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

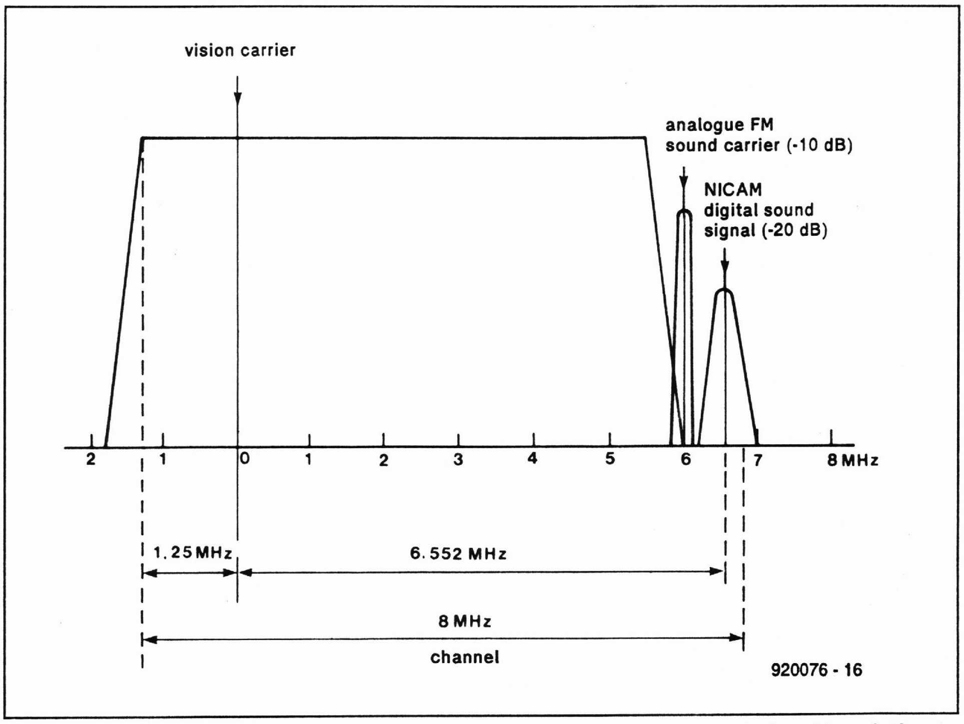


Fig. 1. The frequency band occupied by the NICAM-728 digital sound signal in relation to the picture and mono (analogue FM) sound signal components in the TV baseband.

'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.

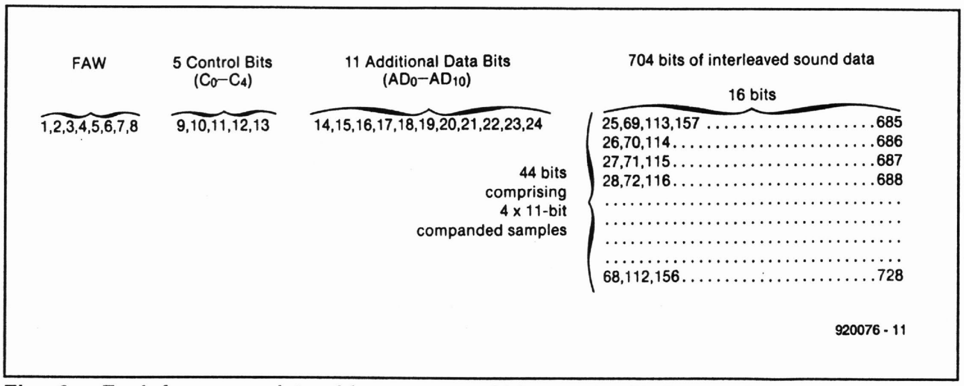


Fig. 2. Each frame consists of four groups of bits, each with its own function.

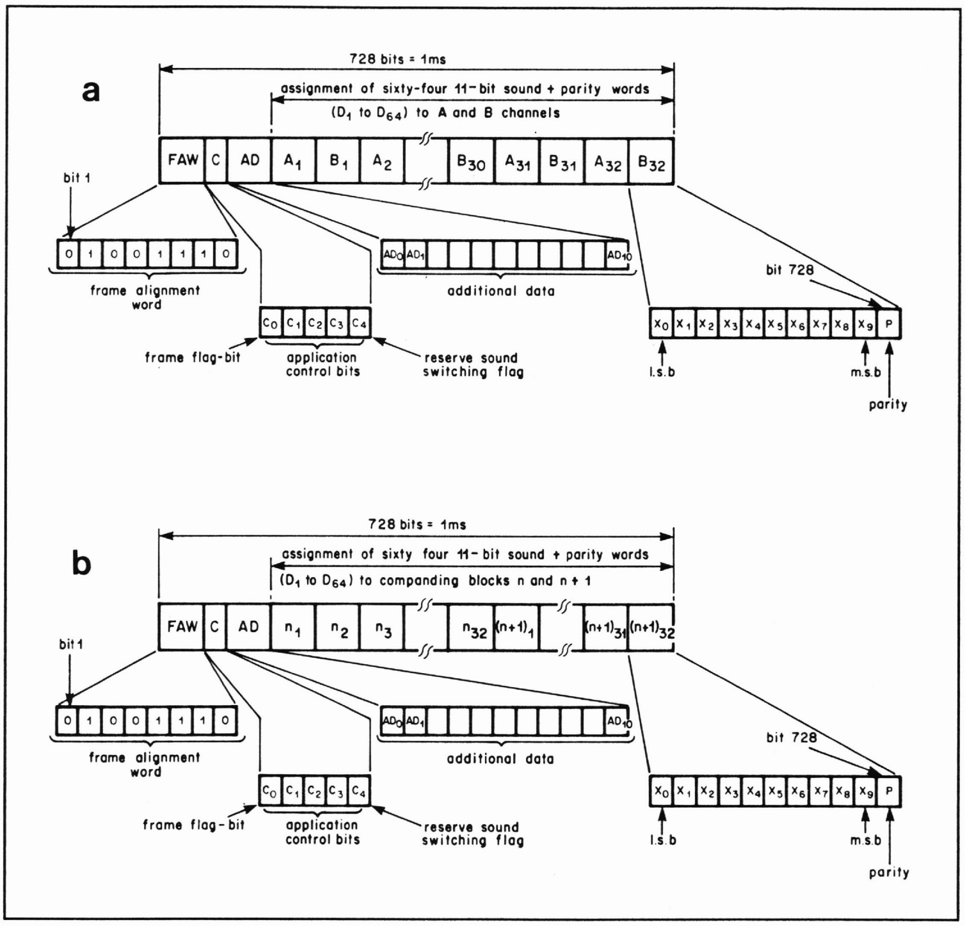


Fig. 3. (a) Structure of a 728-bit frame containing a stereo sound signal (before interleaving); (b) the same for a mono sound signal (also before interleaving).

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 C_1	C_2	C_3 *	Contents of 704-bit sound/data block
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

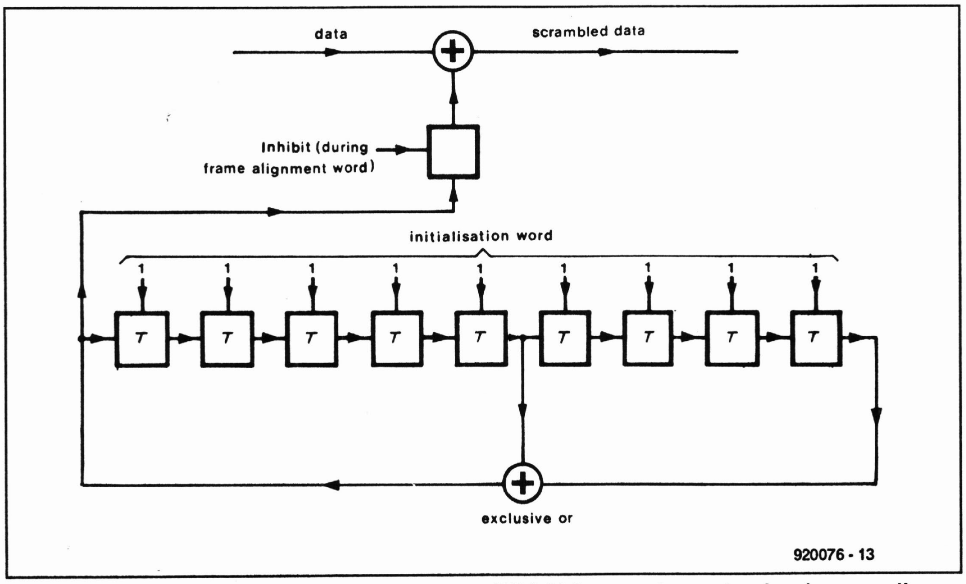


Fig. 4. Pseudo-random sequence generator (PRSG) for spectrum shaping (energy dispersal scrambling).

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

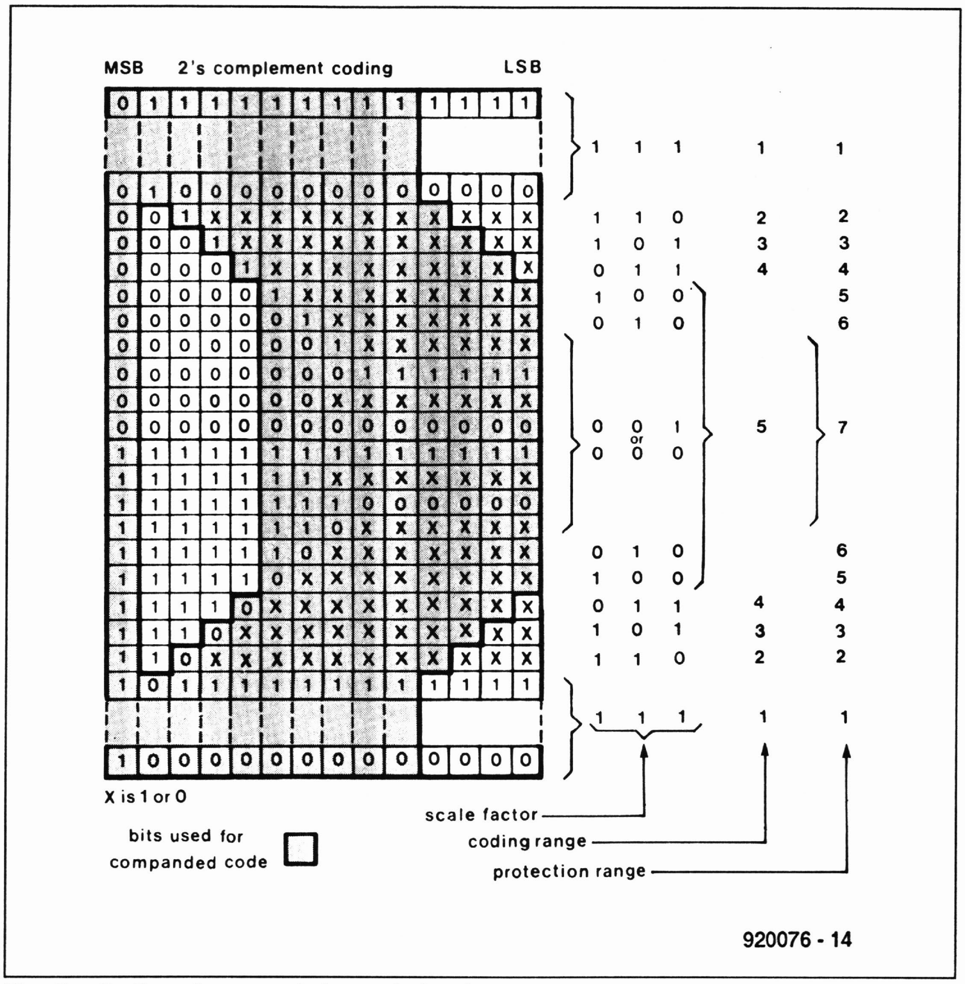


Fig. 5. Coding of companded sound signals.

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 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

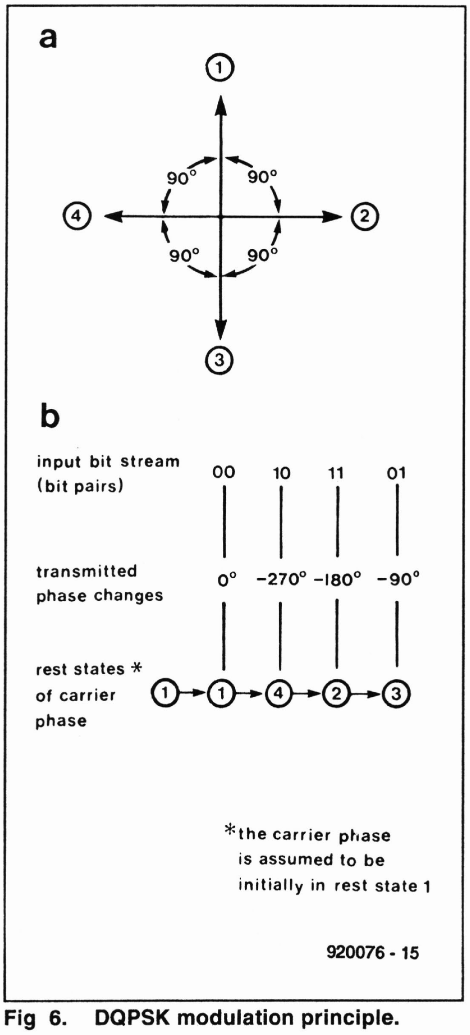


Fig 6. DQPSK modulation principle.

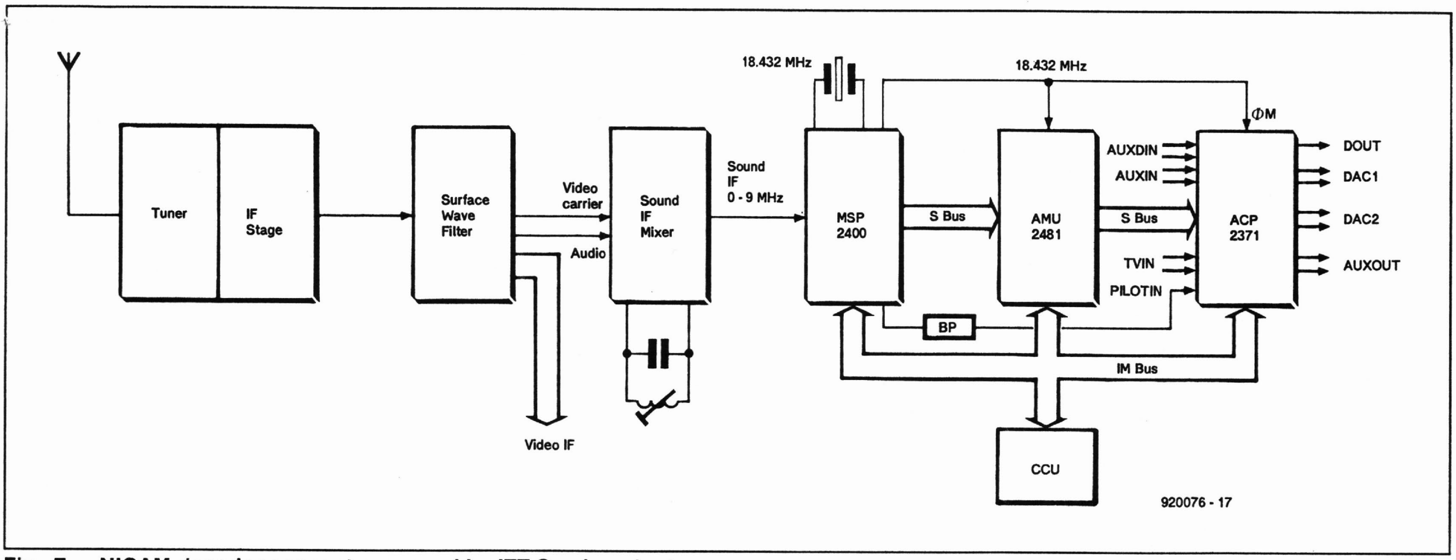


Fig. 7. NICAM decoder concept proposed by ITT Semiconductors.

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

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

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

and the filter has a constant group delay for all frequencies $\leq 1/t_s$. The filter made on the basis of the above transfer characteristic has a 100% cosine rol-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.

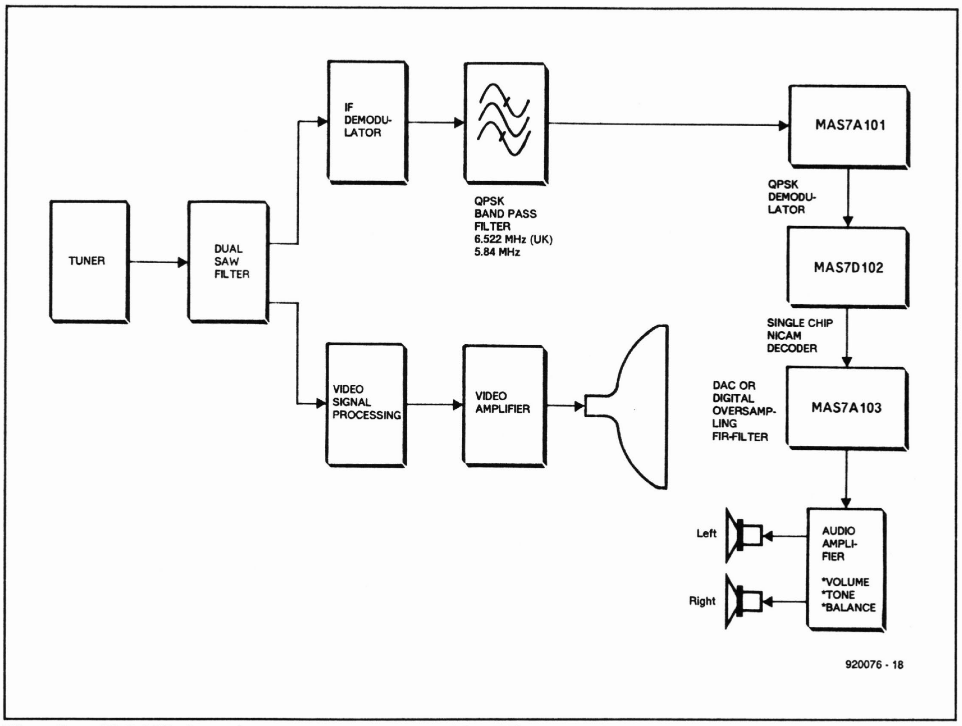


Fig. 8. NICAM decoder concept proposed by Micronas Inc.

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. ■

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'*.

3. *Specification of Television Standards for 625-line System-I transmissions in the United Kingdom*. Department of Trade and Industry, Radio Regulatory Division, London, 1984.

NICAM DECODER

The decoder described here is aimed at the experienced radio and TV enthusiast who wants to upgrade an existing TV set or video recorder with NICAM digital stereo sound. Suitable for PAL TV systems 'I' (UK) and 'B/G' (Scandinavia, Belgium, Spain and others), the decoder is a compact and simple to control circuit that can either be built as a set-top extension, or incorporated into a TV set.

Design by Rob Krijgsman PE1CHY

THIS decoder is based on a NICAM chip set developed by Micronas Inc. of Finland. The set consists of the MAS7A101 QPSK demodulator, the MAS7D102 NICAM decoder, and the MAS7A103 dual D-A converter. The chip set allows two high-quality audio channels (stereo or dual-language mode) to be recovered from a NICAM signal at 5.85 MHz or 6.552 MHz (if broadcast, and depending on the PAL system used) in the TV baseband spectrum. All that is needed to be compatible with either of the two PAL systems is to fit the correct input filter, a jumper and a quartz crystal for the demodulator clock.

Three ICs

As shown by the block diagram in Fig. 1, the upper part (say, above 5 MHz) of the TV baseband spectrum is first filtered to extract the NICAM signal centred around 5.85 MHz (system B/G) or 6.552 MHz (system I). The insertion loss of the band-pass filter is compensated by an amplifier.

MAS7A101 QPSK demodulator

The NICAM signal is applied to the MAS7A101 QPSK demodulator IC. This is a pretty complex integrated circuit, whose internal architecture is given in Fig. 2. The QPSK signal at the input is buffered before it is applied to a multiplier circuit which consists of analogue switches. The switches are opened and closed by a signal derived from a phase-controlled quartz crystal oscillator. The crystal frequency equals four times the NICAM subcarrier frequency, i.e.,

$$5.850 \times 4 = 23.400 \text{ MHz}$$

for PAL systems B and G, or

$$6.552 \times 4 = 26.208 \text{ MHz}$$

for PAL system I. The quartz oscillator is locked to the received NICAM signal by means of a PLL. The demodulated signal is taken through a switchable low-pass filter, and subsequently split into two.

One signal is sent to a second PLL which serves to recover the 728-kHz NICAM bit clock from the demodulated signal. The crystal-controlled VCO in this PLL operates at eight times the NICAM bit clock, or 5.824 MHz. This VCO also provides the central clock signal for the other ICs in the decoder.

The other demodulated signal is sent to a slicer circuit where it is converted into a bi-

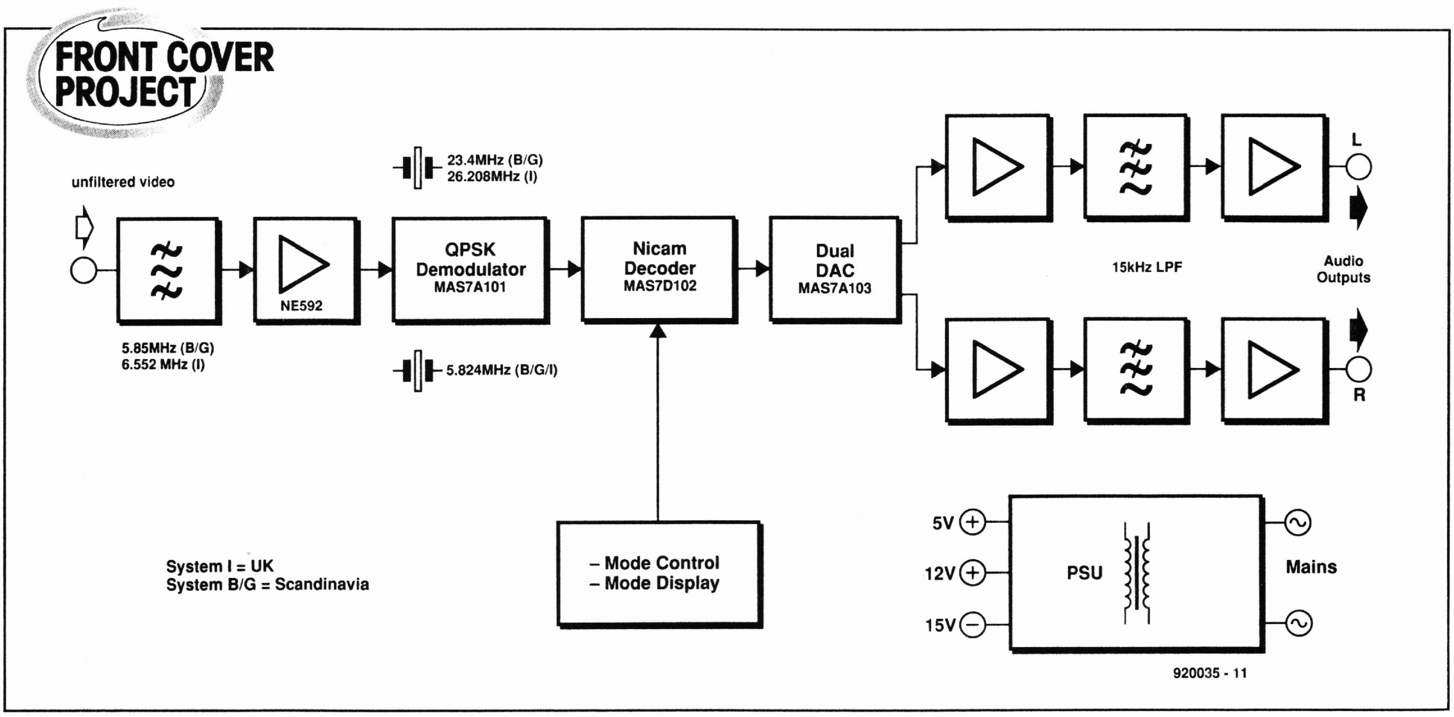


Fig. 1. Block diagram of the decoder.

nary digital signal. The recovered clock signal and the binary signal are available at the corresponding outputs of the MAS7A101.

MAS7D102 NICAM decoder

The MAS7D102 NICAM decoder (Fig. 3) uses the recovered NICAM bit clock to tackle the decoding proper of the bitstream supplied by the QPSK demodulator. The decoding process involves quite a lot: descrambling, de-interleaving, error detection and correction, and reconstruction of the original 14-bit sound samples in both channels. The MAS7D102 can be programmed or wired to supply digital output signals suitable for one of three different bus systems: the I²S-bus (Philips), the S-bus (ITT), or the DAC-bus (Toshiba). Many functions of the IC can be controlled either via an I²C link, or by means of external hardware. The latter option is exploited here, and has the advantage of obviating a microcontroller and a dedicated control program.

With reference to the IC architecture shown in Fig. 3, it is seen that the digital signal supplied by the QPSK demodulator is split into two. One signal is fed to a synchronisation logic section where the FAW (frame alignment word) is detected and extracted. The FAW is never scrambled. The other copy of the digital signal is sent to the descrambler circuit, which serves to counteract the energy dispersal (spectrum-shaping) scrambling applied at the transmitter. When the decoder chip is first switched on, it uses the standard descrambling initialisation word '11111111', which enables reception of non-encrypted NICAM broadcasts. External hardware is required to be able to change the initialisation word (or 'seed') 'on the fly' when the system is used for reception of Pay-TV transmissions using encrypted NICAM audio.

Returning to the operation of the MAS7D102, the control information bits C_1-C_2-C_3-C_4 are extracted from the datastream. These bits enable the receiver to determine the type of programme material: i.e., dual-language or stereo. The decoded control bits are available in an I²C register as well as on an output port. The latter allows a simple display to be connected that indicates the receiver mode. The sound samples are fed to the de-interleaver, and from there to the error detection/correction circuit. Finally, they are de-companded to their original 14-bit resolution, and fed to the output of the IC according to the selected signal format (I²S-bus, S-bus or DAC-bus). The format selection is effected via the I²C bus, or via logic levels applied to the configuration (CONFIGx) pins, which in addition allow you to select between mono-A or mono-B during dual-language broadcasts. The functions of all registers contained in the MAS7D102, and the configuration options that can be set in hardware, are given on page 41.

MAS7A103 dual DAC

This IC converts the 14-bit sound samples furnished by the decoder into two analogue audio signals. Since the output datastream of

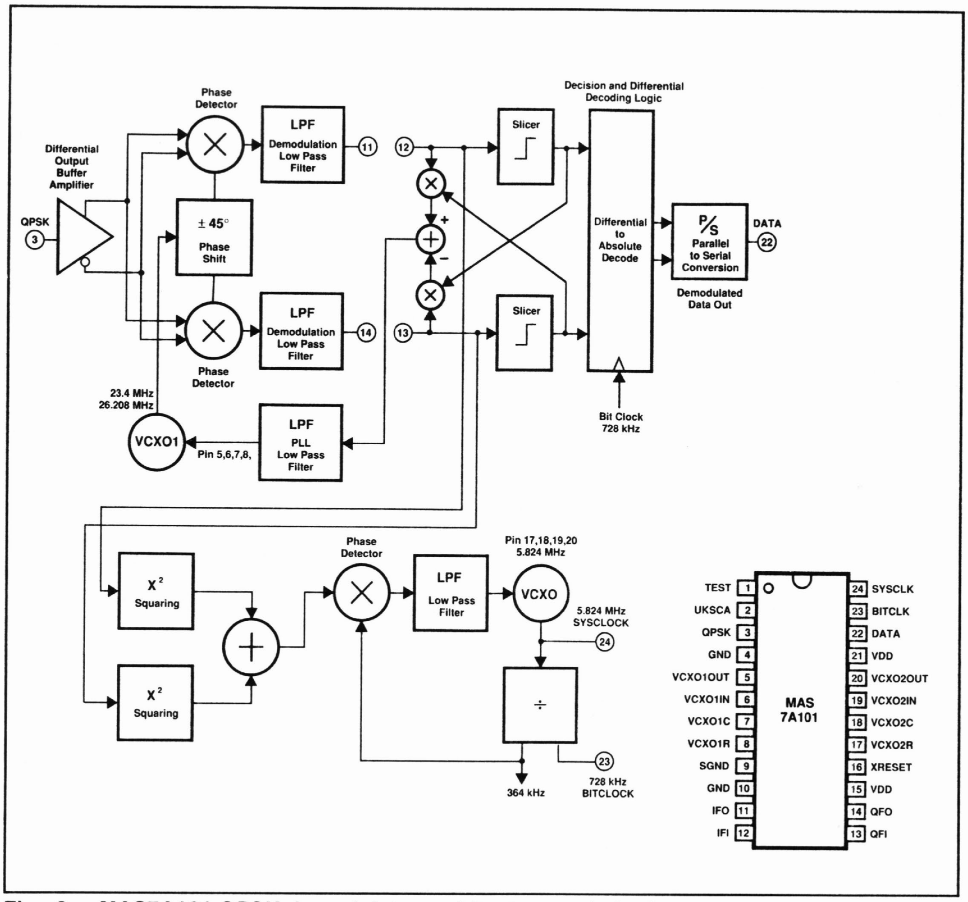


Fig. 2. MAS7A101 QPSK demodulator architecture and pinning.

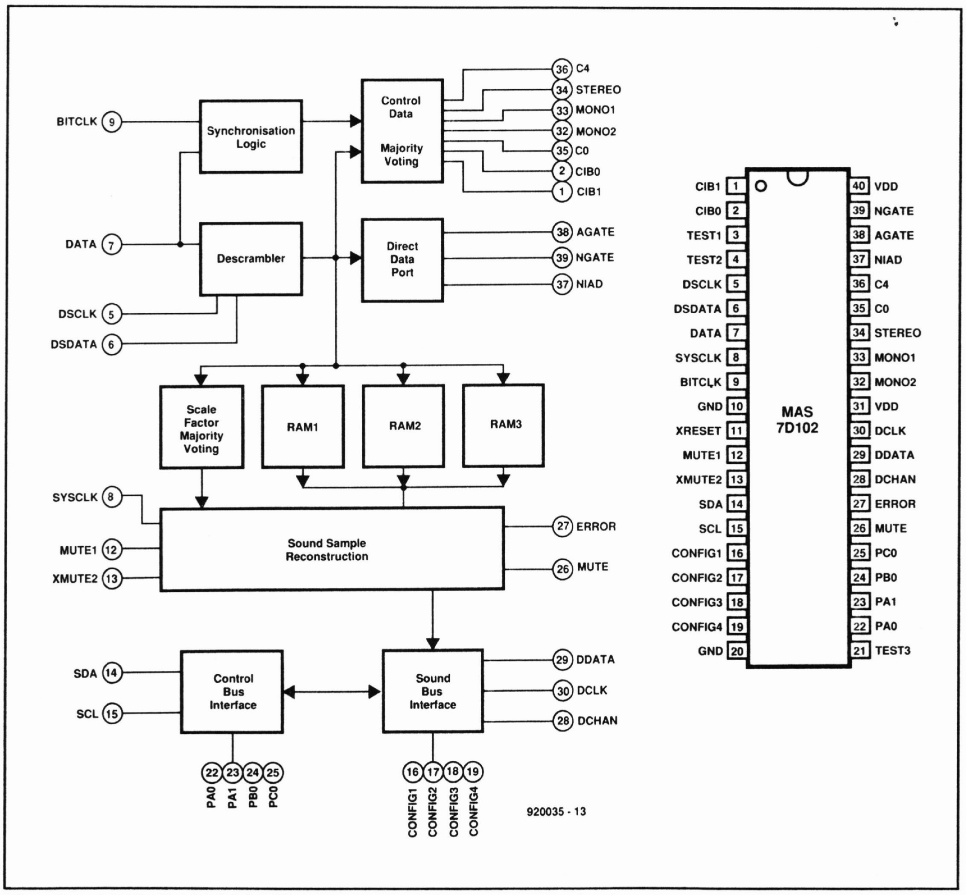


Fig. 3. MAS7D102 NICAM decoder architecture and pinning.

the decoder IC is multiplexed, the first task of the DAC is to extract and separate the information that belongs with each channel. Next, the two digital signals are converted into analogue ones by *R*-2*R* ladder networks. These supply output currents rather than voltages, so that two external opamps are required to obtain audio signals that can be fed to an amplifier. Before that can be done, however, the audio signals need to be taken through a 15-kHz low-pass filter to remove the residue of the 32-kHz sampling signal.

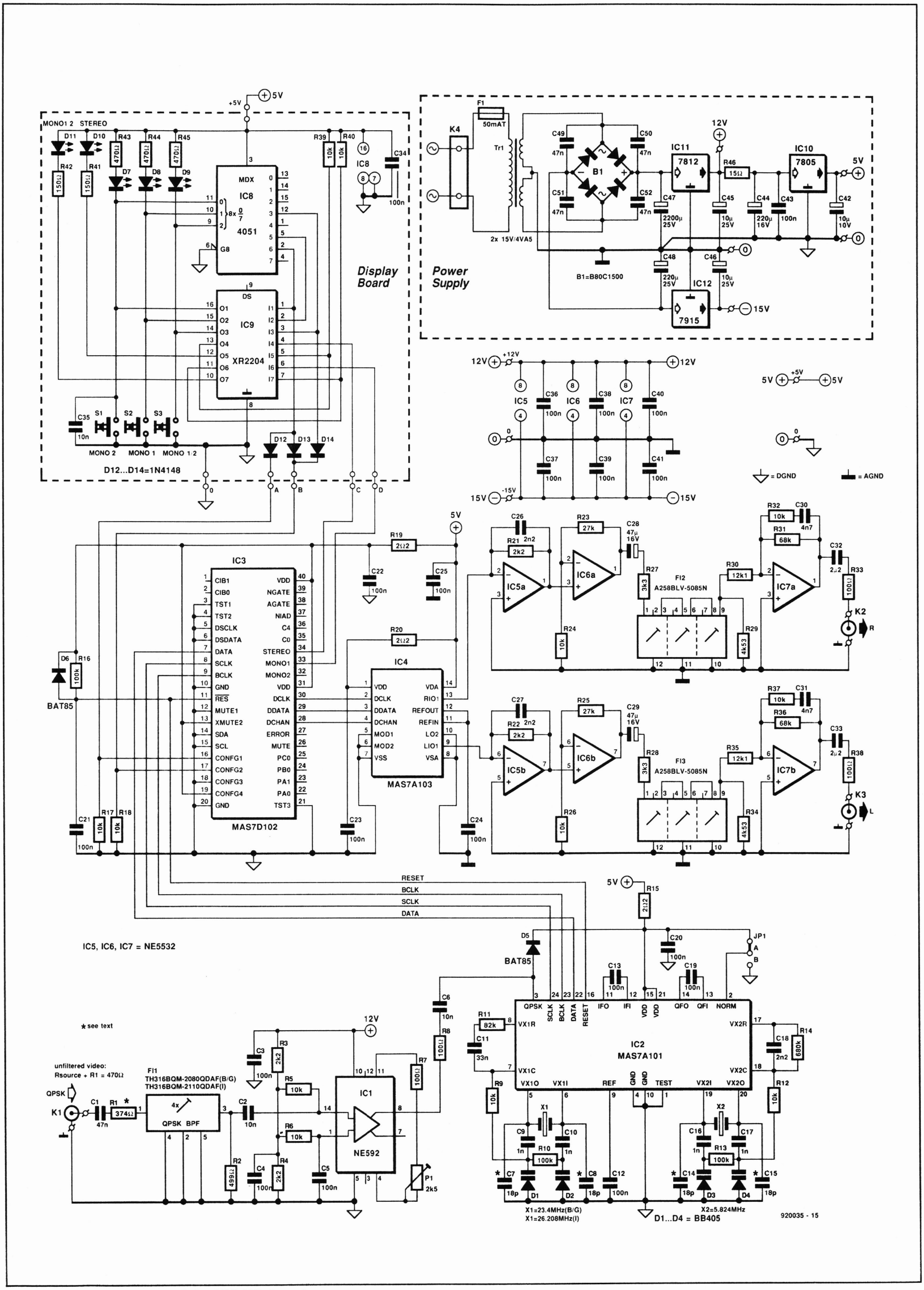


Fig. 4. Circuit diagram of the NICAM decoder.

Country	TV system	Stereo sound	Main sound subcarrier	Stereo subcarrier	QPSK filter Fl1	Quartz X1	Jumper JP1
Scandinavia	PAL B/G	Digital; NICAM-B	5.5 MHz	5.850 MHz	TH316BQM-2080QDAF	23.400 MHz	A
United Kingdom	PAL I	Digital; NICAM-I	6.0 MHz	6.552 MHz	TH316BQM-2110QDAF	26.208 MHz	B
Germany; Switzerland; Benelux	PAL B/G	Analogue	5.5 MHz	5.740 MHz	—	—	—
Italy; Spain	PAL B/G	Digital; NICAM-B	5.5 MHz	5.850 MHz	TH316BQM-2080QDAF	23.400 MHz	A

Table 1. The choice of two components in the NICAM decoder, and the position of a jumper, depends on the country you live in.

This filter takes us back to the block diagram in Fig. 1, with the final remark that J17 de-emphasis is applied on the audio signals.

Practical circuit

After studying some of the background theory on NICAM (to be found elsewhere in this issue), and having acquired samples and datasheets of the NICAM chip set, the author set out to work, and was able to design and build a simple NICAM decoder that was tested with the aid of NICAM broadcasts received from the Belgian national TV station BRT (these broadcasts were experimental at the time, and are currently regular). The BRT transmits NICAM-728 according to PAL standard B/G. Initially, the application circuits suggested by Micronas were built, and from there on further experiments evolved to produce a repeatable decoder.

The final result is an uncluttered circuit shown in Fig. 4. The unfiltered video signal taken from a suitable point in the TV tuner (more about this further on) is applied to the input of a four-section bandpass filter tuned to 5.85 MHz (6.552 MHz for the UK system-I). The input impedance of the decoder is about 900 Ω. To ensure that the input of the bandpass filter is correctly terminated, the sum of the source impedance and resistor R1 must be 470 Ω, as indicated in the circuit diagram. The Type TDA2541 demodulator IC, for instance, has an output impedance of about 100 Ω. The bandpass filter used is a ready-made, pre-aligned module from Toko (note that different types are required for systems B/G and system I). Its insertion loss lies between 8 dB and 16 dB. This is compensated by amplifier IC1, whose gain can be set as required with the aid of preset P1 to give a signal level of 200 to 800 mV_{pp} at the input of the QPSK demodulator, IC2.

As indicated in the diagram, the frequency of quartz crystal X1 is determined by the PAL TV system used in your country. Jumper JP1 should also be fitted in accordance with the system used, to select the appropriate low-pass characteristic in the demodulator. Information on the options in the circuit depending on the TV system used is summarized in Table 1.

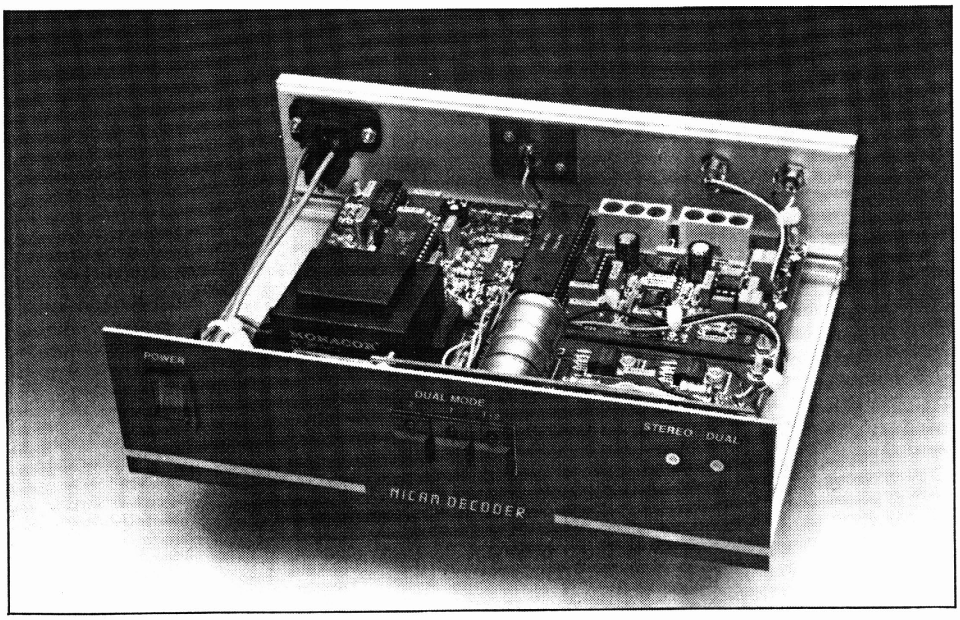
Depending on the characteristics of the crystals used in positions X1 and X2, the exact values of C7-C8 and C14-C15 may have to be changed from those shown in the circuit diagram. Given that the quartz crystals probably have to be cut to order (the frequencies being non-standard as far as we have been able to find out), some experimenting may be required to obtain the correct oscillator frequencies.

The demodulator, IC2, supplies the recovered 728-kHz bit clock, the digital NICAM signal, and the 5.824-MHz system clock to the decoder, IC3. An *R-C* network, R16-C21, resets the demodulator and the decoder ICs at power-on.

Mode selection is effected with configuration bits config1 and config2. The available options are mono-2, mono-1 and mono-1/2 (dual language mode). The logic bit combinations required for these settings are supplied by IC8, IC9 and three push-buttons, S1, S2 and S3. The combination of these parts forms a kind of three-position flip-flop with a built-in latch function, a debounce circuit and an indication (on five LEDs). Capacitor C25 ensures that the 'mono-2' mode is automatically selected at power-on.

Diodes D12, D13 and D14 provide the required logic levels at the CONFIG inputs of the decoder IC. LEDs D10 and D11 indicate the currently transmitted mode: dual-language (mono-1/2) or stereo. This indication can not be changed by pressing the MODE switches.

Like the QPSK demodulator IC, the NICAM decoder, IC3, is used in a standard application circuit as suggested by the manufacturer. Similarly, few surprises are found in the link to the dual DAC, IC4, and the subsequent two-stage opamp-based current-to-voltage converters/amplifiers. It will be noted, though, that the opamps work from a symmetrical (+12 V/−15 V) supply. The gain of IC6a and IC7a in the right (R) output channel is set such that the loss introduced by the 15-kHz low-pass filter, Fl2, is overcome whilst ensuring an audio output level that is compatible with other equipment driving a amplifier 'line' input. The same goes, of course, for the corresponding components in the left (L) channel. The low-pass filters are, again, ready-made pre-aligned modules from Toko. Here, we are dealing with two A258BLV-5085N three-section *L-C* filters (the designer apologizes for the type numbers). Finally, the J17 de-emphasis networks in the right and left audio channels are formed by R32-C30 and R37-C31 respectively. The outputs of the NICAM decoder are capable of driving amplifier 'line' inputs.



A look into the completed prototype of the decoder.

Construction

First, cut the printed circuit board (Fig. 5) into three to separate the power supply

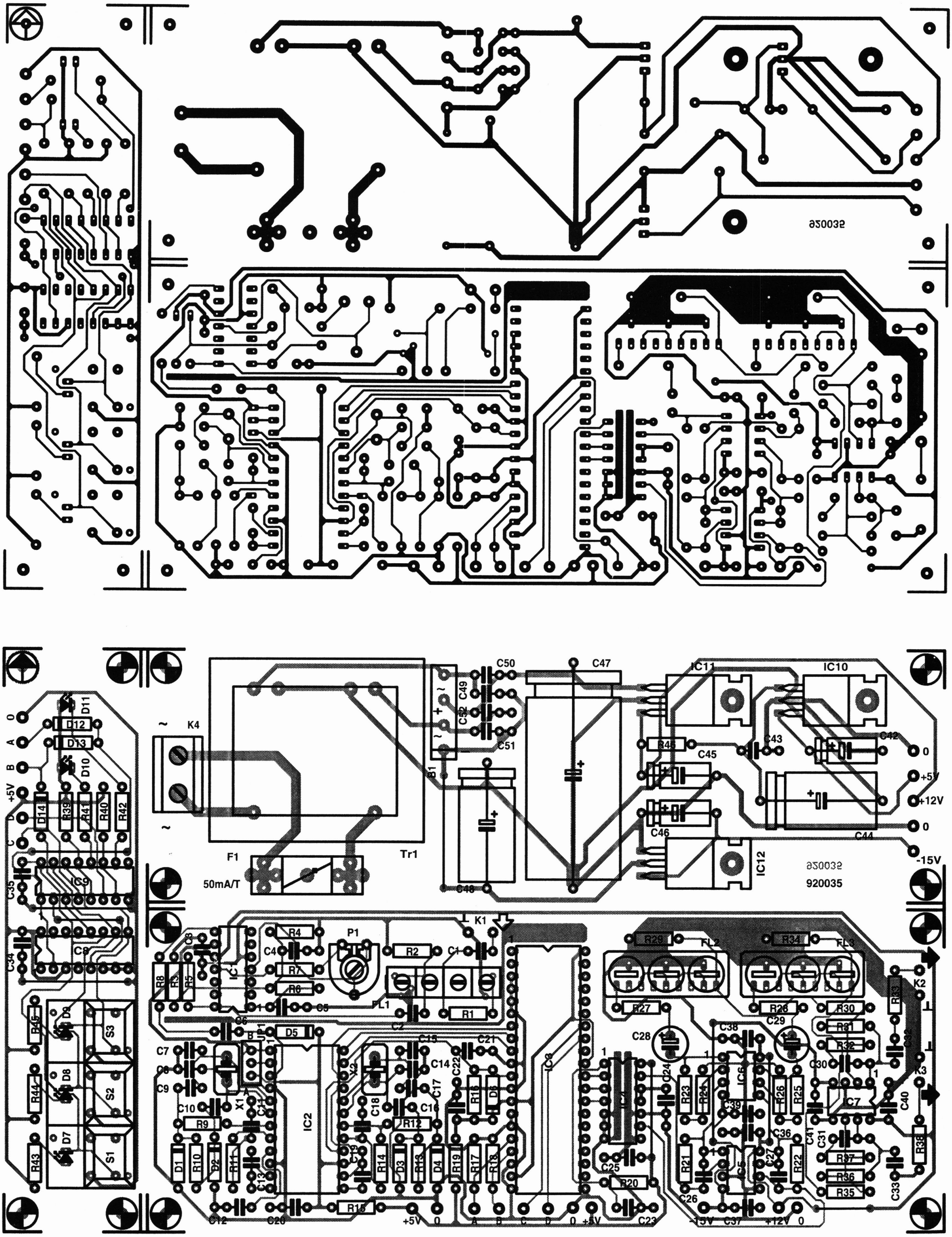


Fig. 5. Track layout and component mounting plan of the single-sided PCB for the NICAM decoder.

board, the decoder board and the keyboard. The population of these boards is entirely straightforward, and should not present problems. It is recommended to use IC sockets. The voltage regulators are bolted straight to the power supply board, and do not need heat-sinks. The fuse is fitted in a holder with a protective plastic cap. On the decoder board, the section with the blue (or red) core in the QPSK bandpass filter, Fl1, is at the side of the NICAM decoder chip, IC2.

The keyboard/display section of the printed circuit board has on it three Digitast press-keys with a built-in LED. The front panel of the enclosure for the NICAM decoder (if used) must be cut and drilled to allow the push-buttons and the two LEDs to the right of the board to protrude—more about this further on.

For an initial test, the completed boards are interconnected. Switch on, and check the presence of the correct supply voltages at a number of points. Press the keys and see if the associated LEDs light. If this works all right, stop, and start thinking very hard about

Finding the input signal

The present NICAM decoder is intended as an upgrade for existing TV sets, set-top TV tuners or video recorders. In nearly all cases, this equipment will have to be opened or modified to find or create a point where the NICAM signal can be 'tapped' and fed to the decoder. The following points should be taken into account:

1. Opening your TV set or VCR in most cases voids your warranty on this equipment.
2. The chassis of most older TV sets is connected direct to the mains. **Never** work on such a TV set without using an isolating transformer.
3. Make sure you have the service documentation (or at least a circuit diagram) of the equipment.

The intrepid among you should be looking for a for a point at the input of the sound demodulator where a signal is available that contains as little video information as possible. In most cases, the input signal of the main FM demodulator (5.5 MHz for system-B/G, or 6.0 MHz for system-I) is taken through a ceramic band-pass filter to suppress the components in the video spectrum. In general, it is best to 'tap' the signal ahead of this filter. The minimum level of the signal to be fed to the NICAM decoder is about 50 mV. In all cases, the load presented by the input of the NICAM decoder should be as small as possible. This may require an emitter follower to be fitted as discussed below.

A little more complex, but certainly more convenient as far as the filtering is concerned, is a TV set or a VCR with a so-called quasi-parallel sound demodulator system. The designer used his HR-S5000E video recorder from JVC to supply the NICAM signal. After studying the service documentation that came with the VCR, it

COMPONENTS LIST

Resistors:

1	374Ω (see text)	R1
1	499Ω 1%	R2
4	2kΩ2	R3;R4;R21;R22
12	10kΩ	R5;R6;R9;R12;R17; R18;R24;R26;R32; R37;R39;R40
4	100Ω	R7;R8;R33;R38
3	100kΩ	R10;R13;R16
1	82kΩ	R11
1	680kΩ	R14
3	2Ω2	R15;R19;R20
2	27kΩ	R23;R25
2	3kΩ3	R27;R28
2	4kΩ53 1%	R29;R34
2	12kΩ1 1%	R30;R35
2	68kΩ	R31;R36
2	150Ω	R41;R42
3	470Ω	R43;R44;R45
1	15Ω	R46
1	2kΩ5 preset H	P1

Capacitors:

5	47nF ceramic	C1;C49-C52
2	10nF ceramic	C2;C6
20	100nF	C3;C4;C5;C12;C13; C19-C25;C34; C36-C41;C43
4	18pF	C7;C8;C14;C15
4	1nF ceramic	C9;C10;C16;C17
1	33nF	C11
3	2nF2	C18;C26;C27
2	47µF 16V radial	C28;C29
2	4nF7	C30;C31
2	2µF2 50V solid MKT	C32;C33
1	10nF	C35
1	10µF 10V	C42
1	220µF 16V	C44
2	10µF 25V	C45;C46
1	2200µF 25V	C47
1	220µF 25V	C48

Semiconductors:

4	BB405	D1-D4
2	BAT85	D5;D6
5	LED red 3mm	D7-D11
3	1N4148	D12;D13;D14
1	B80C1500	B1
1	NE592	IC1
1	MAS7A101	IC2
1	MAS7D102	IC3
1	MAS7A103	IC4
3	NE5532AN	IC5;IC6;IC7
1	4051	IC8
1	XR2204 or ULN2004	IC9
1	7805	IC10
1	7812	IC11
1	7915	IC12

Miscellaneous:

1	3-way pin header with jumper	JP1
3	RCA (phono) socket	K1;K2;K3
1	2-way PCB terminal block; pitch=7.5mm	K4
3	Digitast push-button (narrow) with integral LED	S1;S2;S3
1	mains transformer 2x15V @ 4.5VA; Monacor (Monarch) type VTR-4215	Tr1
1	TH316BQM-2080QDAF *	Fl1
1	TH316BQM-2110QDAF **	Fl1
2	A258BLV-5085N	Fl2;Fl3
1	Quartz crystal 23.400 MHz *	X1
1	Quartz crystal 26.208 MHz **	X1
1	Quartz crystal 5.824 MHz	X2
1	Fuse 50mA slow; with PCB mount holder and cap	F1
1	Printed circuit board	910035
1	Front panel foil	910035-F
1	Metal enclosure Telet 55205	

* PAL TV system B or G
** PAL TV system I

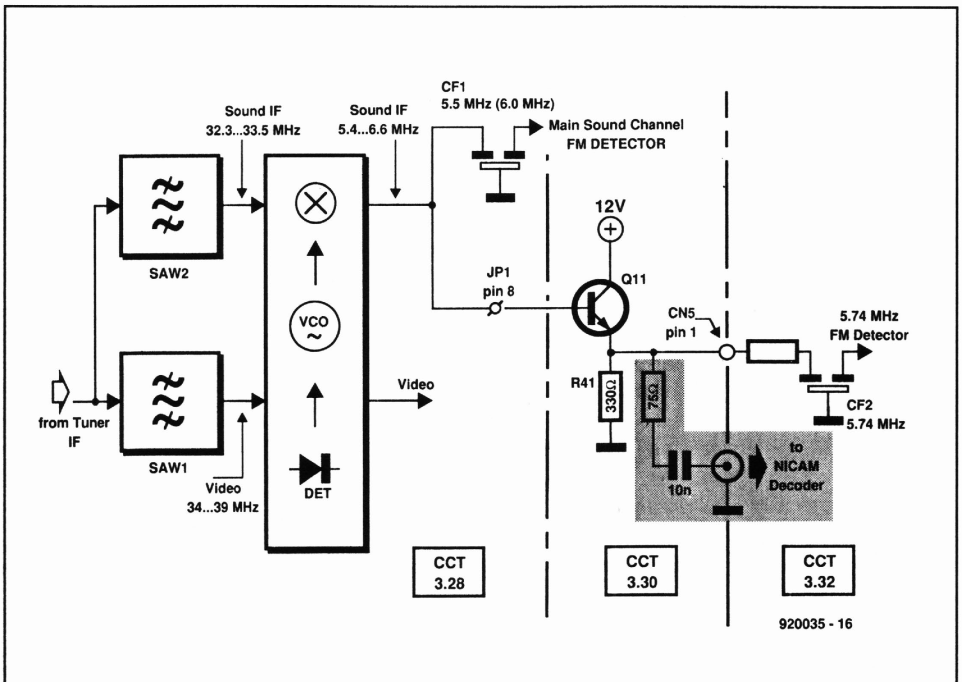


Fig. 6. This drawing illustrates how a suitable decoder input signal was found in the JVC HR-S5000E video recorder.

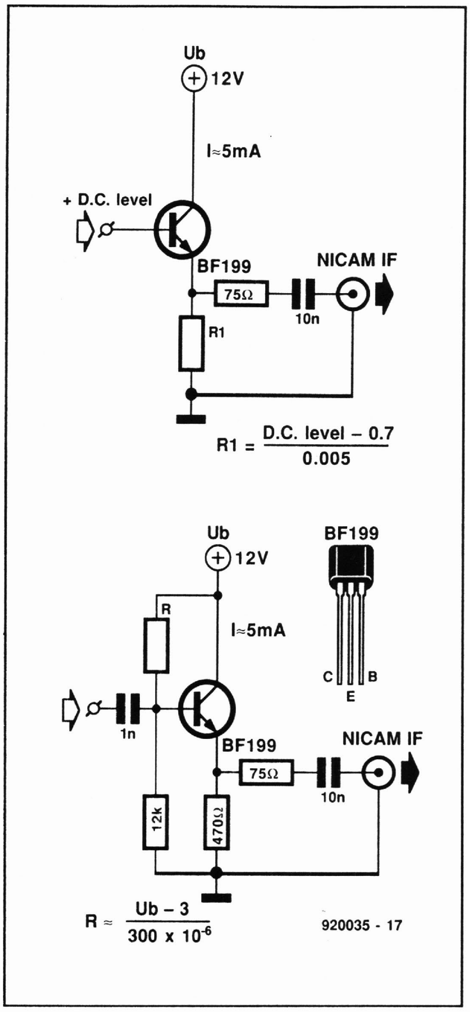


Fig. 7. Emitter followers for NICAM signals on a relatively small d.c. component (7a), and NICAM signals on a d.c. component so large that a.c. coupling is required (7b).

was decided to try the output signal of an emitter follower located between the 'sound IF' output and the input of the 5.74 MHz ceramic filter fitted for the German 'dual-language' demodulator. The search for this emitter follower, Q11, was complicated by the fact that it happened to 'reside' between three pretty large circuit diagrams. Figure 6 shows essentially what has been added to the VCR: one resistor, a coupling capacitor and a 'phono' socket do a perfect job.

As already mentioned, an emitter follower may have to be used to prevent the input signal of the sound demodulator disappearing when the NICAM decoder is connected. One of the circuits shown in Fig. 7 will be adequate. The first, Fig. 7a, may be used when the signal is superimposed on a d.c. level between 0.3 and 0.7 times the supply voltage. The other, Fig. 7b, has an input coupling capacitor, and is used in all other cases. Remember, you are dealing with signals of 5 MHz and higher here, so keep component wires as short as possible.

Testing

The input impedance of the NICAM decoder is fairly high: about 900 Ω. This means that conventional coax cable with an impedance of 50 Ω or 75 Ω can not be used unless its length remains below 50 cm or so. Longer cables of either type will cause reflections and serious mismatches, resulting in attenuation of the NICAM subcarrier. If you can not go round the use of a relatively long, low-impedance, coax cable between the TV set and the NICAM decoder, be sure to fit a terminating resistor across socket K1. This resistor prevents reflection and high-frequency loss to some extent. When a 50-Ω cable is used, fit a 52.9-Ω terminating resistor, and change R1 into 444 Ω. Similarly, when a 75-Ω cable is used, terminate it with 81.8 Ω, and change R1 into 431 Ω. In some cases, ordinary screened cable as used with audio equipment, or car radio coax cable (if you can get it), is the best alternative. In any case, do not fit BNC or similar low-impedance RF sockets at the TV side and the decoder input. On the prototype we used an insulated 'phono' (RCA-style) socket for chassis mounting. An insulated socket is required to prevent an earth loop between the analogue and digital ground rails.

Demodulator input level

Switch on the NICAM decoder, and tune the

PROGRAMMING THE MAS7D102 NICAM DECODER

The bus format can be selected either by applying logic levels to pins Config4 and Config3, or by programming control bits Config4 and Config3 via the I^2C microprocessor interface.

Config 4	Config3	DAC bus format
0	0	High-Z
0	1	S-bus
1	0	I^2S bus
1	1	Toshiba DAC bus

The pins Stereo, Mono1 and Mono2 are active-low outputs that indicate the current NICAM transmission mode.

Stereo	Mono1	Mono2	Type of transmission
0	1	1	Stereo signal
1	0	1	Dual language transmision
1	1	0	One mono sound channel and 352 Kbit/s data channel
1	1	1	No sound signal. Transparent 704 Kbit/s data transmission, or no NICAM encoded transmission

During dual-language transmissions, the main language selection is controlled by input pins Config2 and Config1.

Config2	Config1	DAC bus	Sound sample order
0	0	I^2S/Toshiba	M1 M1
0	1	I^2S/Toshiba	M1 M2
1	0	I^2S/Toshiba	M2 M1
1	1	I^2S/Toshiba	M2 M2
0	0	ITT	M1 M1 M1 M1
0	1	ITT	M1 M1 M2 M2
1	0	ITT	M2 M2 M1 M1
1	1	ITT	M2 M2 M2 M2

The decoder has two addresses on the I^2C bus. Address 4E (hex) is for writing to the decoder, and address 4F (hex) for reading from the decoder.
There are three status registers (read) and three control registers (write) that can be accessed. The three control registers can be addressed individually by the two most significant bits of each control word. The three status registers can be addressed as a complete set only.

The function of the Da, Db and Dc control bits is to define external ports Pa, Pb and Pc as inputs or outputs, as shown below.

Da	Db	Dc	Pa1	Pa0	Pb0	Pc0
0	0	0	out	out	out	out
0	0	1	out	out	out	in
0	1	0	out	out	in	out
0	1	1	out	out	in	in
1	0	0	in	in	out	out
1	0	1	in	in	out	in
1	1	0	in	in	in	out
1	1	1	in	in	in	in

The **status registers** of the MAS7D102 have the following structure:

Status register	D7	D6	D5	D4	D3	D2	D1	D0
1	Osn	CI1	CI0	C4	C3	C2	C1	C0
2	Ser10	Ser9	Mute	TestS	Pa1	Pa0	Pb0	Pc0
3	Ser8	Ser7	Ser6	Ser5	Ser4	Ser3	Ser2	Ser1

- The Osn status bit goes high when the decoder is not synchronised.
- CI0 and CI1 are the two CI bits extracted from each NICAM frame.
- C4-C0 are the C bits associated with the current NICAM transmission, and they indicate the mode as shown below.

C1	C2	C3	NICAM transmission mode
0	0	0	Stereo transmission
0	1	0	Dual language transmission
1	0	0	One mono channel plus data transmission
1	1	0	One 704 Kbit/s data channel

- C0 is the Frame Flag bit that indicates the super frame pattern of the NICAM transmission.
- C4 is the Reserve Sound Switching flag, which goes high when the FM mono signal carries the same programme as the digital stereo signal.
- The Mute status bit goes high to indicate that the decoder has been muted for some reason.
- TestS is a test status indication bit reserved for test purposes.
- Pa1 and Pa0, Pb0 and Pc0 indicate the status of the corresponding external pins, when they are configured as input ports.
- The Ser10-Ser1 bits show the value contained in the sample error counter. This counter is incremented whenever an erroneous sample is detected. The control processor can read the error count at suitable time intervals, and take decisions depending on the error rate.

Control register	D7	D6	D5	D4	D3	D2	D1	D0
1	0	0	Test1	Test2	MuteS	MuteA	x	Reset
2	0	1	x	x	Config4	Config3	Config2	Config1
3	1	Da	Db	Dc	Pa1	Pa0	Pb0	Pc0

- Test1 and Test2 are reserved for test purposes, and must be set low.
- The MuteS control bit mutes sound output. Active high.
- The MuteA control bit mutes sound output and resets the synchronisation of the decoder completely. Active high.
- The Reset control bit resets the decoder completely. Active high.

TV set or the VCR to a station transmitting NICAM sound. Use an oscilloscope to check the signal level at pin 3 of the QPSK demodulator, IC2. The level should be between 200 mV_{pp} and 800 mV_{pp}. If necessary, adjust preset P1 to achieve a level of about 500 mV_{pp}.

QPSK demodulator PLL adjustment

Connect the scope to pin 11 of the QPSK demodulator IC. You should see a so-called 'eyes' waveform (which may be very difficult for the scope to trigger on). Adjust P1 so that the tops of the waveform are just below the supply voltage; i.e., they are just not clipped. This gives a signal level of about 5 V_{pp}. Move on to pin 7 of IC2. This supplies the error voltage of the demodulator PLL. It is a fairly 'messy' signal superimposed on a direct voltage, which will look like a broad band on the scope. Tune to a non-NICAM station, and back to the NICAM station again, to see how the PLL responds by locking on to the NICAM signal. For best performance of the PLL, the d.c. component in the error signal should be at about half the supply voltage, i.e., 2.5 V. When it is too close to either 0 V or +5 V, change the crystal matching capacitors, C7 and C8, until the centre of the band is at about 2.5 V. Increase the capacitor values (to 22 pF or 27 pF) when the d.c. component is too low, and decrease them (to 15 pF or 12 pF) when the d.c. component is too high. Try to get as close to 2.5 V as you can. The exact oscillator frequency will be very difficult to measure at pin 5 of IC2 because the impedance is high locally. This means that any capacitive load, however small, formed by a test probe will detune the crystal oscillator to some extent.

Clock recovery PLL adjustment

The 5.824-MHz PLL for the NICAM clock signal recovery is adjusted in a similar manner to the QPSK PLL as discussed above. Connect the scope to pin 18 of IC2, and check that the error voltage has a d.c. component of about 2.5 V. If not, change the values of C14 and C15. It will be found that this error voltage is much 'cleaner' than the one used for

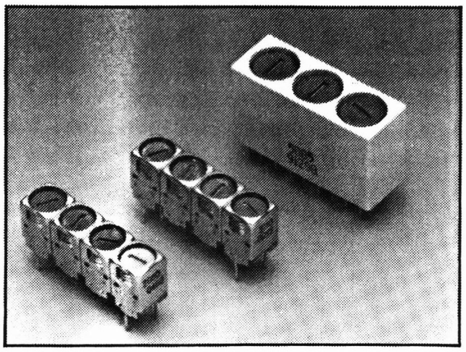


Fig. 8. The Toko QPSK bandpass filter (either for system I or B/G) and the 15-kHz low-pass filter used in the decoder.

controlling the first PLL. If the second PLL frequency is correct, pin 23 of the demodulator IC supplies a clock signal of 728 kHz, which is easily measured with a frequency meter.

That completes the adjustment of the NICAM decoder. If you have not already done so, connect a stereo amplifier to the outputs, and enjoy the programme!

Finishing touch

Some of you may want to fit the decoder permanently inside a TV set, while others may want to use it as a self-contained unit.

The prototype of the decoder was housed in an aluminium enclosure Type 55205 from Telet. The decoder and supply boards were fitted on a perspex plate that could be slid horizontally into the railings provided along the inside of the front and rear panels.

The keyboard PCB and the mains switch are fitted on to the front panel, for which a ready-made self-adhesive foil is available. This foil is used as a template to determine the locations of the holes to be cut in the front panel. A jig-saw is used to cut the rectangular clearances for the mains switch and the three push-buttons.

The keyboard PCB is mounted on four screws of which the (countersunk) heads are glued to the inside of the front panel. Plastic stand-offs are used to fit the PCB at the right distance behind the front panel.

The rear panel is drilled to hold the mains socket, the NICAM input socket and the two audio output sockets.

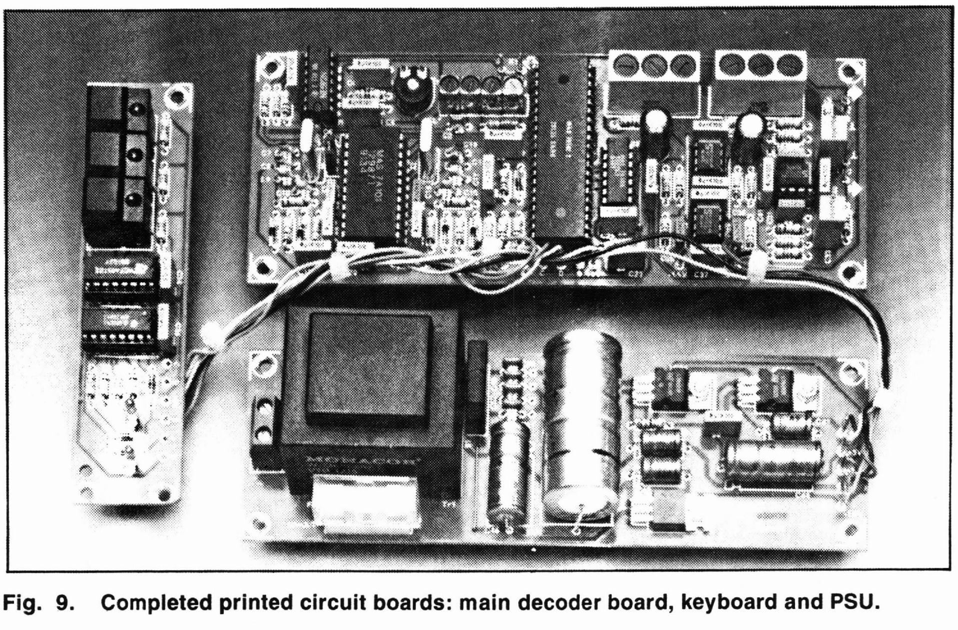


Fig. 9. Completed printed circuit boards: main decoder board, keyboard and PSU.

Conclusion

The NICAM decoder described here has been in use for some time now, and provides excellent stereo sound on broadcasts received from BRT1 and BRT2. Regrettably, the unit could not be tested in the UK, although suitable components (a 6.552-MHz QPSK bandpass filter and a 26.208 MHz quartz crystal) were available.

Although the construction and adjustment of the unit are fairly simple, finding a suitable input signal may be daunting if you have little experience in TV and VCR technology. We feel, therefore, that it is fair to warn beginners not to undertake this project until a dedicated TV tuner is available, which will be described in a future issue of *Elektor Electronics*.

Postscript for advanced users

As already mentioned, the MAS7D102 NICAM decoder has optional I^2C control, which may be used to access most of the internal registers. The SDA and SCL inputs of this IC are TTL-compatible, and may be connected to an I^2C bus via appropriate interfaces. If you have a PC available fitted with an I^2C interface (Ref. 1), you may use the information given in the MAS7D102 inset to implement software control on the NICAM decoder.

The descrambler on board the MAS7D102 can be loaded with a descrambling key other than the standard 'seed' used for non-encrypted NICAM broadcasts. Changes to the scrambling keys must occur synchronously at the transmitter and the receiver(s). The NICAM decoder IC provides a serial data input, Dsdata (pin 6), and a clock input, Dsclk (pin 5) to access an internal shift register. This register contains the descrambler key that is loaded in parallel into the descrambler one per frame. The shift register contents can be updated at any time with a maximum clock rate of 5 MHz. The time interval between the falling edge of the Ngate signal (pin 39) and the rising edge of the Agate signal (pin 38) is not allowed for descrambler key updating. During this interval, Dsclk (pin 5) must be held static. Output signals C0 (pin 35), Agate and Ngate may be useful for synchronisation purposes.

Happy listening! ■

The co-operation of Mr. Matti Antman of Micronas Inc., Mr. Peter de Vroome of Arcobel b.v., and our photo model Miss Diony Erven is gratefully acknowledged.

Reference:

1. I^2C interface for PCs. *Elektor Electronics* February 1992.

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MV601	12.50
455GC resonator	3.50
SLB0586	9.85
J309	3.95
2N5109	6.50
2SK146V	13.75
BF469/470	1.95
2SA968	8.25
2SC2238	8.25
2SA1216	27.50
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Policom RTP58	79.00
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78Lxx	1.45
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79Lxx	1.45
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BC517	0.40
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BF961	2.75
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80C32 SBC (MAY 91) HFL 139.00

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Complete HC/HCT, CMOS, LSTTL ranges available ex-stock

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32.768 kHz	1.50
100 kHz	11.50
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6.000 MHz	4.10
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9.216 MHz	6.50
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10.444 MHz	19.50
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24.000 MHz	9.50
100 MHz	19.50
116 MHz	14.75

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1n – 47n	0.65
56n – 100n	0.75
120n – 470n	0.85
560n – 820n	0.95
1µ0 – 1µ5	1.35
4µ7	4.50
10µ	9.50
Polystyrene 2.5%/5%; all values to 4n7	0.75
Polystyrene 1% radial; all values	2.45
1n feedthrough	0.75
1n coffin	0.75
PTFE foil trimmers	
5/10/20pF	1.85
40/65/100pF	2.60

DISPLAYS

HD1105	4.95
HD1107O	5.00
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HD1131O	5.00
7760R	3.75
LM016L	99.00
3½-digit LCD with LO-BAT	11.50

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Complete DIN, IDC, BNC, phono (gold-plated) and D-series available.

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PCB-mount rotary types: 3×4; 2×6; 1×12; 4×3 3.35

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E48 0.1% metal	2.25
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40µH/3A toroid	3.50
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Neosid:	
7A1S/7F1S/7T1S	3.00
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Toko:	
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SH10-683 6.25

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We supply all SMA HC, HCT, CD, LS, linear & memory ICs, R's, C's, and transistors used in EE projects. **One-off no problem.**

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excl. PCB, incl. enclosure see EE February 1990

Spectrum shift DEcoder kit incl. PCB and case. Works f.b. on Intelsat 27.5° West. Price incl. P&P 99.00

15-way Amstrad SRX interface connector 3.00

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FOCAL 10V516 SUBWOOFER (P&P 40.00) 295.00

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Monoblock unit incl. protection cct.	235.00	P&P 15.00
Case: the beautiful 2-HU 19-inch rack with integral heatsinks at the sides, as shown on EE Nov. 91 front cover:	325.00	P&P 30.00*
Ancillary PSU, incl. Tr1 (220V 50Hz only):	69.00	P&P 15.00
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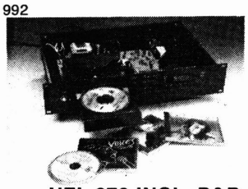
Set: Stereo LFA150-A, excl. Tr2:	975.00	P&P 50.00*

* surface mail only. P&P prices outside Europe on request.

BUILD A COMPACT DISK PLAYER

Elektor Electronics January 1992



Kit includes:
- CD mechanism CDM-4/19.
- CD loader 88.
- Main board with servo and decoder electronics.
- Controls and display PCB, populated, with connecting cables.
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- Headphone connector
- Extensive documentation, assembly instructions, etc. (English).

HFL 279 INCL. P&P

These kits are sold as 'electronic surplus', and are not refunded when found to be incomplete or defective. **LIMITED SUPPLY**.

Also available for this project:

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NICAM DECODER

IC set: MAS7A101, MAS7D102 and MAS7A103	99.00	P&P 8.50
Filter set: Toko QPSK filter* and 15-kHz LPF	23.50	P&P 8.50
Kit*: incl. PCB, front panel foil, mains transformer (220V) and Telet case	275.00	P&P 20.00

* please state System B/G or I

All kits INCLUDE PCB(s) (if produced by Elektor), components that fit the PCB, software, EPROM(s) and front panels if used in the particular project, UNLESS OTHERWISE STATED. Contents of kits strictly to Elektor's parts lists. Supply time of kits: approx. 4 weeks.

April 1992

Multi-purpose Z80 card	ask
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March 1992

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February 1992

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January 1992

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Fast, precise thermometer	ask
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December 1991

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Digital function generator	ask
Small projects in issue 12-91	ask

November 1991

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October 1991

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September 1991

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