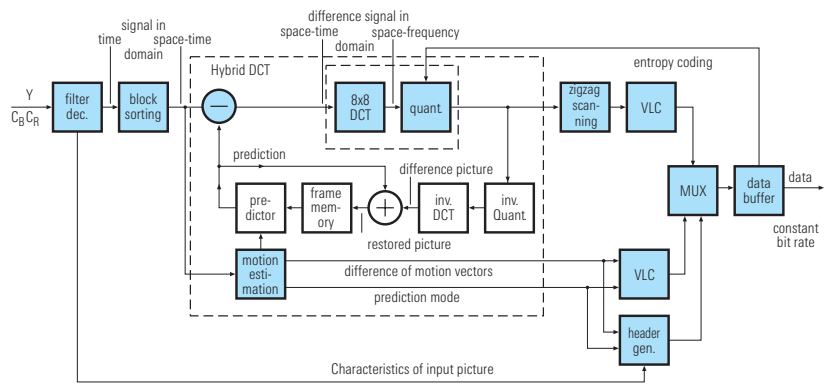


Fig 99 Principle of MPEG2 video encoder.

only performed for the luminance signal. Macroblock or block sorting is required because the necessary number of lines has to be present before a block or macroblock can be formed.

Hybrid DCT has already been described in section 10.2. The data obtained from DCT, the motion vectors and the prediction mode (I, P or B frame) depending on the picture content are forwarded to the data buffer memory via a multiplexer. Depending on the occupancy level of the buffer, a criterion is chosen for selecting the quantization used for the DCT. The buffer then outputs a constant bit rate.

In the MPEG2 decoder essentially the same procedure is performed in the reverse order. Motion estimation is not required in this case as the transmitted motion vectors are used for motion compensation.

The stages of data reduction for the video signal are as follows:

- DSC 270 Mbit/s \rightarrow 10 bit/pixel reduced to 8 bit/pixel \rightarrow 216 Mbit/s,
- 216 Mbit/s \rightarrow active video signal only \rightarrow 166 Mbit/s,
- 166 Mbit/s with 4:2:2 format \rightarrow 4:2:0 format with 125 Mbit/s,
- 125 Mbit/s \rightarrow DPCM with motion compensation, approx. factor 4:1,
- DCT with quantization, zigzag readout of coefficients and RLC, VLC, factor 12 \rightarrow up to 2.6 Mbit/s without a significant loss in signal quality.

The MPEG2 standard **Main Profile @ Main Level** was selected for the transmission of data-compressed video signals to viewers in line with the European DVB standard. Depending on the picture content, the resolution is selected to ensure that at least PAL quality is achieved. With the resolution being limited to 5.1 MHz, this yields 544 pixels horizontally. The bit rate to be transmitted for obtaining PAL quality is between about 4 Mbit/s and 5.2 Mbit/s [12].

11.3 Layers of audio data stream

Data compression in the audio signal is based on **MUSICAM** described in section 10.3. This technique has already been used by the MPEG1 standard for stereo audio signal coding.

Three layers have been defined for audio data compression which differ in the degree of compression and the complexity of the circuitry involved.

The data of a studio stereo audio signal with a source data rate of 2×768 kbit/s = 1536 kbit/s can be reduced in:

- Layer I** to 384 kbit/s, minimum outlay, used for digital compact cassette (DCC)
- Layer II** to 256 kbit/s, medium outlay, used for digital audio broadcasting (DAB), ASTRA digital radio (ADR)... and for MPEG2

Layer III to 128 kbit/s, maximum outlay.

The MPEG2 standard extends audio signal coding to multichannel audio transmissions, which is defined by **Layer II mc**.

The following coding methods are thus available for the audio signal with MPEG2:

- Single channel coding (mono signals)
- Dual channel coding (bilingual mono signals)
- Stereo coding (right, left signal)
- Joint stereo coding (stereo, high frequencies mono only)
- Multichannel coding (multichannel sound, 5 channels)

For compatibility with stereo coding, the

5 audio channels (L, C, R, LS, RS)

are matrixed, i.e. linearly combined to form the two

stereo signals L_0 and R_0

and the

three signals LS^W , C^W , RS^W

for transmission in the MPEG2 data stream [13,14].

11.4 Packetized program and transport stream

The video and the audio encoder generate an **elementary stream** each which, after packetizing, is available as a **packetized elementary stream (PES)**.

A PES packet of the video signal contains a data-compressed picture. PES packets may be of variable length, the maximum length being $2^{16}-1 = 65\,535$ bytes.

Each packet starts with a **header** of fixed length (6 bytes). The first three bytes signal the packet start. The next byte identifies whether video or audio information is transmitted and the remaining two bytes indicate the packet length.

A **program stream (PS)** is formed from the packetized video and audio data streams (**video PES and audio PES**) together with additional data, e.g. teletext, and **program-specific information (PSI)** by adding a common time base, the **program clock reference (PCR)**.

Time stamps are inserted into the sequence of PES packets as the **system clock reference (SCR)** required by the decoder for synchronization (Fig 100).

Since program stream packets are relatively long, they can only be transmitted in a largely error-free environment. Examples of such an environment are program production in the studio or storage media. The longer the packet, the more difficult the resynchronization and restoration of erroneous data.

A **transport stream (TS)** is formed for data transmission over long distances, e.g. for TV signals to be transmitted to the viewer. Since transport streams of up to 40 Mbit/s can be transmitted in a conventional TV channel using channel coding in line with the **digital video broadcasting (DVB)** standard, the transport stream contains several program streams. In order to fully utilize the permissible data rate, several transport streams of different program providers can be combined to form a **transport stream multiplex**.

Because of possible interference to be expected during transmission, the packets should be as short as possible. The transport stream therefore contains consecutive packets with a fixed length of **188 bytes**. A packet contains data of only **one signal component** (video, audio or other data). The **transport packet** starts with a 4 byte header leaving 184 bytes for transmitting information data (**payload**).

A transport stream length of 4 bytes + 184 bytes = 188 bytes was chosen because an ATM cell in global communication networks using the **asynchronous transfer mode (ATM)** contains 47 bytes (188 bytes = 4×47 bytes) and scrambling is based

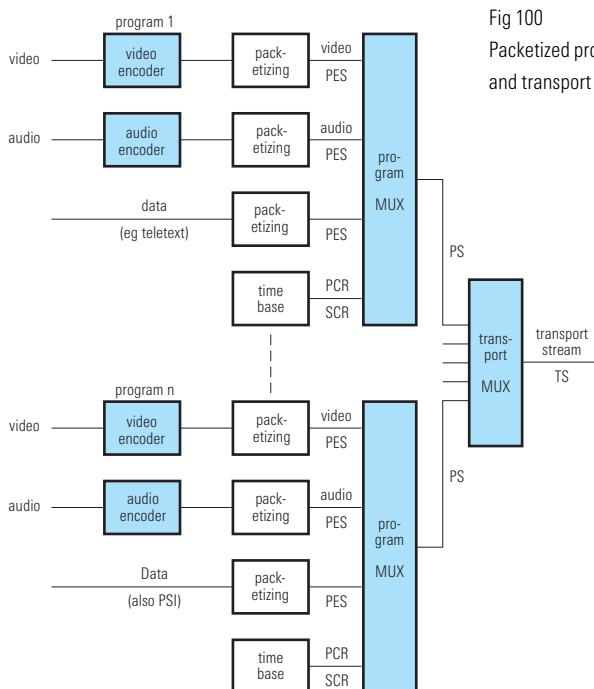


Fig 100
Packetized program (PS)
and transport (TS) data stream.

on an 8 byte sequence (184 bytes = 23×8 bytes).

For **program reproduction** in the receiver the individual elementary streams are first decoded and then synchronously output as **video and audio signals**. To allow synchronization, **time stamps** are transmitted in the transport stream. There are two types of time stamps:

- **Absolute time stamps** derived from the **system time clock** for synchronizing the system time base in the receiver are inserted in the program stream as **system clock reference** (SCR) and in the transport stream as

program clock reference (PCR) with reference to a specific program.

- **Relative time stamps** for the correct timing of decoded video and audio signals. They are inserted in the packetized elementary stream as **decoding time stamps** (DTS) and **presentation time stamps** (PTS). These stamps are indispensable because the decoding and reproduction time is not identical in the case of bidirectional prediction.

The system clock of the receiver delayed with respect to that of the transmitter due to the transmission channel can be restored from the transmitted **system time clock**. Delay time variations can be

compensated for by updating the time stamps, i.e. re-stamping [15].

12 Transmission of DVB signal

In **digital TV**, the terms MPEG2 and DVB are not always clearly distinguished. Processing of the digital transport stream as described above is in line with the

MPEG2 standard and defined worldwide by ISO and IEC in the standards

- ISO/IEC 13818-1 (multiplex),
- ISO/IEC 13818-2 (video coding) and
- ISO/IEC 13818-3 (audio coding)

With European **digital video broadcasting (DVB)**, the digital transport stream is transmitted in the TV broadcasting channels as stipulated by the guidelines of the European Telecommunications Standards Institute (ETSI) in the standards

- ETS 300 421, DVB-S, satellite transmission
- ETS 300 429, DVB-C, cable transmission
- ETS 300 744, DVB-T, terrestrial transmission

In all three cases, the channel used for signal transmission is affected by interference so that appropriate measures have to be taken for the transport stream to arrive at the MPEG2 decoder with a minimum of errors. Scrambling of data for pay TV will not be dealt with in the following.

12.1 Error correction

Due to the complexity of the transmitted data, one faulty bit in the transport stream may cause a complete picture failure. So, effective error correction methods are required in addition to channel-matched carrier modulation. The aim of error correction is to reduce picture failures to no more than one a day, which corresponds to a permissible BER of max. 10^{-11} .

The data stream transmitted in the channel should show an evenly distributed power density spectrum. To ensure this, periodically recurring bit patterns causing discrete spectral lines should be avoided.

The transport stream is therefore first subjected to **energy dispersal**. The data stream –without the **sync byte** at the beginning of each 188 byte packet –is linked bitwise to the data stream of a **pseudo-random binary sequence (PRBS) generator** using an exclusive OR element (Fig 101).

The PRBS generator is initialized after every eighth 188 byte packet with the aid of a fixed bit pattern. The beginning of every eighth 188 byte packet in the data stream is marked by an inversion of the currently present sync byte. This informs the receiver that the random data scrambling following the known pseudo-random bit pattern of the transmitter should be removed.

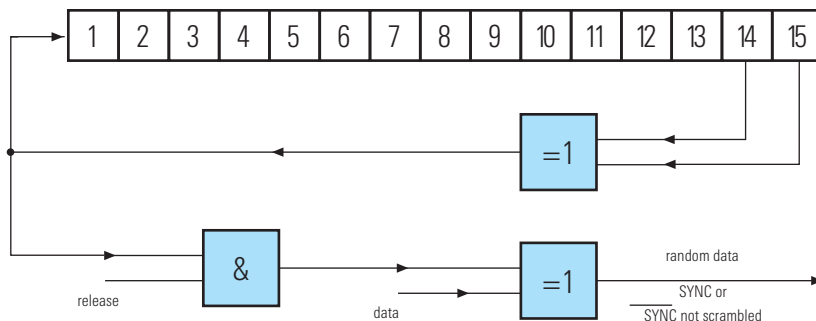


Fig 101 Scrambling of transport stream for energy dispersal.

After energy dispersal,

channel encoding

is performed in the form of concatenated error protection. To optimize error protection efficiency, **outer coding** based on the **Reed-Solomon block code (RS204, 188)** plus **inner coding** by means of **punctured convolutional coding** are used. The symbols obtained after outer coding are reorganized in an **interleaver** (Fig 102) before inner coding is performed. Interleaving allows long burst errors to be corrected in addition to single bit errors and short burst errors.

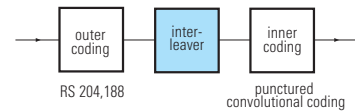


Fig 102 Concatenated error protection with interleaving.

The Reed-Solomon error correction code adds a sequence of 16 redundant bytes to each 188 byte transport packet so that the packet protected by the RS204, 188 code has a total length of 204 bytes (Fig 103). This allows correction of up to 8 faulty bytes in the packet.

SYNC	info data	redundancy
1 byte	187 byte	16 byte
188		
204		

Fig 103 Transport packet with Reed-Solomon error protection.

As specified by the DVB standard, Reed-Solomon encoding is followed by **inter-leaving** with a depth of $l = 12$. This means that the first bytes of 12 successive transport packets are transmitted in succession, then the second bytes and so

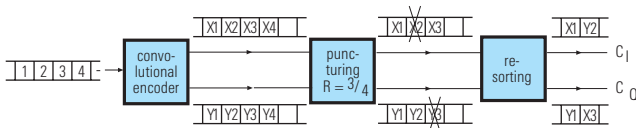


Fig 104
Convolutional coding with puncturing.

on. Neighbouring bytes of the original data stream are thus spaced by at least 205 bytes in the output data stream.

Convolutional coding is a powerful method of error correction but doubles the input data rate. A **code rate** of $R = 1/2$ is thus obtained. The code rate can be increased by way of **puncturing** but this reduces the error correction capabilities. Using a code rate of $R = 3/4$ as an example, Fig 104 shows the conversion of the serial input data stream to the parallel output streams required for further transmission by modulating a 0° and 90° carrier component.

Error correction in the receiver is performed with the aid of a **Viterby decoder**. To analyze the redundant data streams with the aid of the Viterby algorithm, a great variety of possible errors is calculated with the aid of a **Trellis diagram**. The result with the greatest probability is taken as the decoded value.

A bit error rate of less than 2×10^{-4} is required for error correction by means of the Reed-Solomon block code RS204,188. This means that the Viterby decoder has to reduce the maximum permissible BER of approx. 10^{-2} to 2×10^{-4} . After evaluation of the outer error correction code (RS204,188), a quasi error-free data signal with a BER of approx. 10^{-11} can be obtained.

12.2 Satellite channel

The transport stream containing error correction information is transmitted in the satellite channel with **4PSK modulation** which is optimal regarding interference and bandwidth. The DVB system uses absolute phase coding with direct

bit assignment to the vector positions because the absolute phase required for carrier recovery in the receiver can be set from the evaluation of the transmitted sync byte. The bit-vector mapping with reference to the in-phase component (I) and the quadrature component (Q) of the carrier signal in line with the Gray code takes into account that only one bit of the two-bit combination is changed between two vector positions (Fig 105).

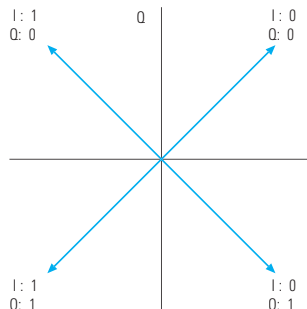


Fig 105
Vector diagram for 4PSK with Gray coding.

The data signals c_1 and c_0 obtained after convolutional coding and subsequent puncturing are pulse-shaped in the base-band filter which, in conjunction with the input and output bandpass filter in the satellite transponder, limits the spectrum of the 4PSK modulation product to a maximum permissible range. In most cases a transponder bandwidth (-3 dB) of 33 MHz is used as a reference. The baseband fil-

ter has a square-root raised cosine roll-off characteristic with a factor of $r = 0.35$. The Nyquist frequency is approx. 14 MHz.

The 4PSK modulation product is processed in an IF range at 70 MHz or 140 MHz. Fig 106 shows the block diagram of the transmitting end of a satellite channel in the 11 GHz or 12 GHz range.

The symbol rate of 27.5 Mbaud transmitted with 4PSK via a 33 MHz transponder corresponds to a bit rate of 55 Mbit/s. Taking into account the Reed-Solomon code (RS204,188) and the punctured convolutional code with an assumed code rate of $R = 3/4$, a useful net data rate of $R_U = 55 \text{ Mbit/s} \times 3/4 \times 188/204 = 38.015 \text{ Mbit/s}$ is obtained for the transport stream.

A carrier-to-noise ratio $(C/N) \geq 7$ dB is needed for a BER of $\leq 10^{-11}$ required at the demodulator input.

12.3 Cable channel

Transmission in the cable channel is fairly undisturbed compared to that in the satellite channel. This allows a higher carrier modulation level to be selected for transmitting about the same data stream as in the satellite channel at the narrower bandwidth of the cable channel. Elaborate error correction as provided by punctured convolutional coding is not needed either.

The data signal arriving as a DVB transport stream with outer coding based on the RS204,188 code and byte interleaving or coming from a satellite receiver station after evaluation and removal of the inner coding is subjected to byte-to-symbol mapping. With the **64-order quadrature modulation** (64QAM) used for the cable channel, the bytes are mapped into 6-bit

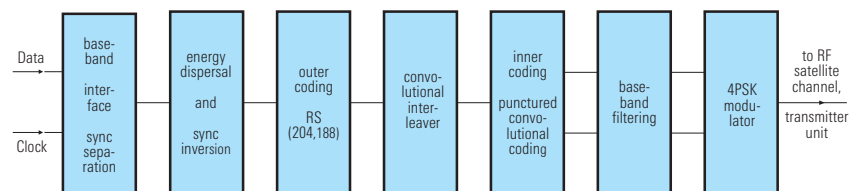


Fig 106
Block diagram of transmitter for satellite channel.

symbols. The bits are grouped into two differentially coded, most significant bits, and four least significant bits. Differential coding is used to simplify carrier recovery in the receiver. A reference carrier of correct frequency is obtained, the phase of which has to be referenced to the quadrants. The vector ends within the quadrants are assigned to the four least significant bits so that the same values are recovered on the I or Q axis when the axes are rotated by the absolute reference phase deviating from the transmitted carrier by multiples of 90° . Fig 107 shows a section of the 64QAM constellation diagram.

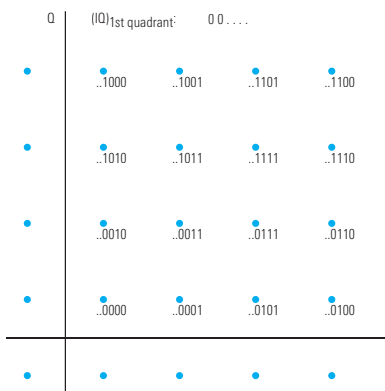


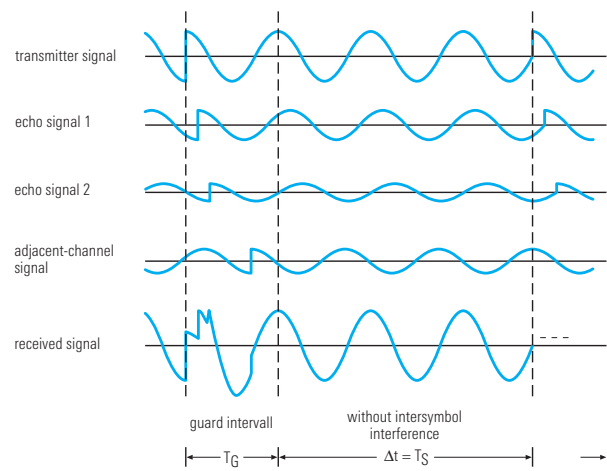
Fig 107 Section of 64QAM constellation diagram.

The 6-bit symbols are then assigned to the I and Q components in the constellation coordinates. Pulse shaping is performed by baseband filtering using a square-root raised cosine filter with a roll-off factor of $r = 0.15$. The spectrum of the 64QAM signals thus occupies a band of 7.935 MHz in an 8 MHz cable channel.

A bit rate of 41.4 Mbit/s can therefore be transmitted with 64QAM in the 8 MHz cable channel. Taking into account the RS204, 188 encoding, a net bit rate of 38.015 Mbit/s is obtained. Thus transport streams received from a satellite channel can also be transmitted in the 8 MHz cable channel to the viewer.

With 64-order quadrature amplitude modulation (64QAM), a carrier-to-noise

Fig 108 Superposition of original and multipath signals and evaluation of receive signal after guard interval.



ratio considerably higher than the 7 dB value for 4PSK is required at the demodulator input. It can be assumed however that the minimum C/N ratio of 28 dB necessary for obtaining an error-free picture with 64QAM is well met by cable distribution systems which must have a carrier-to-noise ratio of 45 dB for analog TV channels.

12.4 Terrestrial transmission channel

When TV signals are distributed via terrestrial transmitters, the complete coverage of the areas to be supplied is of foremost importance. This can be attained by setting up a sufficient number of transmitter stations. Using directional instead of omnidirectional receiving antennas improves the gain and reduces undesired reflections. Multipath reception producing ghost images and multiple image contours in the picture cannot always be avoided however. These echoing effects may be very disturbing in digital signal transmission.

Therefore, a **multicarrier transmission technique**, which has proved very successful in digital audio broadcasting (DAB), is used for DVB via terrestrial transmitters. This **OFDM** (orthogonal frequency division multiplex) **method** makes use of the 1705 or 6817 modulated carriers intended for DVB. It permits 4PSK, 16QAM or 64QAM modulation of the individual carriers. According to the DVB-T standard the data stream can be split up in high-priority and low-priority compo-

nents, which is the basis for **hierarchical modulation** (with 16QAM or 64QAM). This ensures that at least an acceptable image is obtained under unfavourable receiving conditions and the picture does not fail completely.

Multipath propagation is deliberately utilized by inserting a **guard interval** into the data stream for data evaluation to be started after a period that takes into account the delay of the reflected signals and allows the receiver to settle (Fig 108). Thus local or large-area **single frequency networks** (SFN) can also be implemented.

With **orthogonal frequency division multiplexing** (OFDM), the data signal is distributed to a large number of subcarriers. A low data rate with a symbol period of Δt is transmitted on each subcarrier. The spacing Δf of adjacent carrier frequencies is selected so that orthogonality is obtained as defined by

$$\Delta f = 1/\Delta t \quad (37)$$

The number N of subcarriers depends on the available channel bandwidth BW_{RF} as given by

$$N = BW_{RF}/\Delta f \quad (38)$$

A frequency-time constellation of subcarriers is thus obtained within the available channel bandwidth BW_{RF} (Fig 109).

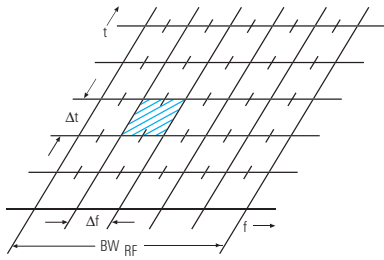


Fig 109
Frequency-time constellation with OFDM.

The maximum theoretical data rate is

$$r_{bit,max} = m \times N \times \Delta f \text{ bit/s} \quad (39)$$

depending on 2^m -level modulation.

With the modulation methods used for DVB-T (DVB, terrestrial transmission channel), the following values of m apply:

- 4-order modulation, 4PSK: $m = 2$
- 16-order modulation, 16QAM: $m = 4$
- 64-order modulation, 64QAM: $m = 6$

The permissible number N of subcarriers is determined using the procedure employed for OFDM by way of **discrete inverse Fourier transform** (DIFT) performed on the basis of $n \times k$ memory locations with digital signal processing. The following two modes can be selected for DVB-T:

- 2k mode with $2 \times 1024 = 2048$ memory locations
- 8k mode with $8 \times 1025 = 8192$ memory locations.

While in the case of **Fourier transform** a function is transformed from the time to the frequency domain, as is shown in the top part of Fig 110, **inverse Fourier transform** performs transformation from the frequency to the time domain (Fig 110, bottom).

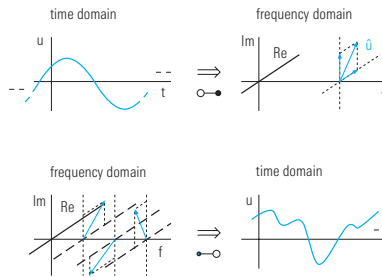


Fig 110
Principle of Fourier transform (top) and inverse Fourier transform (bottom).

To generate the OFDM signal, the data signal consisting of the parallel data streams c_1 and c_0 is successively assigned to N subcarriers by **mapping** of m bits at a time (2 bits with 4PSK, 4 bits with 16QAM, 6 bits with 64QAM), each subcarrier being described by a vector with real and imaginary component for the symbol period Δt . Within the period Δt , inverse Fourier transform is performed to yield a continuous time function. This function corresponds to the complex envelope of the OFDM signal in the baseband. The OFDM signal is then converted to the center frequency of an IF channel or directly to the RF. Fig 111 gives a simplified illustration of this procedure using 4PSK for the 1536 subcarriers commonly used in DAB, which are assigned data from a total of 2048 subcarriers (2k mode).

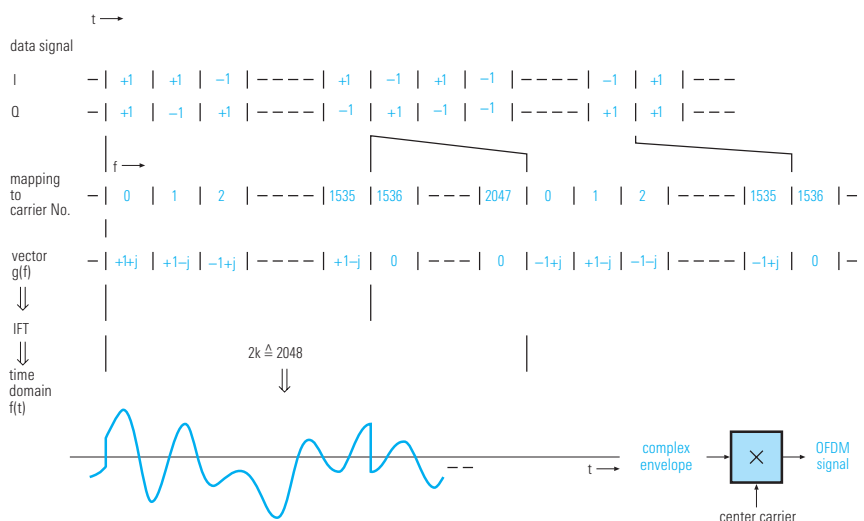


Fig 111
Mapping of I/Q data signals to vector positions of individual carriers in frequency domain and inverse Fourier transform to complex envelope of time domain with OFDM.

In the receiver, the time function is retransformed to the frequency domain by means of **discrete Fourier transform** (DFT) with assignment of data to the carrier positions and reading out of the real and imaginary components of the individual positions.

DVB-T uses 1705 of a total of 2048 carriers in the 2k mode and 6817 of 8192 carriers in the 8k mode. 1512 (2k) or 6048 (8k) carriers contain information data, and the remaining carriers transmit reference signals with information known to the receiver or are used for transmitting signalling or sync patterns.

The symbol period T_S in the continuously transmitted information data stream is extended by the guard interval T_G . This however reduces the transmitted information data rate. Depending on the maximum permissible distance between the transmitters of a single frequency network, the guard interval may have a length of $1/4$, $1/8$, $1/16$ or $1/32$ of the symbol period T_S . A guard interval of $1/4$ of a symbol period is required for the largest distance between transmitters. The total symbol period

$$T_{total} = T_S + T_G \quad (40)$$

is thus extended to 1.25 times the symbol period and the effective transmitted data rate reduced to 0.8 times the value determined by the symbol period.

Digital signal processing in OFDM signal generation is based on a sampling clock with the period $T = 7/64 \mu\text{s}$, which corresponds to a sampling frequency of approx. 9.14 MHz. Thus the useful symbol period is

$$T_S = 2048 \times 7/64 \mu\text{s} = 224 \mu\text{s} \text{ in the 2k mode and}$$

$$T_S = 8192 \times 7/64 \mu\text{s} = 896 \mu\text{s} \text{ in the 8k mode.}$$

Based on orthogonality, the carrier spacing is therefore

$$\Delta f = 1/224 \mu\text{s} = 4.464... \text{ kHz in the 2k mode or}$$

$$\Delta f = 1/896 \mu\text{s} = 1.116... \text{ kHz in the 8k mode.}$$

A total symbol duration of $T_{\text{total}} = 896 \mu\text{s} + 224 \mu\text{s} = 1120 \mu\text{s}$ is obtained in the 8k mode to be introduced in Germany, which uses a maximum transmitter distance of approx. 67 km and a guard interval of $T_G = 1/4 \times 896 \mu\text{s} = 224 \mu\text{s}$. The total bit rate of the transport stream to be transmitted on the 6048 carriers used for information data is therefore

$$r_{\text{bit}} = 6048 \times 4 \text{ bit} / 1120 \mu\text{s} = 21.6 \text{ Mbit/s for 16QAM}$$

and

$$r_{\text{bit}} = 6048 \times 6 \text{ bit} / 1120 \mu\text{s} = 32.4 \text{ Mbit/s for 64QAM.}$$

The total RF bandwidth occupied by the OFDM signal up to the first null positions outside the main spectrum as given by

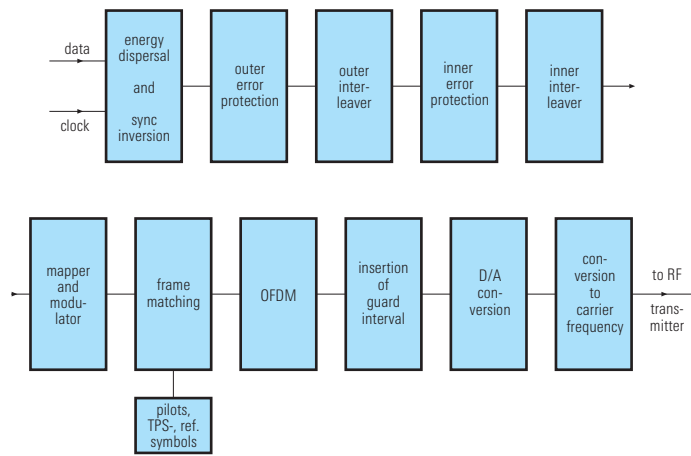
$$1705 \times 4.644... \text{ kHz} = 7.611 \text{ MHz for the 2k mode,}$$

and by

$$6817 \times 1.116... \text{ kHz} = 7.608 \text{ MHz for the 8k mode}$$

is approx. 7.61 MHz and thus lies within the 8 MHz TV channel.

Fig 112
Block diagram of transmitter for DVB-T using non-hierarchical modulation.



As shown in the simplified block diagram of Fig 112, processing of the DVB-T data signal is at first the same as with satellite transmission (DVB-S), i.e. energy dispersal, outer error protection based on RS204,188 code, interleaver and error protection by punctured convolutional coding. With DVB-T an inner interleaver using combined bit and symbol interleaving follows for distributing longer errors of individual or adjacent carriers across the whole data stream so that errors are already corrected by the bitwise inner coding of the Viterby decoder [16]. The data stream of the I and Q channels is mapped to the vector position of the carriers transmitting the information signal. Reference and sync signals are assigned to the pilot carriers, and the total number of carriers used is then converted into a complex time function by means of OFDM. Each symbol with the duration T_S is extended by a guard interval T_G . Finally digital/analog conversion and conversion to the center carrier frequency are carried out.

The defined mixed distribution of data to the individual carriers by means of inner interleaving changes OFDM to **COFDM** (coded orthogonal frequency division multiplex).

The aim of the DVB system is a high transparency of data stream transmission from the satellite channel to the cable channel or to the terrestrial broadcasting channel. Cable channel transparency requirements are fully met as the net information data stream of 38 Mbit/s from the satellite can be fully fed into the cable channel. This is not quite so with the terrestrial 8 MHz channel where, given the same total data rate, the transmitted net data rate is lower than in the 8 MHz cable channel as a result of inner error protection, guard interval and because only 6048 of the 6817 carriers are used. It can be assumed that a data rate of around 24 Mbit/s is transmitted in the terrestrial 8 MHz channel.

This data rate is sufficient for the simultaneous transmission of 4 to 8 SDTV programs of PAL quality or 3 EDTV programs of PALplus, i.e. of almost studio quality [17], in comparison with a single analog program.

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Extracts from the following book were used for chapters 1 to 7 of this brochure:

- [18] Mäusl, R.: Fernsehtechnik - Von der Kamera zum Bildschirm. Hüthig Buch Verlag, Heidelberg, 1991



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