quadro in practice

In response to an earlier article on quadrophony ('Quadro 1-2-3-4 ...', December 1974), we received many requests for the complete circuit of a quadrophony decoder — preferably one which is suitable for all systems. However, the problem here is that the patentees for QS and CD-4 will not consent to the ICs developed for their systems being supplied by the retail trade*. They are only to be supplied to original equipment manufacturers.
As a result, a home-made quadro decoder will have to consist of discrete components, so that it becomes far more complex than would otherwise be necessary. Nippon Columbia has now put such a design at our disposal, so that we can fulfil the wishes of many of our readers. For completeness' sake a survey is first given of the decoder principles of the four different systems: SQ, QS (RM), CD-4 and UD-4.

* It is extremely difficult to write a truly up-to-date article on the subject of quadrophony, as new information is pouring in continuously. Although this article has been up-dated as far as possible (the first draft was written in February ...), it is already slightly out-of-date: according to our latest information, the Signetics CD4-392 is now available via the retail trade. Elektor laboratories are now studying the new possibilities which this offers.

For a good understanding of quadrophony decoders in general, and of the decoder described here in particular, it is necessary to know something about the theory of the various systems. Therefore the systems SQ, QS, CD-4 and UD-4 are first dealt with in this article. For a description of the results that can be obtained with the various systems, the reader is referred to the above-mentioned article 'Quadro 1-2-3-4 ...'.

SQ

This system (by CBS) is a 4-2-4 matrix system, that is to say, that the four original channels (R_F, L_F, R_B, L_B) are combined in a matrix ('encoder') to two channels for transmission (R_T and L_T, where 'T' can be translated as 'transmission'), and then separated again in a second (decoder) matrix into four channels ($R_{\acute{F}}$, $L_{\acute{F}}$, $R_{\acute{B}}$, $L_{\acute{B}}$) for reproduction. With SQ the encoder is described by the formula:

$$R_T = R_F + 0.707\,(jR_B - L_B), \text{ and}$$
$$L_T = L_F + 0.707\,(R_B - jL_B).$$

Here the 'j' corresponds to a 90° phase shift. R_T and L_T are signals which are transmitted via, for example, a gramophone record; for reproduction, four different signals must be derived from them via the following formulae:

$$R_{\acute{F}} = R_T,$$
$$L_{\acute{F}} = L_T,$$
$$R_{\acute{B}} = 0.707\,(-jR_T + L_T),$$
$$L_{\acute{B}} = -0.707\,(jL_T + R_T).$$

These formulae define the SQ decoding matrix. For completeness' sake we can also calculate what the four reproduction channels have in common with the original four channels. For front-right we have:

$$R_{\acute{F}} = R_T = R_F + 0.707(jR_B - L_B).$$

So this signal is composed of the original front-right signal (R_F), the right-back signal attenuated 3 dB and shifted through 90°, and the left-back signal also attenuated 3 dB and shifted through 180°. In the same manner it can be seen that the signal reproduced left-front also consists of a mixture of three signals: the original left-front signal, and in addition the original left-back signal attenuated 3 dB and shifted through 270° (−j), and the original right-back signal attenuated 3 dB and in phase. Expressed in formula:

$$L_{\acute{F}} = L_T = L_F + 0.707\,(R_B - jL_B).$$

The signal reproduced at the right-back is:

$$R_{\acute{B}} = 0.707\,(-jR_T + L_T)$$
$$= R_B - 0.707\,jR_F + 0.707\,L_F,$$

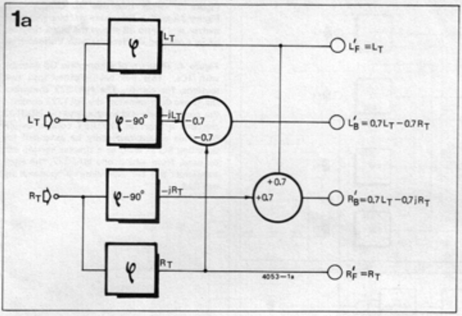
whereas left-back is reproduced as

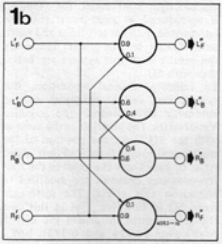
$$L_{\acute{R}} = L_R + 0.707\,jL_F - 0.707\,R_F.$$

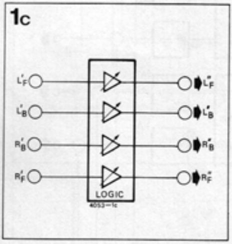
In both cases the crosstalk products are similar to those in the front channels. The channel separation can be improved by using two approaches. The simplest is the addition of two resistors, by means of which extra crosstalk, say 10%, is introduced between the front channels and 40% between the back channels. In formula:

$$R_{\acute{F}}' = 0.9\,R_{\acute{F}} + 0.1\,L_{\acute{F}},$$
$$L_{\acute{F}}' = 0.9\,L_{\acute{F}} + 0.1\,R_{\acute{F}},$$
$$R_{\acute{B}}' = 0.6\,R_{\acute{B}} + 0.4\,L_{\acute{B}},$$
$$L_{\acute{B}}' = 0.6\,L_{\acute{B}} + 0.4\,R_{\acute{B}}.$$

Other 'blend' ratios are also used. This decrease of channel separation at the front and back results in a 6 dB improvement of channel separation between 'centre-front' and 'centre-back'. The second approach is the so-called 'gain-control logic'. In its simplest form this consists of automatic level controls for the front and back channels. The operating principle is that if, for example, the front channels are louder than the back channels (in other words, the main sound sources are at front) the difference is emphasized by extra amplification of the front channels and simultaneous attenuation of the back channels. The more expensive version ('full logic' or 'wave-matching logic') can in a similar manner boost one channel at the expense of the other three. So in that case only the left-front channel, for example, is







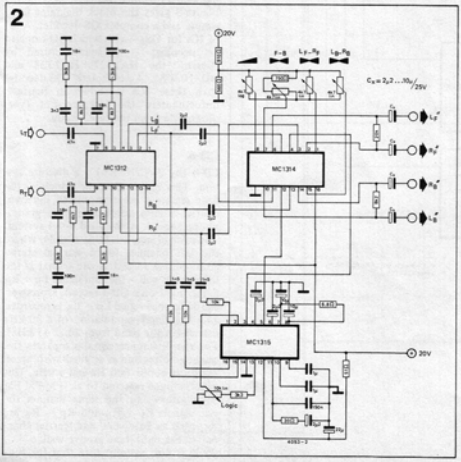


Figure 1. Block diagram for an SQ decoder. The basic decoder corresponds to figure 1A. The phase shifters are so designed that within the audio band there is an almost constant mutual phase difference of 90° (corresponding to ±j) between the two pairs of output signals. Figure 1B gives an addition (so-called blend), with which the channel separation front-back and along the diagonals can be improved at the expense of the channel separation left-right. Alternatively, 'logic' (figure 1C) can be used, for the same purpose.

Figure 2. The diagram of a complete SQ decoder with 'logic'. The MC1312 comprises the phase shifters and matrix, the MC1315 comprises the control circuits for full logic and drives the MC1314 which comprises the buffer amplifiers with gain control.

amplified whilst the other three channels are attenuated.

With suitable demonstration records this 'logic' can lead to an almost discrete reproduction of single channels. The block diagram of an SQ decoder is shown in figure 1. Motorola is one of the firms offering a complete series of ICs for an SQ decoder with full logic: the MC1312, MC1314 and MC1315. The first comprises the basic decoder (except for a number of resistors and capacitors!) whilst the other two contain variable-gain amplifiers and the full-logic control. A practical circuit with these ICs is given in figure 2. In contrast to the other ICs, these types can be obtained via the retail trade.

QS

This system (by Sansui) is also a 4-2-4 matrix system. The matrix itself is standardised in Japan as RM (regular matrix), so that on universal decoders the QS position is sometimes indicated as 'RM'.

In this case the encoder is described by the formulae:

$$R_T = 0.924(R_F - jR_B) + 0.383(L_F - jL_B), \text{ and}$$
$$L_T = 0.924(L_F + jL_B) + 0.383(R_F + jR_B).$$

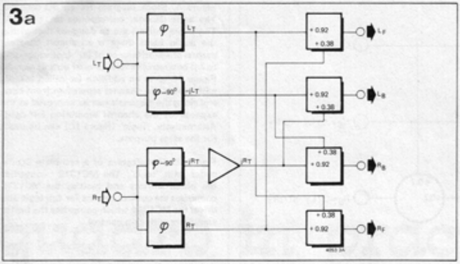
The factors 0.924 and 0.383 are cos 22½° and sin 22½°, respectively. The decoder matrix is as follows:

$$R_F' = 0.924\,R_T + 0.383\,L_T,$$
$$L_F' = 0.924\,L_T + 0.383\,R_T,$$
$$L_B' = -0.924\,jL_T + 0.383\,jR_T, \text{ and}$$
$$R_B' = 0.924\,jR_T - 0.383\,jL_T.$$

After some calculation we find that for reproduction the output signals are composed as follows:

$$R_F' = R_F + 0.707\,L_F - 0.707\,jR_B,$$
$$L_F' = L_F + 0.707\,R_F + 0.707\,jL_B,$$
$$L_B' = L_B + 0.707\,R_B - 0.707\,jL_F, \text{ and}$$
$$R_R' = R_B + 0.707\,L_B + 0.707\,jR_F.$$

In contrast to the results obtained with SQ, crosstalk is almost symmetrical: the



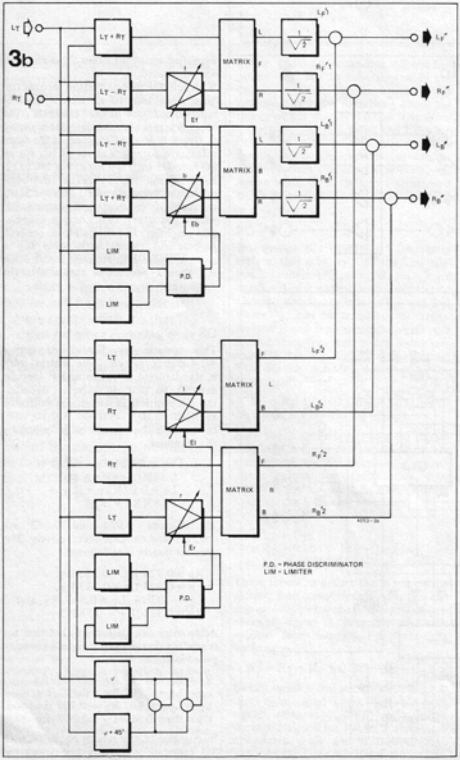


Figure 3. Block diagrams for QS decoders. Figure 3A shows the phase shifters and basic matrix, and figure 3B shows the block diagram of an extended QS decoder with Variomatrix.

Figure 4. Diagram of a complete QS decoder with ICs. This has been divided into two sections, for clarity. The HA1327 comprises the phase discriminator, the HA1328 contains the control logic and matrix, and the HD3103 contains a fourfold MOSFET control amplifier. This circuit can easily be extended for decoding SQ as well; or a simpler version can be used, from which one HA1327, the eight transistors and corresponding components are omitted.

original right-front signal, for example, is reproduced at right-front, and also (attenuated 3 dB) at left-front and right-back. Owing to this greater symmetry the results with this system are better than with SQ.

To improve channel separation, this system, too, can be equipped with an automatic level control: the so-called Variomatrix. The principle is the same as with the SQ logic: the position of the main signal source is located (left, right, front or back) and the gains in the four reproduction channels are modified to emphasise this effect. The difference from SQ is, however, that in this case the gain control influences the matrix coefficients (0.924 and 0.383); hence the name Variomatrix.

Figure 3 gives the block diagrams for a simple and a complex QS decoder. A set of ICs for QS – with which SQ can also be decoded – is manufactured by Hitachi: the HA1327, HA1328 and HD3103PA. A complete QS decoder with these ICs is shown in figure 4; unfortunately the ICs are not (yet) obtainable from the retail trade.

CD-4

CD-4 (by JVC/Nivico) is a discrete system. This means that, in principle, the four channels are transmitted independently of one another via the gramophone record: a so-called 4-4-4 system. The basic principle is fairly simple. Where the left channel for a normal stereo gramophone record is cut – that is the left groove wall – the sum signal $L_F + L_B$ is recorded on a CD-4 record. Moreover, the difference signal $L_F - L_B$ is recorded as a frequency modulation of a 30 kHz carrier (in the band from 20 ... 45 kHz). For the difference signal a separate frequency correction is applied with break points at about 800 Hz and 6 kHz. This is sometimes referred to as 'FM-PM-FM equalisation'. In the same manner the two signals $R_F + R_B$ and $R_F - R_B$ are recorded as 'baseband' and 'carrier channel' in the right hand groove wall.

So it is not entirely true that the four

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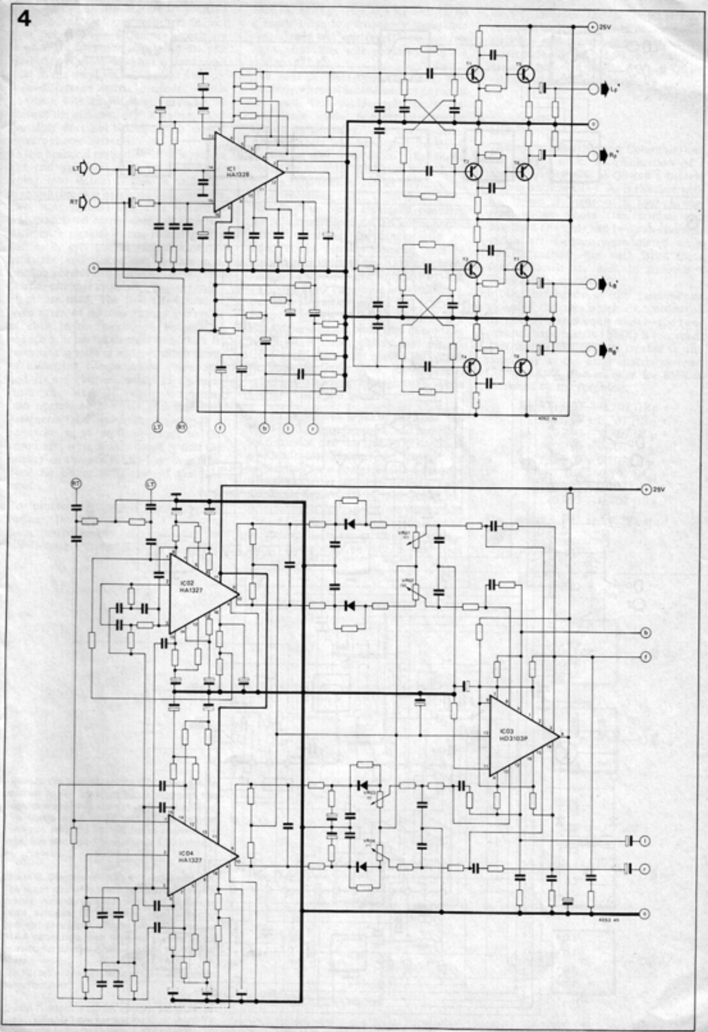

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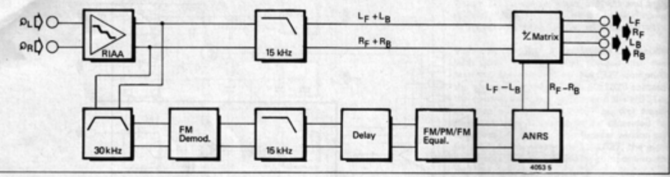
RIAA
15 kHz
L_F + L_B
R_F + R_B
½ Matrix
L_F
R_F
L_B
R_B
L_F − L_B
R_F − R_B
30 kHz
FM Demod.
15 kHz
Delay
FM/PM/FM Equal.
ANRS
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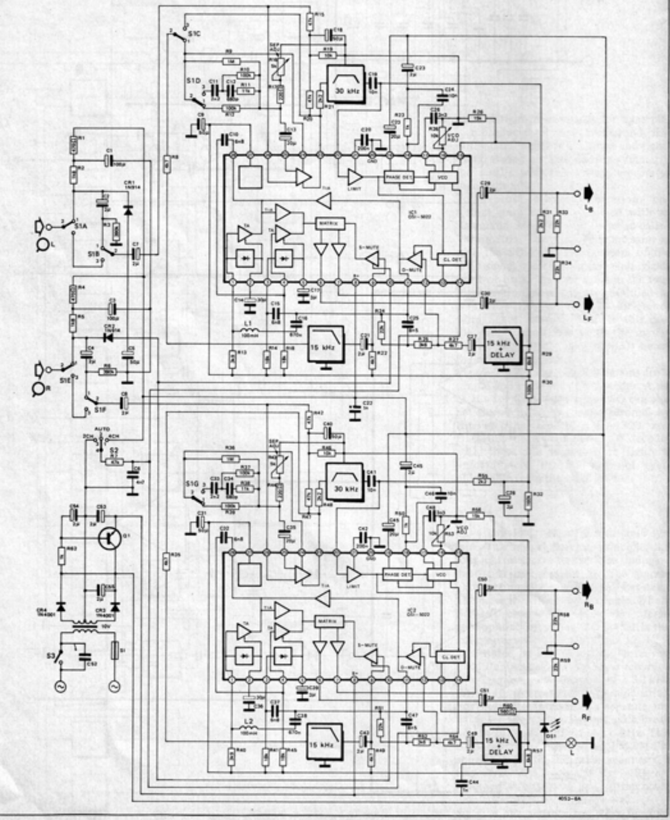
S1A
S1B
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S1D
S1E
S1F
S1G
S2
S3
AUTO
2CH
4CH
30 kHz
15 kHz
15 kHz DELAY
PHASE DET
VCO
MATRIX
CL DET
LIMIT
L1
L2
L_B
L_F
R_B
R_F
4053-6A

channels L_F, L_B, R_F, R_B are transmitted independently of one another: in fact two sum and two difference signals are used. This does not affect the discrete character of the system, but it does mean that in the playback equipment a simple 'sum-difference matrix' is needed – as is the case with an FM-stereo decoder. In spite of the inclusion of this matrix, CD-4 certainly does not belong to the 'matrix quadrophony' systems!
In the practical realisation of this system the real problems crop up ... Without going into details it can be said that transmitting two baseband channels (the two sum signals) and two wideband FM-modulated carrier channels (the two difference signals) in one record groove can easily give rise to various kinds of crosstalk and distortion; in this connection phrases like 'uptalk', 'downtalk', 'carrier-channel crosstalk', 'energy spill', et al. are used. The first CD-4 records were often of inferior quality as a result of this. In the meantime, however, a change-over has taken place to a 'Mark II' recording system in which various forms of distortion compensation (Neutrex I and II) are built in, whilst in the near future the 'Mark III' version is to come into operation. The fact that this gives designers (and recording studios?) a headache is of no importance to the consumer; what is important is that the quality of the newer CD-4 records is indeed far better than that of the first ones.

To 'protect' the carrier channels even further, the later equipment also contains the possibility of suppressing the low-frequency components of the difference signals ($L_F - L_B$ and $R_F - R_B$). As a result these low-frequency signals are reproduced as 'centre-left' or 'centre-right', but this will generally not be noticed.
The next problem concerns noise. Particularly when wideband carrier channels are used, this could easily reach unacceptable levels. A noise suppression system is desirable, and the so-called ANRS system was developed for CD-4. According to JVC/Nivico, the initial problems have by now been solved, by adding an automatic carrier level control at the recording end.
The final problem especially concerns the consumer. Since a CD-4 gramophone record contains frequencies up to 45 kHz, the pick-up element must be able to reproduce these. At the moment suitable cartridges are supplied by a.o. JVC, National Panasonic and Pickering, but they are not cheap.
Figure 5 shows the block diagram of a CD-4 demodulator. Note that in contrast to all other quadrophony systems the CD-4 system does not require 90° phase shifters. This means that the system is suitable for extensive integration, which can save costs and space.

For CD-4, various types of ICs exist: the CD-4 392, manufactured and supplied by Signetics, and the QSI 5022 supplied by Quadracast Systems Inc. The factory application for a complete CD-4 decoder with the latter IC has already been published several times. For completeness' sake this diagram is given once again (figure 6); the necessary low-pass and band-pass filters are supplied by Matsushita. It should be noted that this circuit includes the preamplifiers for magnetic cartridges, as well as changeover switches and biassing arrangements for semiconductor cartridges. These circuits are not included in the SQ and QS decoder designs shown previously.

UD-4

This system (by Nippon Columbia) can be thought of as a combination of a matrix system such as QS and a discrete system such as CD-4. As is the case with CD-4, use is made of a four-channel gramophone record (the normal two baseband channels and two sub-channels which are frequency-modulated on a 30 kHz carrier), but with UD-4 these four channels are used to transmit a four-channel matrix.
In the basebands of the gramophone record – the ones which are reproduced by normal stereo equipment – the two-channel basic matrix (BMX) is recorded. This matrix can first be treated in the same way as the other matrix systems (SQ and QS). The encoder for BMX is described by the formulae:

$$\begin{aligned} R_T = {} & (1.707 - 0.707\,j)R_F + \\ & (1.707 + 0.707\,j)R_B + \\ & (0.293 - 0.707\,j)L_F + \\ & (0.293 + 0.707\,j)L_B; \\ L_T = {} & (1.707 + 0.707\,j)L_F + \\ & (1.707 - 0.707\,j)L_B + \\ & (0.293 + 0.707\,j)R_F + \\ & (0.293 - 0.707\,j)R_B. \end{aligned}$$

The factors 1.707 and 0.293 are $1 + \frac{1}{\sqrt{2}}$ and $1 - \frac{1}{\sqrt{2}}$ respectively.

Figure 5. Block diagram of a CD-4 decoder. In contrast to the previous block diagrams this diagram also shows the dynamic preamplifier. This forms an essential part of the decoder since not only the normal stereo channels are used, but also two FM-modulated 30 kHz carrier channels.

Figure 6. Diagram of a CD-4 decoder with ICs. The input circuit is more complicated than is strictly necessary, because it has also been made suitable for the new semiconductor pick-up cartridges. These do not require RIAA correction, but they do require a bias current; furthermore, the outputs are in antiphase.
The ICs are only supplied to original equipment manufacturers.

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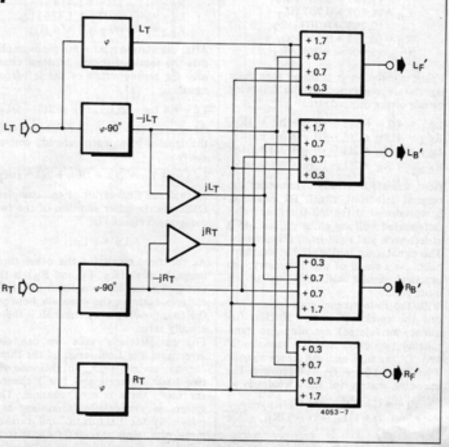

Figure 7. Block diagram of a simple UD-4 decoder, suitable only for BMX.

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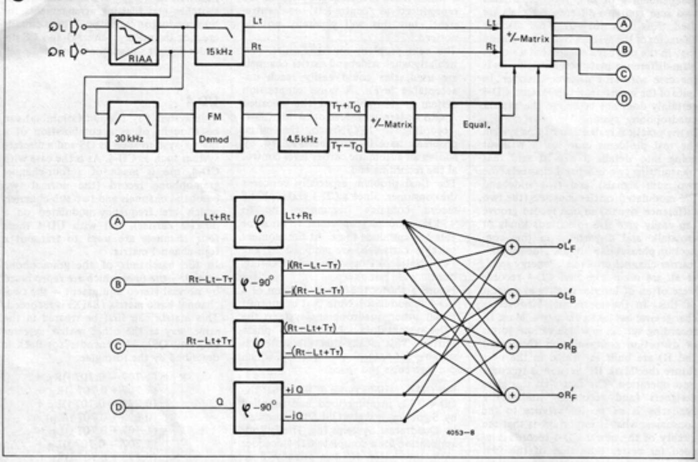

The decoder matrix for BMX is as follows:

$$L_{F'} = (1.707 - 0.707\,j)L_T + (0.293 + 0.707\,j)R_T,$$
$$R_{F'} = (0.293 - 0.707\,j)L_T + (1.707 + 0.707\,j)R_T,$$
$$L_{B'} = (1.707 + 0.707\,j)L_T + (0.293 - 0.707\,j)R_T,$$
$$R_{B'} = (0.293 + 0.707\,j)L_T + (1.707 - 0.707\,j)R_T.$$

From calculation it is found that these signals are composed of the following blends of the original signals:

$$L_{F',1} = 4\,L_F + 2(1-j)L_B + 2(1+j)R_F,$$
$$R_{F',1} = 4\,R_F + 2(1-j)L_F + 2(1+j)R_B,$$
$$L_{B',1} = 4\,R_B + 2(1-j)R_B + 2(1+j)L_F,$$
$$R_{B',1} = 4\,R_B + 2(1-j)R_F + 2(1+j)L_B.$$

Here, crosstalk is fully symmetric: the original left-front signal, for example, is reproduced at the left-front, and also (attenuated 3 dB and phase shifted ±45°) at left-back and right-front respectively. This signal is not reproduced at the right-back. As a result of this symmetry, the practical results with this matrix are quite good.
With the four-channel gramophone record the carrier channels (30 kHz, frequency-modulated) are used for transmitting two extra matrix channels, 'T' and 'Q', as sum and difference signals: $T_T + T_Q$ and $T_T - T_Q$, respectively. The encoding matrix for these channels is:

$$T_T = (j-1)L_F + (j+1)R_F - (j+1)L_B - (j-1)R_B,$$

and

$$T_Q = 1.414\,jL_F - 1.414\,jR_F - 1.414\,jL_B + 1.414\,jR_B.$$

The corresponding decoder matrix for these additional channels is:

$$L_{F',2} = -(1+j)T_T - 1.414\,jT_Q,$$
$$R_{F',2} = (1-j)T_T + 1.414\,jT_Q,$$
$$L_{B',2} = -(1-j)T_T + 1.414\,jT_Q,$$
$$R_{B',2} = (1+j)T_T - 1.414\,jT_Q.$$

After calculation it is found, for example, that the result of these additional channels for reproduction of the left-front signals is:

$$L_{F',2} = 4\,L_F - 2(1+j)R_F - 2(1-j)L_B.$$

Of course, this signal is combined with the signal of basic matrix already derived above:

$$L_{F',1} = 4\,L_F + 2(1-j)L_B + 2(1+j)R_F,$$

so that the final result of the complete QMX matrix (after addition of the two component signals) is:

$$L_{F'} = 8\,L_F.$$

As the final result for the other three original signals (R_F, L_B and R_B) is the same, this can be considered as a fully discrete system: the crosstalk between the four quadrophony channels is theoretically zero!
For completeness' sake we can also investigate the final result of the three-channel matrix TMX. In this case the two basic channels and the T channel are used; there is no Q channel. This system is particularly interesting as a possibility for 'FM-quadro' and, furthermore, the latest theories show that an

Figure 8. Block diagram of an extensive UD-4 decoder, suitable for the four-channel matrix (QMX). The first section can be compared with the CD-4 decoder, and also contains the dynamic preamplifiers; the second section comprises the phase shifters and matrix. To economise on phase shifters, the signals L, R, T and Q are not routed through separate phase shifters (in that case eight of these would be needed); instead the blend signals L + R, R – L – T, R – L + T and Q are formed first, so that four phase shifters are sufficient.

Figure 9. No ICs are available for UD-4 as yet. However, starting from imaginary ICs (derived from existing types, designed for other systems) this diagram has been set up. It gives an impression of what a UD-4 decoder with ICs could look like. Even more thorough integration is conceivable: by integrating both input preamplifiers in IC1, sufficient room would be made available in IC2 to accommodate IC3.

optimal three-channel system is in fact inherently better than a four-channel system (used non-redundantly) for giving four-loudspeaker quadrophonic reproduction! The output signal of the left-front channel is then composed as follows:

$$L_F' = 6\,L_F + 2\,L_B + 2\,R_F - 2\,R_B.$$

So channel separation is far from complete (about 12 dB), but owing to the high degree of symmetry and the lack of phase shift in the final output, the practical results are particularly good. As regards the extra carrier channels, the practical problems encountered with UD-4 run parallel with those of CD-4. Here, however, a different solution has been found for the noise (and crosstalk) problem: the bandwidth of the T_T and T_Q channels is limited to about 6 kHz, so that above this frequency the system is actually reduced to BMX. So localisation at high frequencies becomes less accurate, but this is hardly noticeable because the human ear itself is less directional at these frequencies.

To reduce low-frequency interference between the carrier channels (or to make matters even more complex?) the T_Q signal is attenuated with respect to T_T for lower frequencies. This means that the difference between the two carriers

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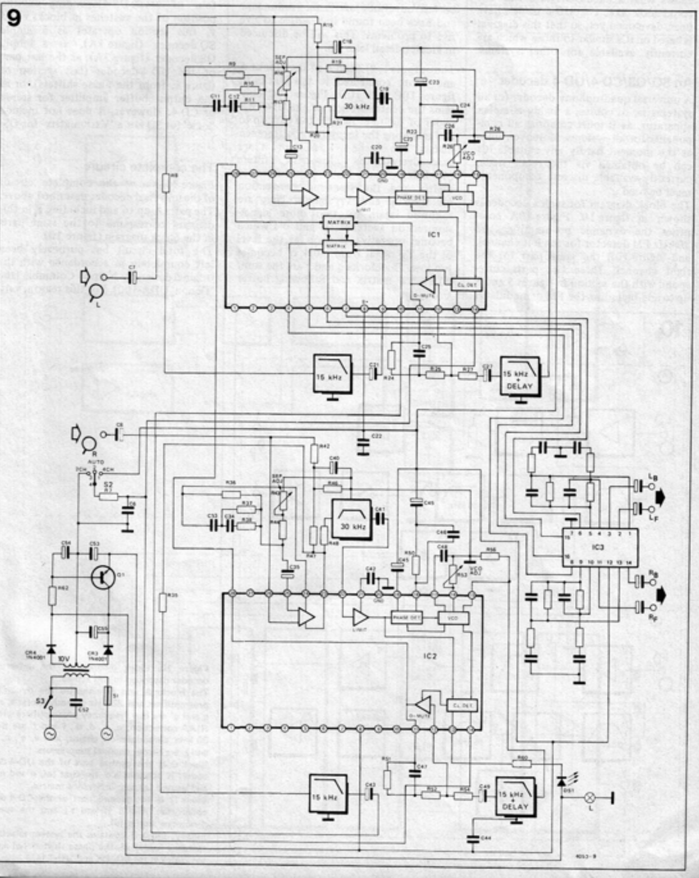

($T_T + T_Q$ and $T_T - T_Q$, respectively) becomes less at lower frequencies, so that there is less chance of mixture products occurring. During playback the T_T and T_Q signals must therefore undergo different frequency corrections to restore them to equal levels. A block diagram for a simple UD-4 decoder (BMX only) is shown in figure 7, whilst figure 8 is the block diagram of a complete four-channel decoder for QMX. Figure 9 shows what a circuit diagram with ICs could look like. For UD-4 no ICs have been developed yet, so that this diagram is based on ICs similar to those which are currently available for other systems.

An SQ/QS/CD-4/UD-4 decoder

A universal quadrophony decoder for all systems is, of course, a highly complex apparatus, as it must comprise all components for all systems. Moreover, since at the moment hardly any suitable ICs can be obtained via the retail trade, currently-available discrete components must be used ...

The block diagram for such a decoder is shown in figure 10. Figure 10A comprises the dynamic pre-amplifier and 30 kHz FM detector for the left channel, and figure 10B the same part for the right channel. These two parts correspond with the section in figures 5 and 8 up to and including the FM demodulator in the carrier channels. In this case, however, the RIAA frequency correction is divided over two blocks, a and g (a′ and g′, respectively). Furthermore, a number of suppression switches for the carrier channels have been added (e, i, e′ and i′) which become operative as soon as the level of the left carrier channel becomes too low. They are driven via block h; this carrier signal detector also drives a pilot lamp. However, it should be noted that the switches e and e′ can be left out: they are in fact redundant and have been found in practice to give rise to problems! This will be discussed in greater detail later on.

The outputs 1, 2, 3 and 4 of the blocks A and B are connected to the blocks in figures 10C and 10D. Figure 10C contains the centre section of the UD-4 decoder of figure 8, namely the part up to and including the second sum/difference matrix. The blocks l, l′, m, n and n′ are successively the low-pass filters (4.5 kHz), the first sum/difference matrix and the frequency correction for the T_T and T_Q signals. They are followed again by two more signal-suppression switches (o and o′) which become operative as soon as the level of the T_T signal from block o′ becomes too low. The blocks q and r are the sum/difference matrix and following buffer amplifiers.

Figure 10D comprises the second part of the CD-4 decoder. (figure 5). The blocks s and s′ contain the 15 kHz low-pass filters, delay circuits and FM/PM/FM frequency correction networks for the carrier channels.

Finally figure 10E contains the system selection switches, phase shifters and matrices for the four systems or stereo. So it is connected to the baseband channel outputs of blocks A and B, and also to the four outputs of block C and the four of block D. Depending on the position of the switches in blocks v and y, this section operates as a simple SQ decoder (figure 1A), as a simple QS decoder (figure 3A), as the last part of the UD-4 decoder (last section of figure 8, from the phase shifters), or as the output buffer amplifier for stereo or CD-4. However, it does not include 'logic' for SQ nor a 'Variomatrix' for QS.

The complete circuit

Figure 11 shows the complete circuit of the universal decoder described above. The parts A up to and including E in this diagram correspond to the same parts in the block diagram (figure 10).

The total circuit has purposely been left completely in accordance with the original design by Nippon Columbia (the 'Denon' UDA-100). For this reason, vari-

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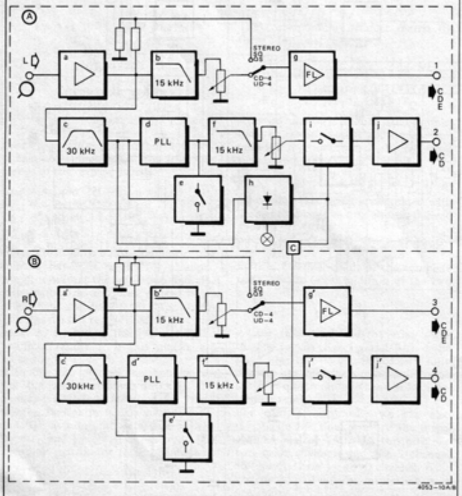

Figure 10. Block diagram of the universal decoder described in this article.
The blocks A and B comprise the dynamic preamplifiers and 30 kHz demodulators: a, a′, g and g′ are low frequency preamplifiers with RIAA correction; c, c′, d, d′, f and f′ are the 30 kHz demodulation chains, and e, e′, h, i and i′ are carrier channel suppressors.
Block C is the central part of the UD-4 decoder; it contains a.o. the first (m, n and n′) and second (q) sum-difference matrix.
Block D is the second part of the CD-4 decoder: the ANRS (t and t′) and the sum-difference matrix (u).
Finally, block E contains the system selector switches (v and y), the phase shifters (w) and the matrices for SQ, QS and UD-4 (x).

ous resistors from the E-24 series occur in it. For the resistors and capacitors it is recommended that 5% types should be used. The choice of alternative semiconductors is generally not critical. The transistors 2SC1000BL can be replaced by TUNs, the 2SC1000BL/CR, 2SC1000GR and 2SC1213 by BC 107B, the 2SC732GR by BC 109C and the 2SA493GR by BC 179B or C. The diodes can be replaced by DUSs, whilst for the FETs almost any type of N-channel FET is suitable (E 300, BF 245, etc.).

The ICs TA7122AP are low-noise differential amplifiers; when replacing these types by alternatives, the connections and some of the resistor values will, of course, have to be changed. For IC3 and IC6 two possible types are indicated: the NE565 (Signetics) and the HA1173, with slightly deviating resistance values. Various adjustment points are also indicated in the circuit (figure 11A/B). The trimmers near IC3 and IC6 (VR2 and VR5) serve for adjusting the centre frequency of the PLL (30 kHz). VR3 and VR6 (near FET2 and FET4) set the level of the demodulated carrier channel separation for CD-4 and UD-4. The potentiometers ('B') in the output circuit of the first low-pass filters serve to adjust the level of the baseband channels; in principle, their function is the same as that of the adjustment resistors VR3 and VR6. Trimmer VR7 (near TR16/17 adjusts the trigger level of the suppression switches (blocks i and i'). In figure 10C only one trimmer occurs: VR5 (near TR26/27). This serves to ad-

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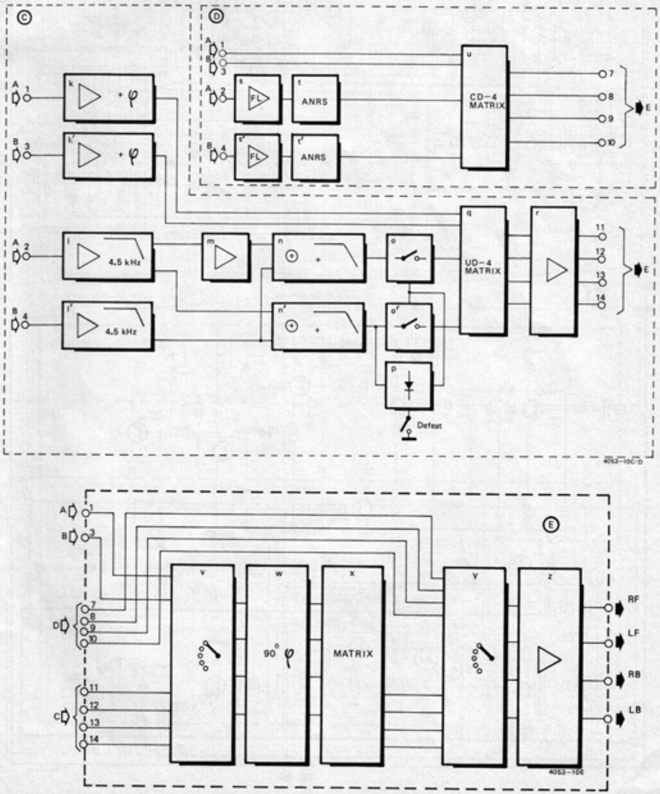

just the trigger level at which the suppression switches for the T and Q signals (blocks o and o') become operative. Finally, in block D there are four trimmers for adjusting the ANRS. VR1 and VR2 serve for zero adjustment: they set the control signal level at which the ANRS begins to function. VR3 and VR4 adjust the ANRS control range; they also influence the zero setting mentioned above. Furthermore, it should be noted that the trimmers in the carrier channels (VR3 and VR6 in blocks A and B) also influence this zero setting.

Final notes

As will be clear from what has been said above, the complete apparatus is not required for every system. For SQ and/or QS block E is sufficient; at the most an SQ logic with MC1314 and MC1315 (figure 2) or extra 'blend' resistors could

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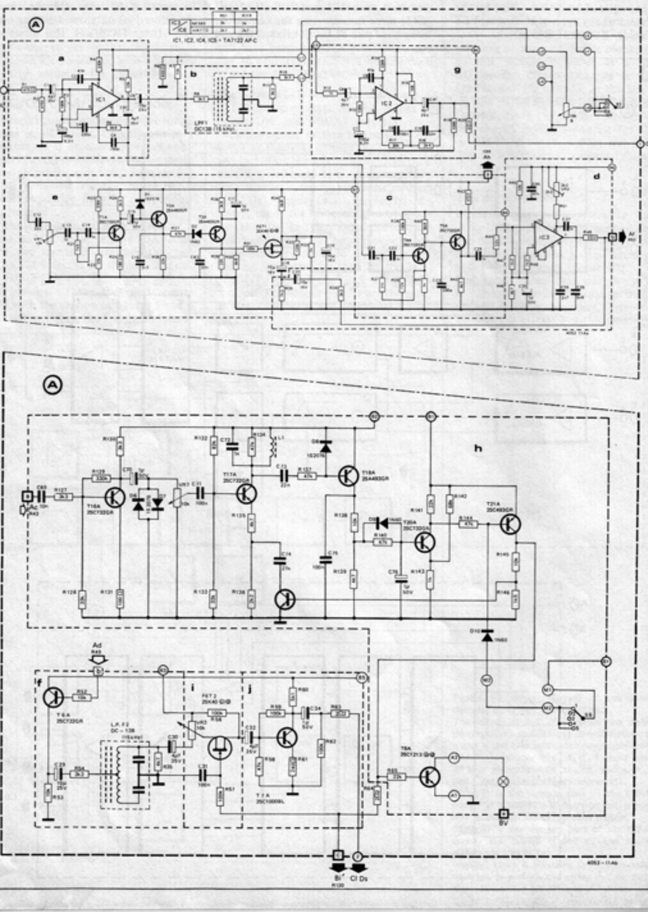

Figure 11. Complete diagram of the universal decoder. The blocks A up to and including E and the blocks a up to and including z correspond to the indications in the block diagram (figure 10). The blocks e and e′ (in A and B) may be left out (see text). Replacement types for the semiconductors are indicated in the text.

be added and also a Variomatrix for QS. For the 'blend' resistors (see figure 1B) one can experiment with a resistor of about 68 k between the outputs L_F and R_F and a resistor of about 10 k between the outputs L_B and R_B.

CD-4 requires the blocks A, B and D, in combination with a simple CD-4/stereo switch and output amplifiers as in block E. For UD-4 only block D is redundant; if only UD-4 is required, the system selection switches can be simplified and various matrix resistors can be omitted from block E.

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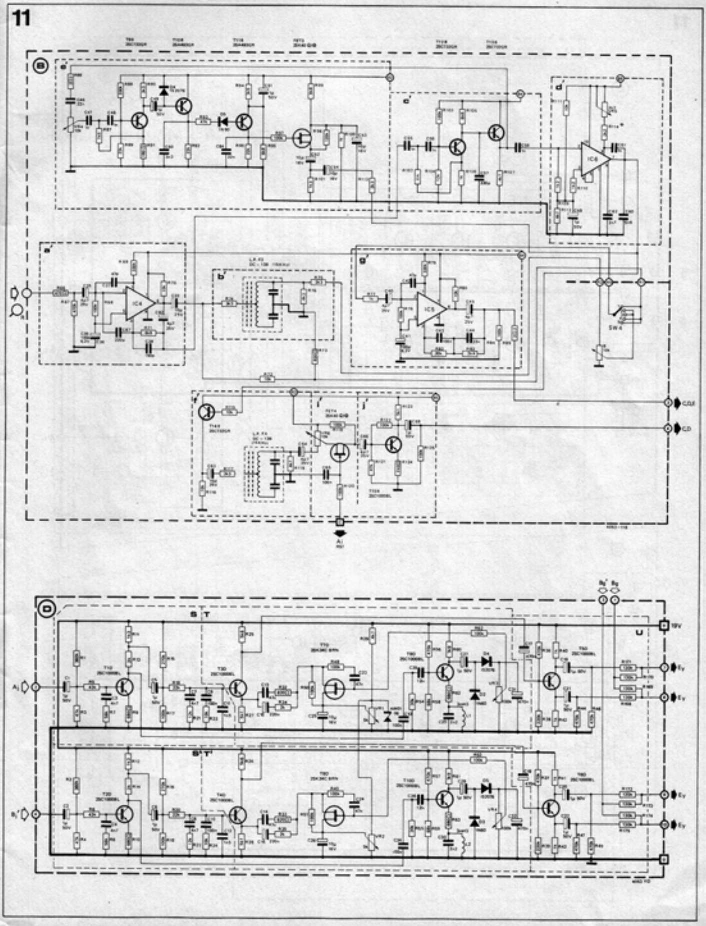

Finally it should be noted that two of the four signal suppression switches in blocks A and B can be left out. This concerns the part from C12 up to and including C18 (block e) and the part from R86 up to C52 (block e'). The resistor combinations R35/36 and R101/113, respectively, can then each be replaced by one 4k7 resistor.

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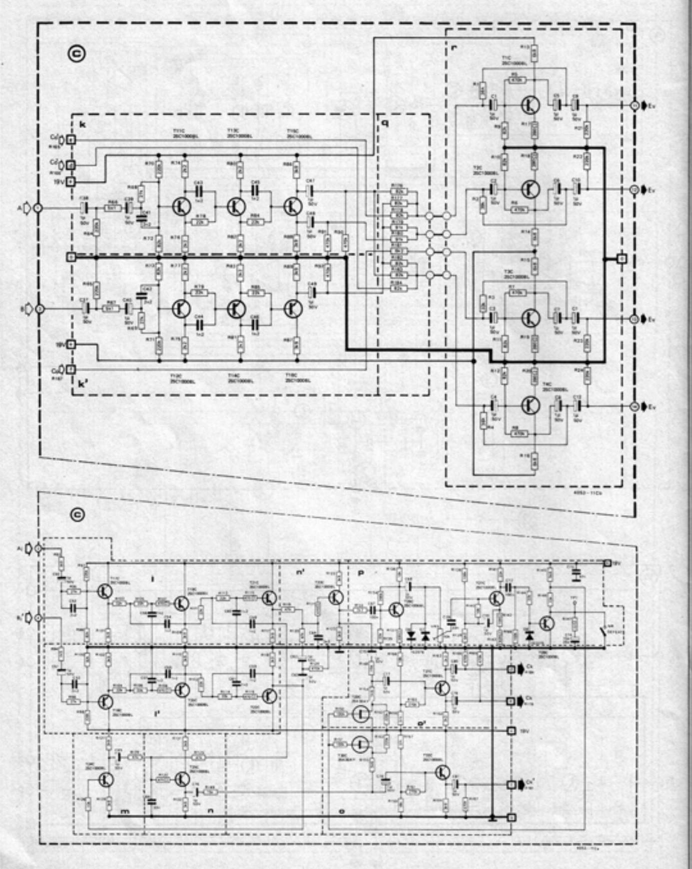

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