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NICAM 3: near-instantaneously companded digital transmission system for high-quality sound programmes

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SUMMARY

The paper describes a system, known as NICAM 3 (Near-Instantaneously Companded Audio Multiplex, Mark 3), which is being developed by the BBC for the transmission of sound programmes on digital circuits designed for multi-channel telephony.

Subjective tests which led to the choice of this system are referred to, and the system design, which is based on a compromise between efficiency and flexibility, is discussed in some detail.

Various applications are mentioned, the main one being for six channels in a dedicated bit-stream of 2048 kbit/s, but other arrangements are possible, including mixed telephony and sound-programme working.

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1 Introduction

Throughout the world, digital transmission is being used increasingly for telephony, with the result that analogue circuits will in time become scarce or even non-existent. For this reason, and also to obtain improved performance, work is being carried out by various organizations including the BBC, with the aim of using the new digital circuits for sound-programme transmission. The CCITT has recommended[1] several different bit-rates for international transmission, the primary and secondary rates favoured in Europe being 2048 kbit/s and 8448 kbit/s.

In order to make the most efficient use of the available bit-rate most of the systems currently being proposed use some form of companding, i.e. compression at the sending end and expansion at the receiving end of the digital circuit. In systems using companding the output noise depends on the signal level, the noise being least for the lowest signal levels, and highest when the signals reach their maximum levels. This may be acceptable owing to the masking effect of high-level signals, but the audibility, and hence the annoyance value, of this programme-modulated noise varies with the particular system, depending on such things as compression ratio and pre- and de-emphasis. Companding may be employed on both analogue and digital audio transmission systems, and the form of companding used may be either analogue or digital.

The BBC has had considerable experience of digital sound-programme transmission over the last 10 years.[2,3] Accordingly one aim for the new development was that the overall quality should be at least as good as that currently being obtained from a 13-bits-per-sample, linearly-encoded p.c.m. system, designed nearly a decade ago, and used since 1972 for the network distribution of radio sound-programmes. With this limitation the most efficient use of the available bit-rate has been sought, bearing in mind such things as error rates which may be expected on digital circuits, and the wish to keep the instrumentation reasonably simple.

2 Consideration of Companding Laws

Companding systems currently being proposed are either instantaneous or near-instantaneous.

2.1 Instantaneous Companding

In instantaneous companding each sample digital word from the analogue-to-digital converter is changed to another word with fewer bits. The transfer characteristic, representing the companded quantizing level as a function of the magnitude of the input signal, is a segmented straight-line graph, adjacent quantizing levels being equivalent to smaller signal level differences at low input levels than at high input levels.

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2.1.1 *13-segment A-law*

An instantaneous companding law which is used for telephony is known as the A-law.[4] This has a segmented form which approximates to the curve defined by:

$$y = \frac{1+\ln(Ax)}{1+\ln A}, \quad \text{for } \frac{1}{A} \leqslant x \leqslant 1,$$

$$y = \frac{Ax}{1+\ln A}, \quad \text{for } 0 \leqslant x \leqslant \frac{1}{A}.$$

A value for A of 87·6 has been selected as optimum for telephony,[5] and the straight-line transfer characteristic approximation has thirteen segments, as shown in Fig. 1. The transfer characteristic is skew-symmetrical about zero, the positive quadrant only is shown here.† The finest discrimination, i.e. the difference between adjacent quantizing levels for very low signals, is equivalent to that which would be obtained with a linearly-coded n-bit system. For telephony, $n = 12$ is used, and hence 8 bits are actually transmitted, but for sound programme use, $n = 14$, which in at least one proposal, is reduced to 10 bits for transmission. It may be seen that in this case the highest level input signals are effectively coded to 8 bits per sample.

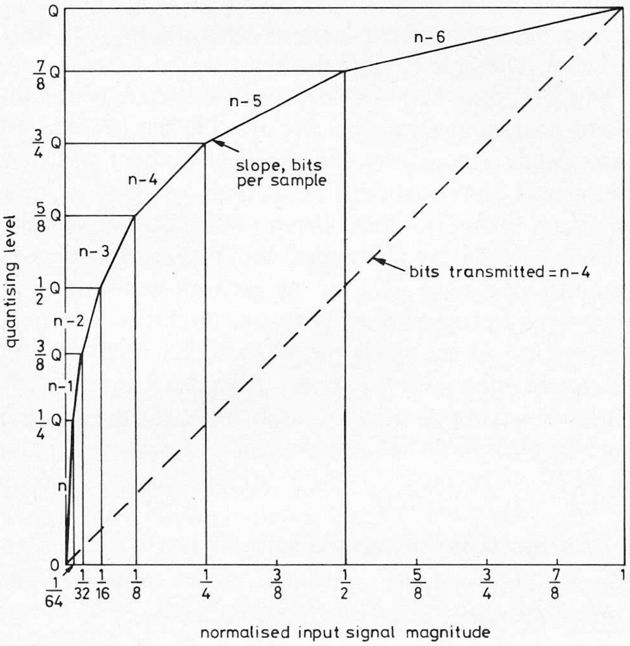

Fig. 1. 13-segment A-law.

2.1.2 *11-segment A-law*

Another approximation to the A-law curve has been proposed for sound-programme use. This has eleven segments with compression from 14 to 11 bits, and in this case the highest level signals are coded to a discrimination of 9 bits. The compression transfer characteristic is similar to that shown in Fig. 1, except that there are only six straight-line segments in each quadrant instead of seven. Also the quantizing level scale is divided into sevenths instead of eighths, the slope changing from n to $n-1$ at $\frac{2}{7}Q$ instead of $\frac{2}{8}Q$.

† The segment through zero is common to both the positive and negative quadrants.

2.2 Near-instantaneous Companding

In near-instantaneous companding blocks of 32 samples are examined to discover the maximum sample value in the block. The coding slope is set according to the amplitude of the maximum sample and all the samples in a block are coded to an accuracy determined by that largest sample value. For each block it is necessary to derive a scale factor word which is sent to the decoder and used to control the expander. It will be seen that, at low audio levels, a particular sample in a block may be represented by several different output words, depending on which coding slope or range is being used for that block.

Several different near-instantaneously companded systems have been proposed. The BBC has considered the development of three, which are known as NICAM 1, NICAM 2, and NICAM 3; in which the acronym NICAM stands for 'Near-Instantaneously Companded Audio Multiplex'.

2.2.1 NICAM 1

NICAM 1 is a four-range system, with compression from 13 to 10 bits per sample. In this system the samples are initially represented by linearly-coded 13-bit words. Consecutive samples are grouped in blocks of 32, and the highest value sample in each block is determined. If any values exceed half the permitted peak audio level the top range is used for all the samples in that block. Maximum values between half and a quarter will employ the second range, and so on. If all the sample values in a block are less than one-eighth of the permitted peak audio level the lowest range is used. A range code signal is sent with each block. It should be noted that the highest level signals are coded to a discrimination of 10 bits.

2.2.2 NICAM 2 *and* NICAM 3

NICAM 2 and NICAM 3 are very similar to NICAM 1, but they both start with samples before companding coded to 14-bit accuracy, in order to overcome the basic quantizing noise limitation of a 13-bit system. NICAM 2 is a four-range system with compression to 11 bits per sample, whilst NICAM 3 is compressed to 10 bits per sample, and hence five ranges are required. Figure 2 shows the positive quadrant of the compression transfer characteristic for NICAM 3, and it will be seen that signal samples which are higher in level than half the maximum amplitude (and any other signals in the same 32-sample block as these high level signals) will be coded to a discrimination of 10 bits, whilst if all the samples in a block are less than one-sixteenth of the maximum amplitude, the coding accuracy is 14 bits per sample.

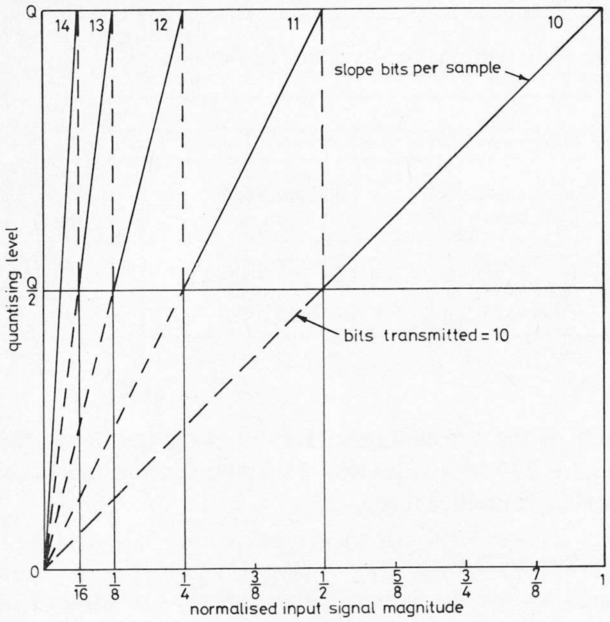

Fig. 2. NICAM 3.

2.3 Subjective Tests

Subjective tests have been carried out comparing various digital companding systems including the 13-segment *A*-law, NICAM 1, NICAM 2, NICAM 3, and a similar near-instantaneous system companding from 14 to 9 bits which had been proposed by Telediffusion de France.[6] These tests, which are fully described in References 7 and 8, involved both single codecs (coders and decoders) and four codecs in tandem.

The material used for the subjective tests consisted of three items, these were: an electronically-generated musical phrase comprising the first four notes of the well-known tune 'Frere Jacques', a short piece of piano music, and a glockenspiel arpeggio. These items, which were all monophonic, were chosen because it was known that they were likely to show up any programme-modulated noise effects. The piano and the glockenspiel items were taken from an analogue tape-recording using Dolby-A noise reduction.

The listeners for these tests consisted of 17 experienced technical staff, and a high-quality monitoring loudspeaker was used in a listening room which was considered to be representative of a good domestic environment. The equipment used for carrying out all the tests comprised a 14-bit linear analogue-to-digital converter and digital-to-analogue converter, the digital signals being processed by a very fast microprocessor to simulate the system being tested.

Each test involved a comparison of two of the systems, A and B, which were presented twice in each test in the sequence ABAB. Tests in the reverse order were also included. Listeners were asked to grade the systems according to the CCIR 7-point comparison scale reproduced here:

+3 A much better than B
+2 A better than B
+1 A slightly better than B
0 A same as B
−1 A slightly worse than B
−2 A worse than B
−3 A much worse than B

Prior to the test sequences the listeners attention was drawn to an example of severe programme-modulated noise, and they were asked to listen for this effect as well as for other impairments such as background noise.

The results of the subjective tests are shown in Tables 1 and 2. Table 1 gives the mean subjective grades obtained with single codecs, and Table 2 gives results obtained when four codecs of some of the systems were put in tandem (a 14-bit digital tape recorder was used for this purpose).

From Table 1 it can be seen that NICAM 2 gave the lowest impairment levels, followed by 13-bit linear, NICAM 1 (and NICAM 3 which is very similar to NICAM 1), TDF, and *A*-law. Note that the particular *A*-law system tested was the 13-segment one which compands from 14 to 10 bits. The 11-segment version was not tested, but since the effective number of bits for

Table 1

Mean grades awarded by listeners in subjective tests comparing single codecs

Test item / Systems compared: A	B	Electronic 'Frere Jacques' test signal	Piano music (Schubert)	Glockenspiel arpeggio	Mean result for the 3 items
A-law	TDF	−1·24 (0·18)	−0·35 (0·26)	−0·88 (0·24)	−0·82
NICAM 1	TDF	+1·29 (0·32)	−0·15 (0·27)	+1·24 (0·25)	+0·72
NICAM 2	NICAM 1	+0·32 (0·16)	−0·09 (0·17)	+0·47 (0·23)	+0·23
13-bit linear p.c.m.	NICAM 1	+0·71 (0·21)	+0·12 (0·21)	— —	+0·42
13-bit linear p.c.m.	NICAM 2	−0·32 (0·11)	−0·03 (0·21)	— —	−0·18
NICAM 3	NICAM 1	0·00 (0·16)	−0·22 (0·26)	+0·03 (0·18)	−0·06

Note: The standard error of the mean grade is given in brackets.

Table 2

Mean grades awarded by listeners in subjective tests comparing 4 codecs in tandem

Systems compared A	Systems compared B	Electronic 'Frere Jacques' test signal	Piano music (Schubert)	Glockenspiel arpeggio	Mean result for the 3 items
A-law	TDF	−1·76 (0·14)	−0·12 (0·26)	−0·26 (0·33)	−0·71
NICAM 1	TDF	+2·24 (0·26)	−0·41 (0·24)	+1·88 (0·32)	+1·24
NICAM 2	NICAM 1	+0·88 (0·36)	+1·24 (0·25)	+0·65 (0·26)	+0·92

Note: The standard error of the mean grade is given in brackets.

the highest level samples is nine, it is expected that the results would be similar to those obtained with the TDF system. Unfortunately, it would not be possible to obtain more than five sound-programme channels in 2048 kbit/s with this system.

It was concluded as a result of these tests that all the NICAM systems would provide a higher quality than would either the 14- to 10-bit *A*-law or the 14- to 9-bit TDF system. Neither of the latter two systems were considered to be completely satisfactory for the BBC national distribution network, but there was little difference between the subjective grades, with respect to programme-modulated-noise, given to the three NICAM systems. However, the adoption of NICAM 3 as the preferred system was felt to be justified, as it is compatible with the performance which is expected from the rest of the broadcasting chain, bearing in mind modern good-quality domestic receivers, and future digital techniques such as recording, which will eventually be used in studio centres. Also with NICAM 3 it is possible to obtain six high-quality sound-programme channels in 2048 kbit/s, whilst NICAM 2 would only allow five such channels to be obtained. Note that amongst the systems considered only NICAM 1 and NICAM 3 provided coding of the highest level signals to 10-bit accuracy. Programme-modulated noise depends particularly on the accuracy to which high-level signals are coded, whilst the background noise with no, or low-level, signals, depends on the accuracy to which such signals are coded. NICAM 2 would give 11-bit accuracy to the highest level signals, but it was felt that this was not necessary as the performance of the 10-bit systems in respect of programme-modulated-noise was satisfactory.

3 Bitstream Format

Where possible it was decided to conform to internationally agreed proposals[6] for the digital transmission of high-quality sound programmes. These include the sampling frequency of 32 kHz. If 20 kbit/s of the 2048 kbit/s are allocated for the multiplex 'housekeeping', i.e. for an overall framing pattern and special synchronizing or justification bits, 2028 kbit/s remain, which means that 338 kbit/s are available for each of the six channels. Ten-bit programme samples require 320 kbit/s, leaving 18 kbit/s for the individual channel housekeeping.

3.1 Single Channel

As explained in Section 2 the samples are grouped in companding blocks or frames of 32 samples, and a range identifying word (to identify which of the five possible companding slopes is appropriate to the particular block) has to be determined. Three bits would be necessary to identify one of five states, but by integrating the identification for three successive blocks, a small reduction has been achieved in the number of bits required for this purpose. The number of possible states for three blocks is 5^3 or 125, and this number requires seven bits to identify it. An error in the range code word could cause a very audible impairment, and hence an additional four bits per word have been allocated, which will enable single errors in any seven-bit-range code word to be corrected. It has been estimated that this should enable satisfactory operation at error rates of at least 1 in 10^5.

These three sample blocks, together with various housekeeping bits, form a frame, and since there are 96 samples in a frame, the total time available is 3 ms. The number of bits per frame is thus $3 \times 338 = 1014$ bits, and these have been allocated as follows:

	Allocation/ frame	Bit-rate/ channel
Sample words	960 bits	320·0 kbit/s
Range coding (with error protection)	11 bits	$3{\cdot}\dot{6}$ kbit/s
Sample word error protection	32 bits	$10{\cdot}\dot{6}$ kbit/s
Signalling	4 bits	$1{\cdot}\dot{3}$ kbit/s
Framing	7 bits	$2{\cdot}\dot{3}$ kbit/s
Total	1014 bits	338·0 kbit/s

Figure 3 shows the arrangement for two frames, known as a multi-frame. The housekeeping bits are shown vertically, but in the serial output bitstream they occur sequentially at the end of each sub-frame, or group

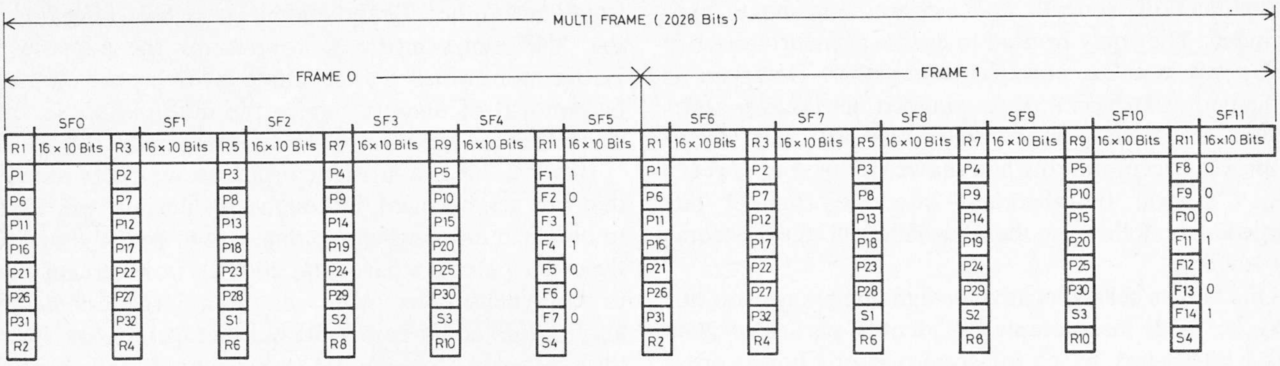

Fig. 3. NICAM 3, single-channel format.

of 16 samples. This arrangement has been chosen as a compromise between simplicity and the need to separate and spread out the sample word error protection bits and the range coding bits. Thirty-two bits have been allocated for sample word error protection. A relatively simple error concealment system has been developed, in which one parity bit deals with three successive samples. In the event of an error being detected at the d.a.c. all three samples are replaced by the previous correct sample. With such a simple system only the five most significant bits in each sample are protected as occasional errors in the five least significant bits are not very serious. Satisfactory operation at error rates up to 1 in 10^5 should be achieved, but if required, a more complicated system giving single error correction may be employed. Further details of these techniques are given in Section 5.6.

Seven bits per frame are used as a framing pattern. These are inverted in alternate frames so that the complete pattern comprises 14 bits in a multiframe of 2028 bits. This has been chosen to minimize the probability of spurious recognition of the pattern in the sample and other housekeeping bits. A reframing time, after individual channel frame loss, of about 12 ms is expected.[9]

The four bits per frame which are left have been allocated for signalling. This facility is available for a variety of purposes including local opt-out and transmitter switching.

3.2 Six-channel Multiplex

Six high-quality sound-programme channels will commonly be multiplexed up to 2048 kbit/s, and Fig. 4 shows how this will be achieved. Pairs of coders (and decoders) will use common housekeeping cards for the reasons explained in Section 4, and hence the bit-rate per coder-pair may be regarded as being 676 kbit/s. The 2048 kbit/s frame has been set at 1 ms, so that 20 bits per frame are available for housekeeping purposes. Seven bits per frame have been allocated as an overall framing pattern, together with seven bits from an adjacent frame, and it is expected that the 2048 kbit/s reframing time after frame loss will be about 2 ms.

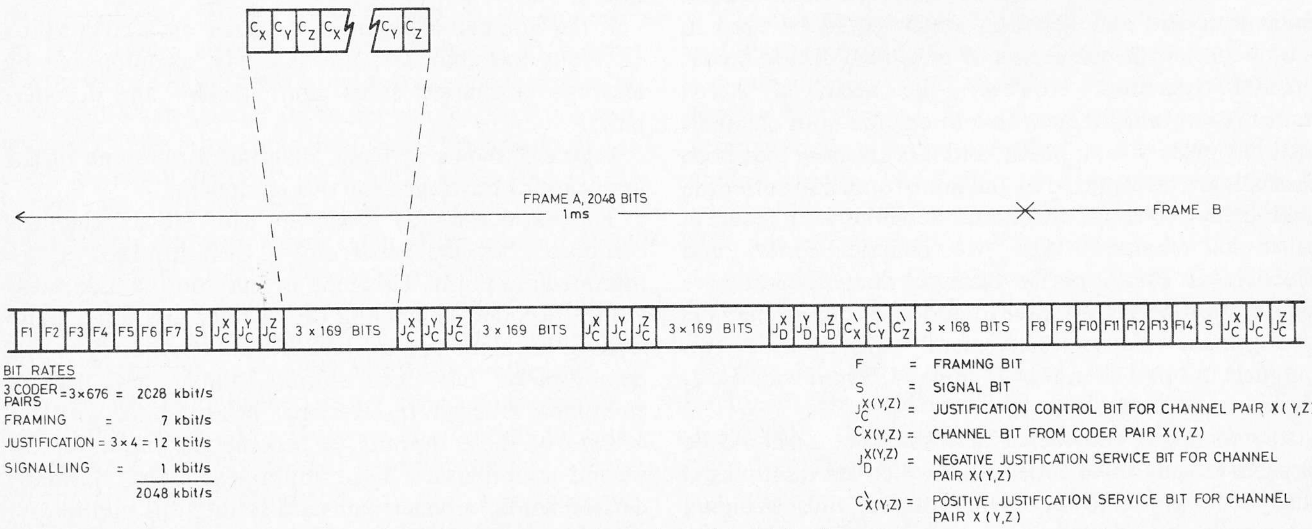

Fig. 4. NICAM 3, six-channel multiplex format.

Justification control and service bits have been included. They may be used to enable asynchronous 676 kbit/s inputs to be multiplexed together. However, as explained in Section 4, if, as planned, all the 676 kbit/s inputs are synchronous, these bits will not be so employed except that the bits shown in Fig. 4 as $C'x$, $C'y$ and $C'z$, will be absorbed into the channel bits immediately following them, which will then become 3×169 bits.

One bit has been allocated for signalling purposes; this may be used, for example, to identify particular 2048 kbit/s bitstreams, which can form part of a higher order multiplex such as 8448 kbit/s.

4 Design Constraints and System Structure

4.1 Requirements

The BBC's existing p.c.m. distribution system multiplexes 13 high-quality sound-programme channels onto a 6336 kbit/s bitstream, which is transmitted over analogue monochrome television links.[3] Although all the coder channels are fully equipped at the sending point in London, decoders are equipped with channel units according to the number of programmes required. Duplicate equipment is installed and powered to provide immediate reserve facilities. This is a somewhat inflexible system but was considered worthwhile in view of the number of channels required on the main distribution network.

Such a system would be very inefficient for outside broadcast, local opt-out, and contribution purposes, in view both of the bandwidth occupied and of the large amount of equipment required. Thus it was decided that any new design should be capable of operation on 2048 kbit/s digital circuits, and that it should be possible to insert or extract channels digitally at intermediate points. Such circuits could be multiplexed up to 8448 kbit/s or even higher multiplexes if required.

A careful study of the BBC's needs showed that optimum flexibility would be achieved with single-channel coders and decoders which could be used in conjunction with bitstreams of one, two, six or higher channel capacities. However, the needs of stereo transmission make it desirable to encode both channels with one piece of equipment, and it is essential that both channels are transmitted by the same route and suffer the same delays. The loss in general flexibility as a result of pairing-off channels into two channel coders and decoders is small, partly because many circuits are required for stereo anyway. In addition, the mechanical arrangement chosen conveniently deals with two channels in one 175 mm (4 U) high, 483 mm wide card-frame.

At first it was considered desirable that it should be possible for any three coder-pairs which are multiplexed onto 2048 kbit/s to be asynchronous, thus avoiding synchronizing problems when the coder-pairs are remote from one another. To this end bits have been allocated in the 2048 kbit/s bitstream format for the purpose of justification, which is a technique involving the addition or removal of dummy bits in the multiplexer, so that asynchronous signals may be synchronized.

However, after a further consideration, it was realized that this arrangement, although very flexible, was likely to be much more expensive than would be the case if all the coder-pairs in a particular 2048 kbit/s bitstream were to be synchronous. After all, in a large number of applications coder-pairs will be close together, and hence the expense of providing for asynchronous working at all the coders and decoders is largely unnecessary anyway. Another consideration which was taken into account was that at digital mixers it will be necessary for all input bitstreams to be synchronous, and accordingly it was agreed to work on the assumption that all the coder-pair outputs would be synchronous, and to deal with the problem of synchronizing remote coders separately. Thus the basis for design eventually chosen was that separate two-channel coders and decoders would be provided, with clock sources included in the multiplexers and demultiplexers. A single master clock source is all that is needed in a 2048 kbit/s coder equipment (similarly for 2048 kbit/s decoder equipment), subsidiary clocks in the coder- and decoder-pairs being derived from these master clocks.

4.2 Typical Facilities

The need for a six-channel coder site is met by using three coder-pairs, together with a clock source and multiplexer which are included in the card-frame of one of the coder-pairs. A similar arrangement is employed at a six-channel decoder site, but using a demultiplexer, etc., instead of a multiplexer. The output from a coder-pair in this case is obtained from a parallel to serial converter, and consists of an intermittent bitstream clocked at 2048 kHz, but with a mean bit rate of 676 kbit/s.

If the number of channels required on a 2048 kbit/s circuit is less than six, this is easily accomplished by leaving out channel cards and/or coder- and decoder-pairs.

Figure 5 shows in block diagram form some of the applications envisaged for this equipment.

The basic link (a) is shown with all six channels equipped, but the bitstream is demultiplexed at an intermediate point. This may or may not be associated with a decoder, depending on whether any local audio feeds or aural monitoring facilities are required. The first demultiplexer has three outputs, and a new piece of equipment labelled 'Inserter' may be associated with one or two of these outputs to become the inputs of the second multiplexer. Since multiplexers are normally carried within a coder-pair card-frame, this multiplexer would be carried within the inserter. At the decoder site

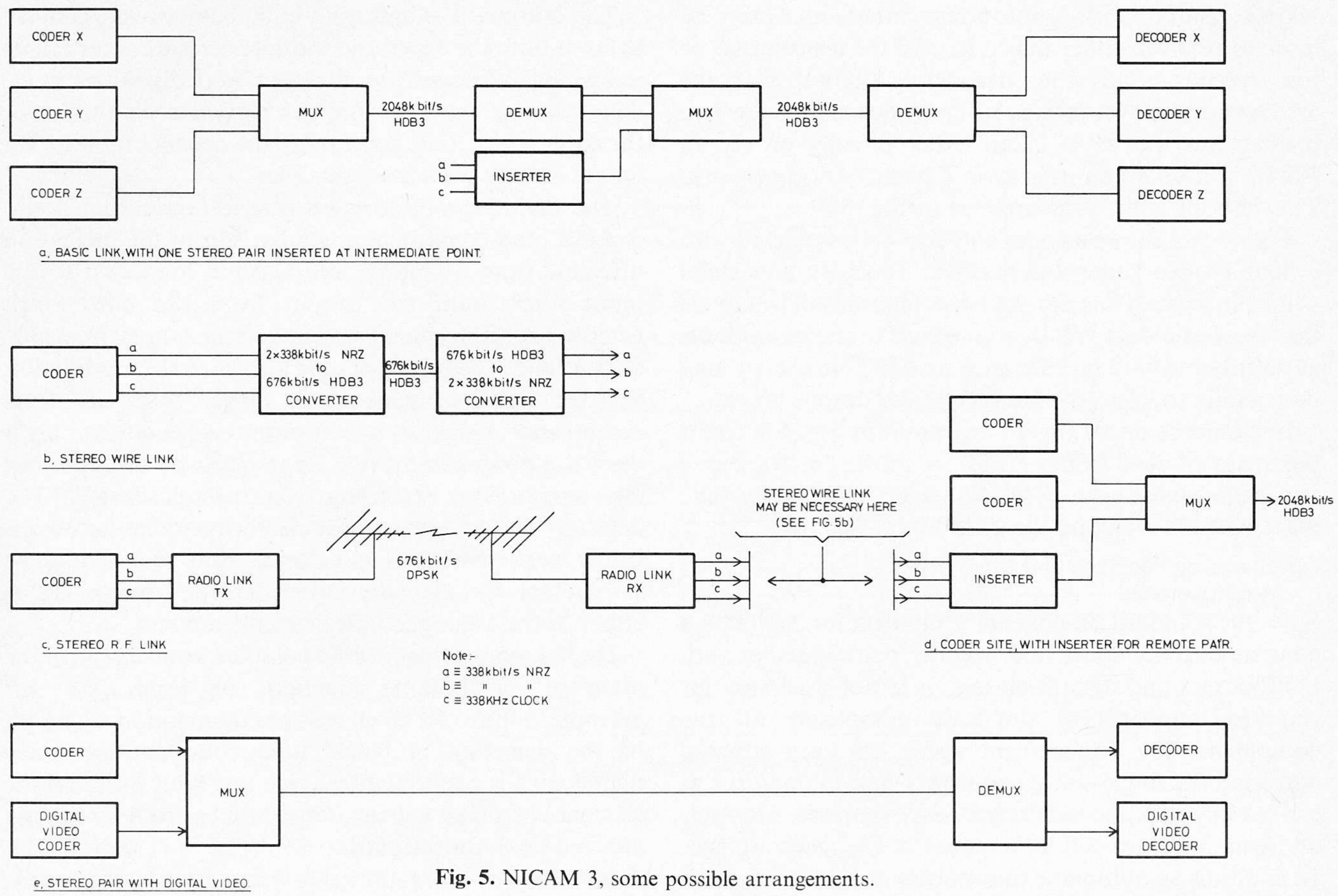

Fig. 5. NICAM 3, some possible arrangements.

'Decoder Z' would decode the inserted part of the bitstream. Normally, for insertion of a stereo-pair it is expected that two synchronous 338 kbit/s non-return-to-zero (NRZ) bitstreams and clock will be the form chosen for the interface between the coder and the inserter, to avoid the complication of HDB3 coding and clock recovery. High Density Bipolar (HDB) codes ensure that only a predetermined maximum number of zeros may be transmitted between successive marks; the codes are known as HDB*n* where *n* is the maximum allowable number of zeros. HDB3 is the code normally employed on 2048 kbit/s circuits. For distances up to tens of metres, the 338 kbit/s NRZ bitstreams and clock may be used, but when greater distances are involved a stereo wire link, Fig. 5(b), could be used. This comprises the clock generator required for 676/338 kbit/s and HDB3 coding and clock recovery. In practice the arrangement will be as shown, with a coder-pair at the sending point. The 676 kHz clock, and the clock for the coder, will be generated on a card in the coder, and a parallel-to-serial converter used to generate 676 kbit/s HDB3 directly. The receiving end may well be integral with the inserter.

Where coders and decoders are installed at a terminal or intermediate point on the main distribution network, the clocks may all be locked together. One technique which is being considered is the possibility of synchronizing all the source clocks to the Droitwich 200 kHz Radio 4 carrier, since this is derived from a rubidium standard, and it, or other co-channel transmitters which are locked to it, may be received anywhere in the United Kingdom. This system should enable remote coders to be synchronized with the main network, any low-frequency phase jitter being taken care of by means of digital storage at the inserter. Another possibility would be to employ a crystal oscillator at the remote coder with a stability high enough to keep clicks on the decoded audio down to, say, one every two hours, as this may be acceptable. Future developments which are planned include a sample-rate synchronizer[10] for use in conjunction with an inserter, in which additional storage is provided and the equipment accepts a plesiochronous input, or one which is only nominally the same frequency. The output of such a synchronizer is synchronous with the accessed bitstream, the sample rate difference being absorbed in the store until a short period of silence occurs in the programme. 'Silent' samples are then omitted or repeated until the store is reset to its half-full condition.

Figure 5(c) shows a stereo r.f. link which is being developed. The signal will be modulated onto an r.f. carrier using differential phase-shift keying, occupying about 1 MHz of bandwidth. The inputs to, and outputs from, the r.f. link, will be the two 338 kbit/s signals and clock previously referred to, although if necessary an

HDB3 section with appropriate interfaces, may be inserted between either the coder and the transmitter, or the receiver and the decoder. Alternatively the arrangement shown in Fig. 5(d) may be employed where the receiver output is taken either directly or via an HDB3 section, to an inserter in a contribution network. This inserter is the same item as in Fig. 5(a).

Figure 5(e) shows a coder and decoder associated with a digital video coder and decoder. Precisely how these will be interfaced has not yet been determined, it may be that the audio data will be transferred to the multiplexer at an intermediate bit rate such as 676 kbit/s, or it may be possible to effect the transfer at the output bit rate.

It should be noted that at any point in Fig. 5 where a bitstream of 2048 kbit/s HDB3 is shown, multiplexing and demultiplexing to 8448 kbit/s or higher may take place without affecting the 2048 kbit/s signals.

4.3 Stand-by Facilities and Monitoring Arrangements

Since the standard six-channel equipment for 2048 kbit/s consists of three coder and decoder pairs together with multiplexers and demultiplexers, it is not necessary for stand-by purposes to duplicate completely all the equipment. The arrangement which has been adopted consists of a single spare coder-pair and multiplexer at the sending end, and similarly a single spare decoder-pair and demultiplexer will be required at the receiving end. In addition an automatic monitoring and switching unit will be required at each end.

The automatic monitoring equipment is very similar as far as both the coder and the decoder are concerned. It is divided into separate analogue and digital sections, and although the following description deals only with the coder, the details concerning the decoder monitoring follow exactly the same principles.

The analogue monitoring is carried out on each audio channel, and consists of a comparison of the indication obtained from a modulation detector connected to the input filter, with the output from the unit which determines the particular companding range applicable at that time. The length of time for which the modulation detector holds a response is longer than the 1 ms compression block, so a fault output is generated only if there is a disagreement (e.g. appreciable audio input but no compression) extending over several seconds. The monitor should operate satisfactorily to levels 30 dB below peak, and it is considered that this should be satisfactory for the purpose of detecting major faults, either in the analogue unit or in the a.d.c.s.

Digital monitoring is more complex but consists in the main of self-checking functions on each card; for example, failure of a clock feed or command recognition, or the detection of illegal data combinations. Each digital card is connected to a one-line fault bus, and the existence of a fault will also be shown by the illumination of a red l.e.d. on the card.

In the event of a fault being detected in a coder-pair, the spare coder-pair has to have the appropriate inputs

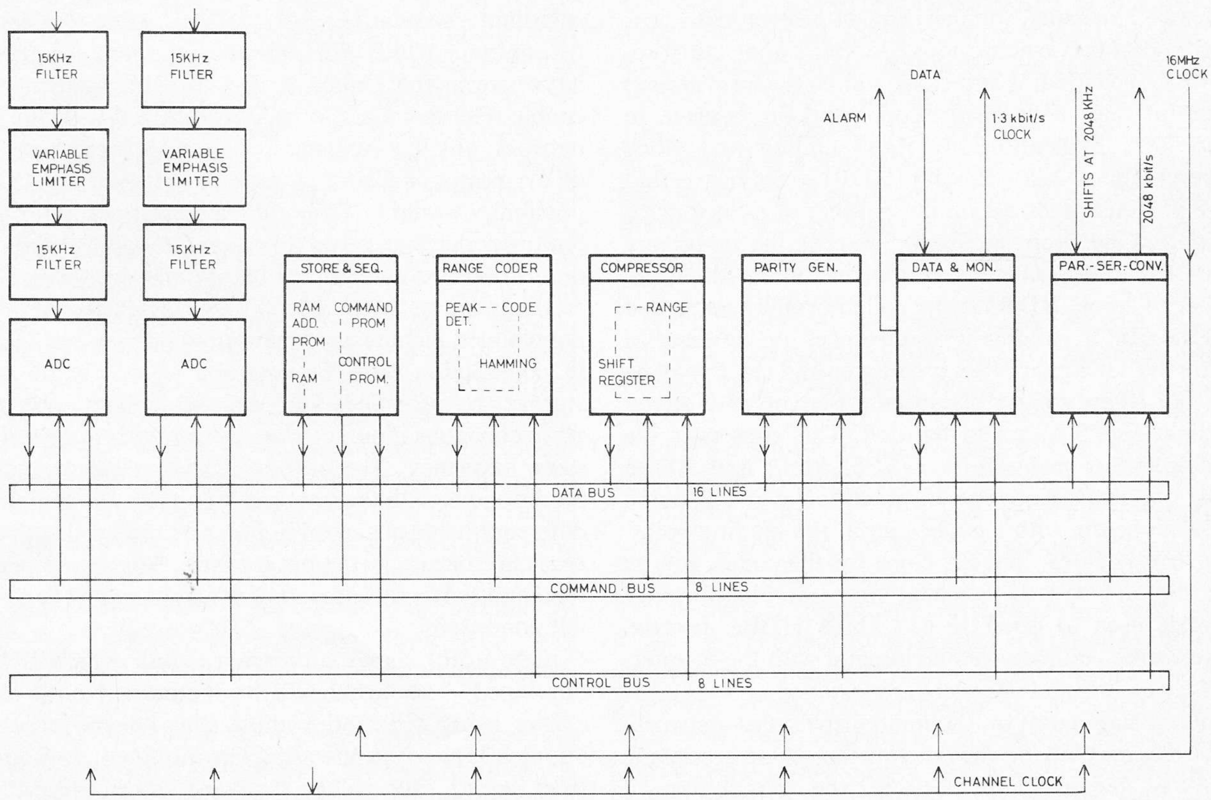

Fig. 6. NICAM 3, dual-channel coder block diagram.

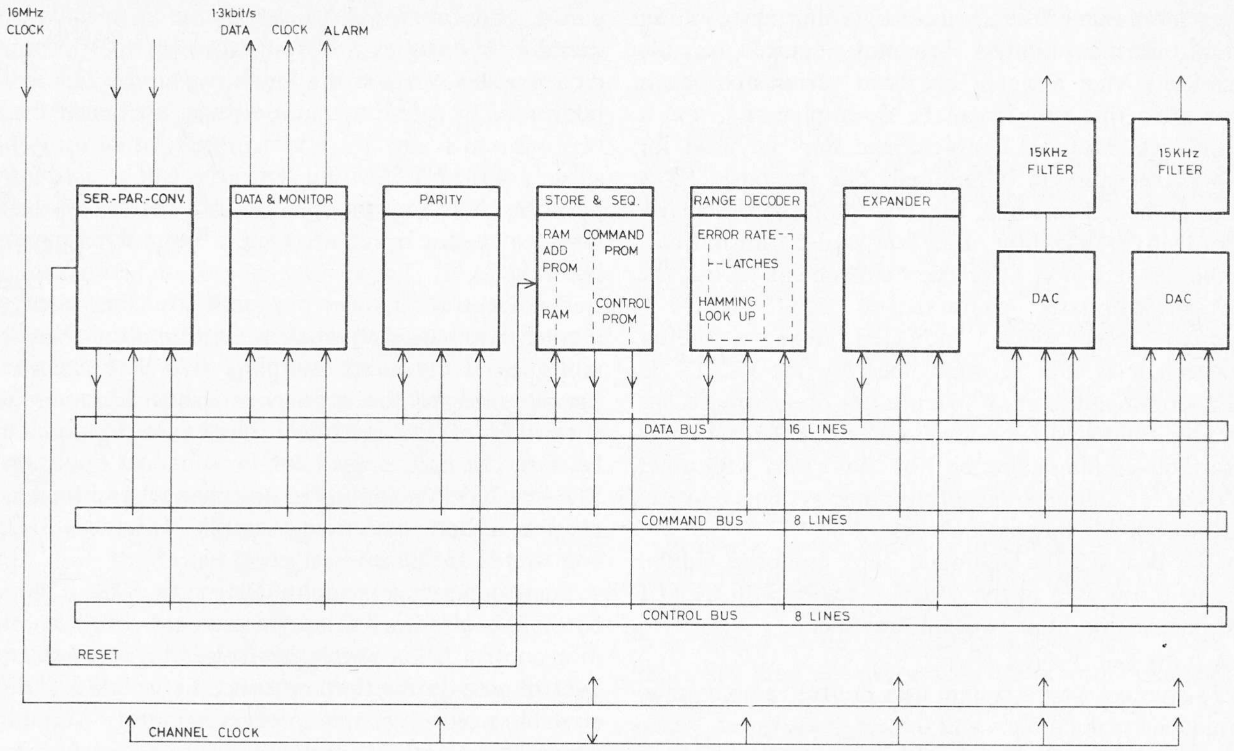

Fig. 7. NICAM 3, dual-channel decoder block diagram.

switched to it, and its output also has to be routed to the multiplexer in place of the output from the faulty coder. These operations, and, if necessary switching in the spare multiplexer, are taken care of by the automatic monitoring unit.

4.4 Maintenance

Maintenance of the equipment has been carefully considered. The original p.c.m. distribution system is maintained successfully on a card replacement basis in the field, with a well-equipped maintenance base in London to effect repairs. For NICAM 3, the same policy could be adopted, the detection of faulty cards in the field being assisted by on-card monitoring. Apart from traditional instruments the field technician may also be equipped with a 'Signature Analyser'. This instrument is particularly useful to detect whether a complex logic waveform is correct or incorrect by displaying a four-figure hexadecimal number. By annotating the handbook with these 'signatures' in the same manner as circuits which include waveforms, mainframe checks can be carried out, particularly the correct operation of the command and control busses. Plugging in a dummy a.d.c. card which produces repetitive dummy sample words would extend the test to include the data bus and hence r.a.m. operation, companding and housekeeping generation.

5 Circuit Description

Figures 6 and 7 are block diagrams of the dual-channel coder and decoder respectively. It will be seen that a bus structure has been adopted for interfacing between the digital cards, similar to that commonly used for microprocessor systems.

5.1 Analogue Units

At the coder the analogue input units consists of filters, limiters, and delay lines, although if the signals have previously been limited the last two items may be omitted.

The audio signals are filtered to avoid components above 15 kHz thus preventing aliasing problems. It is undesirable to permit limiting action to occur on components which will not be transmitted anyway, so there is one three-element filter before the limiter, and, as the limiter may produce harmonics, a second similar filter immediately precedes the a.d.c. The limiter is a two-stage type known as a variable-emphasis limiter.[11]

The first stage is a flat delay-line-type limiter with automatic control of recovery time-constant. In this section no clipping should occur because gain reduction is effected on the delayed programme signal. The recovery time-constant is rapid if a quiet interval follows limiting action, but is extended up to ten seconds gradual recovery of gain if the programme level is high. The second stage of limiting has no delay line and does not alter the gain of the main part of the spectrum. Instead, the circuit applies 50 μs pre-emphasis as required for v.h.f. broadcasting and television sound, and if an overload occurs due to this, the 50 μs time constant is reduced thus reducing the gain at all frequencies above 1 kHz. By employing a variable-emphasis limiter of this type at the input of the sound-programme distribution network, further limiters are not required at the transmitter. After a signal has been variable-emphasis protected in this way it can be de-emphasized, and a different pre-emphasis characteristic may be used for network transmission if required, but the basic 50 μs protection is still retained, thus avoiding the possibility of serious over-deviation of an f.m. sound transmitter.

Although it would have been convenient to use the same pre-emphasis characteristic for NICAM 3, subjective tests[3] have indicated that a better characteristic is that recommended by the CCITT in their Recommendation J17. Hence this pre-emphasis has been used following 50 μs de-emphasis at the output of the variable-emphasis limiter. For conformity with other proposals for digital sound-programme systems a loss of 6·5 dB at 800 Hz has been chosen.

In the decoder the analogue units comprise similar filters to those used in the coder, together with CCITT de-emphasis, and audio output amplifiers.

5.2 A.D.C.s and D.A.C.s

The 13-channel p.c.m. system uses double ramp counter techniques for the a.d.c.s and d.a.c.s.[12] However, whilst this was quite satisfactory for 13 bits, it was considered that instrumental difficulties were likely to be too great to employ the same principle for 14 bits, since the time between samples with a sampling rate of 32 kHz is only 31·25 μs.

An a.d.c. and a d.a.c., which operate on the floating point principle, have been developed for studio applications, and a version of each of these may be employed in NICAM 3. They are capable of 16-bit resolution and are based on hybrid 12-bit a.d.c.s and d.a.c.s. In the a.d.c. a buffered feed of audio is applied to a full-wave rectifier, and this drives a sample peak-level detector. The peak amplitude of a sample is stored in digital form, and is used to control a programmable gain amplifier, which adjusts the amplitude of the sampled audio. Samples are converted into 12-bit words in the hybrid a.d.c., and for NICAM 3 two extra bits are added at the appropriate end or ends of a shift register, as determined by the sample peak detector.

In the d.a.c. the two most significant bits of each 14-bit sample word are used to address a p.r.o.m., which controls a shift register containing the sample word, so that the 12 most significant active bits are presented to a multiplying d.a.c. At the same time the p.r.o.m. drives a ranging d.a.c. which produces a reference voltage proportional to the range. This reference voltage is applied to the multiplying d.a.c. The decoded audio signal from the multiplying d.a.c. is passed through a sample-and-hold circuit which will remove any 'glitches' that are present.

5.3 Store and Sequencer

The purpose of the store and sequencer unit is to provide control signals to the processing units, i.e. compressor, parity, and frame code generator, etc., in a coder or decoder. It includes a central store for the purpose of data transfer between the individual units. The actions performed by the store and sequencer are determined by two p.r.o.m.s and by altering the contents of these, subsets of the NICAM 3 instructions can be used for test purposes. The sequencer could also control processing units which may be required in the future development of NICAM 3.

The control signals produced by the store and sequencer are at rates which are both multiples and sub-multiples of the audio sampling rate. For each audio sample interval the sequencer issues 32 commands, consisting of two identical blocks of 16 commands because the unit is used for two-channel applications. The first block is applied to one channel and the second to the other. As each sample occupies 31·25 μs commands are issued every 0·98 μs.

Each command is identified by an 8-bit command byte. Two bits from this byte are used to select one of four control fields which determine the detailed timing control used during the command. Three bits are used to control which of the processing units should be activated. The remaining three bits determine the particular board function that must be completed during the activation time period.

The card is driven from a 16·384 MHz clock. This is derived from 32 kHz sampling × 16 commands × 2 channels × 8 control intervals × 2 (division for control interval clock). A counter drives a command p.r.o.m., control p.r.o.m. and an address p.r.o.m. The address p.r.o.m. drives a 512 location by 16-bit wide r.a.m. onto a bi-directional data bus. The command p.r.o.m. produces 1536 commands of 64 different types.

The store and sequencer unit in the coder is similar to that in the decoder, but the p.r.o.m.s are programmed differently.

5.4 Range Coder and Decoder

The range coder card provides range code information for each of the pair of channels being coded. NICAM 3 employs five ranges to compress from 14 to 10 bits, and the range coder examines the four most significant bits of each sample to determine which range is required. A 3-bit word is produced which indicates the highest range (least slope) required by each group of 32 sample words. Three consecutive 3-bit words are then integrated, and a 7-bit word is derived, to determine uniquely which of the five ranges each of the 3 groups require.

Four bits are added to each 7-bit word to provide single error correction according to a system proposed by R. W. Hamming.[13] The generation of the resulting 11-bit word is carried out by addressing two p.r.o.m.s with the three 3-bit words defining the three ranges.

This procedure is the same for both channels in a coder-pair, and hence only one range coder is needed, multiplexed between the two channels.

The range decoder performs the inverse of the above operation. First the 11-bit word is applied to the address of a p.r.o.m. and a corrected 7-bit word is produced. This is then passed to another p.r.o.m. which provides the required three 3-bit words.

5.5 Compressor and Expander

The compressor consists basically of a shift register, which is sequentially loaded with each 13-bit (magnitude only) sample. The 3-bit range code word for each group of 32 samples determines which four bits of each sample are discarded by arranging for the shift register to be clocked, so that the correct number of bits are removed from the appropriate end of each sample.

The expander is also a shift register, which is loaded sequentially with the nine magnitude bits of each 10-bit compressed sample word. Again the 3-bit range code word causes the shift register to be clocked so that the required number of most significant bits (m.s.b.s) and/or least significant bits (l.s.b.s) are added to produce the appropriate 13-bit magnitude word.

M.s.b.s that were removed in the compression process only need to be replaced by logic '0's in the expansion process. However, when the l.s.b.s are removed the result is a continual rounding down of the samples. It is therefore necessary to replace these l.s.b. zeros with a correction factor equivalent to '0·5'. This will produce an average result between 'rounding up' and 'rounding down', and should reduce the distortion introduced by companding to a minimum. The correction factor added is '0111' after the last bit transmitted. Table 3 below indicates the extra bits which are added to any compressed 10-bit word for ranges 0 to 4 during the expansion process. The particular 10-bit word chosen for the table is 1011001101 but this has no special significance.

Table 3

Showing the m.s.b.s and l.s.b.s added during expansion from 10 to 14 bits

Range	10-bit word: sign bit	10-bit word: l.s.b.	14-bit word: sign bit	14-bit word: Additional m.s.b.s … l.s.b. (Correction factor)
0	1	0 1 1 0 0 1 1 0 1	1	0 0 0 0 0 1 1 0 0 1 1 0 1
1	As range 0		1	0 0 0 0 1 1 0 0 1 1 0 1 0
2	,,	,, ,,	1	0 0 0 1 1 0 0 1 1 0 1 0 1
3	,,	,, ,,	1	0 0 1 1 0 0 1 1 0 1 0 1 1
4	,,	,, ,,	1	0 1 1 0 0 1 1 0 1 0 1 1 1

The expansion process is identical for the other audio channel, the same hardware being multiplexed between the two channels.

5.6 Sample Parity

Sample error protection can assume two different forms in NICAM 3, depending upon the type of bearer circuit available. Where Post Office digital bearer circuits are employed a simple form of sample parity which provides error concealment is proposed. This is because typical error rates on these circuits are expected to be not worse than about 1 in 10^5. However, in the case of radio-link circuits where the link is subject to severe fading, a more sophisticated form of error protection could be used. An error correction system employing a Wyner–Ash 16, 15 code,[14, 15] has been devised, and this could replace the error concealment system.

Both the above methods of sample error protection use 32 parity bits per frame. Only the 5 m.s.b.s of each 10-bit compressed sample are protected, and this has been found to be satisfactory subjectively. Protecting more bits does not necessarily improve the performance of the audio signal in the presence of errors.

There are 32 bits available to protect 96 samples, hence for the simple error concealment system a detected error will result in the replacement of three samples at the d.a.c. by three repetitions of the previous correct sample. If long bursts of errors are encountered, the held voltage at the d.a.c. output will gradually decay to zero, in order to minimize the disturbance on recovery.

Thus error concealment will be most effective in dealing with relatively isolated single bit errors, whereas the Wyner–Ash technique has been designed to correct bursts of up to 8 consecutive protected bits in error.

The bits of the sample words are re-ordered for transmission so that the protected 5 m.s.b.s are interleaved with the unprotected 5 l.s.b.s. Furthermore, as shown in Fig. 4 the 2048 kbit/s bitstream commutates between the six channels. Thus protected bits are 12 bits apart providing considerable burst error protection, and since 5 bits of three samples are protected together, a burst as long as 180 bits could result in only one parity failure and sample repeat process, in each audio channel.

5.7 Parallel-to-Serial and Serial-to-Parallel Converters

The operation of these units is fairly straightforward. In the case of the parallel-to-serial converter in the coder, sample and housekeeping data, which are present intermittently in parallel form on the data busses of each coder-pair, are clocked out in serial form at a rate suitable for the particular application, for example, bursts of 2048 kbit/s, or a continuous bitstream of 676 kbit/s.

The serial-to-parallel converter in the decoder is required to reverse this process, but to achieve this, the unit must first recognize the individual channel frame alignment words and use this information to reset the store and sequencer unit.

5.8 Multiplexer and Demultiplexer

The 2048 kbit/s multiplexer accepts the three tributary bitstreams from the coder-pairs and combines them so that the output data rate is continuous at 2048 kbit/s. A fourth input to the multiplexer accommodates the output from the reserve coder-pair, as explained in Section 4.3. An associated clock card also includes a binary-to-HDB3 converter.

The demultiplexer performs clock recovery on the incoming bitstream and converts it from HDB3 to binary. It separates the 2048 kbit/s frame word in order to decode the signalling channel and lock the shift pulse generator. The bitstream is fed to the three decoder pairs and reserve and each is also fed with a set of shift pulses which clock in the appropriate bits to each decoder from the complete 2048 kbit/s signal. (Note that the shift pulses sent to the reserve decoder pair will correspond to those sent to one of the main decoder pairs.)

5.9 Signalling Facilities

As explained above, signalling bits are included both in the individual channel bitstreams, and at the multiplexed 2048 kbit/s level. Appropriate input/output ports are provided together with appropriate clock feeds, but the coding and decoding arrangements are not part of the NICAM 3 equipment, and will not be described here.

6 Conclusions

A digital processing system known as NICAM 3 has been described, which is designed to be used for many of the BBC's future high quality sound-programme circuits. The design has concentrated on flexibility, in order to accommodate most of the likely needs. In particular, six 15 kHz channels can be provided on dedicated 2048 kbit/s digital circuits. The sound quality obtained with NICAM 3 is expected to be better than any other companded system so far examined.

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John James Jarrett (Member 1957, Graduate 1951, Student 1949, age 52) joined EMI Research Laboratories in 1945 and has continued with the company until the present, having been appointed Head of the Audio and Acoustics Department in 1977. Between 1945 and 1949 he studied at Southall Technical College for the Higher National Certificate with endorsements and in 1949 was promoted to Engineering status. Mr Jarrett's earlier work was concerned with television camera and studio equipment and for some 14 years he was with EMI Electronics working on various military projects and later was a member of a group working on pattern recognition. In 1970 he re-joined the central research laboratories as Head of the Television Department where the projects he was concerned with included single tube colour cameras and display equipment for computer tomography X-ray machines. He holds patents on a wide range of subjects.

John Jarrett has served on the Communications Group Committee since 1973 and on the Education and Training Committee from 1974 until 1976. He was also a member of the Organizing Committee for the 1977 conference on Digital Processing of Signals in Communications.

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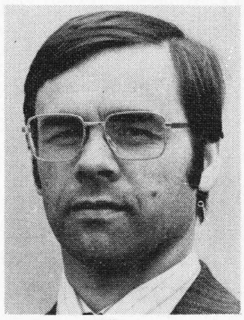

Alan English studied at Harrow College of Technology and was awarded the H.N.C. in electrical engineering in 1968. He joined the Engineering Designs Department of the BBC in the same year, where initially he worked on the design of receivers for re-broadcast applications. Since then, he has concentrated on various forms of signal multiplexing, including the high-quality transmission of two video signals over the same circuit, data with sound, and more recently, time-division-multiplexed p.c.m. sound systems using companding and error-correcting techniques.

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* See also page 518.