Digital Transmission of Two Television Sound Channels in Horizontal Blanking

By MANFRED MAEGELE

The conventional way of transmitting television sound information on subcarriers leads to a number of difficulties in satellite communications systems using frequency modulation, because of the limited power available. In particular, when high-quality sound is required, modulation products can only be avoided by going to large bandwidths and high power compared with the actual picture transmission — and this is expensive. A method has been found for transmitting two high-quality (15-kHz bandwidth and better than 60-dB SNR) sound signals with no mutual interference between sound and video signals. No additional RF bandwidth and little additional power is required. The technique involves using a part of the line-blanking interval after shortening the synchronizing pulse. The sound signals are sampled and companded to a given number of bits in a non-return-to-zero code. At the receiving end the pulses are stored, decoded and read out with the original sampling frequency. After passing through a low-pass filter, the signals are available in the original position. Tests have been promising, and a practical application seems possible soon.

Introduction

The generally known television systems provide separate transmission links for video and sound signals. Although this ensures a satisfactory service for most televiewers, the further expansion of television coverage faces, in addition to a number of organizational and economic questions, the following technical problems:

(1) In the case of separate transmission paths, the sound information belonging to the video signal requires additional bandwidth for the connection from the studio to the transmitter whether on radio-relay links or on satellite links.

(2) The additional transmitter power required for separate sound transmission is of great disadvantage, particularly for satellite systems.

(3) The number of television channels offered to the televiewers is on the increase, but the available frequency bands are already very densely occupied.

(4) The transmission of programs in several languages, or of television programs with accompanying stereophonic sound, necessitates several sound channels.

The existing systems cannot provide the bandwidth that is needed to satisfy all mentioned requirements. For all service networks at a planning stage, there must therefore be an attempt — by integration of the sound channels into the video signal — to reduce the demand for additional bandwidth by 15 to 20% and that for increased transmitter power by up to 40% when direct-broadcast satellites are concerned. An additional point in favor of integrated sound channels is that digital-coding circuits for the sound signals replace complex analog filters for separating picture and sound.

Presented on 14 November 1974 at the Society's Technical Conference in Toronto by Manfred Maegele, Fernmeldetechn. Zentralamt der DBP, D-61 Darmstadt, Am Kavalleriesand 3, Germany. (This paper was first received on 17 June 1974 and in final form 25 September 1974.)

In a contract awarded by the Federal Ministry of Research and Technology and by the Federal Ministry of Posts and Telecommunications the SEL Research Center has investigated the possibilities and methods of integrating the transmission of sound signals into the video signal. The result of these investigations is given in the following sections.

Possible Methods

As a consequence of the line interlacing method which is used to avoid flicker, the television picture (Standards B and G) is subdivided into two fields being arranged one above the other. This video signal comprises time sections during which picture information is transmitted and time sections needed for synchronizing and color reference signals, as well as for national and international test signals. In the field of 20-ms duration (Standards B and G) the picture flyback (vertical blanking interval) takes 1.6 ms. The line period is 64 μs, with 12 μs of it being consumed by the line flyback (horizontal blanking interval).

In the picture flyback up to 10 lines can be used for the integration of sound signals. This corresponds to a relative duration of 3% referred to the total picture duration. For an integration of sound signals into the line flyback, the horizontal synchronizing pulse with a duration 4.7 μs and the back porch with 5.6 μs are available in Standards B and G. By suitable modification of these intervals, a maximum of 7.5 μs can be utilized for sound transmission, which corresponds to 12% of the line period of 64 μs. An integration of the sound signals into the line flyback thus offers the following advantages compared to an integration into the picture flyback:

(1) longer periods per picture available for the transmission of sound information so that simple and clear coding methods can be used;

(2) shorter storage time of the sound sampling values (64 μs instead of 20 ms); and

(3) required storage capacity lower by about 300 times.

Both in the picture flyback and in the line flyback the time sections mentioned are already partly used for characteristic signals of the PAL, NTSC or SECAM systems. Thus the investigations of the sound transmission methods had also to include substitute signals or signal offsets which might possibly be required, as well as the restoration of the standard video signal in the receiver.

The application of integrated sound transmission has already been investigated by several P.T.T. administrations and broadcasting companies. All methods known so far either allow the transmission of only one sound channel, or do not meet all requirements on the sound quality, or are not compatible with the three color television systems.

At the present time, the main interest is focused on the integrated transmission of two high-quality sound channels with a bandwidth of 15 kHz each, a distortion factor of less than 1%, crosstalk attenuation of more than 70 dB and a weighted SNR (signal-to-noise ratio) of over 60 dB. The required SNR of the sound channel cannot be obtained with the usual analog-modulation methods if that of the video channel, according to long-distance specifications, is only 52 dB. The best SNR achieved with the analog methods, pulse-phase modulation (PPM) and pulse-duration modulation (PDM) which are most suitable for sound integration, is 56 dB in the sound channel.

The required SNR can be obtained much more easily with a digital-modulation method: pulse-code modulation (PCM). The sampling frequency for a signal with a bandwidth of 15 kHz must be at least 30 kHz. In television systems the recommendation is to take double the line frequency, e.g. 31.25 kHz for the 625-lines standard. The specification of the high-quality sound channels calls for a high resolution of the analog/digital converters. According to investigations performed by the Fernmeldetechnisches Zentralamt (FTZ) of the Deutsche Bundespost, which have been made known to the CCITT, A/D converters with linear quantizing with 14 bits should be chosen in order to guarantee the required sound quality in the wanted dynamic range. On the transmission link a compandor system can reduce the 14-bit words to 10-bit words when preemphasis and deemphasis are used at the same time.

Our considerations are thus confined to a digital-modulation method which transmits the sound signals during the line flyback. Figure 1 shows the standard

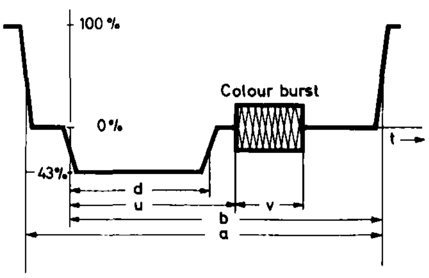

Fig. 1. Blanking interval in a PAL-System, 625 lines; (d = 4.7 μs; μ = 5.6 μs; v = 2.25 μs; b = 10.5 μs; and a = 12.0 μs).

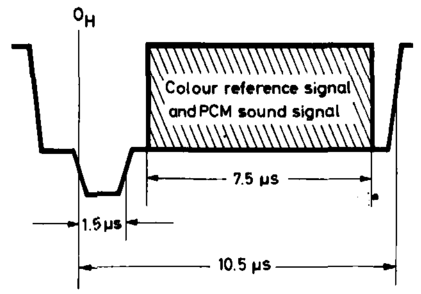

Fig. 2. Blanking interval, modified for integrated sound transmission.

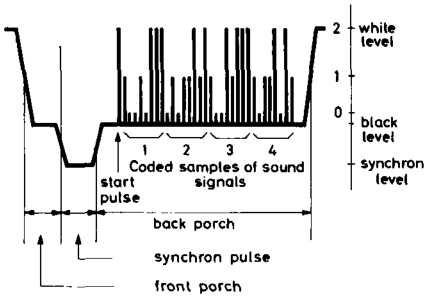

Fig. 3. Integration of ternary R Z pulses.

horizontal blanking interval for Standards B and G with the PAL color burst. Whereas the NTSC signal contains a comparable burst, that in the SECAM signal is modified. The limited time interval in the line flyback requires, for the integrated PCM sound pulses, as high a repetition frequency as possible within the band limitation of the video signal. In the color video signal, there already exists the color subcarrier frequency which for PAL is 4.43 MHz and is close to the band limit of 5 MHz.

The transmission time required when two sound channels are integrated into the video signal can be calculated if the sampling frequency, the resolution of the sampling values and the applied coding method are taken into account. When color subcarrier frequency and line frequency, as well as 14-bit resolution are used, the components of two binary-coded sound channels, which are assigned to one line period, take a transmission time of 12.7 μs. After companding to the permissible minimum of 10 pulses per sample there still remain 9.1 μs. This means that binary coding of the sound pulses is impossible, especially since a start pulse of 0.23 μs for controlling the digital units in the receiver has to be added. These periods can be shortened by ternary coding of the sound samples, or by a binary non-return-to-zero (NRZ) code. Nevertheless, the standard and unmodified video signal does not have a time section which would be long enough for the integrated transmission of two sound channels.

If, however, the horizontal blanking interval is changed as shown in Fig. 2, it is possible to integrate the coded sound signals into the video signal. With the horizontal synchronizing pulse being reduced to 1.5 μs, line synchronization and clamping circuits can still be operated without faults. The lengthened back porch of 8.5 μs allows the transmission of sound pulse packages which may be either ternary R/Z or binary NRZ coded.

Development and Optimization of a Method

On a model with two ternary coded PCM channels, which was set up for the PAL system, the feasibility of the method was proved. Figure 3 shows how the sound signals in the line flyback are arranged. The characteristic features of the model are:

(1) two sound samples per channel and line;

(2) source encoding 12 bits linear without preemphasis;

(3) conversion of the 12-bit words into 8 ternary coded pulses;

(4) pulse repetition frequency equal to color subcarrier frequency with correct phase position of the burst;

(5) amplitudes of the ternary pulses at 100%, 50% and 10% of white level;

(6) auxiliary pulses in the vertical blanking interval for continuous sound transmission; and

(7) regeneration of the color burst at the receiver end from sound pulses by limiting the sound pulse packages to the 10% amplitude level and by subsequent amplifying and blanking.

The model was tested on different transmission links, particularly on modems for radio-relay links. In each case attenuation and delay on the simulated links were varied in order to measure the transmission characteristics. The received sound signals have satisfied all requirements made with regard to sound quality and channel separation. The dynamic range reached the theoretically determined limits of a linear resolution of the sound samples with 12 bits. It is easily possible to extend the model to the mentioned resolution of 14 bits with companding to 10 bits and thus to increase the SNR to 69 dB within the desired dynamic range and reduce the distortion factor to 0.3%. Any impairments caused by crosstalk from the sound channels into the picture and vice versa were not perceptible.

Although the investigations of the described method yielded good results, the requirements to be met by an optimum television system with two integrated high-quality sound channels were not satisfied in all respects. Here we must, above all, mention the applicability to all video systems, the complexity of the regenerative circuits and the independence between sound and color information. In consideration of these arguments, the sound transmission system described above was further developed in

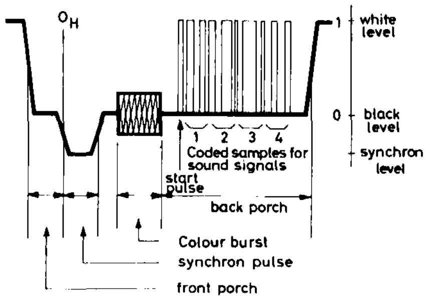

Fig. 4. Integration of binary NR Z pulses.

the SEL Research Center, meanwhile submitted to the CCIR for discussion within the scope of a Study Program and, at the Geneva meeting in the spring of 1974, proposed for being included in CCIR Report No. 488. Figure 4 illustrates in which form and in which time sequence the sound signals are written into the horizontal blanking interval of the video signal. The principal differences compared with the method described above are:

(1) source encoding 14 bits linear;

(2) digital companding from 14 bits to 10 bits by means of the 13-segment companding law with CCITT preemphasis;

(3) transmission of 10-bit words with NRZ pulses;

(4) binary transmission code;

(5) pulse repetition frequency equal to double color subcarrier frequency (auxiliary frequency used in the system only);

(6) transmission time of 4.7 μs on extended back porch of the video signal;

(7) pulse amplitudes with 0% and 70 to 100% of white level;

(8) separation of color burst and sound signals; and

(9) color burst shortened by two cycles and shifted within the extended back porch.

An integrated sound transmission according to the outlined method offers the following advantages without impairment of the picture and sound quality:

(1) applicability to all television standards;

(2) saving of bandwidth in the case of satellite transmissions, radio-relay links and cable television networks;

(3) possibility of transmitting two high-quality sound channels (or several of lesser quality);

(4) one transmission link for picture and sound;

(5) considerable reduction of transmitter power, particularly in the case of satellite systems; and

(6) applicability to PAL, NTSC and SECAM color television systems by slight modifications.

In view of the theoretical and practical experience gained with integrated sound transmission systems, the use of the chosen method on transmission links appears to be possible in the very near future because no change of the equipment or the television standard will be required. A modification of the standard will be necessary when the method is extended to television systems including the home receiver. This would require a longer period with interim solutions for the simultaneous reception of integrated and nonintegrated sound signals.

Discussion

Robert J. Butler (NBC TV): I very much am in favor of integrating sound with the television picture. However, I question the quality when taking only two samples of the highest frequency of interest. Our studies indicate that the quality of signals that you can achieve by taking only two samples of the highest frequency would be objectionable. How does your experience bear on this?

Mr. Maegele: As outlined in the paper we have tested the system in the first method described and the results were very good. I must say that all the requirements were fulfilled in all respects on quality.

Mr. Butler: Were these subjective measurements that you made?

Mr. Maegele: These were objective measurements, some of them being automatic measurements.

Mr. Butler: Did you get any figure on distortion of signals as a result of low sampling rate?

Mr. Maegele: Yes I have some graphs with me which I can show you afterwards if you like. All the results were within the limits.

Charles W. Rhodes (Tektronix, Inc.): Would you comment on the rise time of these pulses and the energy outside of the video bandpass generated by these rather fast narrow pulses used? Have you tested the applicability over satellite channels with this bandwidth strictly limited?

Also, what is the means provided for field sync in the system so that no field frequency interruption in audio information arises?

Mr. Maegele: Yes, this is a real problem and we have noticed on tests via satellite link that we must somewhat reduce the amplitudes of the pulses; otherwise too much energy is lost outside the video bandwidth. The second problem is solved by inserting three additional pulses in the vertical blanking.

Andrew Kufluk (Ryerson Institute): In view of the fairly high level white going signals during the tail end of the vertical blanking, have you noticed any effect on the monitor or the receiver screen towards the left of the screen in the form of vertical bar pattern during retrace? I am thinking that the blanking period isn't always black and it does vary between white and black during the encoded time. Therefore, there is the possibility during the retrace time that the pattern will show as or at least the coding will show as a vertical pattern of stripes towards the lefthand of the screen as you look at the monitor or to the receiver. Has that been a factor or noticeable?

Mr. Maegele: Yes, this is a problem of clamping circuits. You must always have the same reference level in order to have a good relation between black and white. This is a problem of restoration of the standard blanking by clamping.

James D. Kirklin (Western Michigan University): My impression of what you are proposing is that for use on a network system once the program is received by a cable system or a local television station this blanking area would be restored to its normal state and the pictures seen on a home receiver would no longer contain this information. Therefore, this encoded information would not be a problem if you had poor blanking in the receiver. It wouldn't show on the retrace, would it?

Mr. Maegele: That is right.

The Development and Application of Metal Halide Lamps for Color Filming and Television

By ROBIN C. ALDWORTH

The development of high-pressure discharge lamps over the years has been aimed at improving color rendering, efficiency and life. The compact-source iodide (CSI) lamp was initially developed in 400-W and 1,000-W ratings for projector applications, but it was the new demands of outside broadcast color television in 1969 that led to the introduction of this lamp housed in a PAR-64 sealed-beam globe for floodlighting sports stadia from high corner towers. Sports areas with sidelighting were not so well suited to the use of the symmetrical beam CSI floodlights, so another metal halide lamp was introduced — this one with an unjacketed linear arc tube — in 750-W and 1,600-W ratings. For filming, the 1-kW lamp performs exceptionally well as a fill-in source with daylight. At 90 lm/W, it is five to six times more efficient than tungsten-halogen lamps used with blue filters to correct the daylight; four lamps are approximately equivalent to a 225-A brute arc. Control gear is being made simpler and more versatile, and the problem of "beat," which is already controllable in filming situations, is expected to be overcome completely with techniques now being developed.

Filament Lamp Development

Light source development over the years has had three main objectives — to improve efficiency, life and color rendering. For lighting units which are to provide a controlled beam, compact high-brightness sources are also necessary. Beyond pure lighting requirements, lamps also need to be robust, simple to replace and, of course, economical. It is evident that all these aims have influenced the development of filament lamps over the years. The color rendering of these lamps for filming has never been a problem because the spectral energy distribution is continuous; the film manufacturers were able to produce film stock to match it, but the early lamps were certainly neither robust nor compact. Over the years the introduction of single coil, coiled coil, bunched and low voltage filaments all added significant improvements, and recently the tungsten-halogen lamp in linear and compact forms has brought dramatic improvements in efficiency, lumen maintenance and life. The 10-kW studio lamp, giving 28 to 34 lm/W, with lives of 400 to 150 hours at around 3200 K, represents the culmination of 100 years of filament lamp development.

Discharge Lamp Development

Most discharge lamps, on the other hand, start with a distinct advantage in terms of efficiency—the amount of light they produce per watt of energy consumed. Plain mercury lamps operate at 50 lm/W with lives in excess of 7,000 h, but they are relatively large, low brightness sources and their color rendering is poor. Compact versions of mercury lamps have also been developed giving similar efficiencies, but attempts to improve color rendering by increasing current loading and adding substances such as cadmium and zinc to the arc tube filling have been of limited success.

Xenon lamps, in the familiar double-ended construction, can provide a very compact source which aids optical design and the color quality is very good. Lives of 1,000 h are common, but the efficiency is lower (30–35 lm/W and the lamp is expensive to make, fragile in operation and complicated to mount in lighting units. All of these lamps have found applications in specialized fields, but their various shortcomings have prevented their general acceptance in the spheres of film and television.

Compared with filament lamps, all discharge lamps have the disadvantage of requiring control gear. Nevertheless the film and television industries have from the beginning accepted grids and dc generators to run their carbon arcs, so a few items of control gear may not be regarded as an impossible burden if there are sufficient other benefits. Such benefits are now recognized as a practical possibility.

Presented on 12 November 1974 at the Society's 116th Technical Conference in Toronto by Robin C. Aldworth, Thorn Lighting Ltd., Thorn House, Upper St. Martin's Lane, London WC2H 9ED, England. (This paper, first received on 7 October 1974, was received in final form on 20 January 1975.)