



RESEARCH DEPARTMENT

**Pulse-code modulation for high-quality sound signal distribution:
incorporation of the sound signal
in the video signal**

TECHNOLOGICAL REPORT No. EL-19

UDC 621.397.238

1968/36

**THE BRITISH BROADCASTING CORPORATION
ENGINEERING DIVISION**

RESEARCH DEPARTMENT

**PULSE-CODE MODULATION FOR HIGH-QUALITY SOUND SIGNAL DISTRIBUTION :
INCORPORATION OF THE SOUND SIGNAL IN THE VIDEO SIGNAL**

Technological Report No. EL-19
UDC 621.397.238 1968/36

This Report is the property of the British Broadcasting Corporation and may not be reproduced in any form without the written permission of the Corporation.

It uses SI units in accordance with B.S. document PD 5686.

J.R. Sanders, M.A.



Head of Research and Development

**PULSE-CODE MODULATION FOR HIGH-QUALITY SOUND SIGNAL DISTRIBUTION :
INCORPORATION OF THE SOUND SIGNAL IN THE VIDEO SIGNAL**

Section	Title	Page
	SUMMARY	1
1.	INTRODUCTION	1
	1.1. General	1
	1.2. Advantages and Disadvantages of Combined Sound and Video Distribution	1
2.	SUMMARY OF PREFERRED METHODS OF CODING THE SOUND SIGNAL	2
	2.1. Pulse-Position Modulation	2
	2.2. Pulse-Code Modulation	2
3.	TIME AVAILABLE IN THE VIDEO WAVEFORM FOR TRANSMISSION OF THE SOUND SIGNAL	2
4.	THE SPECIFICATION OF THE PULSE-CODE MODULATION SYSTEM	3
	4.1. Sampling Frequency	3
	4.2. Number of Digits	3
	4.3. Digit-Pulse Amplitude	3
	4.4. Digit-Pulse Shape and Spacing	4
5.	CONCLUSIONS	4
6.	REFERENCES	5
7.	APPENDIX	6
	7.1. General Considerations	6
	7.2. The $\text{Sin}x/x$ Pulse	6
	7.3. The Cosine^2 Pulse	8
	7.4. Intermediate Pulse Shapes	9
	7.5. Conclusions	10

PULSE-CODE MODULATION FOR HIGH-QUALITY SOUND SIGNAL DISTRIBUTION : INCORPORATION OF THE SOUND SIGNAL IN THE VIDEO SIGNAL

SUMMARY

A study has been made of the possibility of combining the sound and video signals in a 625-line programme distribution network, thus avoiding the necessity for separate sound circuits. A somewhat non-standard video waveform would be used, which would be restored to standard form before being broadcast; the sound signal would be separately recovered from the combined signal.

1. INTRODUCTION

1.1. General

At the present time the video and sound signals of all television programmes are carried between the point of origination and the transmitter by a network of contribution and distribution circuits in which the video and sound signals use separate circuits and equipment. Recently, however, the expansion of both sound and television networks has led to a growing shortage of high-quality sound circuits; as a result it was decided to investigate the possibility of relieving this situation by combining associated sound and video signals to form a composite 'programme signal' capable of being carried by a normal 625-line video link. This report describes the results of this investigation.

Essentially the proposed system uses time-division-multiplexing of the video and sound signals, the two signals being combined by suitable terminal equipment so that the sound information is included in the video signal at times when no essential picture information is being sent. The resulting 'programme waveform' is then of necessity non-standard, but is nevertheless of suitable form to be sent over 625-line video circuits. The terminal equipment at the receiving end, or at any point on a route, recovers the sound and video signals in their standard forms.

The system considered in this report is intended primarily for the BBC's 625-line distribution network and possibly for the contribution network also; the use of such a system could be extended to the contribution network and temporary links as long as these circuits were free of gross distortions and excessive noise.

1.2. Advantages and Disadvantages of Combined Sound and Video Distribution

With the sound signal incorporated in the video waveform, the cost of a separate high-quality sound distribution network for the 625-line television service could be avoided and the possibility of operational errors would be reduced since the sound could not become dissociated from the appropriate video signal between terminal points. Overall reliability of the sound distribution system should be improved, since failure of video links is comparatively rare and solid-state circuitry will ensure high reliability of the terminal equipment. Sound quality would be consistent and independent of the quality of the video link over wide limits and a uniform audio-frequency response over a bandwidth of some 14 kHz would always be available¹.

Against the above-mentioned advantages must be set the need for additional terminal equipment and the fact, already mentioned, that failure of the video link would cause loss of both sound and vision. Precautions will have to be taken to avoid loss of operational flexibility as the non-standard video waveform could not be routed through vision mixers or other equipments which depend upon the presence of standard synchronizing information. Picture monitors on the route would have to be provided with devices to remove the sound information and restore the normal synchronizing pulses.

With the system as envisaged, separation and re-combination of the video and sound signals would be carried out only where it is essential that both signals be available separately, since in each separation and re-combining operation small spurious signals might be introduced into the video waveform. Further, and perhaps more important,

repetitive demodulation and remodulation of the sound signal must be avoided as this would result in a loss of sound signal-to-noise ratio. These factors place a restriction on the number of times that the signal may be so processed; consequently, no studio centre through which the signal passes should carry out these operations unnecessarily and some revision of operational methods at studio centres may be required.

2. SUMMARY OF PREFERRED METHODS OF CODING THE SOUND SIGNAL

2.1. Pulse-Position Modulation

This is the most advantageous of the systems which rely upon analogue modulation of pulse parameters. A previous report² has shown that by using pulse-position modulation with two pulses in each line-blanking interval, it would be possible to achieve a ratio of peak sound signal-to-peak weighted* a.f. noise of 55 to 58 dB with a video link of good performance having a peak signal-to-r.m.s. noise ratio of 46 dB. The audio bandwidth would be about 14 kHz.

The principal disadvantage of such a system is that the signal-to-noise performance of the sound circuit is a direct function of the noise present in the video link; for short periods this performance may well deteriorate below the agreed acceptance limits and cause unacceptable sound quality while remaining acceptable for video transmission. Another disadvantage which became apparent during the development of a pulse sound system for broadcasting³ was that unwanted phase modulation of the pulses at their source, due to phase modulation of the synchronizing pulses, etc., would need to be kept below the equivalent of ± 0.5 ns timing error in order to obtain a sound peak signal-to-peak weighted noise ratio of 54 dB due to this cause alone. It is felt that this requirement might be difficult to meet.

2.2. Pulse-Code Modulation

Recent experiments have demonstrated the feasibility of coding high-quality sound signals using pulse-code modulation^{1,4} (p.c.m.); a 10-digit system, providing 1024 possible levels for the sound signal, will have a theoretical peak signal-to-peak weighted quantizing noise ratio of 53 dB. This ratio of signal-to-quantizing noise is that which would be obtained with a conventional p.c.m. system; however, by the use of a moderate amount of

pre- and de-emphasis¹ this figure can be improved by 3 dB. This figure is still below the requirements for high-quality sound signals⁵, but it is expected that the use of specially developed forms of companding will improve the signal-to-noise ratio by 10 dB or more thus making 10 digits sufficient. If these hopes are not realised it would be necessary to use at least 11 digits and the parameters stated in this report, which considers a 10-digit system, would need to be modified accordingly.

P.C.M. is the preferred method of coding principally because this form of modulation is substantially immune to moderate amounts of noise and distortion in the transmission circuit^{6,7}; the system fails only when the combined effect of noise and distortion cause the value of a digit, i.e. a one or a zero, to be incorrectly determined at the receiving terminal. Phase modulation of the synchronizing pulses is also relatively unimportant. The use of coding systems in which the individual pulses could have more than two levels was also considered, but was rejected due to the susceptibility of such systems to the effect of group delay distortion and the inferior resistance to noise.

The principal disadvantage of p.c.m. is the addition of fairly complex a.d.c.* and d.a.c.* apparatus to the terminal equipment. However, this additional apparatus can be made with integrated logic circuits and will therefore be both compact and reliable.

3. TIME AVAILABLE IN THE VIDEO WAVEFORM FOR TRANSMISSION OF THE SOUND SIGNAL

In a monochrome signal, the total time allocated to the line and field synchronizing and blanking intervals occupies some 25% of the total time required to send a complete picture; the field blanking interval occupies 6.5% of each field period, and 19% of each line period is occupied by line-blanking. In a colour signal, the colour burst occupies the post-line-synchronizing blanking interval, leaving the line-synchronizing pulse as the most suitable interval for the transmission of other information; this period is therefore chosen for the p.c.m. sound signal. The line-frequency of the 625-line system is 15.625 kHz, a figure which is comparable with the upper limit of the audio-frequency range to be transmitted.

The chief disadvantage of the line-synchronizing period is its brevity and consideration has been given to ways in which it could be temporarily extended to accommodate the sound signal. Fig. 1(a) shows the line-blanking interval of a 625-line

* This is the standard method of specifying signal-to-noise ratios obtaining in the BBC sound distribution system. The figures quoted, originally given in terms of unweighted r.m.s. noise, have been converted by a correction factor, the derivation of which is given in an appendix to Reference 1.

* a.d.c.: Analogue-to-digital converter.
d.a.c.: Digital-to-analogue converter.

waveform. It will be seen that, excluding the 'front porch' and colour burst, the total time available for line-synchronizing and other information is $8.2 \mu\text{s}$, this being made up of $4.7 \mu\text{s}$ of line-synchronizing pulse, $0.8 \mu\text{s}$ of black level from the trailing edge of the line-synchronizing pulse to the start of the burst, and $2.7 \mu\text{s}$ of black level from the end of the burst to the start of the picture; the figures given are all nominal half-amplitude values. If the longest possible period for transmitting the sound signal is desired, the waveform could be modified to the form shown in Fig. 1(b), thus making available an uninterrupted $7.8 \mu\text{s}$ period of synchronizing level (200 ns guard-space is allowed at each end of the burst). This modification was rejected since high precision would be needed to move the burst to its new position and subsequently replace it; it was considered that the process would involve a significant increase in equipment complexity, a complication justified only if absolutely necessary. The conventional form of line-synchronizing waveform was, therefore, retained and the sound information inserted in the line-synchronizing intervals.

An examination of the tolerances proposed* for the 625-line waveform as distributed to transmitters shows that the minimum line-synchronizing pulse duration is $4.6 \mu\text{s}$ with maximum rise-times of 300 ns, which leaves a minimum of $4.3 \mu\text{s}$ of undisturbed level available. The equipment which combines the video and sound signals could, however, be designed to ensure that the duration of the line-synchronizing pulse measured at its output never fell below $4.7 \mu\text{s}$, so that a period of $4.4 \mu\text{s}$ is potentially available. Since the leading edge of the line-synchronizing pulse is the primary timing edge in the waveform, it was decided to leave a guard space of 300 ns between the centre of the leading edge and the sound signal; the timing edge should then never be disturbed by the sound signal, even when non-uniform group-delay distortion is present on the circuit. Prior to the trailing edge of the line-synchronizing pulse, a guard space of 100 ns is sufficient so that the interval remaining for the sound signal is $4.0 \mu\text{s}$. For reasons indicated in the Appendix**, the actual space taken up by the coded sound signal may be less than this figure.

4. THE SPECIFICATION OF THE PULSE-CODE MODULATION SYSTEM

4.1. Sampling Frequency

Converting an analogue signal to digital form requires regular sampling at a frequency that is at least twice the highest frequency present in the

* P.I.D. Drawing No. 1000.7.33A1

** The Appendix was contributed by Dr. P.C.J. Hill.

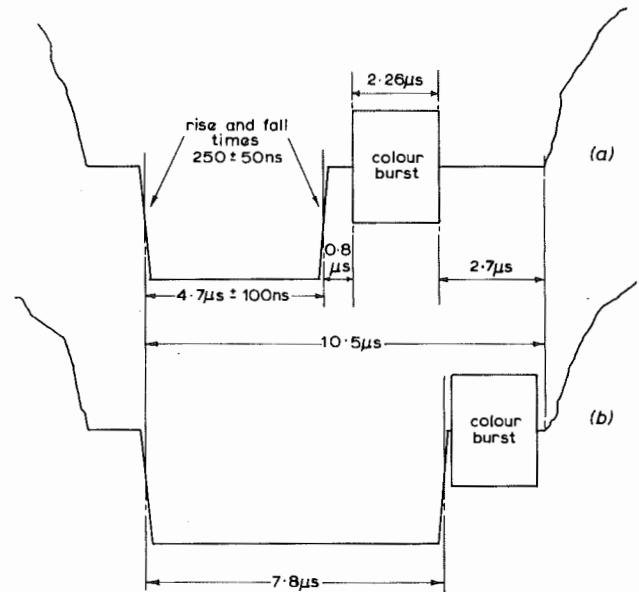


Fig. 1 - Standard and modified 625-line sync waveforms

(a) Standard (b) Modified

original signal. Since the sound signal is to be included in a 625-line television waveform a simple multiple of television line-frequency would be convenient as a sampling frequency and 31.25 kHz, twice line-frequency, was adopted. This results in an effective audio bandwidth of 14 kHz when the imperfections of practical filters are taken into account.

4.2. Number of Digits

The minimum number of digits likely to prove satisfactory for a practical system is 10; this corresponds to 1024 quantum levels. Such a p.c.m. system has a theoretical peak signal-to-peak weighted quantizing noise ratio of 53 dB as mentioned in Section 2.2.

In order to combine groups of 10 digits that are produced at a frequency of 31.25 kHz, with a 625-line video signal, alternate groups must be stored for half the duration of a television line ($32 \mu\text{s}$) so that two groups can be inserted in each line-synchronizing interval. In addition an extra pulse must be inserted immediately before each combined group of 20 digits, to act as a time reference for the decoding process.

4.3. Digit-Pulse Amplitude

The maximum pulse amplitude which can be reliably transmitted through a video circuit is equal to the excursion from synchronizing level to peak-white level. Pulses of this amplitude will provide the greatest possible protection against noise and interference and since these pulses will never be transmitted or displayed on a picture monitor there is no reason why they should not be used.

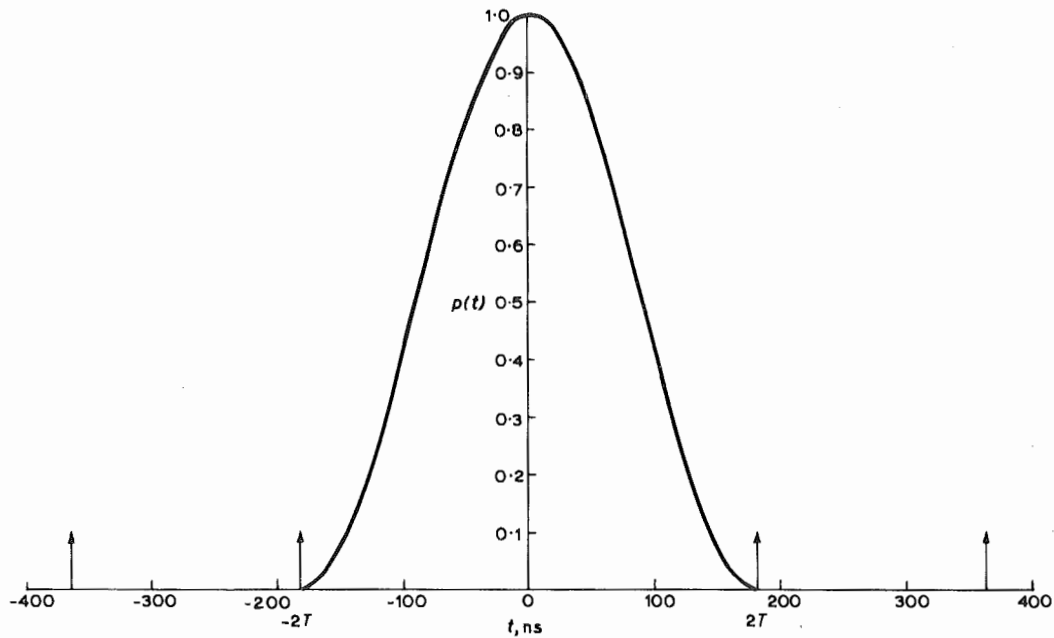


Fig. 2 - Ideal $2T$ -cosine² pulse (bandwidth approximately 5.5 MHz)

$$p(t) = \frac{1}{2}[1 + \cos(\pi t/2T)]; \quad -2T \leq t \leq 2T; \quad 2T = 182 \text{ ns}$$

↑ Minimum spacing of pulses without reducing noise immunity

4.4. Digit-Pulse Shape and Spacing

Individual pulses of the 21-pulse combination (one marker pulse plus two 10-digit words) must each have a well-defined start and finish since any spurious disturbance outside the $4.0 \mu\text{s}$ interval could cause interference with other parts of the waveform. An obvious choice for the pulse shape is that formed by one cycle of the cosine² function; this is the shape of the well-known $2T$ -test pulse shown in Fig. 2. The practical pulse has insignificant minor lobes and a spectrum which falls to zero at 5.5 MHz with negligible power at higher frequencies. The choice of pulse shape is discussed in detail in the Appendix.

The closest spacing of pulses which can be used without reducing their immunity to noise is that for which the maximum of one pulse exactly coincides with the zero of an adjacent pulse as indicated by the arrows in Fig. 2. Using this spacing, the full noise immunity can be realised only if the determination of the presence or absence of a pulse is carried out at precisely the time at which its maximum occurs. This spacing is also the closest which may be adopted if the amplitude of a block of pulses is not to be greater than the amplitude of an individual pulse, a desirable condition, since it allows pulses of the greatest possible amplitude to be used. The loss of noise immunity which would be incurred by the use of a pulse code containing more than 10 digits, and therefore requiring a closer

spacing between pulses, is considered in detail in the Appendix.

5. CONCLUSIONS

It has been shown that there is sufficient redundant time available in the 625-line video waveform to include the accompanying sound signal in the form of a 10-bit p.c.m. signal. If companding systems are not able to improve the signal-to-quantizing noise ratio as is hoped it may be necessary to adopt an 11-bit system in order to achieve a sufficiently good signal-to-noise ratio for the sound signal. If this is the case the digit spacing would be reduced with a consequent small loss in noise immunity.

A system such as that investigated in this report could almost certainly provide a high-quality sound circuit when used in conjunction with permanent 625-line vision links, provided that the limitations imposed by the use of a non-standard waveform can be tolerated. The most obvious advantage of such a system is the elimination of a separate sound circuit with its associated cost and risk of interruption or incorrect routing.

A further report will describe experimental equipment constructed to test the feasibility of such an arrangement.

6. REFERENCES

1. Pulse-code modulation for high-quality sound signal distribution: appraisal of system requirements. BBC Research Department Report No. EL-10, Serial No. 1967/50.
2. Television sound transmission by modulated pulses in the line blanking intervals. BBC Research Department Report No. T-176, Serial No. 1966/58.
3. Pulse sound: a system of television sound broadcasting using pulses in the video waveform. BBC Research Department Report No. T-181, Serial No. 1966/73.
4. Pulse-code modulation for high-quality sound signal distribution: coding and decoding. BBC Research Department Report No. EL-18, Serial No. 1968/29.
5. The assessment of noise in audio-frequency circuits. BBC Research Department Report No. EL-17, Serial No. 1968/8.
6. OLIVER, B.M., PIERCE, J.R. and SHANNON, C.E. 1948. The philosophy of p.c.m. *Proc. Inst. Radio Engrs.*, 1948, **36**, 11, pp. 1324 – 1331.
7. GOODALL, W.M. 1947. Telephony by p.c.m. *Bell Syst. tech. J.*, 1947, **XXVI**, 3, pp. 395 – 409.
8. DRAJIĆ, D.B. 1967. New approach to the choice of the amplitude function for pulse transmission. *Electron. Letters*, 1967, **3**, 9, pp. 437 – 438.

7. APPENDIX

The Choice of Digit Pulse Shape

Contributed by P.C.J. Hill, Ph.D., B.Sc., D.I.C.,
M.I.E.E.

7.1. General Considerations

The spectrum of the sound pulses must be confined to a bandwidth of 0 – 5.5 MHz so that the composite signal can be transmitted over vision links having this nominal bandwidth. The range of pulse shapes which were investigated included the $\sin x/x$ function and the \cos^2 function together with some intermediate shapes. Although it is normally assumed that the \cos^2 function is the optimum choice for pulse transmission it can be shown that the optimum function depends very much on system criteria⁸.

The factors which decide the usefulness of a particular pulse shape concern the relative resistance to inter-pulse interference (crosstalk); this determines the effective noise immunity of the transmitted signal. Firstly, the amplitude of an individual pulse should be as large a proportion of the peak-to-peak magnitude of a group of pulses as possible. Secondly, the pulse zeros should be

equally spaced on each side of the main lobe so that they can be made to coincide with the maxima of adjacent pulses; this allows optimum synchronous detection of the pulse group. In addition, the time occupied by the subsidiary lobes should be short since this will extend the effective duration of the group. It should be noted that the pulses produced by practical shaping networks would depart somewhat from the theoretical form outside the main lobe.

7.2. The $\sin x/x$ Pulse

Fig. 3 shows part of a theoretical $\sin x/x$ pulse corresponding to spectrum (a) in Fig. 4. Assuming that each pulse is sampled at its maximum, the closest spacing which could be allowed without crosstalk between pulses in a group is 91 ns; with this spacing, shown by the lower set of arrows in Fig. 3, the maximum of any one pulse occurs at the position of the first zero-crossing of the next one and so on.

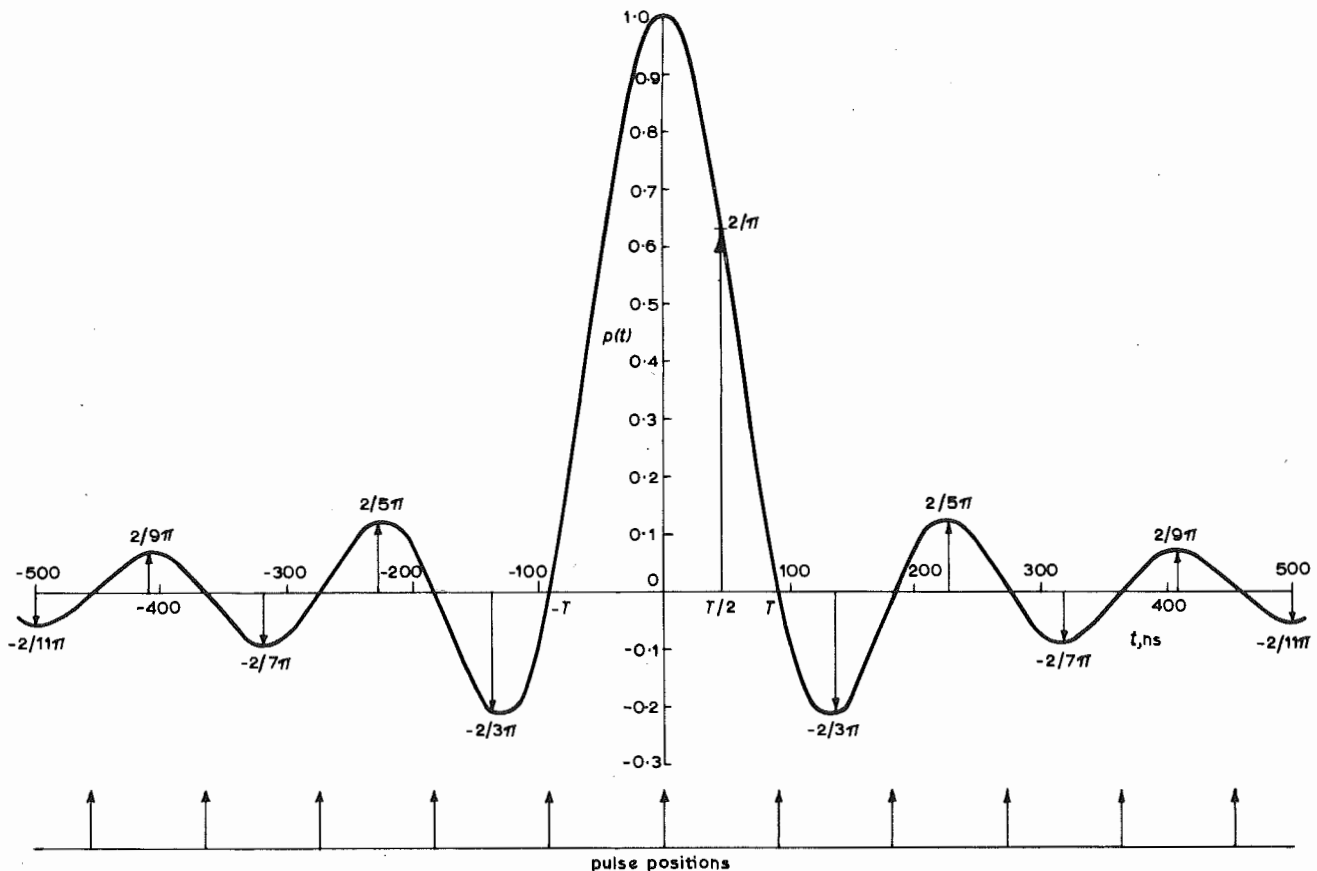


Fig. 3 - 5.5 MHz $\sin x/x$ pulse

$$p(t) = \frac{T \sin(\pi t/T)}{\pi t}; \quad T = 91 \text{ ns}$$

The peak-to-peak amplitude of a group of equi-amplitude pulses depends upon the length of the group and on the code sequence; the maximum positive and negative peaks, however, will occur in general with different code sequences. The extreme cases for a 7-digit pulse group found by inspection, are illustrated in Fig. 5. The maximum positive excursion, \hat{P}_+ in Fig. 5(a), is very close* to the point of crossover between the main lobes of the two adjacent pulses nearest to the centre of the group, the remaining pulses being alternately present and absent in such a way that only positive minor lobes contribute to the peak value. Similarly, the maximum negative excursion, \hat{P}_- in Fig. 5(b), occurs at a point half way between the two pulses nearest to the centre of the group; these pulses are three digit-spaces apart, the remaining pulses being alternately present and absent in such a way that only negative minor lobes contribute to the peak value. In general the code sequence(s) to be considered for evaluating the peak-to-peak amplitude depend upon the number of digits in the group as is shown in the examples given in Table 1 in which m denotes the number of digits per word.

TABLE 1

Code Sequences for Determining Peak-to-Peak Amplitudes of Pulse Groups : the sign \wedge or \vee shows the position of the maximum or minimum

No. of Digits	Positive Peak	Negative Peak
$2m$ (even)		
m even	$101010\hat{1}10101$ $1010\hat{1}1010101$	$1010100\vee10101$
m odd	$101010\hat{1}1010101$	$01010100\vee10101$ $1010100\vee101010$
$2m + 1$ (odd)		
m even	$01010\hat{1}1010101$ $101010\hat{1}101010$ $1010\hat{1}10101010$ $0101010\hat{1}10101$	$1010100\vee101010$ $01010100\vee10101$
m odd	$1010\hat{1}101010$ $01010\hat{1}10101$	$010100\vee10101$ $1010100\vee1010$ $10100\vee101010$ $01010100\vee101$

* The minor lobes are not quite symmetrical in shape.

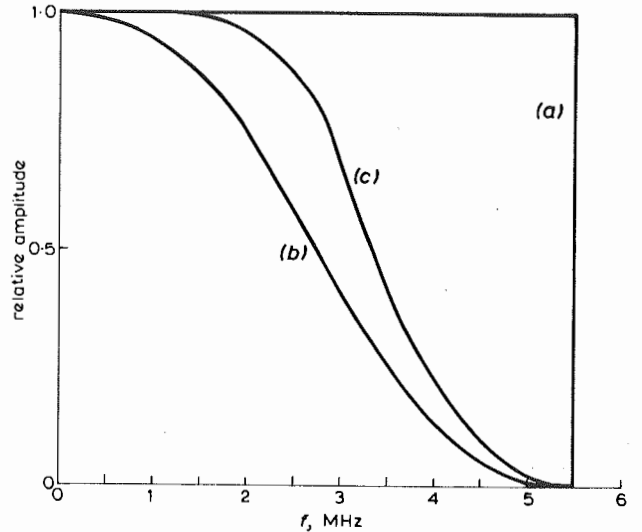


Fig. 4 - Pulse spectra
 (a) sinc/x pulse (b) cosine^2 pulse
 (c) intermediate 'convolved' pulse

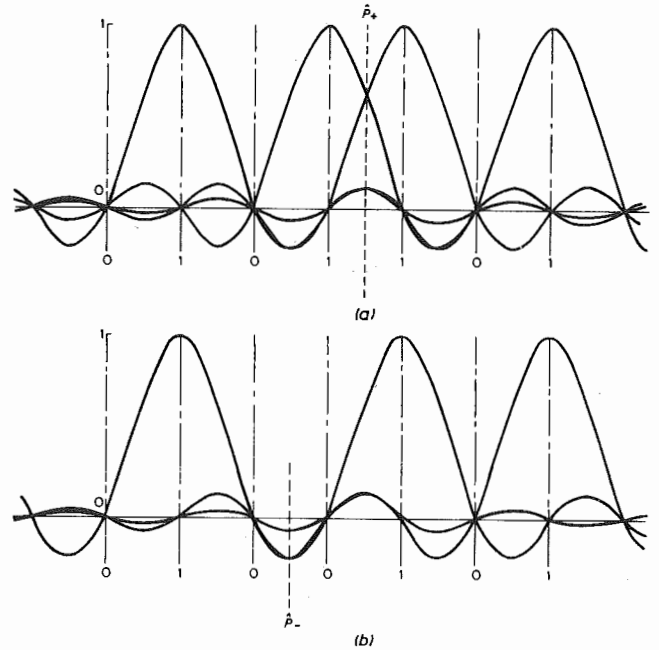


Fig. 5 - Code sequences which determine the maximum peak-to-peak amplitude for a 7-digit code with sinc/x pulses
 (a) Positive peak (b) Negative peak

For a 21-pulse group (one marker pulse plus two 10-digit words, i.e. $m = 10$) the peak-to-peak excursion is given by the following sequences:

- $10101010\hat{1}1010101010$
- or $1010101010\hat{1}10101010$ } positive
- $10101010100\vee10101010$ negative

Since the positive peaks of $\text{sinc}(t/T)/(t/T)$ occur at $t = 0, t \approx \pm 5T/2, \pm 9T/2, \dots$ and the negative

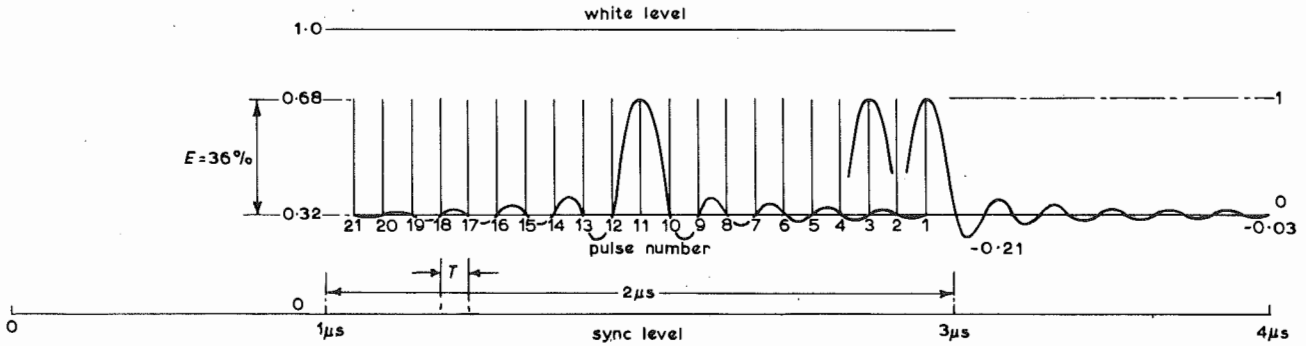


Fig. 6 - Time-amplitude diagram for 21-sinx/x pulse group

peaks at $t \approx \pm 3T/2, \pm 7T/2, \dots$ (see Fig. 3), the peak-to-peak amplitude of the 21-pulse group is given quite accurately by the summation of the following series (Fig. 3):

$$\frac{4}{\pi} \left(1 + \frac{1}{5} + \frac{1}{9} + \frac{1}{13} + \frac{1}{17} + \frac{1}{42} \right) + \frac{4}{\pi} \left(\frac{1}{3} + \frac{1}{7} + \frac{1}{11} + \frac{1}{15} + \frac{1}{19} \right)$$

$= 1.87 + 0.87 = 2.74$. The pulse amplitude at the points where there is maximum protection against noise is only 1.0, however, so that restricting the peak-to-peak amplitude of the group to the peak video excursion (sync-to-peak white level) gives an effective modulation depth of only $1/2.74$ or 36%; this is the amplitude of the so-called 'discrimination eye-pattern' (the significance of this term will become clear in Section 7.3, (Fig. 7)).

The major lobes of 21 $\text{sin}x/x$ pulses would occupy a time $22T (2 \mu\text{s})$ as shown in Fig. 6. Minor lobes, however, are very extensive; a single pulse at the end of the group has a minor lobe 3% of its major lobe amplitude at a point $1 \mu\text{s}$ out from the major lobe. The pulse group will therefore have significant minor lobes extending outside the permitted $4 \mu\text{s}$ total duration; the maximum peak-to-peak excursion of the pulse group in this region can be shown to be 12% of peak video. The $\text{sin}x/x$ pulse shape is thus inefficient both in regard to overall duration and noise protection. Moreover, the shape of the pulse at the zero-crossings would be very sensitive to channel degradations; this leads to inter-pulse crosstalk and a further reduction in noise immunity.

7.3. The Cosine² Pulse

The ideal cosine² pulse, shown in Fig. 2, corresponds to spectrum (b) in Fig. 4. It has been pointed out (Section 4.2) that the closest pulse spacing which can be employed without reducing the noise margin is exactly $2T (= 182 \text{ ns})$. The width t_s of the time slot required by an n -digit pulse group is given by:

$$t_s = (n - 1)s + 4T$$

where s is the pulse spacing. For $n = 21$ and $s = 2T$

we have $t_s = 44T (= 4 \mu\text{s})$ which just fills the available time-slot within the sync period. In this case the effective modulation depth is 100%; this is shown in Fig. 7(a) which illustrates the formation of an eye-pattern with an amplitude $E = 100\%$.

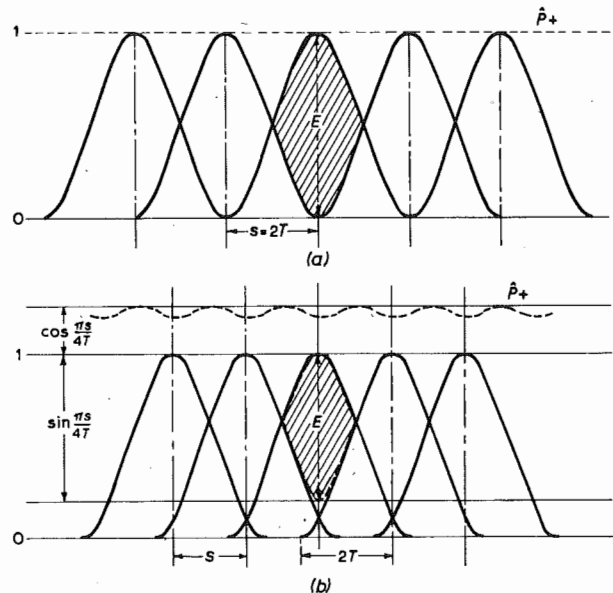


Fig. 7 - Illustrating the formation of discrimination eye-patterns with cosine² pulse groups
eye-patterns represented by shaded areas
(a) $s = 2T$ (b) $s < 2T$

In practice the cosine² pulse is formed by impulse-driving a frequency-shaping network and this inevitably introduces a delay of approximately $2T$. It is therefore convenient from a practical standpoint to reduce the duration of the pulse group by about 200 ns, i.e. $t_s = 3.8 \mu\text{s}$; this requires a reduced pulse spacing $s = 1.89T (172 \text{ ns})$ with a 20 ns pulse overlap. For unit pulse amplitude, the peak height \hat{P}_+ of the pulse group occurs at each main crossover point and is given by $1 + \cos \pi s / 4T$, i.e. 1.09; this is illustrated diagrammatically in Fig. 7(b). At the peak point of a wanted pulse the interfering skirts of the two adjacent pulses can add to give a total crosstalk $1 - \sin \pi s / 4T$, i.e. 0.004. The 'discrimination eye-pattern' amplitude

E is therefore 0.996/1.090 or 91% of the peak amplitude of the group.

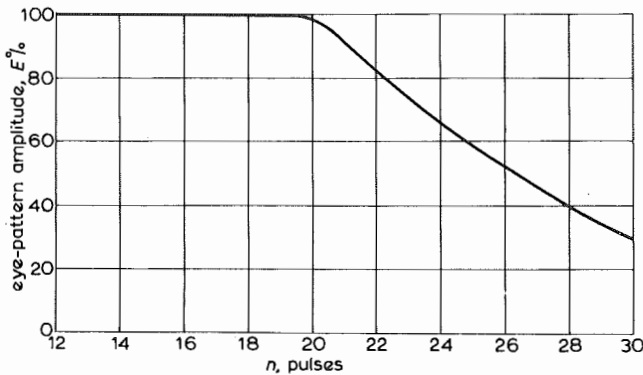


Fig. 8 - Variation of discrimination eye-pattern amplitude with number of pulses per sync-slot ($t_s = 3.8 \mu s$)

Assuming a fixed pulse-group duration of $3.8 \mu s$, Fig. 8 shows the variation of the discrimination eye-pattern amplitude E with the number of pulses n contained in the group; for $n = 27$, i.e. one marker pulse plus two 13-digit words, this amplitude is only just below 50%. In interpreting these results it must be borne in mind that channel degradations will further reduce the size of the eye-pattern. For example, it can be shown that with a channel K_2T -rating of 2% the eye-pattern amplitude E of a 21-pulse group falls from 91% to a minimum of 50% after transmission; i.e. the presence of other impairments reduces the immunity of the system to noise (and vice versa).

7.4. Intermediate Pulse Shapes

Intermediate pulse shapes were investigated with the aid of a computer programme* which calculated a number of functions obtained by convolving the rectangular spectrum of a $\sin x/x$ pulse with that of a Gaussian function, $\exp(-f^2/2\sigma^2)$, the latter being a close approximation to the spectrum of a cosine² pulse; the variable parameter was the relative width (i.e. standard deviation σ) of the Gaussian spectrum. One of these pulses, the form of which appeared to show a good compromise between duration of subsidiary lobes and main pulse width, and which also gave an acceptable eye-pattern, will now be examined. The pulse shape is shown in Fig. 9 and the corresponding spectrum by curve (c) in Fig. 4. It will be seen that the spectrum is broader than that of the cosine² pulse and that the main pulse width is significantly narrower at the expense of producing about three significant minor lobes on each side of the pulse; the optimum pulse spacing is $1.26T$ (115 ns) since this allows synchronous detection of the pulse group.

The maximum positive and negative signal peaks of a 21-pulse group with the above pulse shape are given by considerations similar to those outlined for the $\sin x/x$ pulse group (Section 7.2); in this case, however, only two positive and four negative minor lobes per pulse need be considered.

* by W.K.E. Geddes.

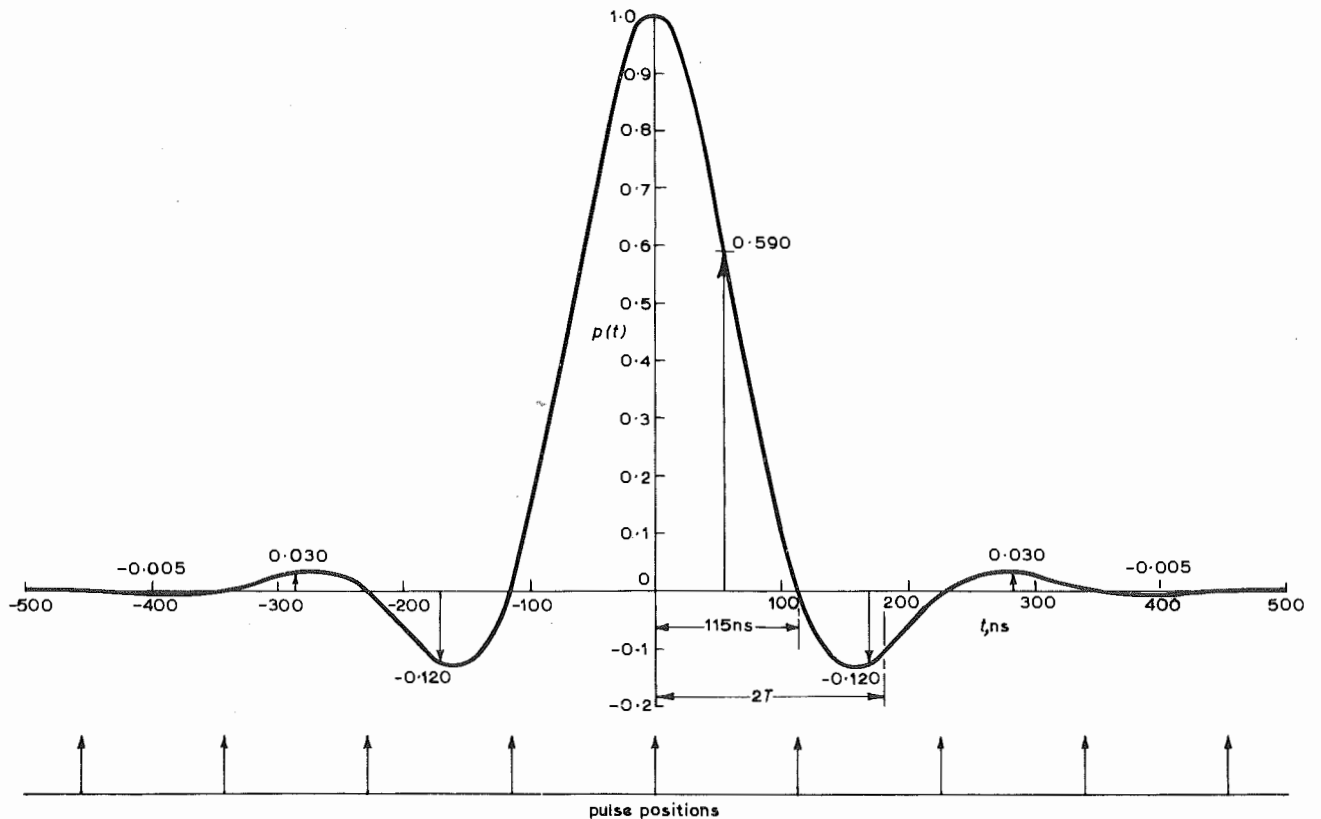


Fig. 9 - An intermediate pulse shape obtained by spectrum convolution
 $2T = 182 \text{ ns}$

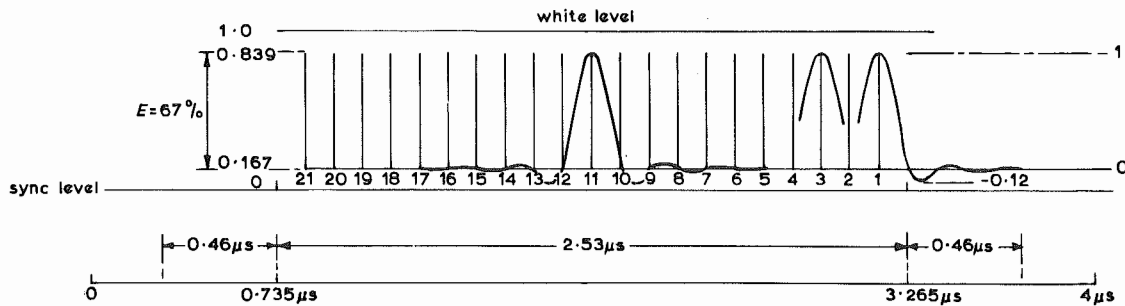


Fig. 10 - Time-amplitude diagram for 21-pulse group with intermediate pulse shape

Referring to Fig. 9 and recalling Fig. 5 it should be evident that the following 8-digit code sequences correspond respectively to the positive and negative signal peaks:

----- 01011010 ----- Positive
 and ----- 10100101 ----- Negative

Such short code sequences would have a high probability of occurrence within a 21-pulse group. The peak-to-peak amplitude of the group is given quite accurately by the summation (Fig. 9):

$$2(0.590 + 0.030) + 2(0.120 + 0.005) = 1.49$$

The pulse amplitude at the points where there is maximum discrimination against noise is only 1.0, however, so that restricting, as before, the peak-to-peak amplitude of the group to the peak video excursion gives an eye-pattern amplitude of $1/1.49$ or 67%; this is almost twice that provided by the $\sin x/x$ pulse group but only three-quarters of that provided by a \cos^2 pulse group.

The major lobes of 21 of these intermediate pulses would occupy 22×115 ns or $2.53 \mu\text{s}$ as shown in Fig. 10. Minor lobes are not very extensive however; a single pulse at the end of the group has negligible minor lobe amplitude at the extremities of the line-synchronizing period; in fact these lobes become insignificant beyond a point four pulse intervals away from the ends of the pulse group. We therefore allow an additional $0.46 \mu\text{s}$ at the end of the main group so that the effective duration of the pulse group is about $3.45 \mu\text{s}$.

In comparison with the \cos^2 pulse group, the intermediate pulses give an economy of $0.55 \mu\text{s}$ in pulse-group duration but this has been achieved at the expense of modulation depth which is reduced from 91% to 67%. It is worth noting that with the same group duration of $3.45 \mu\text{s}$, \cos^2 pulses can be employed provided that the spacing is reduced from $2T$ to $1.7T$ (155 ns); crosstalk between pulses would then limit the effective modulation depth to 72% but this figure would still be greater than that given by the intermediate pulse and could be achieved with a 23-digit \cos^2 pulse group (1 marker plus two 11-digit words) requiring a total duration of $3.8 \mu\text{s}$.

This result is typical of all pulse shapes which tend towards \cos^2 shape and indicates that no appreciable advantage is obtained by using pulses narrower than $2T$ - \cos^2 even if minor lobes of small magnitude can be permitted outside the duration of the main pulse group.

7.5. Conclusions

Theoretical considerations have justified the choice of the \cos^2 pulse for the 10-bit p.c.m. sound-in-video syncs system. This pulse shape provides a greater noise immunity than the $\sin x/x$ function or pulses of intermediate shape without producing sound pulse interference outside the allotted period. A subsidiary but important advantage of using this pulse is that it is identical to the standard $2T$ -test pulse; the K -rating of the test pulse could therefore provide an immediate objective measure of system performance with degraded video channels.