TECHNICAL INSTRUCTION

P.15

SOUND IN SYNCS

CONTENTS

SOUND IN SYNCS

Introduction

The distribution of BBC television programmes from studio centres to transmitting stations involves a network of cable and microwave links which are leased from the British Post Office. In the past the sound and video components of the programmes have been routed over separate circuits and have often followed different routes and used different media. The sound-in-syncs system combines the sound and video signals by sending the sound information in pulse-code-modulated (see Instruction GP.2) form during the line synchronising periods of the video waveform and thus enables both sound and video signals to be routed via the same link.

The sound-in-syncs system is a time-division multiplex system in which the video circuit is made available to the associated sound signal during line-sync periods. The sound signal is sampled at twice line frequency and the samples are converted into 10-digit pulse-code modulation (p.c.m.) signals. Alternate p.c.m. samples are stored for half a line and the samples are then inserted, two at a time, into the following line-sync pulse.

The complete sound-in-syncs system is shown in simplified block diagram form in Fig. 1. The sound input signal is compressed in volume, sampled at twice line frequency and then coded in an analogue-to-digital converter. From the converter the coded signal is applied, together with the video signal, to a combiner unit in which the video signal is clamped at the bottom of syncs and coded sound signals are inserted in the sync pulses. The sync pulse leading edges are preserved because they are the main timing references of the video signal.

At the receiving terminal the procedure is reversed. The sound and video components of the combined signal are separated and the sound pulses are decoded in a digital-to-analogue converter. The audio-frequency output of the converter is then applied to a volume expander in which the original sound signal is reproduced.

Any fault in the distribution circuit which causes a loss of video will also remove the sound. However, should the video signal fail prior to the distribution circuit, a standby feed of line-sync or line-drive pulses is provided automatically at the combiner unit so that sound signals can still be sent through the link.

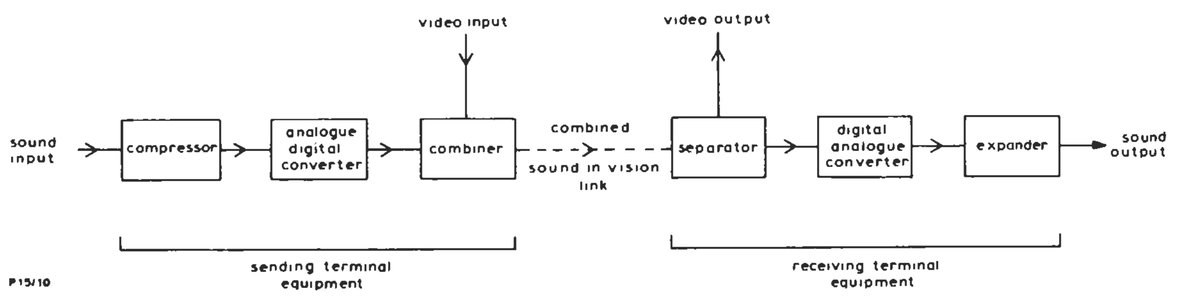

Fig. 1. Simplified Block Diagram of Sound-in-syncs System

The system provides a means of transmitting both sound and video signals over a single distribution network, so economising on lines, and also reducing the possibility of operational errors because the sound cannot become separated from its associated video between terminal points. Overall reliability is greater than that of systems where the sound signal is transmitted via a separate distribution network, because the failure of video links is comparatively rare and solid-state circuitry ensures high reliability of the sound-in-syncs terminal equipment.

The insertion and separation of the coded sound pulses is made with reference to the leading edge of syncs and, because sound information is carried only in the sync-pulse period, the transmission of the sound pulses is independent of the amplitude of the video signal. Noise and distortion on the video link, unless of such magnitude as to produce serious degradation of picture quality, has no effect on the quality of the sound signal.

Sampling and Quantising

Sampling

Converting an analogue signal to a series of pulse-amplitude-modulated samples requires regular sampling at a frequency at least twice that of the highest frequency present in the original signal. For the sound-in-syncs system sampling is carried out at twice line frequency and (on the 625-line standard) this gives a sampling frequency of 31·25 kHz which, after allowing for the imperfections of practical filters, provides an audio bandwidth of about 14 kHz.

Quantising

Samples of analogue signals are taken only at discrete intervals of time, but their amplitudes can have any value within a wide range. Each sample is given a discrete amplitude by comparing it with a scale which is graduated into a finite number of levels and specifying the sample by the interval on the scale into which it falls. The process is known as quantising and the scale intervals are called quantising levels. The sound-in-syncs system uses 10 binary digits per sample and thus provides 2^{10} quantising levels.

Quantising Noise

If the sample amplitude falls between two quantising levels there is an error in the coded signal. This error, the difference between the sample amplitude and the quantising level below it, gives rise to noise which is known as quantising noise. The greater the number of levels used the smaller the quantising error and the lower the level of quantising noise. The coded output signal can be regarded as a perfect representation of the input signal plus quantising noise. If the number of quantising levels used is large (the sound-in-syncs system provides 2^{10} = 1024 quantising levels) the audible effect of quantising noise on a sound programme resembles that of white noise. For further information on quantising noise see Instruction GP.2: *Pulse Code Modulation.*

However the mean voltage level of the combined pulse group does vary with the sound modulation and, if the distribution circuit has low-frequency attenuation distortion, these mean-level variations can be impressed on the back porch of the video waveform. Subsequent clamping operations transfer these variations to the picture period and produce an objectional sound-on-vision effect. To alleviate this the digits are arranged in reverse order so that the least significant digit appears first, instead of the most significant. Thus those digits that are subject to the most frequent change are furthest in time from the back porch.

Because the first pulse in any combined pulse group could be either a logic 1 or a logic 0, an extra (marker) pulse, which is always at logic 1, is inserted immediately in front of each combined pulse group. The marker pulse acts as a reference for the decoding process.

Digit Pulse Space and Shaping

The individual digit pulses are shaped into pulses of sine-squared form with a half-amplitude duration of 182 ns and an inter-pulse spacing of 182 ns. The

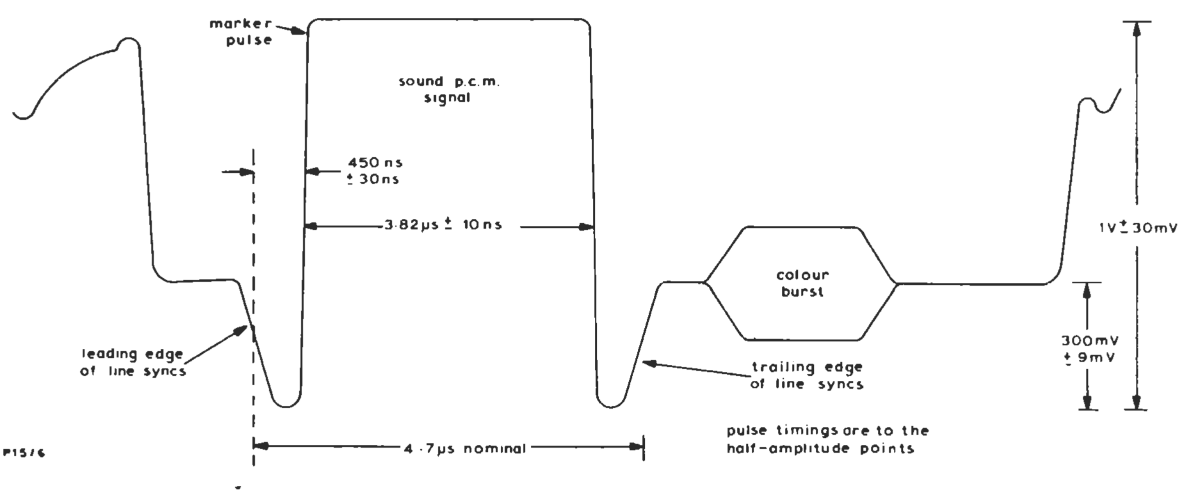

Fig. 2. Idealised Waveform Showing Combined Pulse-group in the Line-sync Period

Binary Pulse Groups

Formation of Pulse Groups

To combine the twice-line-frequency coded samples so that two samples can be inserted in each line synchronising pulse, alternate coded samples are stored for half a line. One binary group from each pair of samples is complemented (a logic 1 is exchanged for a logic 0 and conversely) and the digits of the two samples are interleaved so that similarly numbered digits from the two samples appear consecutively. This arrangement ensures that the mean level of the combined pulse group varies as little as possible.

sine-squared pulses have negligible energy content above 5·5 MHz and so very little distortion of the pulses occurs when they are handled by video circuits with this cut-off frequency. The total duration of a combined pulse group (i.e. two interleaved coded samples plus a marker pulse) is 3.82 µs (±10 ns) at the half-amplitude point and 4 µs (nominal) at the base of the group. An idealised representation of a combined pulse group positioned in the line-sync period is given in Fig. 2 and a detailed waveform of such a pulse group is given in Fig. 3. The half-line samples contained in Fig. 3 each have the digital code 1101100100 and before the samples are inter-

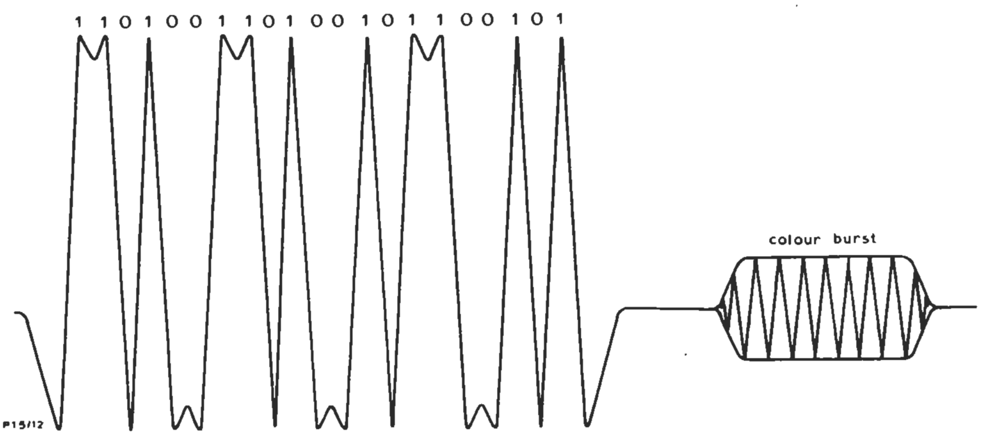

Fig. 3. Details of Pulse-group Waveform

leaved the digits of one sample are complemented; therefore the code of the displayed group is made up of:

1 1 0 1 1 0 0 1 0 0	unmodified sample
0 0 1 0 0 1 1 0 1 1	complemented sample
1 1 0 1 0 0 1 1 0 1 0 0 1 0 1 1 0 0 1 0 1	pulse group plus marker pulse

During the field-blanking period the sound pulse groups are still inserted in the video waveform at line rate and alternate equalising pulses are widened to accommodate them. Modified field-blanking waveforms which show the widened equalising pulses are given in Fig. 4.

Digit Pulse Amplitude

The maximum amplitude of signal which can be reliably passed through a video circuit is equal to the excursion from the bottom of syncs to white level. Pulses of this amplitude are used for the sound-in-syncs digits to ensure that the sound signal is immune to all but the most severe interference and distortion.

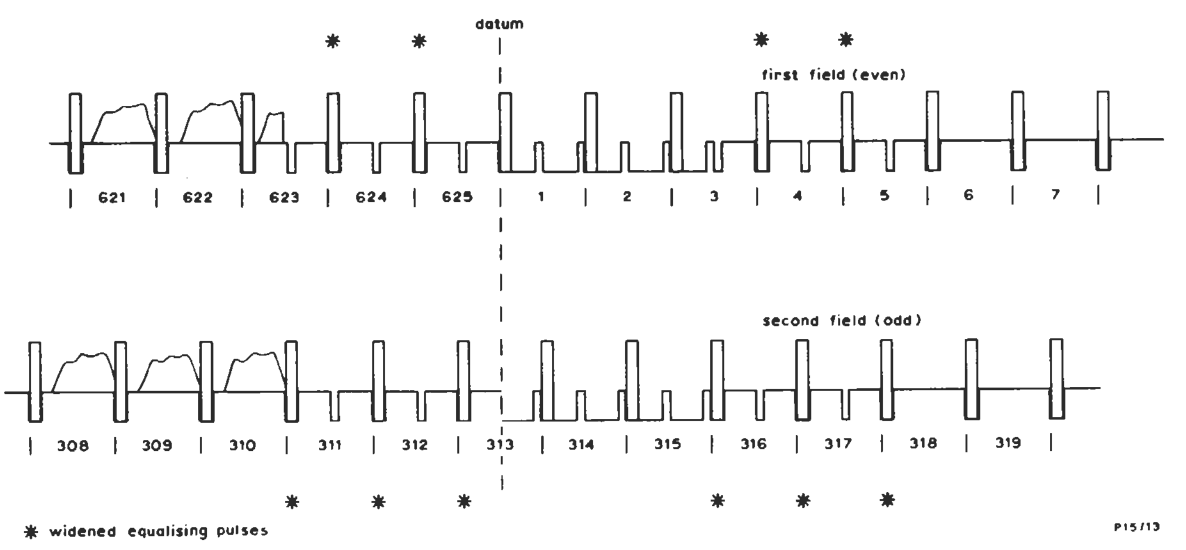

Fig. 4. Modified Field-blanking Waveform

Coding and Decoding

Coding (See also Instruction GP.2)

The sound-in-syncs equipment uses a counter-type coder as shown in Fig. 5. The sample to be coded is compared with the voltage developed across a capacitor which is charged from a constant-current source; the combination of capacitor and constant-current generator is known as a ramp generator.

Initially the capacitor is short-circuited and the counter is held in the reset condition (all outputs at logic level 0). When a sample is applied to the comparator, signals from the logic circuits remove the capacitor short-circuit and, simultaneously, allow clock pulses to be fed to the counter. The capacitor charges linearly and the counter accumulates a number which is proportional to the charging time. During the counting process the sample-and-hold circuit ensures that the sample amplitude is held constant.

the AND gates which link the counter to the register. The counter is then cleared and the capacitor short-circuited in preparation for coding the next sample. The shift register performs a parallel/series conversion and the application of shift pulses to the register causes the digital information to be fed to the output in time sequence.

The time interval between successive sampling pulses is 32 μs but, because the counter must be reset between the completion of one count and the start of the next, it is desirable to use a clock-pulse frequency that allows the count to be completed in about two-thirds of this time. A ten-bit system is used and so the maximum possible count is $2^{10} = 1024$. The clock-pulse frequency required to complete this count in, say, 21 μs is $1024/(21 \times 10^{-6}) = 49$ MHz. In practice a frequency of 50 MHz is used and the maximum possible count is completed in about 20 μs.

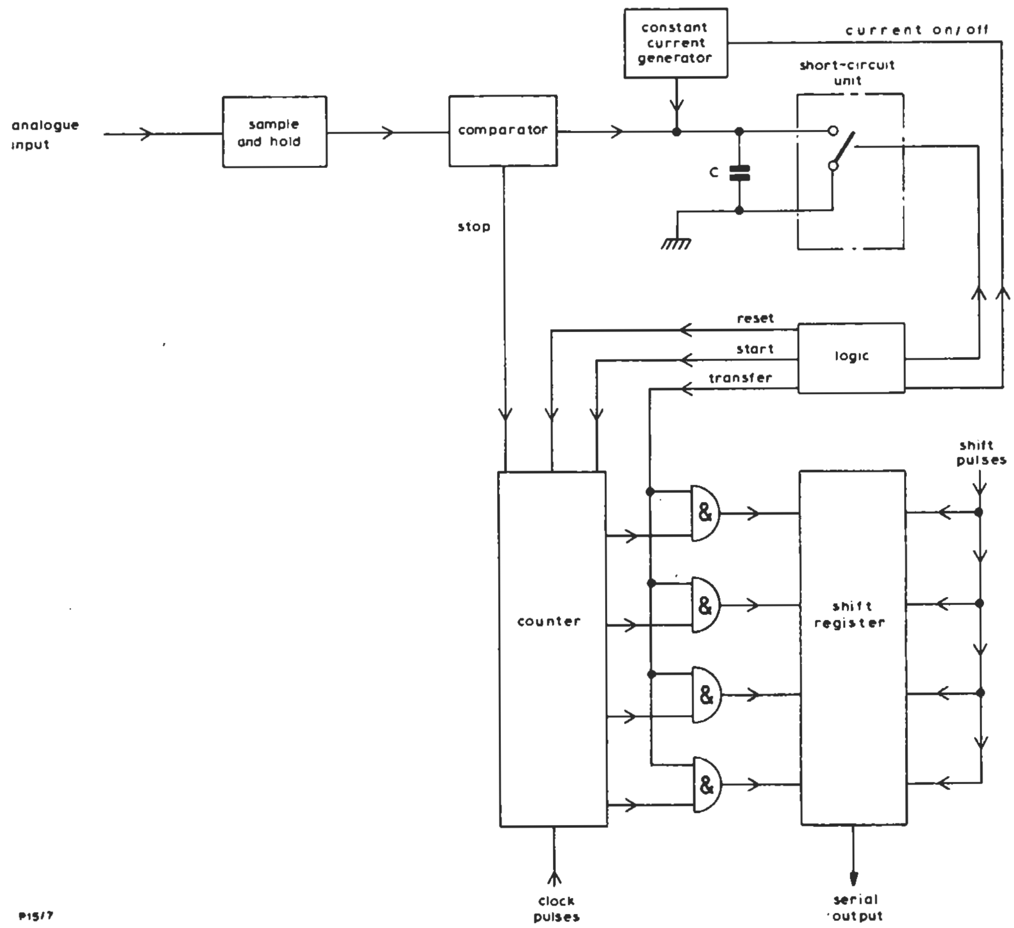

Fig. 5. Counter-type Coder Used for S-I-S

When the capacitor voltage equals the sample voltage, the comparator produces a pulse which stops the counter; the number held in the counter when it stops is the binary representation of the sample. The binary digits held by the counter are fed to the shift register by the application of transfer pulses to

The clock-pulse frequency for the shift-register is determined by the number of pulses in each pulse group and the time allocated for these pulses to be inserted into the sync pulse. There are 21 pulses in each group (one marker pulse plus two ten-bit coded samples) and the time allocated for the insertion of these pulses is 3·82 μs. The time spacing between consecutive pulses is 182 ns; therefore the shift-register clock-pulse frequency is $10^9/182 = 5{\cdot}5$ MHz.

Decoding (see also Instruction GP.2)

A simplified block diagram of a counter-type decoder is given in Fig. 6. The coded input signal is assembled in a shift register and is then transferred in parallel form to a counter. Clock pulses are applied to the counter to make it count down towards zero and, simultaneously, a ramp generator is started.

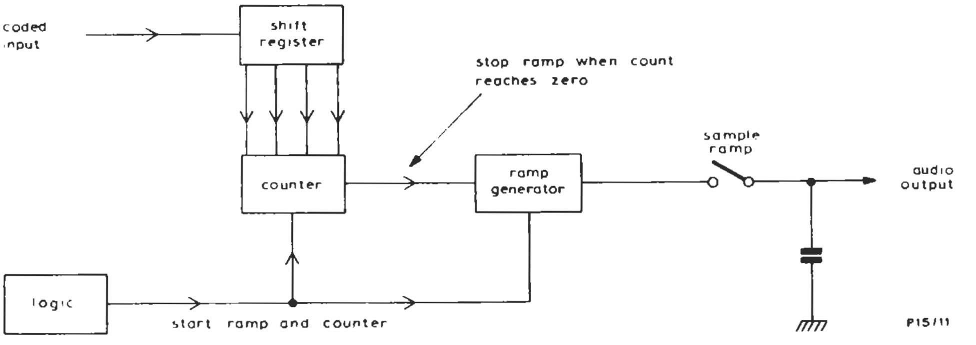

Fig. 6. Counter-type Decoder Used for S-I-S

When the counter is cleared (all sections at logic 0) the ramp generator stops. The ramp voltage is then sampled by a capacitor and the capacitor voltage at the end of the sampling period is equal to the amplitude of the recovered p.a.m. sample.

The counter and shift-register clock-pulse frequencies are the same as those used in the coder.

Companding

Function of the Compandor

The sound-in-syncs system provides codings for only a finite number of signal amplitudes and therefore has a well-defined overload point that must not be exceeded. To prevent overloading a quick acting limiter, or compressor, is inserted in the system prior to the coder to reduce the gain whenever the level of the incoming signal exceeds that corresponding to full modulation of the system. By arranging for the compressor to be in almost constant operation, the dynamic range of the signal is compressed and the mean signal-to-quantising noise ratio is increased.

To restore the signal to its original form at the receiving terminal an expander is provided to compensate for the gain variations introduced by the compressor. The complete system of compressor and expander is known as a compandor.

Pilot Tone

Because most of the fluctuations in the level of the incoming analogue signal are suppressed by the action of the compressor at the sending terminal, some of the information required to reconstruct the signal at the receiving terminal is lost. This information is recovered by adding to the signal before it is compressed a low-level pilot tone which has a frequency above that of the highest audio frequency transmitted. The pilot tone amplitude is therefore varied by the action of the compressor. At the receiving terminal the pilot tone component of the compressed signal is extracted by means of a filter and applied to a variable-gain circuit which controls the gain of the expander so as to give a constant level of pilot tone. In this way the overall gain of the system is maintained at a constant value.

To prevent a beat between the pilot tone and sampling frequencies, pilot tone is arranged to be 15·625 kHz (half-sampling frequency) and the phase is such that sampling takes place at the point of maximum amplitude in each half cycle.

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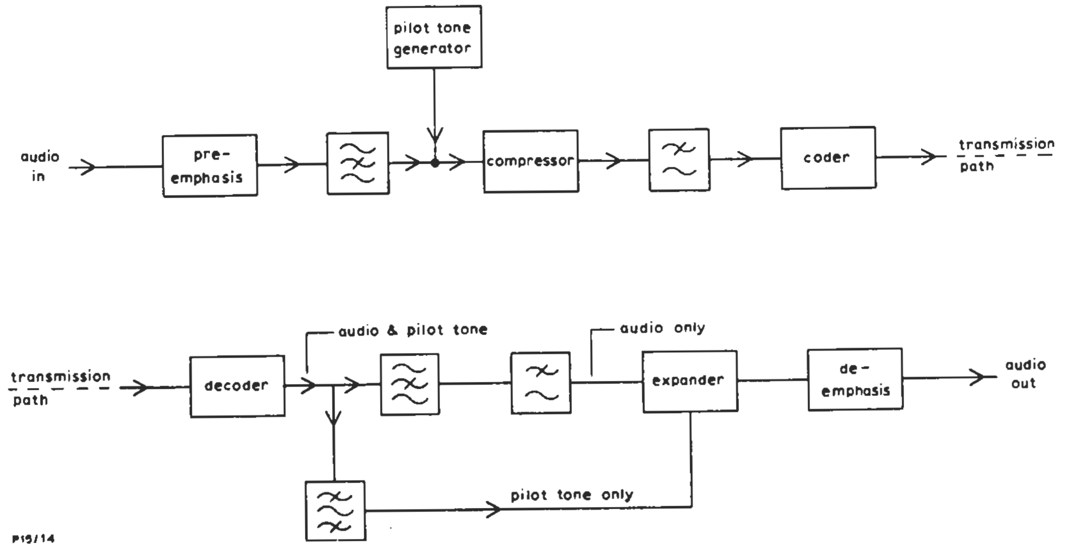

Fig. 7. Simplified Block Diagram of a Compandor

Compandor Operation

A simplified block diagram of a compandor is given in Fig. 7. At the sending terminal the signal is fed via a pre-emphasis network to a band-stop filter which has its rejection band centred on 15·625 kHz to ensure that spurious signals of this frequency are not applied to the compressor. Pilot tone at a frequency of 15·625 kHz is then added to the audio signal and the composite (audio plus pilot tone) signal is applied to the compressor. From the compressor the signal is fed to the coder via a low-pass filter which removes all frequencies above half the sampling rate.

At the receiving terminal the sound-in-syncs signal is decoded and the resulting compressed audio-plus-pilot-tone signal is applied to two separate signal paths. In one of these the pilot tone component is filtered out and the remaining audio signal is applied (via a low-pass filter which removes any spurious components above the required audio frequency) to the expander. In the other path the audio signal is filtered out and the remaining pilot tone is used to control the gain of the expander. From the expander the signal is applied, via a de-emphasis network, to the audio output.

Pre-emphasis and De-emphasis

The compandor characteristics are arranged so that companding takes place only when the signal contains a sufficient proportion of high-frequency components to mask the noise. The pre- and de-emphasis characteristics of a compandor suitable for use with sound-in-syncs circuits are given in Fig. 8. The characteristics are of standard C.C.I.T.T. form but the reference levels have been shifted so that the gain is zero at the lowest frequencies. Without pre-emphasis the maximum signal amplitude into a correctly-aligned coder just reaches the limiting point (1 kHz at zero level). With pre-emphasis programme signal components of high-frequency and large amplitude cause the compressor to reduce gain by up to 18 dB. The improvement in signal-to-noise ratio obtained with this arrangement is 13 dB which is slightly better than that produced by using a 12-bit system without companding.

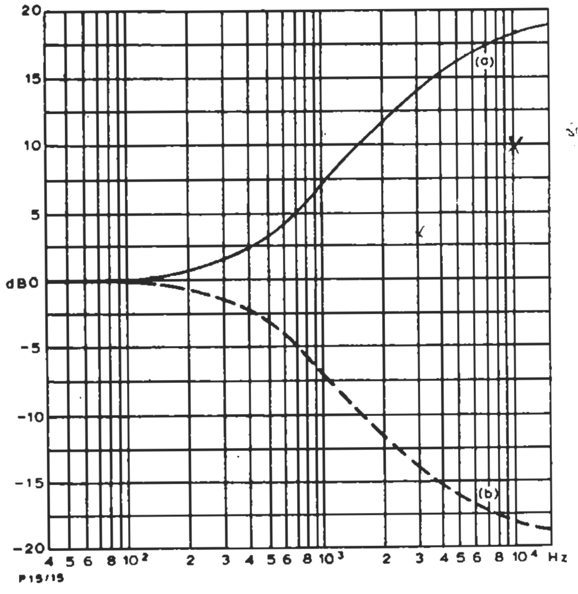

Fig. 8. Pre-emphasis and De-emphasis Characteristics Used in S-i-s Equipment

Monitoring

Video Link Monitoring

Monitoring of the sound-in-syncs signal on the video link can be done simply by checking that the sound pulse groups are present during the sync-pulse periods. The picture monitors which are required at various points along the link will not lock if fed with a sound-in-syncs signal; these monitors are therefore fitted with circuits which remove the coded pulses.

For programme-identification and cueing purposes sound receivers which incorporate decoders are provided.

Decoder Monitoring

Certain disturbances which may affect the combined video and sound-in-syncs signal have only a nominal effect on the observed picture but seriously degrade the sound signal if not corrected. Consequently the combined signal is monitored at the decoder to detect:

(a) The presence of a continuous train of synchronising pulses.

(b) Overswings or flashes in any part of the composite signal which lasts for more than 10 μs and any overswings which occur during the digit periods.

(c) The presence of a marker pulse at the start of each digit group. (The group is also examined to check that an extraneous twenty-second digit is not present.)

In the event of a fault one of two courses of action is automatically taken by the equipment. If the fault is of a transitory nature the information contained in the incorrect sample is discarded and the sample prior to the faulty one is used again. This action is known as holding and provided that the duration of the fault is short (say up to four samples) the resultant aural effect is negligible. If the fault is continuous or of a serious nature, a mute is applied to the audio output of the decoder. The mute is applied in a sophisticated manner so that audible clicks or disturbances are not generated and, provided that the muting period does not exceed 6 ms, occasional interruptions pass un-noticed in many types of programme.

A hold is applied to the signal if:

(a) the marker pulse is absent or does not have the correct time relationship to the leading edge of syncs.

(b) a transition is detected during the pulse group which exceeds the top or bottom of the group by more than 4 dB

(c) a logic 1 (i.e. a 22nd digit) is detected immediately after the pulse group.

A mute is applied if:

(a) sync pulses are absent

(b) the video signal exceeds an amplitude of about 4 dB above white level or about 4 dB below sync bottoms for more than 10 μs during any line period

(c) more than three holds are encountered in less than 10 lines

(d) the timing generator goes out of lock.

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PART 2

RUGGEDISED SOUND-IN-SYNCS

Introduction

Ruggedised Sound-in-Syncs is a sound-in-syncs system for use in television outside broadcasts. A rugged system is required is outside broadcast links because of the possibility of high noise levels, signal reflection and multipath distortion. If standard sound-in-syncs is used under such conditions the sound system may fail even though the vision signal is still acceptable.

Principal Features

The audio signal to be transmitted is pre-emphasised and sampled at line rate. Each sample is converted into a 9-bit binary code and this together with the effect of pre-emphasis and subsequent de-emphasis results in a peak-signal to peak-weighted-noise ratio of 54 dB. The audio bandwidth, which has a theoretical maximum of half the sampling rate, is restricted to 6·8 kHz by low-pass filters.

The standard sound-in-syncs system is susceptible to three forms of failure which are outlined below together with the measures adopted in the ruggedised system to reduce their effects.

(a) Noise spikes, of peak amplitude greater than half-vision amplitude, can be decoded as sound digits if they are coincident with the transmitted sound pulses.
The ruggedised system reduces the effect of such spikes in the sound pulse group by complementing each of the transmitted digits except the least significant. A cross check between the received primary digits and their complements enables the decoder to detect whether the received information is true (complement correct) of whether a false digit has been decoded from a noise spike (complement incorrect). In practice, a check for errors in the five most significant digits of each pulse group is adequate because errors in the remaining four digits are subjectively tolerable. When the complement check shows an incorrect digit, the information from the previous correctly received digit group is repeated; and thus an uninterrupted audio output is maintained. This error-concealment process can be repeated for four consecutive samples before the effect is noticeable. When four consecutive errors occur the sound output is muted until a correctly decoded output is again received.

(b) The timing of the decoding operation uses the leading edge of line syncs as the time reference, any noise which causes an apparent change in the position of this edge can cause erroneous operation of the decoder.
To improve the noise immunity of the decoder time reference in the ruggedised system, timing is derived from the half-height amplitude of the pulse-group maker pulse. Because this pulse amplitude is slightly more than three times that of the line sync pulse the improvement in noise immunity is about 10 dB and is reinforced by using the detected time reference to synchronise a flywheel circuit instead of a hard-lock circuit.

(c) Frequency distortion and echoes in the coder and decoder signal path can produce a sound on vision effect because the mean d.c. variation of the sound pulse group produces apparent changes in the back-porch level.
The digit complementing process mentioned in (a) also reduces the mean d.c. variation. Because all but the least significant digit in each sound pulse group are complemented, the maximum variation is limited to five per cent.

These features of the ruggedised sound-in-syncs system ensure satisfactory performance for a peak signal to unweighted r.m.s. noise ratio of 18 dB and usable performance if the ratio is as low as 15 dB.

Waveforms

Each line-rate sample is represented by a pulse-group of eighteen sine-squared pulses of 200 μs half-amplitude which fit into the same time slot as in the standard system shown in Fig. 2. Insertion of the pulse groups during the field period uses the same method as standard sound-in-syncs and is shown in Fig. 4.

The pulse sequence in each group is:

$MARKER.2^0.\overline{2^1}.2^1.\overline{2^2}.2^2.\overline{2^3}.2^3.\overline{2^4}.2^4.\overline{2^5}.2^5.\overline{2^6}.2^6.\overline{2^7}.2^7.\overline{2^8}.2^8.$

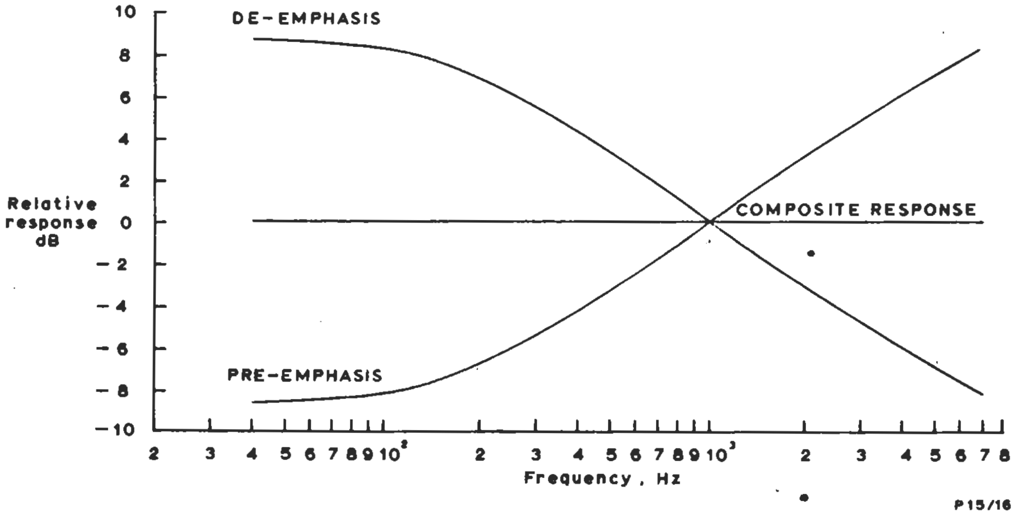

Fig. 9. Basic Pre-emphasis and De-emphasis Curves

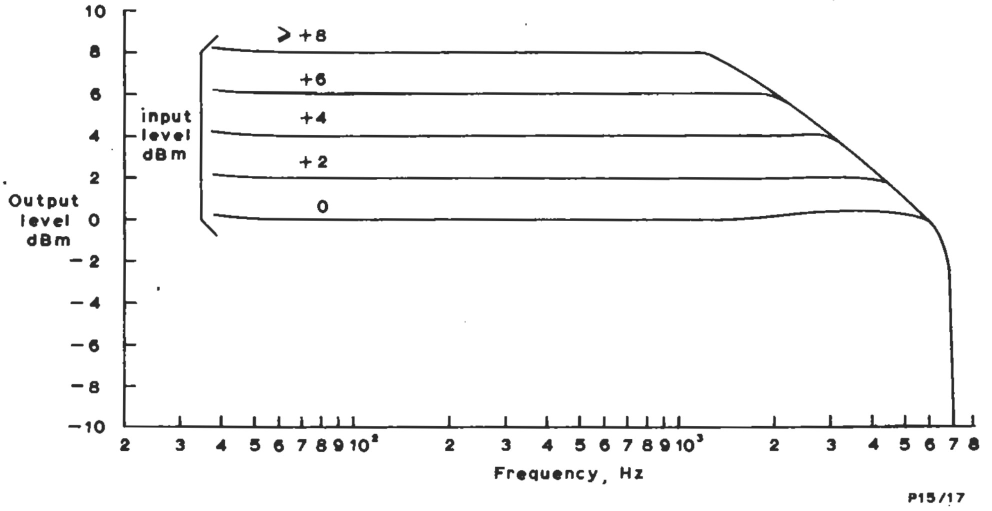

Fig. 10. RSIS Audio System Overall Response

Audio System

The processing of the audio signal comprises pre-emphasis in the coder and de-emphasis in the decoder. This gives a 6-dB improvement of signal to noise ratio. The audio signal is amplitude limited before pre-emphasis.

The basic pre-emphasis characteristic used is similar to that recommended by the CCITT for music-in-band circuits and is shown in Fig. 9. To prevent over-modulation by high amplitude high frequency components the degree of pre-emphasis is automatically reduced when such components are present with a consequent reduction in overall h.f. response because the decoder de-emphasis is constant. The effect is shown in Fig. 10 which shows that a flat response with de-emphasis exactly cancelling pre-emphasis occurs at programme levels below 0 dB. At levels above 0 dB the response at high frequencies reduces with amplitude and frequency.

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