BBC RD 1973/42

1969
THE QUEEN'S AWARD
TO INDUSTRY

RESEARCH DEPARTMENT REPORT

DIGITAL SOUND SIGNALS:
multiplexing six high-quality sound channels for transmission at a bit-rate of 2.048 Mb/s

M.G. Croll, B.Sc., A.R.C.S.
D.W. Osborne, C.Eng., M.I.E.E.
D.F. Reid, B.Sc.

Research Department, Engineering Division
THE BRITISH BROADCASTING CORPORATION

December 1973

BBC RD 1973/42
UDC 534.86
621.376.56

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M.G. Croll, B.Sc., A.R.C.S.
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Summary

This report describes a way in which six companded p.c.m. high-quality sound channels could be multiplexed to form a bit-rate of 2·048 Mb/s, including error protection and about 1 kb/s of signalling information. The predicted performance of such a system is discussed and compared to that of the existing 13-channel multiplex system which employs linear coding.

The proposed system uses a near-instantaneous digital companding technique that permits the number of bits in each audio sample word to be reduced from 13 to 10.

With this proposed 6-channel system, the sound quality would be virtually indistinguishable from that achieved with the existing 13-channel multiplex and the performances of the two systems in the presence of errors would be similar. The only significant difference between the two systems would be that, with the 6-channel system, the mean time taken to re-frame after a break in transmission would be about 6½ ms compared to about ½ ms for the 13-channel system. However, this is considered to be acceptable for most applications of such a 6-channel system.

Issued under the authority of

Head of Research Department

Research Department, Engineering Division,
BRITISH BROADCASTING CORPORATION

December 1973

(EL-85)

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DIGITAL SOUND SIGNALS: MULTIPLEXING SIX HIGH-QUALITY SOUND CHANNELS FOR TRANSMISSION AT A BIT-RATE OF 2·048 MB/S

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1. Introduction

The p.c.m. multiplexed sound system, in service with the BBC for distributing sound signals from the London studios to the main v.h.f./f.m. transmitters, uses a linear p.c.m. code[1] and generates 13 bits from each audio sample. In this system 13 sound channels are multiplexed together to form a bit-rate of 6·336 Mb/s which includes framing and parity check information.

Recently, techniques of bit-rate reduction have been investigated[2] which might permit more sound channels to be multiplexed together within a given bit-rate. In this investigation, 'near-instantaneous' digital companding was found to give virtually no audible impairment of the sound quality when the number of bits per sample was reduced to about 10, using a 4-segment companding law. It is therefore considered that this companding technique could be be used for high-quality sound transmission systems where a worthwhile economy results from the lower bit-rate.

The proposed near-instantaneous digital compandor is described fully in Reference 2 but can be briefly described here as a device in which only 10 bits of the original 13-bit linearly-coded signal are selected for transmission. This selection is made on the basis that the crest signal magnitude occurring within a specified interval of time must be accommodated. If the crest signal magnitude is within 6 dB of the system overload point then the three lowest significant digits will be omitted and only the ten most significant digits transmitted. For lower crest signal magnitudes some of the most significant digits and some of the least significant digits will not be selected for transmission. The exact position, within the original 13-bit words, of the ten bits that are selected for transmission is communicated to the receiving terminal using a separate two-bit scale-factor word which is transmitted once for each time interval.

Error protection schemes for such a system have been investigated[3] and it has been found to be more immune to digit errors than the 13-bit linear system. Hence a ratio of only about 1 parity check digit per two sample words has been proposed for the near-instantaneous companded signal whereas it is necessary to use 1 parity check digit for each sample word of linearly-coded signals. This gives a further reduction in the overall bit-rate required per channel in a multiplexed sound system where companding is employed.

The overall bit-rate per channel for a near-instantaneous digitally-companded signal is 336 kb/s plus a small addition for the scale-factor word and its error protection. To multiplex six such channels into a total bit-rate of 2·048 Mb/s allows 32 kb/s for the scale-factor word with its error protection, framing synchronisation and additional low bit-rate signalling information.

The importance of 2·048 Mb/s is that it is the first-order multiplex bit-rate chosen by the British Post Office (BPO) and digit transmission capacity for this standard will be available in most parts of the UK when the BPO plans for its digital communication network have been implemented.

A system for transmitting six high-quality sound programmes, or three stereo pairs, would be particularly convenient for the BBC as a means of feeding the three main programmes, in stereo, to v.h.f./f.m. transmitters. A possible requirement for such a system has already arisen for feeding programmes from Glasgow to Kirk O'Shotts.

The 2·048 Mb/s standard is also to be adopted by European countries. If agreement could be reached also on a form of coding, the international exchange of sound programmes would be greatly facilitated and it is suggested that the system described in this report might be acceptable as such an international standard.

This report describes the proposed 6-channel multiplex sound system and compares estimates of the performance of such a system with the performance of the existing 13-channel sound multiplex system.

2. A proposed frame structure

As discussed in the Introduction, the total bit-rate that can be allowed for transmission of the scale-factor words (with error protection), frame synchronisation and signalling information is only 32 kb/s. This can be compared with the 320 kb/s which was allowed for synchronisation information alone in the 13-channel linear coding system, where the total transmitted bit rate of 6·336 Mb/s is a little over three times 2·048 Mb/s. Hence each parameter must be considered carefully and its allocation of bits reduced until service requirements are just met.

To derive the proposed frame structure each parameter has been traded against other parameters to fit the signals into a total bit-rate of 2·048 Mb/s. The result is shown in Table 1. The performance and specifications for each parameter are discussed separately in the following Sections.

For simplicity only one-sixth of a complete multiplexing frame is shown in the Table. The complete frame has a 12-bit framing pattern and 6 scale-factor words (one for each channel).

TABLE 1

The Proposed Frame Structure

Frame length	320 bits
comprising:	
30, 10-bit sample words (5 per channel)	300 bits
Parity bits for the sample words	15 bits
Framing bits ($^1/6$ of framing pattern)	2 bits
Scale-factor word (for one channel)	2 bits
Parity bit for scale-factor word	1 bit

In Table 1 no provision has been made for signalling information; methods of accommodating this will be discussed later.

3. Sound quality for each channel

The near-instantaneous 4-segment, 10 bits per sample digital compandor using pre- and de-emphasis having the shape of the CCITT characteristic,[4] was assessed[2] as giving a quality virtually indistinguishable from that obtained with 13-bit linear coding. This assessment was made by critical observers listening to very critical programme material and comparing the two systems by switching between them.

These quality-appraisal tests were carried out with intervals between scale-factor words of 16 or 24 sample words. This change of scale-factor interval made no perceptible difference but, with the proposed frame structure shown in Table 1, the interval would have to be increased to 30 words.

Also, whereas the CCITT pre- and de-emphasis characteristic has been found very effective in reducing the impairment likely to be caused by digital compandors[2] (programme-modulated quantising noise), there would be operational advantages in using instead a 50 μs pre- and de-emphasis characteristic. This is the characteristic used at v.h.f./f.m. transmitters and has been adopted, for convenience, for the 13-channel linearly-coded p.c.m. system. If the 50 μs pre- and de-emphasis characteristic could also be adopted for the 6-channel system, interfacing between systems and with transmitters would be simplified.

To determine whether the scale-factor word interval could be increased to 30 words and whether 50 μs pre- and de-emphasis could be used instead of the CCITT pre- and de-emphasis, a short series of subjective tests was conducted.

The tests were carried out in a listening room of 85 cubic metres having a mid-band reverberation time of 0·3 s and using a high-quality BBC type LS5/5 monitoring loudspeaker. In these tests twelve observers, most of whom were experienced in subjective sound system testing, were asked individually to switch between presentations A and B and to assess the difference using the 7-grade comparative scale shown opposite.

Comparison	Grade
A much better than B	+3
A better than B	+2
A slightly better than B	+1
A equal to B	0
B slightly better than A	−1
B better than A	−2
B much better than A	−3

The programme material used was a low-noise recorded tape loop of a 5-note piano scale which had been used in the general investigation of the performance of compandors.[2]

The results of these tests are shown in Table 2, in which an extract from an earlier series of tests[2] using the same observers has also been included.

These results show that no significant additional impairment is produced when the scale-factor word interval is increased from 16 words to 30 words. Surprisingly the mean results from presentation 1 show a slight preference for the 30-word interval rather than the 16-word interval. However, the results from presentations 3 and 4 (the latter being an extract from previous tests) could be taken to mean that the 16-word interval is preferred. These differences are small compared with the standard error of the mean and it can be reasonably assumed that increasing the interval by this amount causes no measurable impairment.

The mean result for presentation 2 shows a significant preference for use of the CCITT pre- and de-emphasis characteristic rather than the 50 μs pre- and de-emphasis characteristic. It was therefore concluded that, to maintain the quality of the companded sound channel as high as possible, the CCITT pre- and de-emphasis characteristic must be used. This has a bearing on the design of an operational companding system which must be compatible with the existing linearly-coded p.c.m. system. Such interfacing problems are discussed later in Section 8 of this report.

4. Frame synchronisation

4.1. General

In any multiplex system there is a fundamental limit to the minimum re-framing time that is determined by the duration of the frame itself. In the proposed 6-channel system the length of one complete frame may be deduced from Table 1 to be 320 x 6 = 1920 bits. It follows that, at the bit-rate of 2·048 Mb/s, the duration of a complete frame will be about 1 ms. So, regardless of how many bits are used for framing, the minimum framing time would be of this order. Because the number of bits available for frame synchronisation is not optimum, the re-framing time will be even longer than this.

The existing 13-channel 6·336 Mb/s multiplex system has a re-framing time of only ½ ms. However, operational experience with this system has shown that transmission

TABLE 2

Subjective Test Results

Presentation	A	B	Mean grade	Std. error of mean
1	4-segment, near-instantaneous companding law. 30-word interval between scale-factor transmissions. CCITT pre- and de-emphasis.	4-segment, near-instantaneous companding law. 16-word interval between scale-factor transmissions. CCITT pre- and de-emphasis.	+0·18	0·12
2	4-segment, near-instantaneous companding law. 16-word interval between scale-factor transmissions. 50 μs pre- and de-emphasis.	4-segment, near-instantaneous companding law. 16-word interval between scale-factor transmissions. CCITT pre- and de-emphasis.	−0·74	0·14
3	13-bit linearly-coded system. No pre-emphasis.	4-segment, near-instantaneous companding law. 30-word interval between scale-factor transmissions. CCITT pre- and de-emphasis.	+0·46	0·21
4 (Extract from previous series of tests)	13-bit linearly-coded system. No pre-emphasis.	4-segment, near-instantaneous companding law. 16-word interval between scale-factor transmissions. CCITT pre- and de-emphasis.	+0·36	0·26

breaks which cause re-framing are infrequent, even though the signal is carried on microwave links which are prone to fading in certain propagation conditions. Moreover, since the 6-channel 2·048 Mb/s signal would probably be carried on cables, any breaks in transmission would be caused by equipment faults. A relatively long re-framing time is therefore considered to be acceptable for the proposed 6-channel system.

4.2. Re-framing time

The average re-framing time is defined as the average time taken by a digital system to regain frame synchronisation after there has been a break in transmission. This can be subdivided into the time taken by the receiving terminal to verify that there has been a synchronisation slip, plus the time taken to re-synchronise.

In a simple frame synchronisation system the receiving terminal would wait until several, say four, framing patterns had been missed before verifying that a synchronisation slip had occurred. This technique avoids unnecessary attempts at reframing when short disturbances or errors cause some of the bits of the framing patterns to be misinterpreted at the receiving terminal. With such a system applied to the 6-channel multiplex frame structure proposed in Section 2, it is estimated that the time taken to verify a synchronisation slip would be about 4 ms.

Methods of calculating the time taken to regain synchronisation once a slip is verified have been described by Häberle[5] and for the system proposed here it is estimated that the average time would be about 2 ms.

Hence the average re-framing time for a simple synchronisation system would be about 6 ms.

The re-framing time could be reduced by means of more sophisticated synchronisation systems. For example, the time would be approximately halved if two circuits were used to check for framing patterns. If the first circuit misses one framing pattern the second circuit could start examining the data. Synchronisation would then be taken from the second circuit if it finds two framing patterns, the correct distance apart, when no further framing patterns have been found by the first circuit. With such techniques the average re-framing time could probably be reduced to 2 − 3 ms depending on the exact form of the instrumentation.

5. Error protection

The effect of errors on a near-instantaneous companded signal has been investigated and is the subject of a separate report.[3] It was found that such companded signals were more immune than linearly-coded signals to the effects of digit errors. In the course of this work, error protection schemes were also investigated and one was recommended for the proposed near-instantaneous companded signal.

The recommendation was, firstly that each 2-bit scale-factor word should be protected by a single parity bit, erroneous scale-factor words being replaced by the previous correct ones. Secondly, one parity bit should be used to protect two sample words, taken from different channels. Each of these sample-word parity bits should protect a group of five digits, comprising the sign digit and the two most significant digits from one sample-word and the sign digit and the most significant digit from the other sample word. Error concealment for the sample words would be obtained by repeating the previous correct sample in each of the two channels associated with the particular parity group in error. This arrangement for parity checking the sample-words is necessary to enable signal failure conditions to be detected. With odd parity applied to an odd number of digits, if the signal fails to all 'ones' or all 'zeros', parity is violated and an error condition is detected. Action can then be taken to restore the service if the signal-failure condition persists.

The recommended error-protection system has been assessed subjectively[3] in the presence of transmission errors as giving a performance similar to that of the 13-channel p.c.m. multiplex system which employs linear coding. With the latter system one parity check digit is used to protect the five most significant digits of each sample-word.

6. Signalling information

6.1. General

BBC operational requirements are such that a total of not more than 1 kb/s should be available within the 2·048 Mb/s stream for the transmission of signalling information. Reference to Table 1 shows that there are no spare bits available for this. The signalling information must therefore be conveyed using, for a small percentage of the time, bits within the frame which would normally be used for other purposes.

Methods of deriving capacity for simultaneous signalling in digital sound circuits have been investigated.[6] Although this investigation was mainly applicable to the 13-channel system, the main recommendations have been considered for application to the 6-channel system proposed in this report. These were: (1) occasional use of the framing pattern or (2) use of the lower-significance digits of the sample word. The latter technique is not considered suitable for digitally-companded signals as the penalty of increased programme-modulated quantising noise would probably be too great an impairment of the quality of the sound signals.

For the proposed 6-channel system, only methods which would not impair the quality of the sound signal under normal conditions have been considered. Hence the only bits which are available are the framing bits (use of these tends to lengthen the re-framing time) and the parity bits (use of these might slightly impair the performance in the presence of errors).

6.2. Use of frame synchronisation bits

In the proposed 6-channel system the frame synchronisation bits occupy 12 x 2048/1920 = 12·8 kb/s of the transmitted bit-rate. A fraction of this bit-rate could be used for signalling in such a way that the average re-framing time is affected in an inverse proportion. Hence if 1 kb/s were used for signalling information the average re-framing time might be lengthened from 6 to 6·5 ms for a simple synchronisation system. To do this, the time slot for an entire 12-bit synchronisation word would be used for signalling at intervals corresponding to every 12th complete frame.

An advantage of this method is that the additional instrumental requirements are minor. For example, the simple frame synchronisation circuits, as used in the 13-channel system, ignore losses of the framing pattern unless four such patterns are missed in succession. Hence the normal process of retaining synchronisation would not be significantly affected.

6.3. Use of sample word parity bits

In the proposed 6-channel multiplex system the parity bits for the sample words correspond to a rate of 6 x 15 x 2048/1920 = 96 kb/s. Therefore to derive, from these, a signalling capacity of 1 kb/s would leave about 1% of all sample words unprotected.

The effect this would have on the error performance of the system would be to increase the proportion of undetected errors when transmission errors are present in the received data signal. At high bit error probabilities, above about 10^{-3}, the number of errors passed in the 1% of sample words that are not protected would be less than the number of double errors passed undetected in the six-parity groups. Hence, in this case, there will be no significant increase in the audible impairment caused by errors. At bit-error probabilities lower than about 10^{-5}, undetected errors in sample words will occur less often than about one every three minutes, and, bearing in mind that such errors in digitally companded signal are less audible than those in linearly-coded signals, it can be assumed that these would not cause a significant impairment of the programme. At bit-error probabilities between 10^{-5} and 10^{-3} it is likely that the error performance of the system would be degraded. However, it may be thought that an impairment over this range of error rates could be tolerated and might be preferable to an increase in the re-framing time.

One method of using sample-word parity bits to derive the required signalling capacity would be to allocate three sample-word parity bits from every three complete frames. The first parity bit would be deliberately inverted so that parity is violated for the associated two channels; the two parity bits associated with the other four channels, would then be used for the signalling information. At the receiving terminal this particular group of three data bits can be recognised, removed and replaced with the correct parity bits so that the normal error-concealment procedure is not applied to the associated sample words. At switch on, or after a loss of synchronisation, the receiving terminal can be designed to test all parity violations for signalling information and to treat the violations as transmission errors. When several groups of signalling bits have been recognised the receiver can then be set to operate the normal error-concealment process for parity violations and to correct only those which are being used for signalling. With this technique there would be a small additional impairment of the sound signal in each channel after re-framing arising from several sample-word concealments, which would last a few tens of milliseconds.

6.4. Recommendations for transmission of signalling information

Each of the two systems considered would cause some impairment of the performance of the 6-channel system; either the re-framing time would be increased or the performance in the presence of errors would be somewhat degraded. Neither impairment is particularly serious and either could probably be tolerated in practice. Unless there is some particular reason why a small increase in the re-framing time of the system could not be accepted, it is recommended that the frame synchronisation bits be used for transmitting signalling information on the basis that this technique would be the easier to implement.

7. Multiplex equipment

Methods of multiplexing digitally-coded signals are well-known and are described elsewhere.[7] However, techniques have advanced recently because many of the logic functions required are now available in single Large-Scale Integration (L.S.I.) or Medium-Scale Integration (M.S.I.) circuits where previously many Small-Scale Integration (S.S.I.) circuits had to be used. Also L.S.I. and M.S.I. circuits can be operated at faster speeds than the equivalent array of S.S.I. circuits. Hence many operations on digitally-coded sound signals which were previously carried out using a number of parallel paths could now be performed on a serial bit stream, leading to considerable simplification of the multiplexing equipment.

To multiplex together 6 digitally-companded sound signals to form a transmitted bit-rate of 2·048 Mb/s, it is considered possible to construct a system where the 6 linearly-coded signals are first multiplexed to form a serial bit stream which is then applied to one digital compressor circuit. Such an arrangement would require the use of only one expandor circuit at the receiving terminal. Thus the cost, per channel, of digital companding could be substantially lower than if separate digital compandors were applied to the individual channels. Also, because fewer circuits are involved, the reliability would be higher.

Block diagrams of a system where one compandor circuit is used for six channels are shown in Figs. 1 and 2. Fig. 1 shows a possible multiplexing arrangement and Fig. 2 shows the corresponding demultiplexer. The basic companding action for each channel is similar to that described in Reference 2 for a single-channel system. The full 6-channel system is shown together with provisions for error protection, synchronisation and the transmission of signalling information. The reason for the use of multiple pre- and de-emphasis networks at each analogue input to the equipment is discussed in Section 8.

8. Interfacing with existing 13-channel systems and transmitters

8.1. Audio limiting and pre-emphasis

As discussed in Section 3, it is necessary to use the

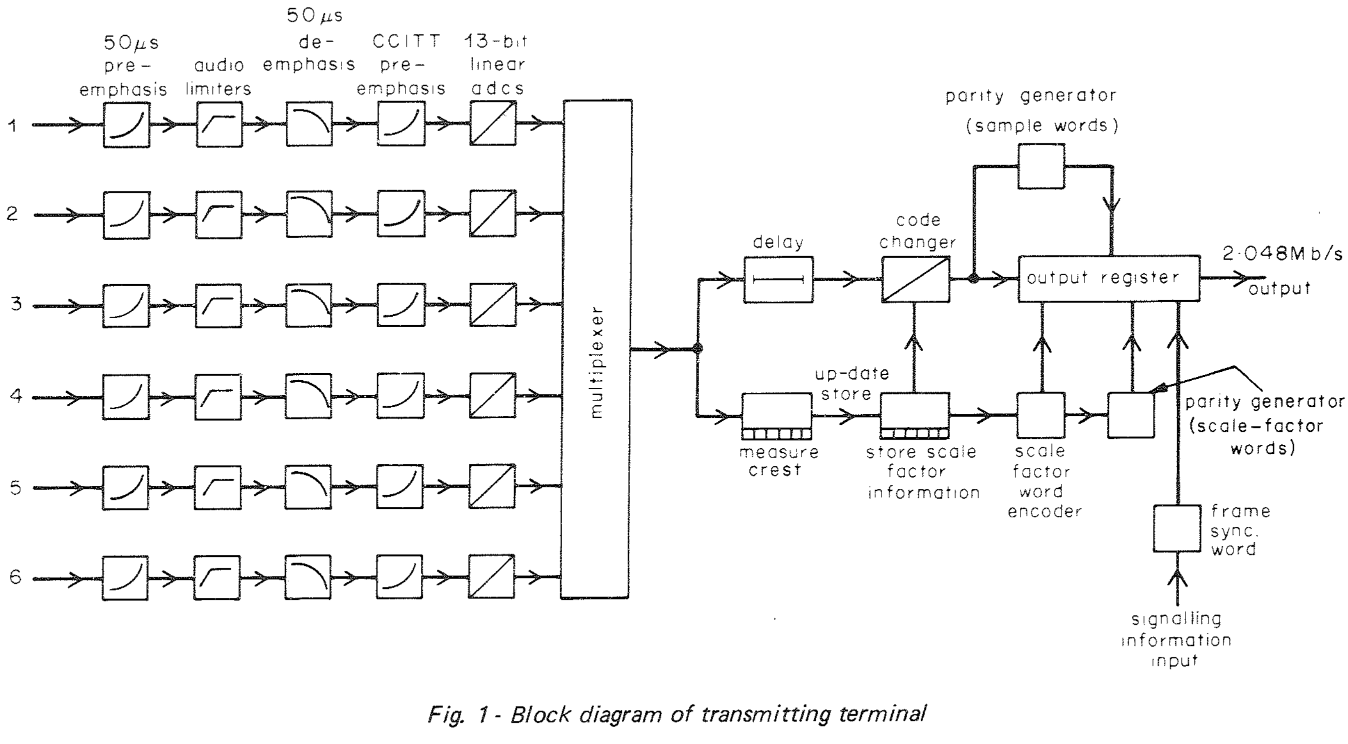

Fig. 1 - Block diagram of transmitting terminal

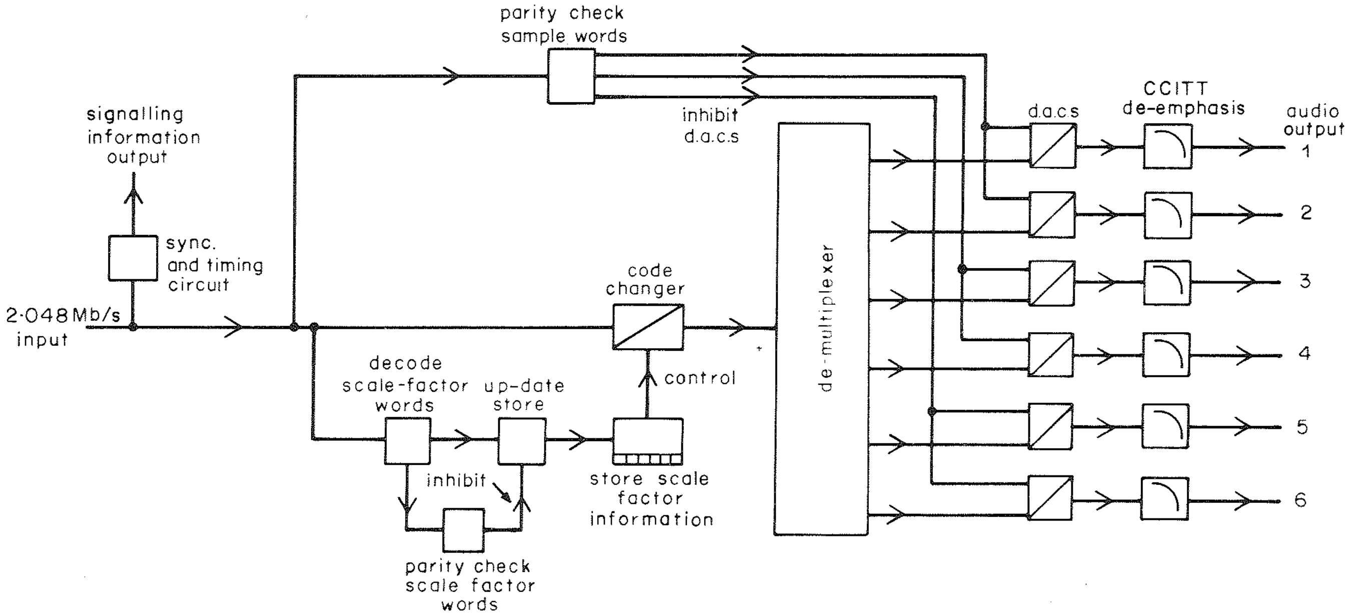

Fig. 2 - Block diagram of receiving terminal

CCITT pre-emphasis characteristic in the proposed 6-channel companded system. However, the 50 μs characteristic is used in the existing 13-channel p.c.m. distribution network and v.h.f./f.m. transmitters.

Furthermore, it is desirable in the interest of economy, and in order to avoid the unpleasant audible effects that result from using a number of audio-frequency limiters in tandem, that only one limiter should be used in any programme channel. This limiter should protect all of the following equipment, including the transmitter, from over-modulation.

The system proposed to satisfy these requirements is illustrated in Fig. 1.

Prior to its entry into the first analogue-to-digital converter, the analogue signal passes through a 50 μs pre-emphasis network followed by an audio-frequency limiter. Provided that the channel gain is correctly adjusted, this is all the spectrum shaping and amplitude limiting that the signal requires for transmission by the 13-channel linear p.c.m. system and for application to a v.h.f./f.m. transmitter.

For transmission by the 6-channel companded system, this pre-emphasised and limited signal requires further processing, provided in Fig. 1 by the 50 μs de-emphasis and CCITT pre-emphasis networks following the audio limiter, and an adjustment of its level.

The level adjustment required is illustrated in Fig. 3. In terms of the signal chain shown in Fig. 1, it is assumed that the nominal peak level of the incoming programme before 50 μs pre-emphasis, the maximum permissible peak input to the a.d.c., and the threshold of limiting at the input to the audio limiter, are all identical, also that the limiter has unity gain for inputs below this threshold. At frequencies below that at which pre-emphasis is effective, say, 100 Hz, the 50 μs pre-emphasis network is required to have an insertion loss of 2 dB and the combination of 50 μs de-emphasis and CCITT pre-emphasis to have an insertion loss of 11·6 dB.

It may appear from Fig. 3 that the reduction of level of low modulation frequencies required by the use of CCITT pre-emphasis entails a considerable noise penalty in the 6-channel system. This is, in fact, not so. The weighted noise level, for low-level low-frequency signals, of the 6-channel system would be within 1 dB of that of the 13-channel system which employs 50 μs pre-emphasis.

8.2. Analogue interfacing

In this case it is assumed that the decoded analogue signal from one system is to be connected to the coder of another system or the analogue output from the 6-channel system fed to a transmitter. No special problems arise and a simple filter would be required to convert from 50 μs pre-emphasis to CCITT pre-emphasis with a low-frequency insertion loss of 11·6 dB or vice versa. Therefore if each complete audio channel includes, at the first digital encoder, a limiter incorporating the 50 μs pre-emphasis characteristic, basic interfacing requirements are simply that the correct frequency responses and gains be set. Each digital circuit would contribute an almost equal amount of noise and four such circuits could be used in tandem whilst retaining an overall peak signal to peak weighted noise ratio of 63 dB.

8.3. Digital interfacing

Occasions might arise when it is desirable to interface digitally between the proposed 6-channel system and the 13-channel system. Such direct digital coupling would remove the need for an a.d.c. and d.a.c. for each channel to be interfaced but would require that conversion from one pre-emphasis characteristic to the other be done digitally.

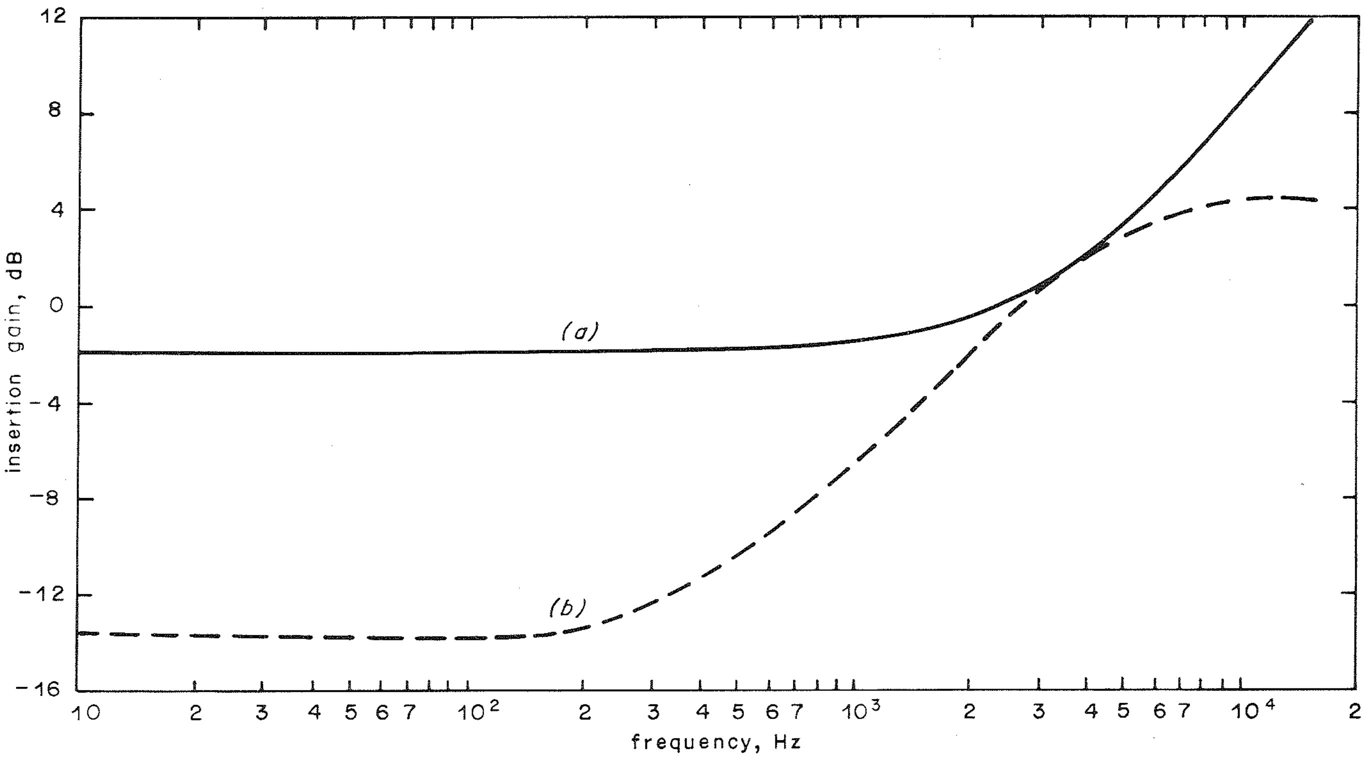

Fig. 3 - Pre-emphasis characteristics

(*a*) 50 μs characteristic as used in existing 13-channel system and at v.h.f. f.m. transmitters

(*b*) CCITT characteristic as proposed for use with the 6-channel system

An investigation[8] has been carried out to assess the feasibility of the necessary digital filter and has concluded that a recursive digital filter of second order would generate the required response with sufficient accuracy and would be economic to provide.

Each digital interface, using such a digital filter, would contribute quantising noise to each audio channel as if the signal were decoded and re-coded. The reason for this is described in Reference 8 and is basically that within the filter more bits per sample word would be generated than the 13 applied to the input. However, when the signal is rounded back to 13 bits per sample at the output this truncation process is equivalent to re-quantisation and thus contributes quantisation noise as in digital coding to 13-bit accuracy.

9. Conclusions

A system to multiplex 6 high-quality sound channels based on a form of near-instantaneous digital compandor could be constructed to generate a bit-rate of 2·048 Mb/s including error protection, synchronisation information and provision for transmitting auxiliary signalling information at a bit-rate of about 1 kb/s. The only significant difference in performance between this and the existing BBC 13-channel multiplex p.c.m. system which uses linear coding, is that the proposed 6-channel system would take, on average, about 6½ ms to regain frame synchronisation after a break in transmission; the 13-channel system requires only about ½ ms. However, 6½ ms is not considered to be an unacceptably long re-framing time, particularly since transmission breaks in a cable 2·048 Mb/s circuit are likely to be rare. Moreover, this time could be reduced by using more sophisticated re-framing circuits.

Although the pre-emphasis characteristic required for the 6-channel system is not the same as that used in existing 13-channel multiplex equipment, there would be no fundamental difficulty in coupling from one system to another (interfacing). In analogue interfaces, simple filters could be used to convert from one pre-emphasis characteristic to the other; for digital interfaces it is considered feasible to use a single, equivalent digital filter which could be inserted in the serial bit stream.

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SMW/VK