Surround-sound Circuits

Build your own matrix circuits using i.cs

by Geoffrey Shorter

Whichever proposal is adopted for all-round sound recording and reproduction, if indeed any one system is, it is a fact that in the U.K. the SQ system is the one for which most records are presently available. And there are many people who are anxious to try them out, some of whom, with limited resources, may not wish to buy commercial SQ decoders for fear that the SQ system may not be standardized and with the consequence that a different decoder would be required.

Two circuits are given to enable constructors to build SQ decoders, one using all discrete components and the other including integrated circuits. If you wish to try out a simple matrix circuit for getting a surround-sound effect from stereo records a simple set-up is possible using a Toshiba integrated circuit. No doubt further i.cs will become available for other systems. One, made by Texas Instruments for Electro-Voice, is compatible with SQ and claimed to be compatible with discs encoded to other systems, but is still not made available outside the U.S.A.

The basis of the SQ system has been covered before in these pages but a resumé is not out of place. Sounds from a pair-wise mixed four-channel master tape are coded into the left and right channels of a stereo disc according to

$$L = L_F - 0.707jL_B + 0.707R_B$$
$$R = R_F - 0.707L_B + 0.707jR_B$$

where j indicates a phase difference of 90° between channels. These two signals are basically the inputs to the front speakers in playback and the two rear signals are derived from $j0.707L - 0.707R$ for the left back speaker and $0.707L - j0.707R$ for the right back speaker.

These equations give rise to unique crosstalk properties (shown on page 56 February 1972 issue of *W.W.* for the four corners), with the feature of little or no crosstalk between the two front "channels" but with the particular penalty of infinite crosstalk for centre front and centre back positions with a "straight" SQ decoder. This makes localization of a centre front sound imprecise. (The simple diagrams of page 56 do not convey how accurately sounds are localized at the corners or other points around the compass.)

The essence of an SQ decoder is shown in the block diagram of Fig. 1. As the j operator shown in the above equations indicates a relative phase difference of 90°, on playback the coded left and right channels are first passed through networks in which phase is an approximately linear function of frequency over most of the audio band. These same signals also pass through similar networks which give a linear phase-frequency response, but shifted in phase by 90°.

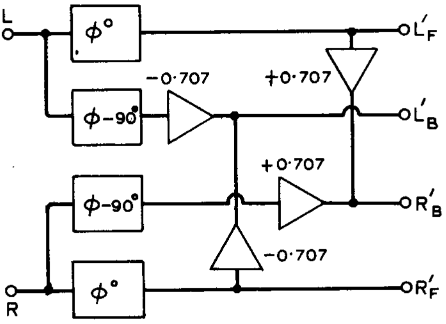

Fig.1. Basic decoder scheme for use with SQ records.

With such a decoder a sound intended to appear at centre front produces equal outputs from all four speakers. And although the rear sounds are in antiphase, they are going to interfere with centre front localization. To alleviate this situation, a certain amount of blending is arranged in some SQ decoders between the two front outputs and the two rear outputs. This has the effect of cancelling some of the antiphase components of the signals, thus reducing the outputs from the rear speakers in the case of a centre front sound. The most common amounts of blend are 10% between front outputs and 40% between rear outputs. In such a decoder the gain of the back channels is reduced by 1dB, giving a total front to back crosstalk of 7dB. Front "channel" crosstalk is increased to 20dB and back channel crosstalk increased to 8dB.

A circuit of a 10-40 blend decoder is shown in Fig. 2, which follows the scheme of Fig. 1. The 90° phase difference provided by these networks is accurate to within ±10° from 100Hz to 10kHz. The 68 and 47kΩ resistors provide the relative gain of 0.7 between front and back signal paths, and two transistors in the second stages of the phase difference networks provide inversion. (This circuit is used in many commercial SQ decoder units. Better phase difference networks are provided in some decoders, like the Lasky's Audiotronics decoder which gives a deviation of ±10% over 20Hz to 18kHz.)

In constructing the circuit of Fig. 2,

resistor values should be $\frac{1}{4}$-watt, 5%-tolerance types, and the eight capacitors in the phase-shift networks should be 10% tolerance. Recommended transistors are 2N3393, except for the output transistors which should be 2N3390. (Both types are made by G-E and Siemens. Motorola have similar devices: MPS 3393 and MPS 6521 respectively.) Input impedance is 20kΩ, output impedance 1.8kΩ and nominal input level 500mV r.m.s., the circuit having unity gain. To convert to a high input impedance the upper and lower bias resistors can be changed to 3 and 1.8MΩ respectively, using 2N5308 (G-E) input transistors.

Integrated circuits are now available from Motorola for this circuit at £1.65. Components need to be added, as indicated in Fig 3, and these should be within 5% tolerance to give a $\pm$8.5% deviation from the 90° norm between 100Hz and 10kHz. With a 20V supply rail (maximum 30V) consumption is 16mA. For a nominal input of 500mV distortion is 0.1%, clipping occurring at 2V. Input impedance is 3MΩ. The blend resistors, shown in broken lines, should be 47kΩ between the two front outputs and 7.5kΩ between the two rear outputs for the 10-40 blend.

Another way of reducing unwanted outputs from speakers is the gain control circuit given on page 597 of the December 1972 issue of *W.W.* Here a discrete component circuit was shown that provides automatic blend and consequent cancellation of antiphase components in the rear signals when a source appears at front centre. When $L+R>L-R$ some cancellation will occur and this also applies in the front "channels" when $L+R<L-R$. Whether this additional complexity is justified depends largely on the programme material. It is very effective for single sources, but multiple sources will defeat the circuit, suppress secondary sources, or cause odd time-varying effects.

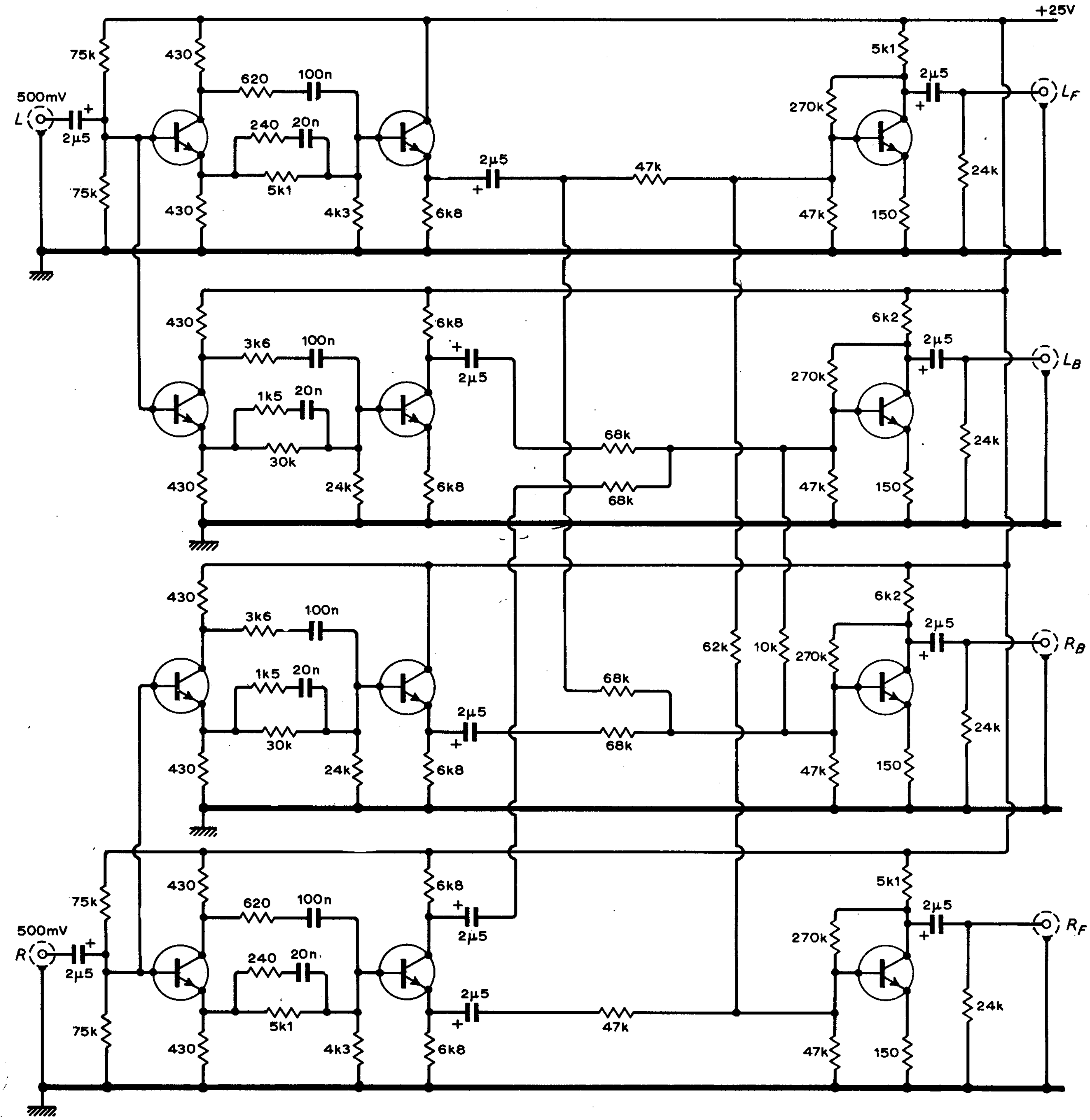

Fig.2. Circuit of SQ decoder used on some inexpensive decoders in which front outputs are blended by 10% and back by 40%.

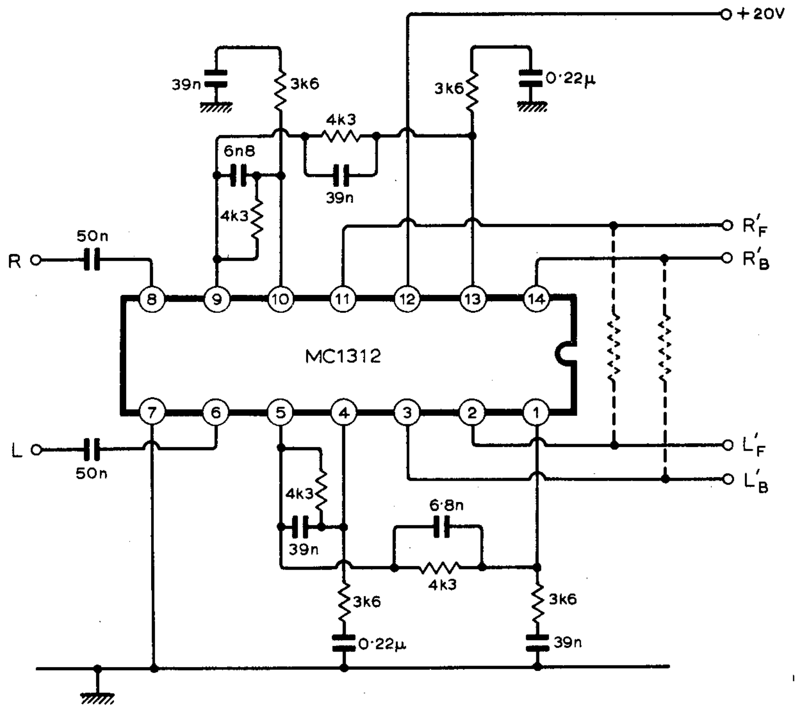

Fig.3. Integrated circuit for SQ decoder. Add resistors of 47kΩ for front pair and 7.5kΩ for rear pair of outputs for '10-40' blend.

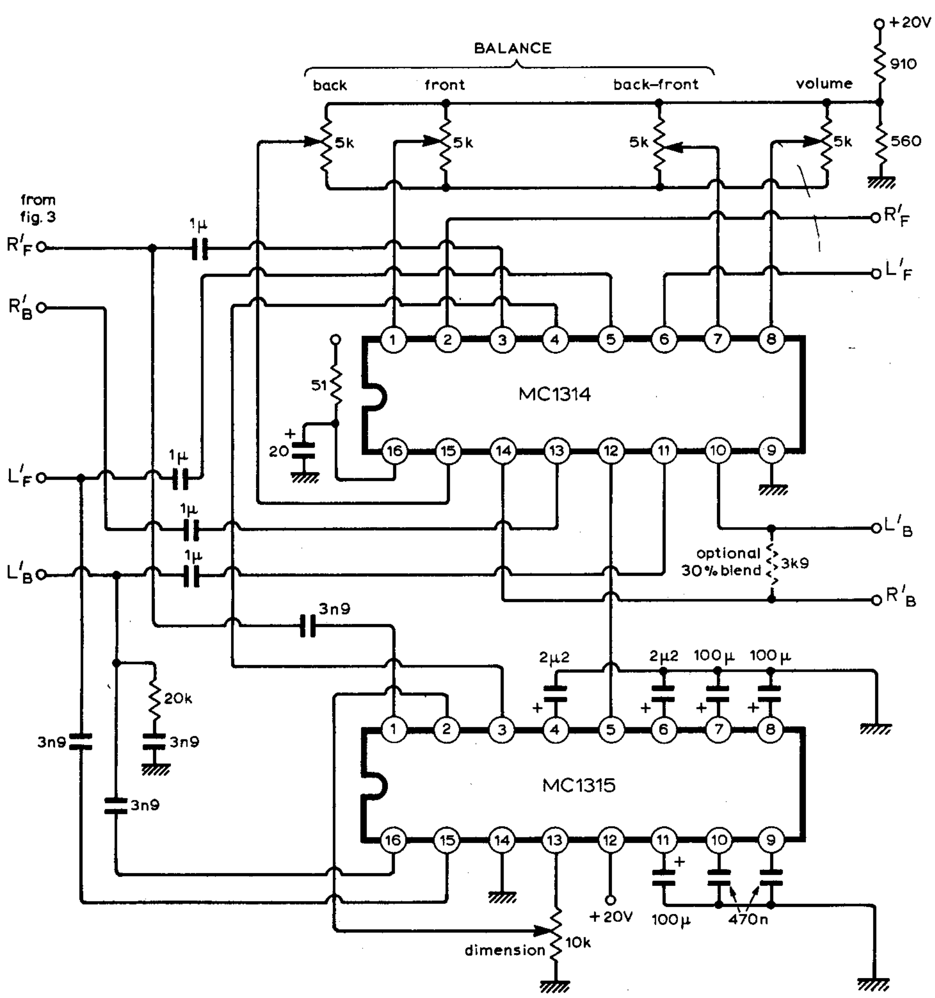

Fig.4. Circuit using i.cs to reduce crosstalk for simple sound sources.

If you wish to try it out chips will be shortly available from Motorola for this and a circuit is shown in Fig. 4, which feeds directly from Fig. 3 with the omission of the two blend resistors. The MC1314 includes the voltage-controlled amplifiers and the MC1315 provides the control voltages. As well as improving centre sounds at front the MC1315 includes circuitry to detect and attenuate unwanted outputs for corner signals.

Components in this circuit should be 5% tolerance in the phase shift networks (around pins 1, 4, 5, 9, 10, 13 on MC1312) and 10% otherwise, excepting electrolytic capacitors. The 5kΩ volume control should have a semi-log law, and the balance controls, which give a 12dB constant-power variation, should be linear. Front and rear balance controls can be "ganged" by connecting pins 1 and 15 on MC1314 and omitting one potentiometer.

If you want to omit volume and balance controls, connect pin 8 of MC1314 to a potential divider giving +6V, and leave pins 1, 7 and 15 open-circuit.

The automatic action can be varied with the linear 10k Ω "dimension" control, which CBS recommend setting at 50%, giving a front-to-back crosstalk of 15dB typically. Signal handling capability of the circuit is reduced at maximum setting unless V_{CC}=30V on the MC1314.

Surround-sound from stereo records

The other readily available i.c. is the Toshiba TA7117P. Phase difference circuits are not a part of this i.c., so the chip contains merely differential amplifiers, matrix circuit and output amplifiers, as indicated in Fig. 5. This chip is fine for getting surround-sound from ordinary stereo records. The two inputs are added and subtracted in varying proportions depending on choice of external resistors. The two added signals, $L+aR$ and $R+bL$, feed two front amplifiers and speakers and the two subtracted signals, $L-CR$ and $R-dL$, feed the two rear amplifiers

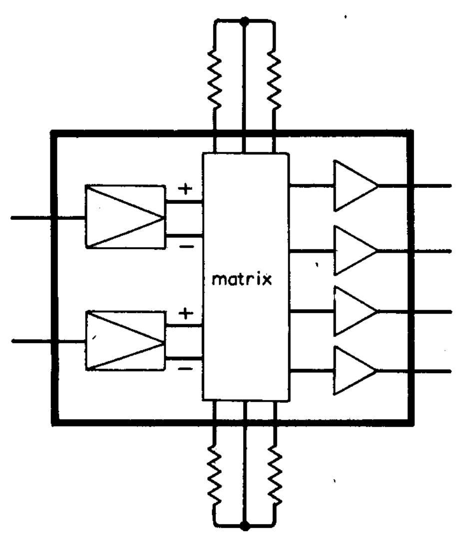

Fig.5. Scheme of Toshiba matrix i.c.

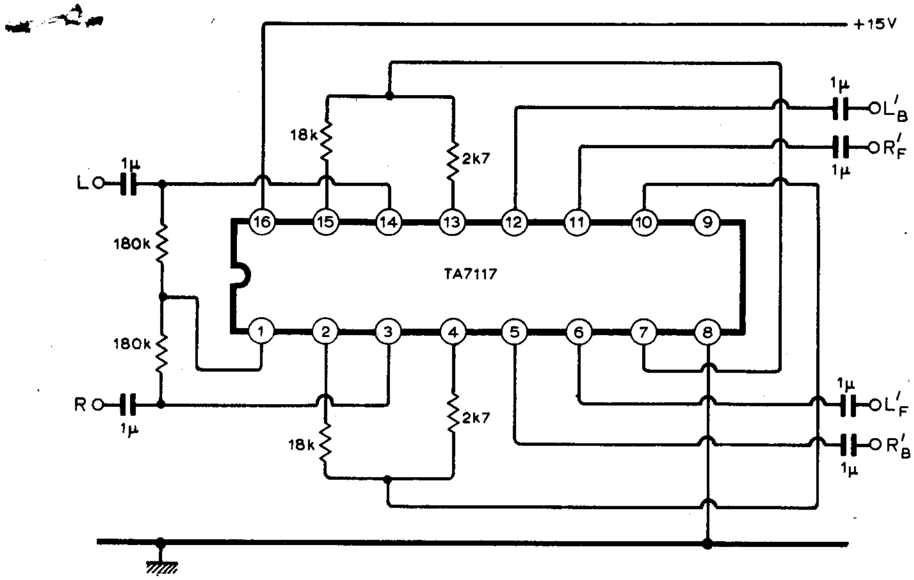

Fig.6. Circuit for getting surround-sound from ordinary records and certain coded records.

and speakers. With such an arrangement the amount of front and rear crosstalk can be experimentally varied, keeping $a=b$ and $c=d$. With $a=b=0$ and $c=d=1$, this gives the equivalent of the simple speaker matrix (obtained with only two power amplifiers by connecting the two rear speakers in antiphase and across the "live" terminals of the two amplifiers, used frequently to enhance the ambience of stereo programmes).

This is not entirely satisfactory, one effect being an increase in apparent width of the stereophonic field. Considering a left-only signal, for instance, sound will emerge from the left front, the left back and right back (in antiphase) speakers, tending to pull the image round anticlockwise from left front. This is counteracted by blending the two front outputs (and in the interests of symmetry the back two). A good starting point is by choosing $a=b=c=d=0.414$, experimenting by varying $a=b$ and $c=d$. The circuit of Fig. 6 has resistor values chosen according to this value. The 18kΩ resistors can be altered to a value of 7.3 divided by the crosstalk fraction required for the front speakers and 7.3 multiplied by the crosstalk fraction required for the rear speakers.

The circuit might give acceptable results for certain coded recordings like the early American Dynaco and Electro-Voice-encoded discs and some Japanese records. Coded QS/RM/Pye records should give acceptable results, but with all records there will be no precise back images and any sounds intended to come from the back speakers will be shifted round toward the nearest front speaker.

With the Toshiba i.c. current consumption is typically 16mA at 15V or 10mA at 8V. Input impedance is 3MΩ. With an input of 100mV r.m.s. harmonic distortion is 0.1% (rising to 0.3% for 300mV and 1% at IV). Price is £1.67* from Eric Electronics Ltd, South Denes, Great Yarmouth, Norfolk.

The Toshiba i.c. should give better results for the Pye QS/RM records than an SQ decoder. An SQ record played through the Toshiba i.c. would not reproduce intended sound directions from the rear speakers.

Quadraphonic two-channel records available in the U.K. total about 100 with around 70 SQ discs from CBS, 15 from EMI and 12 QS/RM discs from Pye. (Total for U.S.A. and Japan is at least 500.)

*Prior to any revaluation of the Yen.

Conferences and Exhibitions

Further details are obtainable from the addresses in parentheses

LONDON

Feb. 26-Mar. 2 Bloomsbury Centre
Seminex
(Evan Steadman and Partners, 4 Lyewood Common, Withyham, Hartfield, Sussex)

Mar. 13-15 Savoy Place
Satellite Systems for Mobile Communications and Surveillance
(I.E.E., Savoy Place, London WC2R 0BL)

Mar. 13-15 Bloomsbury Centre Hotel
Sound 73
(Assoc. of Public Address Engineers, 6 Conduit St, London W1R 9TG)

Mar. 22 & 23 Royal Garden Hotel
Man Made Memories
(Mrs. Rosemary Willson, Mercury House, Waterloo Road, London SE1)

Mar. 27-29 Imperial College
Ultrasonics International
(*Ultrasonics*, 32 High Street, Guildford, Surrey)

Mar. 28-Apr. 1 Excelsior Hotel
Sonex Audio Exhibition
(Federation of British Audio, 31 Soho Sq., London W1V 5DG)

CARDIFF

Mar. 26-30 Sophia Gardens
Aimex 1973 (industrial measurement and control)
(Exhibitions Wales & West Ltd, Holly House, Rhiwderin, Nr. Newport, Mon.)

OVERSEAS

Mar. 6-10 Basle
Medical Electronics and Bio-engineering
(Sekretariat MEDEX 73, CH-4021 Basel)

Mar. 6-10 Basle
INEL 73 — Industrial Electronics
(Sekretariat INEL 73, CH-4021 Basel)

Mar. 19-23 San Francisco