THE NICAM SYSTEM

Stereo TV sound has finally come of age with the progressive introduction over the past few years of a digital system called NICAM. This article aims at providing a background to the operation of the NICAM (near-instantaneously companded audio multiplex) system, which is now in use in most of the UK, Scandinavia, Belgium and Spain. NICAM-728, with subversions for PAL systems B/G and I, is also recommended by the EBU as the system for multi-channel sound transmission with terrestrial television. It has been adopted for use in several countries, including the UK, and now forms part of a draft CCIR recommendation.

By J. Buiting, technical editor.

WHEN we talk about different television standards, the discussion is usually about different ways of conveying the picture to the viewer. Up to ten years ago, the sound was taken for granted, which is remarkable because the stereo age was well under way at that time. Following a German initiative, some European countries introduced stereo TV sound based on an auxiliary subcarrier above the main (mono) FM carrier. Although this works, the NICAM system offers superior sound quality at a roughly equal bandwidth requirement. Originally developed by the BBC, the NICAM-728 specification has been formally approved by the Department of Trade and Industry as the United Kingdom standard for two-channel digital sound with terrestrial television broadcasts.

A brief history of stereo TV sound

Since 1979, a number of stereo TV sound systems have been introduced that were aimed at downward compatibility with the existing mono sound systems. Among the requirements for the new sound systems were:

- minimum interference and crosstalk between the channels;
- quality of existing (main) mono channel must not be affected;
- equipment to upgrade transmitters and receivers must remain as simple as possible.

The need of maintaining downward compatibility, as well as the limited bandwidth available for the new sound systems, have forced the designers of analogue stereo TV sound systems to drop some of their target specifications, and agree on certain compromises that reduce the quality that could have been achieved in theory. Analogue stereo sound systems can be made downward compatible in two ways:

- by modifying the audio signal before it is modulated on to the carrier (single-carrier principle);
- by adding a second sound carrier just above or below the existing (mono) sound carrier (dual-carrier principle)

In both cases, a decoder matrix is required to separate the left and right channels, and produce the stereo sound image. Some systems also require de-emphasis and/or decompanding to improve the signal-to-noise ratio and the dynamic range.

The dual-carrier system is basically analogue, and offers quite reasonable sound quality. However, in this day and age of digital sound, it is not surprising that alternatives have been sought, based on the technology already familiar from CD players and the sound transmission standard developed for the MAC system. In particular the channel separation offered by NICAM is much higher than that achieved by any form of analogue dual-carrier system. Overall, the sound quality of a NICAM broadcast is so close to that of a compact disk that it is hard to tell the difference by just listening.

NICAM-728 digital sound transmission

Strictly speaking, the NICAM-728 system should be classified as a dual-carrier system, because a second sound signal is introduced in the baseband spectrum (see Fig. 1). The spectrum shown is for PAL system-I as used in the UK, with the main sound carrier at 6.0 MHz above the vision carrier, and a total channel bandwidth of about 8 MHz. Most other European countries use PAL system B or G, where the main sound channel is at +5.5 MHz, and the channel bandwidth is about 7 MHz.

The NICAM signal is recovered from a QPSK (quadrature phase shift keying) spectrum with a bandwidth of about 600 kHz. The centre frequency of this

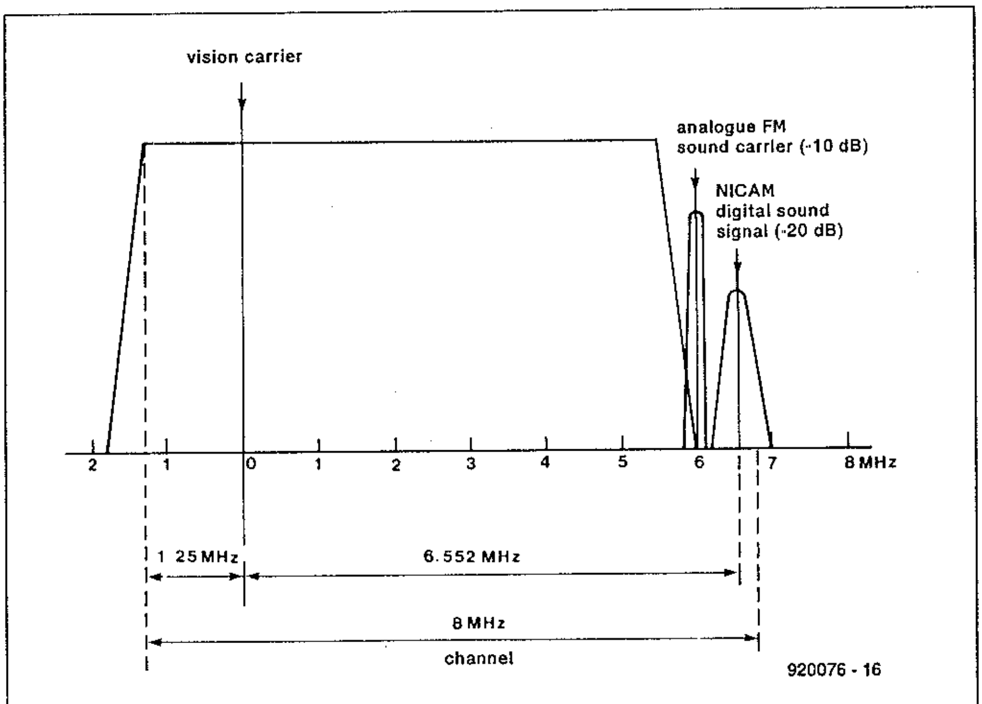

Fig. 1. The frequency band occupied by the NICAM-728 digital sound signal in relation to the picture and mono (analogue FM) sound signal components in the TV baseband.

'molehill' (that is what it looks like on a spectrum analyser) is +6.552 MHz (system I), or +5.850 MHz (system B/G). The level is about −20 dB with respect to the vision carrier. In the rest of this article, we will refer to the UK standard (PAL system-I) only.

Contrary to the analogue dual-carrier systems, the NICAM signal contains all the information necessary to reproduce the two stereo channels, i.e., it is completely independent of the main FM carrier at +6.0 MHz (except for the fixed frequency and phase relation), which is currently transmitted only to ensure downward compatibility with existing TV sets.

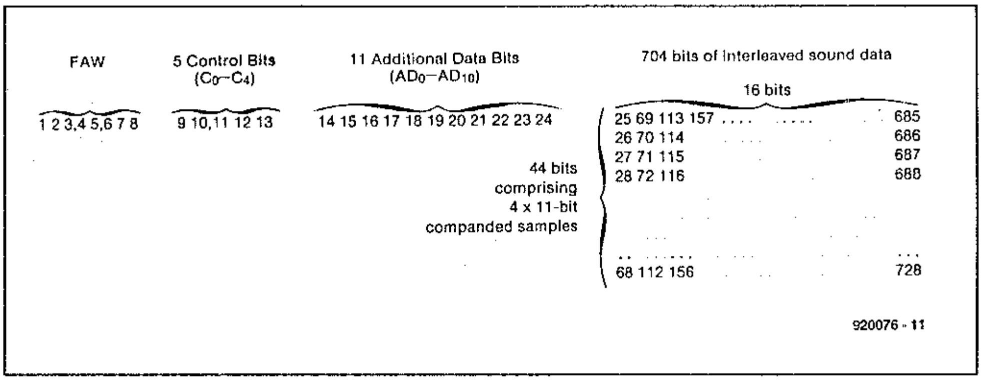

Fig. 2. Each frame consists of four groups of bits, each with its own function.

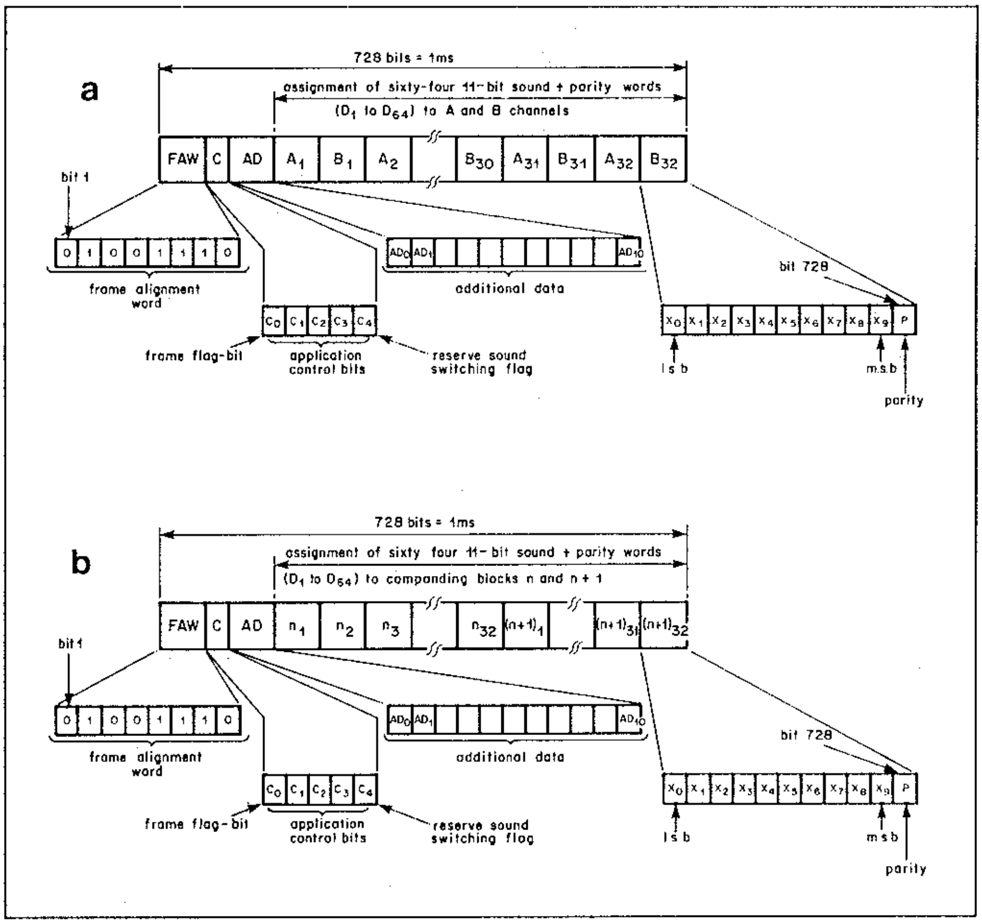

Fig. 3. (a) Structure of a 728-bit frame containing a stereo sound signal (before interleaving); (b) the same for a mono sound signal (also before interleaving).

Sound multiplex and sound coding methods

To understand how the NICAM system works, we will take a look at the structure of the serial data stream at the transmitter side.

Frame structure and bit interleaving

As shown in Figs 2 and 3, the data consists of 728-bit frames which are transmitted continuously without gaps. One frame is transmitted every millisecond, so the overall bit-rate is 728 Kbit/s, whence the system designation NICAM-728.

The 720 bits that follow the frame alignment word (FAW) have a structure that closely resembles that of the first-level protected, companded sound signal blocks in the systems of the MAC family. After the control bits and the additional data bits follows a block of 704 interleaved sound data bits. The interleaving pattern relocates data bits which are adjacent in the frame structure of Fig. 2 to positions at least 16 clock periods apart in the transmitted data stream.

Energy dispersal scrambling

The transmitted bit stream is scrambled for spectrum-shaping purposes (remember the restrictions as regards the baseband bandwidth). The scrambling operates synchronously to the multiplex frame. The FAW is not scrambled, and used to synchronise the pseudo-random sequence generator used for descrambling in the receiver. Figure 4 shows the general layout of the scrambler. The following parameters apply:

- the bit that follows the FAW is the first scrambled bit, and is added modulo-two to the first bit of the pseudo-random sequence;
- the bit that precedes the FAW is the last scrambled bit;
- scrambling takes place immediately after interleaving (and descrambling is therefore prior to de-interleaving in the receiver);
- the pseudo-random sequence is defined by a generator polynomial
  $x^9 + x^4 + 1$
  and an initialisation word ('seed')
  1 1 1 1 1 1 1 1 1

Thus, with reference to Fig. 2, the sequence starts:

0000 0111 1011 1110 0010

FAW and control information block

The FAW is 01001110, which is a series of bits transmitted in that order. The control information converged to the receiver consists of a frame flag bit, C_0, three application control bits, C_1, C_2 and C_3, and a reserve sound switching flag, C_4 (see Fig. 3). The frame flag bit, C_0, is set to '1' for eight successive frames, and to '0' for the next eight frames. The frames are numbered within the sequence as follows: the first frame (Frame 1) of the sequence is defined as the first of the eight frames in which C_0=1. Hence, the last frame (Frame 16) of the sequence is the last of the eight frames in which C_0=0. This frame sequence is used to synchronise changes in the type of information being carried in the channel.

The function of the three application control bits, C_1, C_2 and C_3, is to define the current application of the last 704 bits in each frame, which may be used to convey either sound samples or data. The available options are shown in Table 1. When a change to a new application is required, these control bits change (to define the new application) on Frame 1 of the last 16-frame sequence of the current application. The 704-bit sound/data blocks change to the new application on Frame 1 of the following 16-frame sequence.

The reserve sound switching flag, C_4, contained in the control information block is used to switch back to the output of the conventional FM demodulator when the digital sound decoding system fails. This is, of course, acceptable only if the FM sound channel carries the same programme as the failing digital channel. The means to

Table 1. Applications of 704-bit sound/data blocks

Application control bits			Contents of 704-bit sound/data block
C_1	C_2	C_3 *	
0	0	0	stereo signal comprising alternate A-channel and B-channel samples
0	1	0	two independent mono sound signals (designated M1 and M2) transmitted in alternate frames.
1	0	0	one mono signal and one 352-kBit/s transparent data channel transmitted in alternate frames.
1	1	0	one 704-Kbit/s transparent data channel.

* C_3=1 provides for signalling additional sound or data coding options. When C_3=1, decoders not equipped for these additional options should provide no sound output.

Table 2. Coding/protection range selection.

Coding range	Protection range	Scale factor value		
		R_2	R_1	R_0
1st	1st	1	1	1
2nd	2nd	1	1	0
3rd	3rd	1	0	1
4th	4th	0	1	1
5th	5th	1	0	0
5th	6th	0	1	0
5th	7th	0	0	1
5th	7th	0	0	0

inhibit such switching is incorporated in the control information. Control bit C_4 is set to '1' when the FM channel carries the same sound programme as the digital stereo signal or the digital, mono signal (where two digital mono signals are transmitted, this refers to the M1 signal only). When the FM channel is not carrying the same programme as the digital sound channel, C_4 is set to '0'. In this state, it can be used to prevent switching to the FM sound. Finally, C_4 has no meaning in the case of data transmission.

Additional data and the sound/data block

Data bits AD0 to AD10 (see Fig. 3) are reserved for future applications yet to be defined.

The last 704 bits in any frame form a block of either sound or data (the two types of information are not mixed within one frame). One frame contains 64 sound samples (D1 to D64). The structures of a stereo sound frame and a mono sound frame are shown in Figs. 3a and 3b respectively.

In stereo mode (AC: C_1=C_2=C_3=0), the odd-numbered samples convey the A-channel, and the even-numbered samples the B-channel. Thus, 32 samples of each channel are transmitted in every frame.

If two independent mono sound channels, M1 and M2, are transmitted (AC: C_1=0; C_2=1; C_3=0), M1 is transmitted in odd-numbered frames, and M2 in even-numbered frames.

If one mono sound channel is transmitted (AC: C_1=1; C_2=0; C_3=0), it is contained in odd-numbered frames, and data are transmitted in even-numbered frames.

Thus, for mono sound signals, each frame with sound information in it contains 64 consecutive sound samples, which will span two complete companding blocks, shown as blocks n and (n+1) in Fig. 3. No format has yet been defined for data information.

Sound signals

Sound signals are sampled at 32 kHz, and coded initially with a resolution of 14 bits per sample. Near-instantaneous companding is used to reduce the number of bits per sample from 14 to 10, and one parity bit is added to each 10-bit sample word for error detection and scale-factor signalling purposes.

The companding process forms the 14-bit digital samples corresponding to each of the sound signals into blocks of 32. All of the samples in each 1-ms block are subsequently coded, using a 10-bit 2's complement code, to an accuracy determined by the magnitude of the largest sample in the block, and a scale factor code is formed to convey the degree of compression to the receiver. Figure 5 illustrates the coding of companded sound signals.

Prior to compression, a pre-emphasis to CCITT recommendation J17 (Ref. 2) is applied to the sound signals, either by using analogue pre-emphasis networks before digitisation, or by using digital filters with the digital signals.

For stereo transmissions, the signals of the left and right sound channels are sampled simultaneously. The Channel-A samples convey the left-hand (L) sound signal, and the Channel-B samples the right-hand (R) sound signal.

One parity bit is added to each 10-bit sound sample to check the six most-significant bits for the presence of errors. The parity group so formed is even (i.e., the modulo-2 sum of the six protected sample bits and the parity bit equals 0). Subsequently, the parity bits are modified to signal the 3-bit scale factor word associated with each sound signal block.

In addition to signalling the coding range, the scale factor signals seven protection ranges. This information may be used in the receiver to provide extra protection for the most significant bits of the samples. Table 2 shows the coding ranges and protection ranges associated with each 3-bit scale factor word. The five coding ranges indicate the degree of compression to which the block of samples has been subjected for the near-instantaneous companding process. The 3-bit scale factor R_2-R_1-R_0 associated with each 32-sample sound block is conveyed by modification of the parity bits (see Fig. 5).

When a stereo sound signal is being transmitted, FE1 (facteur échelle; scale factor) is the scale-factor word R_{2A}-R_{1A}-R_{0A} associated with the 'A' samples, and FE2 the scale-factor word R_{2B}-R_{1B}-R_{0B} associated with the 'B' samples. If P_i is the parity bit of the i[th] sample, this is modified

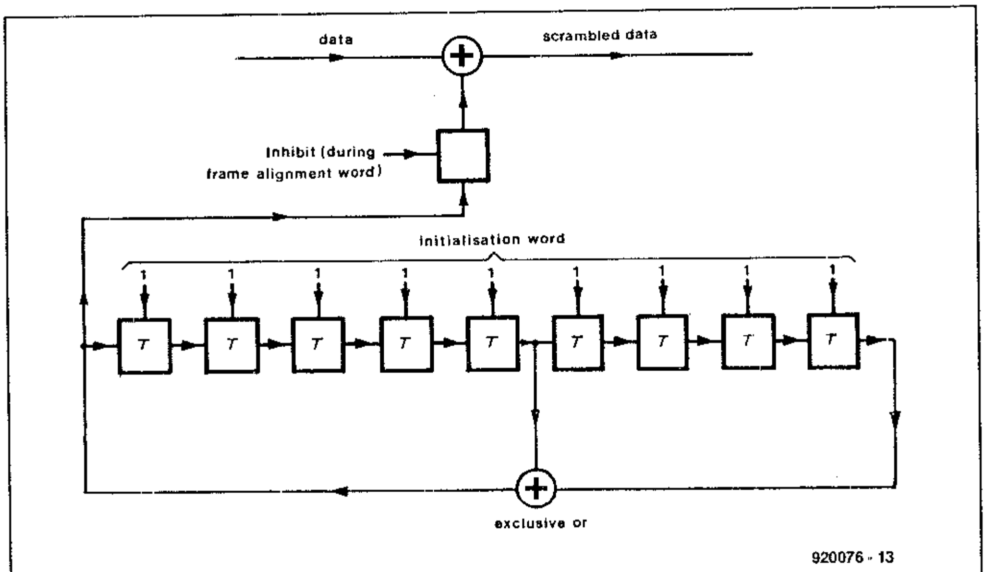

Fig. 4. Pseudo-random sequence generator (PRSG) for spectrum shaping (energy dispersal scrambling).

to P'_i, by modulo-2 addition of one bit of one of the scale-factor words according to the following relationship:

$P'_i = P_i \oplus R_{2A}$ for i = 1.7 13 19 25.31.37 43.49
$P'_i = P_i \oplus R_{1A}$ for i = 3.9,15,21,27,33,39 45,51
$P'_i = P_i \oplus R_{0A}$ for i = 5 11 17 23 29 35 41 47.53

$P'_i = P_i \oplus R_{2B}$ for i = 2.8.14 20.26,32.38,44.50
$P'_i = P_i \oplus R_{1B}$ for i = 4.10,16,22,28.34,40.46.52
$P'_i = P_i \oplus R_{0B}$ for i = 6.12,18,24.30 36.42.48.54

When a mono signal is being sent, FE1 is the scale-factor word R_{2n}-R_{1n}-R_{0n} associated with the first block of 32 samples in the frame, and FE2 is the scale-factor word R_{2n+1}-R_{1n+1}-R_{0n+1} associated with the second block of 32 samples in the frame. As in the case of stereo sound, the parity bit of the i[th] sample, P_i, is modified to P'_i by modulo-2 addition of one bit of one of the scale-factor words. However, in the mono case, the modification of the parity bits relates to the block structure of the mono signal, as follows:

$P'_i = P_i \oplus R_{2n}$ for i = 1 4.7 10.13 16.19 22.25
$P'_i = P_i \oplus R_{1n}$ for i = 2.5.8 11.14 17.20.23 26
$P'_i = P_i \oplus R_{0n}$ for i = 3,6,9,12,15.18 21 24 27

$P'_i = P_i \oplus R_{2n+1}$ for i = 28.31 34.37 40 43 46 49.52
$P'_i = P_i \oplus R_{1n+1}$ for i = 29.32 35 38 41 44 47 50.53
$P'_i = P_i \oplus R_{0n+1}$ for i = 30,33 36.39 42 45,48 51,54

It should be noted that some of the scale-factor information in the second block of samples is conveyed in the parity coding of samples 28 to 32, which are in the first block. This conforms with the specifications for the MAC/Packet family of transmission standards drawn up by the EBU (Ref. 1).

The scale-factor coding range and protection range information are extracted at the decoder by majority decision logic. Subsequently, the original parity is restored for the purpose of error concealment.

The control information described in Section 6.2.3 of Ref. 1 (Chapter 3, Part 3) is not used. However, other information could be transmitted by the same means, i.e., two information bits such that one modifies samples 55 to 59, and the other samples 60 to 64. NICAM receivers should be designed to take account of this facility.

Modulation parameters

The characteristics of the AM vision (vestigial sideband) and FM sound are defined in the UK specification for PAL system-I transmissions (Ref. 3), with the exception that the FM sound carrier power is 10 dB down with respect to the vision carrier, instead of 7 dB. In the case of PAL system-B/G transmissions, the definitions given in CCIR Report 624-3 apply.

The NICAM signal in the baseband is classified as differentially encoded quadrature phase shift keying (DQPSK or 4-phase DPSK). This is a four-state phase

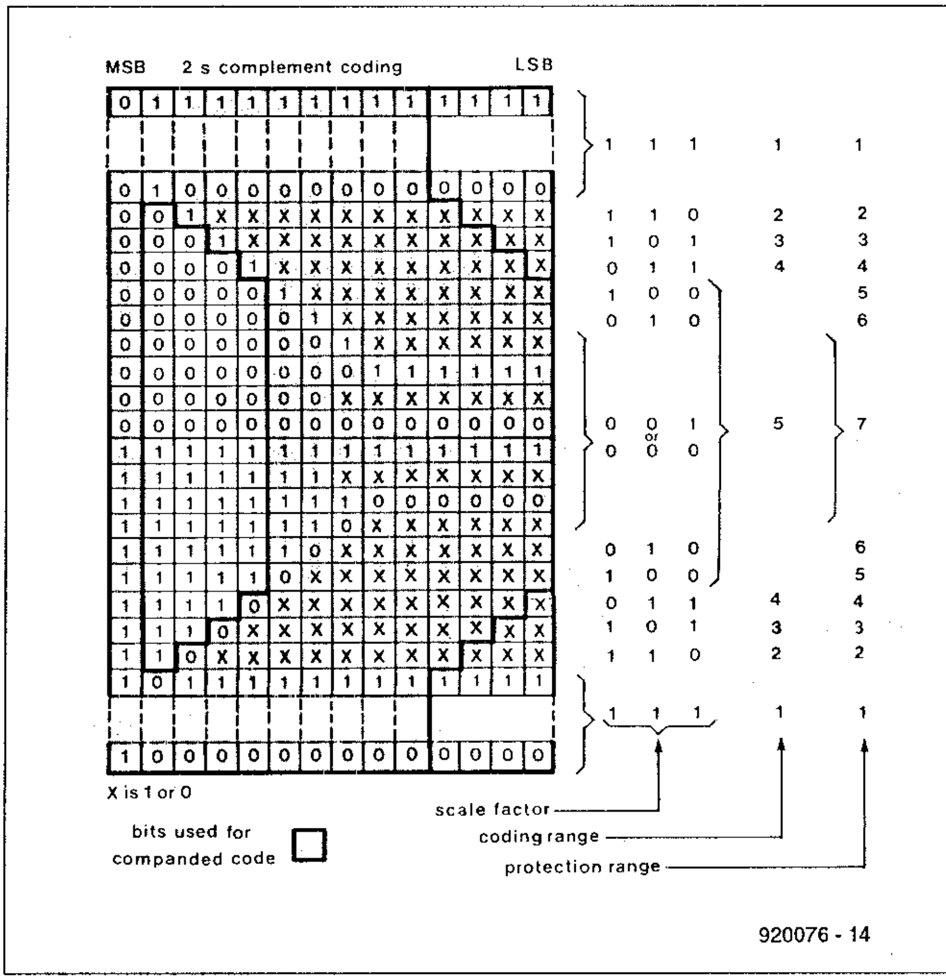

Fig. 5. Coding of companded sound signals.

modulation system in which each change of state conveys two data bits. The input data stream at the modulator is differentially encoded. This is done in two steps: (1) serial to two-bit parallel conversion, and (2) coding of the transmitted phase changes. The amounts of the changes of carrier phase which correspond to the four possible values of the input bit pairs (A_n-B_n) are shown in Table 3.

Table 3. DQPSK carrier state changes.

Input bit-pair		Amount by which the carrier changes phase
A_n	B_n	
0	0	0° (no change)
0	1	–90°
1	0	–270°
1	1	–180°

Thus, the carrier phase can be at one of four rest-states which are spaced at intervals of 90° apart (Fig. 6a). An input bit-pair will shift the carrier phase into a different rest-state by the amount of phase change assigned to that particular value of bit-pair. The transmitted phase-changes and resulting carrier rest-states for the input bit-pair sequence 00, 01, 11 and 01 are illustrated in Fig. 6b. In the receiver the transmitted datastream may be unambiguously recovered by determining the

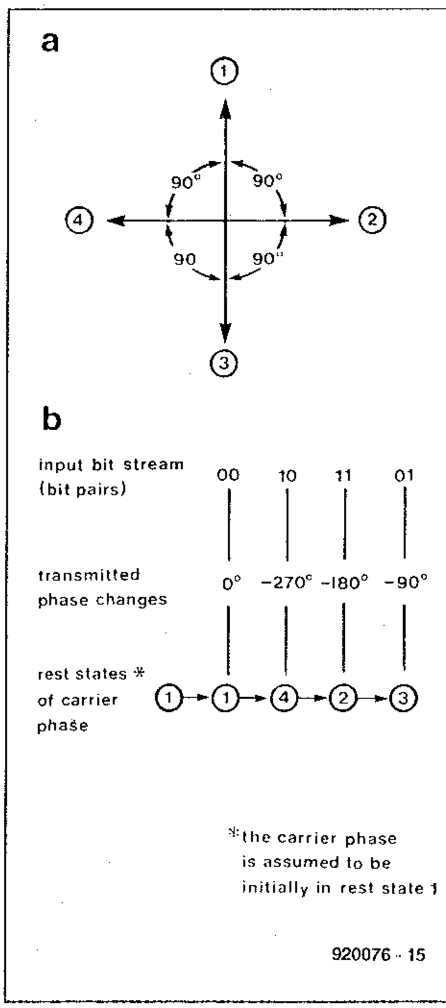

Fig 6. DQPSK modulation principle.

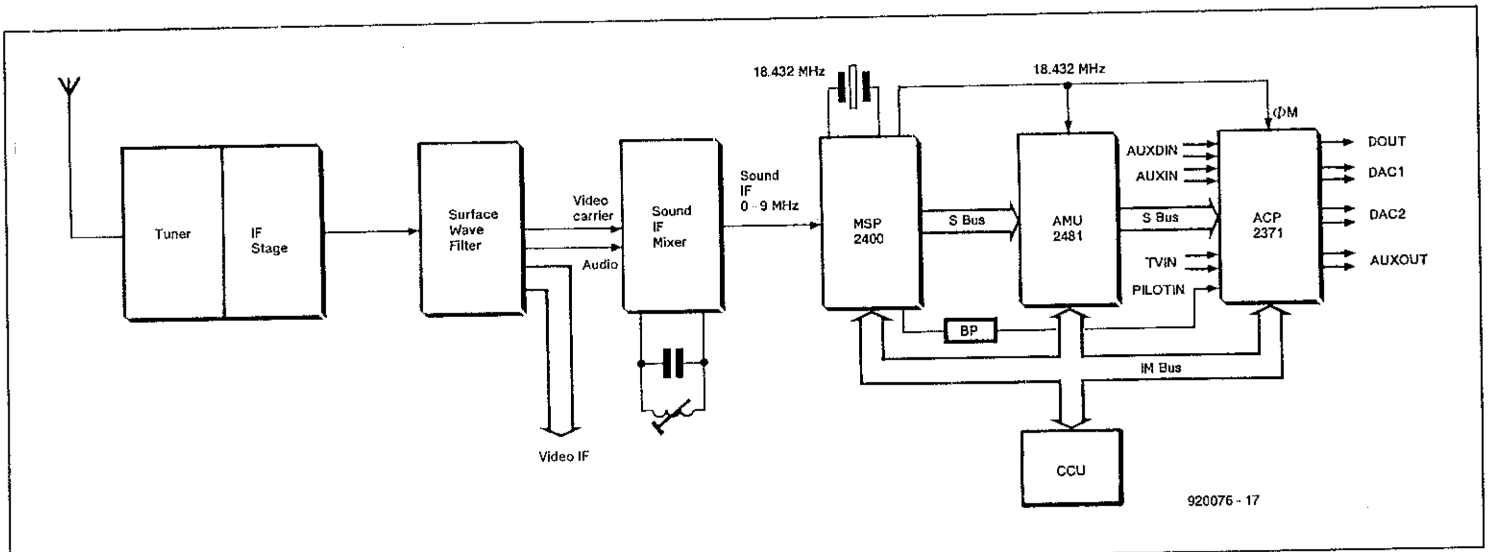

Fig. 7. **NICAM decoder concept proposed by ITT Semiconductors.**

phase-changes between one bit-pair and the next.

It was already mentioned that spectrum-shaping techniques are applied to keep the bandwidth of the NICAM signal in the baseband within limits. For best performance in the presence of random noise, the amplitude-frequency response of data spectrum-shaping filters at the receiver should be identical to that at the transmitter. The target amplitude frequency response, $H_T(f)$, is given by

$$H_T(f) = \begin{cases} \cos\dfrac{\pi f t_s}{2} & \text{if } 0 \le f \le \dfrac{1}{t_s} \\ 0 & \text{if } f > \dfrac{1}{t_s} \end{cases}$$

where $t_s = \dfrac{1}{364{,}000}$ s

and the filter has a constant group delay for all frequencies $\le 1/t_s$. The filter made on the basis of the above transfer characteristic has a 100% cosine roll-off (for PAL systems B and G a filter with 40% cosine roll-off is required).

In the UK, the NICAM subcarrier is located at 6.552 MHz above the frequency of the vision carrier (see Fig. 1). This frequency is obtained by multiplying the transmitted bit-rate (728 Kbit/s) by 9. In countries where PAL system-B or -G is used, the subcarrier frequency is +5.850 MHz.

NICAM decoder concepts

Among the IC manufacturers that have developed NICAM processors for use in commercial-grade receivers are ITT Semiconductors of Germany, and Micronas, Inc. of Finland. A decoder based on ICs from the latter manufacturer is described elsewhere in this issue.

ITT Semiconductors have integrated their NICAM processors, the MSP2400 and MSP2410, into the Digit 2000 TV system. Figure 7 shows the block diagram of the ITT approach. Apart from the MSP2400 or MSP2410, two additional ICs are required, the AMU2481 and the ACP2371. Remarkably, the MSP2400 has a digital filter to extract the NICAM information from the baseband spectrum (0 to 9 MHz). This is in contrast to the Micronas circuit (Fig. 8), which uses a conventional *L-C* bandpass filter tuned to 5.84 MHz (PAL system B/G) or 6.552 MHz (PAL system I). The ITT circuit has a number of interesting options such as multistandard sound processing and automatic standard recognition and switching. The configuration as shown in Fig. 7 is capable of handling mono FM, stereo FM (the German dual-carrier system) and all NICAM modes (a special version of the ACP2371 is available for satellite TV sound). The disadvantage of the ITT circuit is, however, that it can not work without control software, and this is where the Micronas system has the edge on the ITT system: it can work 'stand alone', and offers an optional way of computer control. ■

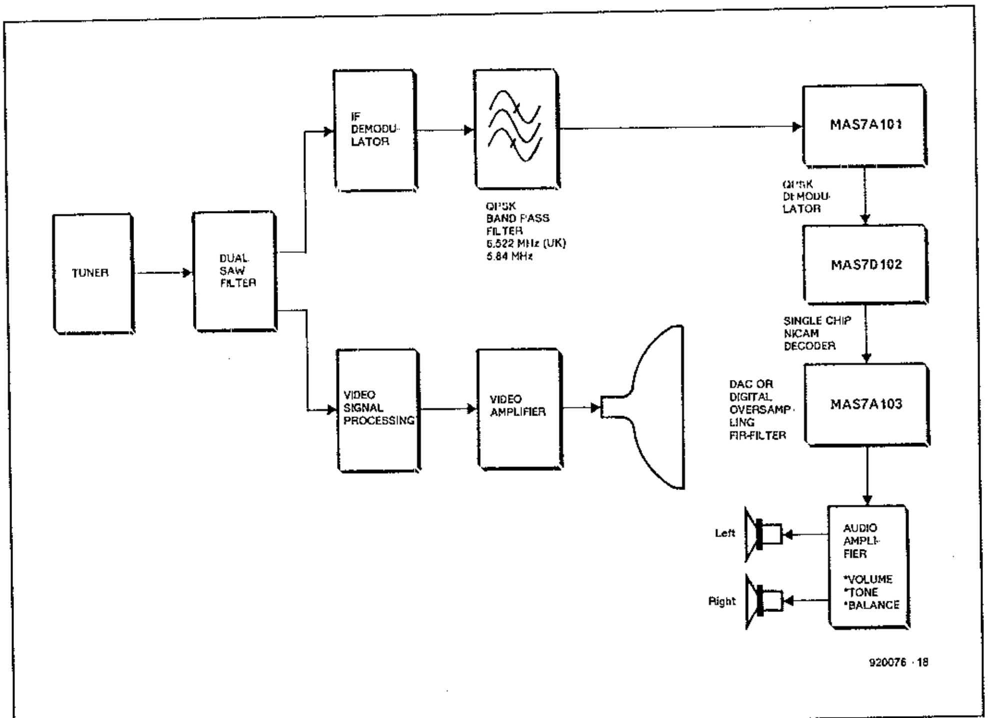

Fig. 8. **NICAM decoder concept proposed by Micronas Inc**

Sources:

(1) NICAM-728: specification for two additional digital sound channels with System-I television.

(2) Document SPB 424, 3rd revised edition, European Broadcasting Union.

References:

1. *Specification of the system of the MAC/Packet family*. European Broadcasting Union (EBU) Technical Document 3285 (1986).

2. CCITT Red Book, Volume III, Fascicle III.4: *Transmission of sound-programme and television signals recommendation J.17 'Pre-emphasis used on sound-programme circuits'*.

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