M8793: A SINGLE-CHIP STEREO DECODER AND AUDIO PROCESSOR FOR TV

ABSTRACT

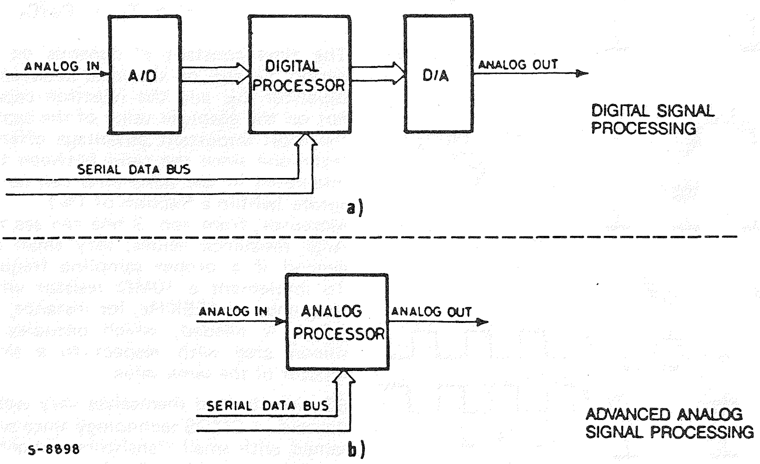
Advanced analog design techniques using switched-capacitor filters allow full integration of audio signal processing. Developing a software-controlled stereo decoder and audio processor IC using analog signal processing techniques, has lead to easy design of compact dual-channel TV-sound receiving systems where analog-processing parameters are controlled by an external microprocessor.

INTRODUCTION

In recent years, a strong interest has arisen for high-quality TV sound, and many countries have developed systems for multi-channel sound TV transmission [1 to 5].
To take full advantage of such a kind of sound transmission, more complex sound signal processing is needed in the receiver end with respect to the standard monophonic sound processing. This processing, of course, should be performed with an efficient and economic system, e.g. by means of a single-chip integrated circuit.

In developing an IC for signal processing, the designer has to make two basic choices, regarding design techniques (analog or digital processing) and wafer fabrication technology.
Recently, digital signal processing (DSP) techniques have been proposed in consumer field [6, 7]. As a matter of fact, digital techniques are very attractive, mainly because of full software programmability and digital controllability of the resulting system.
However, in present TV broadcasting systems, transmitted audio signals are analog, so solutions based on DSP techniques are rather expensive since they require analog-to-digital (A/D) and digital-to-analog (D/A) converters, plus the logic needed for the processing itself (fig. 1a).
In TV-audio application field, solutions based on advanced analog design techniques (fig. 1b) allow drastic reduction in silicon area and in component count, which results in reduction of the cost of the system and improvement of its reliability.
Therefore, a good choice is the use of analog processing techniques for the signal path and the use of a central microprocessor for the digital control of the analog-processing parameters.

Fig. 1 - Signal path in digital (a) and analog (b) signal processing



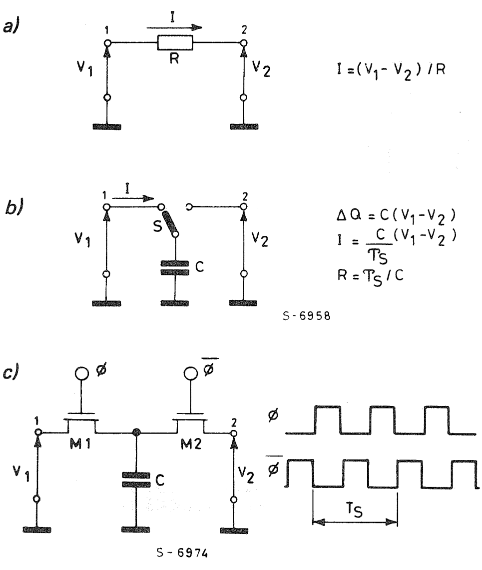
As regards technology, silicon-gate CMOS gives a good integration level and, more importantly, it provides ease in the design of analog circuits such as switched-capacitor (SC) filters, which are basic building blocks for advanced analog processing of audio signals.
CMOS also allows wide dynamic range for the processed signal and is particularly suited to implement analog switches with very high on/off-resistance ratio.
Furthermore, this technology ensures low power consumption, thus improving device reliability and reducing power-supply requirements.

In this paper, we present a single-chip stereo decoder and audio processor IC (M8793) which has been developed following the above criteria, for dual-channel TV-sound transmitted with the German two-carrier system. Target specification of this device were complete analog processing of the demodulated audio signals, where all the functions are controlled by an external microprocessor through a serial bus, thus providing a not-expensive digitally-controlled TV-sound processing system.

BASIC PRINCIPLES OF SWITCHED-CAPACITORS OPERATION

The use of switched capacitors to implement monolithic integrated filters has become more and more popular in recent years, and a great deal of papers is available on this subject.
Here, we give only a very brief outline of the basic principles of operation of a SC circuit. For better understanding, the reader should refer to some of the papers existing in the literature [e.g. 8, 12].

Fig. 2 - Equivalence between time-continuous (a) and switched capacitor (b) resistor. MOS implemention of a SC resistor is also shown (c)



The basic concept of a SC circuit is a charge transfer between two nodes by means of switches and a capacitor (fig. 2).
The switch S is alternatively thrown from the left-hand position to the right-hand one, with a switching frequency $f_S = 1/T_S$. When the switch is in the left-hand position, as shown in the figure, the capacitor C is charged with a charge equal to $C\,V_1$. When the switch is thrown to the right-hand position, its charge becomes equal to $C\,V_2$. So, during a complete switching period T_S, a net charge ΔQ is transferred from node 1 to node 2, where:

$$\Delta Q = C\,(V_1 - V_2) \qquad (1)$$

Over a period which is very large with respect to the switching period T_S, the charge transfer is quasi-continuous, so that an equivalent current flow can be assumed to occur between the nodes 1 and 2. This current is equal to:

$$I = \Delta Q/T_S = C\,(V_1 - V_2)/T_S \qquad (2)$$

Therefore, the block formed by the capacitor C and the switch S is equivalent to a resistor R equal to:

$$R = T_S/C \qquad (3)$$

Fig. 2c shows a simple MOS implementation of a SC resistor. M_1 and M_2 are N-channel enhancement-mode transistors which are clocked with opposite non-overlapping phases of frequency f_S (ϕ and $\overline{\phi}$, respectively).
A basic building block to implement active filters is the integrator. Fig. 3 shows both the time-continuous integrator and its equivalent implementation obtained using SC technique.
The time constant of the time-continuous integrator is given by:

$$\tau = R \cdot C_F \qquad (4)$$

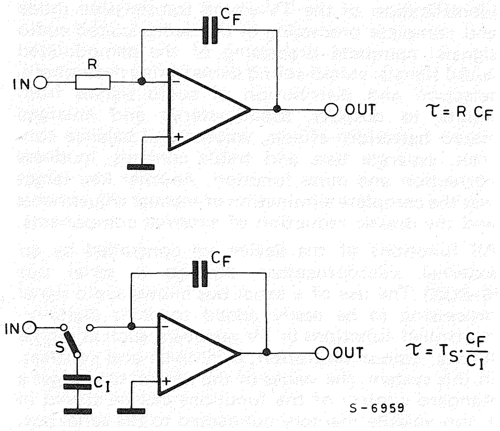
and the time constant of the SC integrator is equal to:

$$\tau' = T_S \cdot C_F/C_I \qquad (5)$$

The time constant τ' depends on the switching period T_S and on the ratio between the feedback capacitor C_F and the injection capacitor C_I, but not on the absolute value of the capacitors. This is the most important advantage offered by the SC technique since the ratio between two capacitors integrated in the same chip can be made very accurate (within a fraction of 1%).
Moreover, from eqn. 3 one can see that to achieve large resistance values, very small capacitors are needed if a proper sampling frequency is used. To implement a 10MΩ resistor with a sampling frequency of 125KHz, for instance, a capacitor of 0.8pF is needed, which occupies a very small silicon area with respect to a time-continuous resistor of the same value.

SC circuits lend themselves very well to be implemented in CMOS technology since switches are obtained with small transistors and good operational amplifiers may be realized.

Fig. 3 - Integrator: time continuous (a) and SC (b) implementation



By means of switched capacitors and operational amplifiers, it is possible to implement many filter configurations.
SC filter can be obtained by replacing time-continuous resistors of RC filters with SC resistors or by following different approaches. A popular design technique starts from passive RLC-filter prototypes.
To obtain precise filters, accurate analysis is required, which utilized the z-transform, and computer simulation is recommended.
In any case, the designer should pay attention to problems, like "aliasing", typical of sampled-data systems.

As a matter of fact, the configuration shown in fig. 2c is not used for SC circuits in practical IC's since it suffers from parasitic effects, and other configurations are implemented. However, the basic principles of operation are still the same as outlined here.

THE GERMAN TWO-CARRIER SYSTEM FOR TV-SOUND TRANSMISSION

The two-carrier transmission system has been introduced in Germany since September 1981 to allow possibility of high-quality stereophonic and bilingual sound TV transmission, while providing compatibility with the existing monophonic TV receivers.

This system has been widely described [5], and here we recall only its basic characteristics.

Two sound carriers are transmitted together with the video signal. The frequencies of the two sound carriers (Sound Carrier 1 and Sound Carrier 2) are displaced of 5.5MHz and 5.742MHz, respectively, with respect to the video carrier.

Both carriers are frequency modulated by sound signal, giving rise to two different sound channels, Channel 1 and Channel 2.
Sound channels are transmitted with different power (-13dB and -20dB, respectively, with respect to the video-signal power).
Frequency deviation due to 500Hz modulation for the maximum volume is ± 30KHz in both channels.
Audio bandwidth is 40Hz to 15KHz.
Transmitted sound signals are preemphasized (time constant = 50μs).
Audio components of the transmitted sound channels are shown in Table 1. It is easily seen that this system ensures compatibility with monophonic TV-sets, which demodulate only Sound Channel 1.

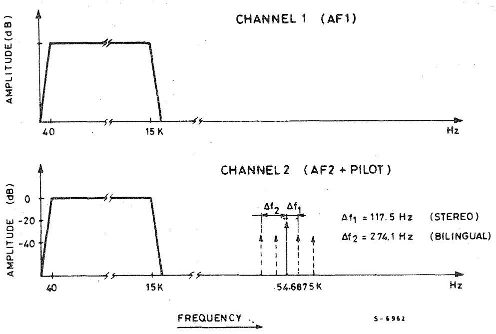
To identify the transmission mode (mono, stereo or bilingual), a "pilot tone" signal is transmitted together with Sound Channel 2. It consists of a pilot carrier (f_C = 54.6875KHz) which is amplitude modulated with 117.5Hz (f_H/133, where f_H is the video horizontal scan frequency) in case of stereophonic transmission and with 274.1Hz (f_H/57) in case of bilingual transmission. Amplitude-modulation depth is 50% . The carrier is unmodulated in case of monophonic transmission.
Frequency deviation of Sound Carrier 2 due to modulation with pilot tone signal is ± 2.5KHz.

TABLE 1 - Audio components of transmitted sound channels and significant outputs of the IC dematrixing network (German two-carrier system)

TRANSMISSION MODE	SOUND COMPONENTS		DEMATRIX OUT		
	CHANNEL 1	CHANNEL 2	A1	A2	A3
MONOPHONIC	MONO	MONO	MONO	–	–
STEREOPHONIC	MONO ($\frac{L+R}{2}$)	R	MONO	L	R
BILINGUAL	LANG. A	LANG. B	LANG. A	–	LANG. B

Frequency spectra of the two sound channels after frequency demodulation are shown in fig. 4, where AF1 and AF2 are the audio signals of Channel 1 and Channel 2, respectively.

Fig. 4 - Spectra of the demodulated sound channels in the German two carrier system



FEATURES OF THE M8793

A block diagram of the sound section of a TV receiver is shown in fig. 5.
The audio signals AF1 and AF2 (frequency range 40Hz to 15KHz) are output by the IF + demodulation section, which operates according to the "quasi-split sound" system [5], and are fed to the M8793, which performs complete audio processing and drives directly the power stages for loudspeakers and earphones.

Specifications of this device (see Table 2) include identification of the TV-sound transmission mode and complete processing of the demodulated audio signals: complete processing of the demodulated audio signals: stereo-sound dematrixing deemphasis, selection and distribution of audio signals from inputs to outputs, pseudo-stereo and enlarged stereo basewidth effects, volume and balance controls, separate bass and treble controls, loudness correction and mute function. Another key target was the complete elimination of manual adjustments and the drastic reduction of external components.
All functions of the device are controlled by an external microprocessor through a serial bus (S-BUS) The use of a serial bus allows audio signal processing to be easily added to other digitally-controlled functions in TV receivers, such as digital tuning, colour saturation, brightness and contrast. In this system, the values of the parameters to get a standard control of the functions can be stored in a non-volatile memory connected to the serial bus.

Fig. 6 shows the block diagram of the M8793. One can easily identify the two main sections for the IC, i.e. the pilot-tone decoding section and the audio signal precessing section. Both of them are controlled through the microprocessor interface. The timing is generated starting from an integrated oscillator which works with an external 4-MHz quartz crystal. Alternatively, and external 4-MHz clock can be used.

Fig. 5 - Block diagram of the sound section of a TV receiver

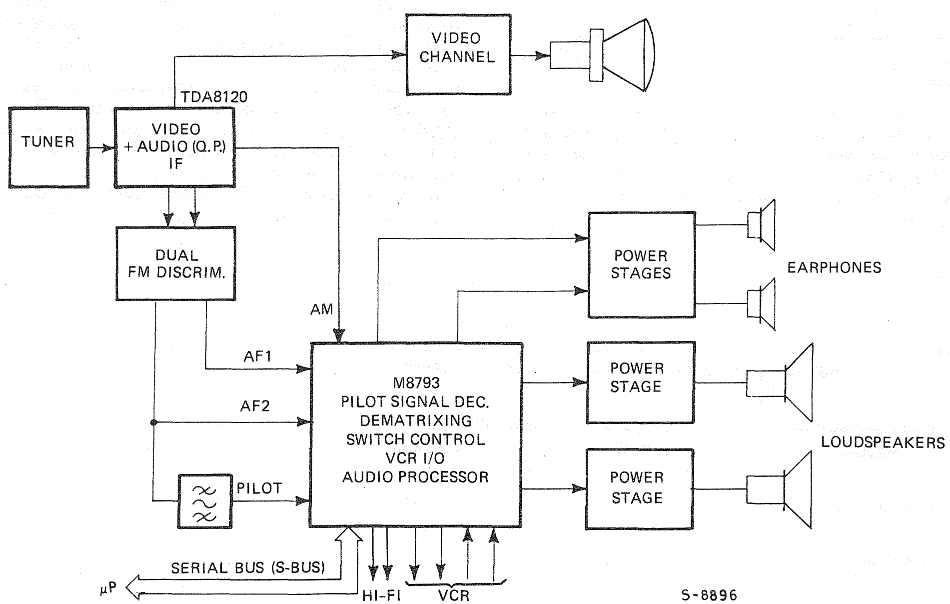


Fig. 6 - Block diagram of the stereo decoder and audio processor M8793

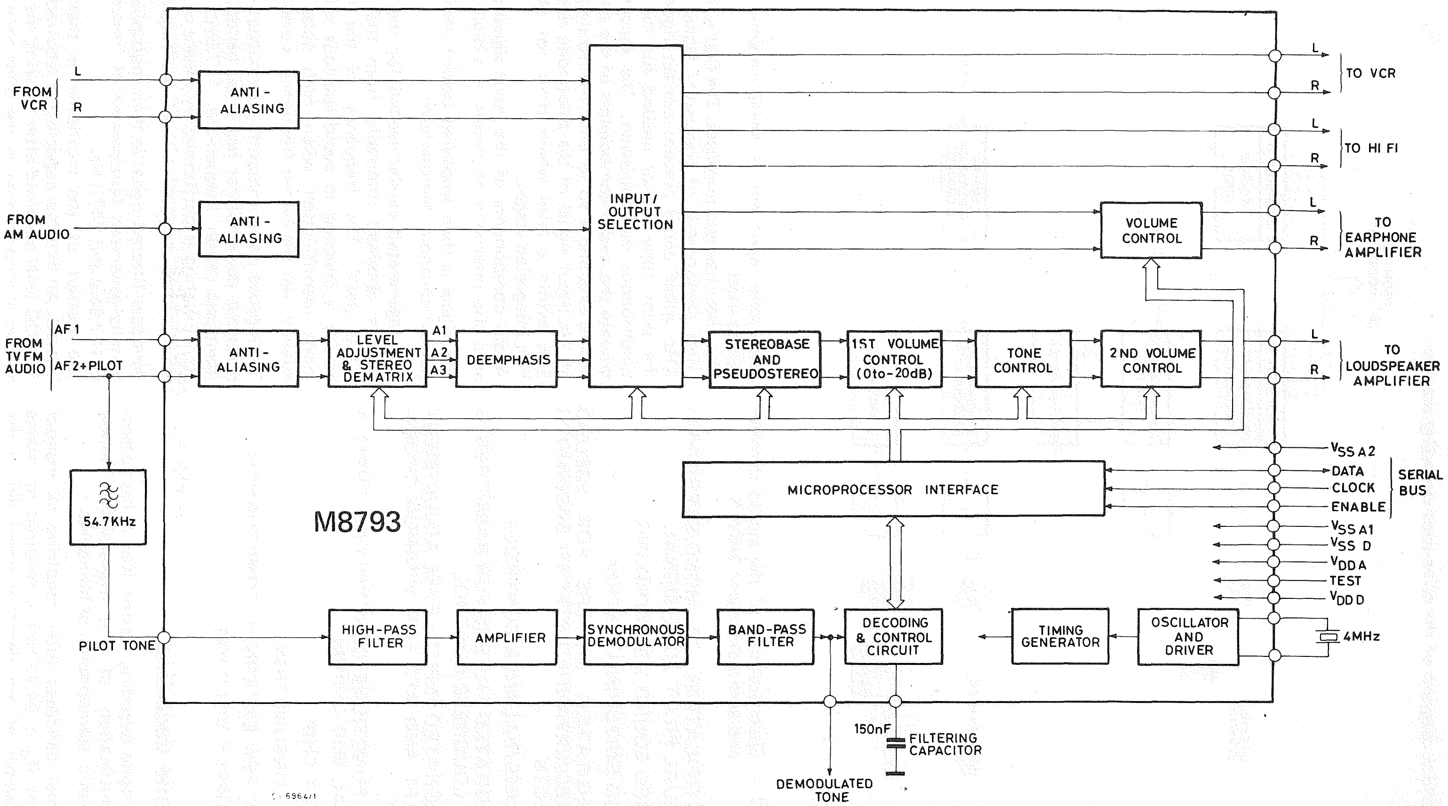


Fig. 7 - Block diagram of the pilot-tone decoding section

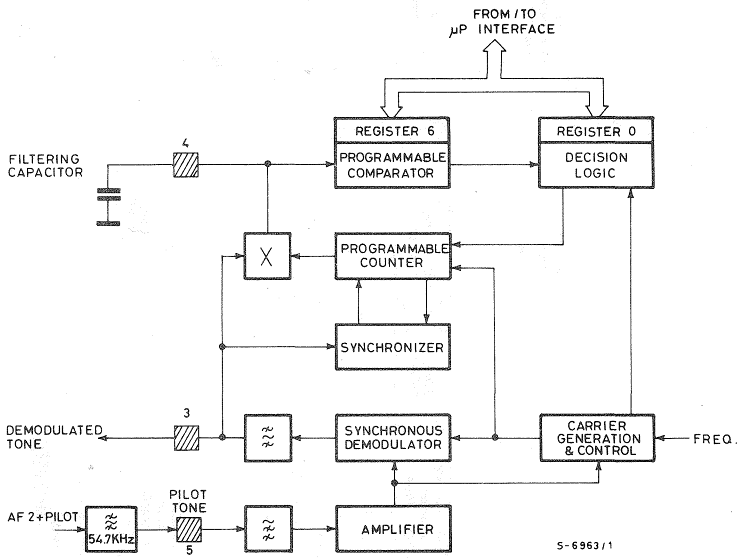


TABLE 2 - Specifications of the stereo decoder and audio processor M8793.

- IDENTIFICATION OF MONO/STEREO/BILINGUAL PILOT TONE SIGNAL
- STEREO SOUND DEMATRIXING
- AUDIO SWITCHING MATRIX
- INTEGRATED FILTERS FOR PSEUDO STEREO/EXPANDED STEREO BASEWIDTH EFFECTS
- VOLUME/BALANCE CONTROLS
- INTEGRATED FILTERS FOR BASS/TREBLE AND LOUDNESS CONTROL
- INTEGRATED INPUT LEVEL ADJUSTMENT OF AF1 AND AF2 FOR MINIMUM CROSSTALK
- ALL FUNCTIONS CONTROLLED THRU A SERIAL BUS (S-BUS)
- SINGLE CHIP
- LOW CONSUMPTION
- VERY FEW EXTERNAL COMPONENTS
- 5V SUPPLY VOLTAGE

Pilot tone decoding

The pilot-tone decoding section (fig. 7) has to provide identification of the transmission mode (monophonic, stereophonic or bilingual).

An external bandpass filter requiring no manual adjustment (f_o = 54.7KHz) is required to reduce the bandwidth of the electrical signal fed to the pilot tone input pin. A cheap ceramic reasonator can be used.
This filter has two purposes. The first is to attenuate noise components of frequency higher than 54.7 KHz which could prevent correct rigeneration of the pilot tone carrier needed for the successive synchronous demodulation. The second is to attenuate the audio components of the demodulated Sound Channel 2 so to limit the maximum voltage of the input signal of the pilot-tone decoding section within a level which does not saturate the first integrated stages.
Audio components of the input signal are attenuated also internally be means of a high-pass filter.

The signal is then amplified (about 24dB) and fed to a synchronous demodulator.

The regenerated carrier needed for the demodulation is obtained internally from the pilot tone signal itself. The frequency of the rigenerated carrier is checked to avoid spurious transmission-mode identification when high noise with frequency near the carrier frequency is present.

To improve noise immunity, the output of the synchronous demodulator has to be bandpass filtered (bandpass center frequencies: 117.5Hz and 274.1 Hz). Filtering is performed by means of integrated circuitry implemented with SC technique.

Bandpass-filtered signal is then multiplied with internally-generated squarewaves of frequency equal to 117.5Hz and 247.1Hz.
The output of the multiplier is then low-pass filtered. An external capacitor is required.
The DC level obtained after filtering the multiplier output is proportional to the pilot tone amplitude

so it is used in the decision logic to identify the transmission mode.
To optimize decoding with respect to the signal level output by the sound IF + demodulation section of the TV-set, a software-programmable comparator is provided.

The information decoded by this section is loaded into a register contained in the microprocessor interface. This register is read, via the serial bus, by the external microprocessor which will then select the operating mode of the audio signal processing section.

Audio signal processing

As it is well known, sampling process generated sidebands located around the multiples of the sampling frequency. Therefore, antialising network is needed for all input audio signals to remove high-frequency components which could be "aliased", i.e. folded down, into the audio base-band by the sampling process. Sampling is performed with a frequency of 125KHz so signal components above about 62.5KHz have to be removed.
Antialising circuits are basically fully-integrated low-pass filters. The best position to place these filters is at the beginning of the integrated signal path so signal processing with SC technique can be optimized.

Stereo audio signal AF1 and AF2 need at first to be dematrixed to obtain the left stereo channel. Dematrixing has to be performed according to the matrix coding of the German two-carrier system so the function to be performed to extract the left channel L'is the following:

$$L' = 2AF1 - AF2 \tag{6}$$

In case of stereophonic transmission, the right channel R' is equal to

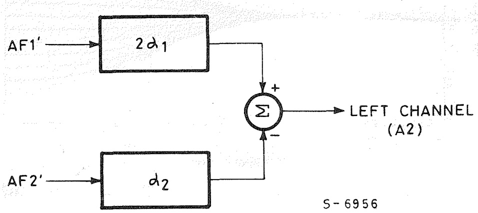
$$R' = AF2 \tag{7}$$

To achieve satisfactory stereo channel separation, levels of the two input signals AF1 and AF2 have to be adjusted so that crosstalk from the right to the left channel is reduced to a minimum. The actual equation to implement in order to extract the left channel is:

$$L' = 2\alpha_1 \cdot AF1 - \alpha_2 \cdot AF2 \tag{8}$$

(see fig. 8, where AF1' and AF2' are the antialised audio signals AF1 and AF2, respectively).

Fig. 8 - Extraction of the left stereo channel



By using microprocessor-programmable active capacitive divides, input level adjustment can be performed digitally on each TV-set in the factory. Values of coefficient α_1 and α_2 are then stored in an external non-volatile memory and will be recalled, e.g., at each mains-on. If no non volatile memory is used in the system, coefficients α_1 and α_2 are set to a standard value, and an external potentiometer is used in the factory to achieve the required input level adjustment.

Table 1 shows the significant outputs of the dematrixing network(A1, A2 and A3) in case of monophonic, stereophonic and bilingual transmission.

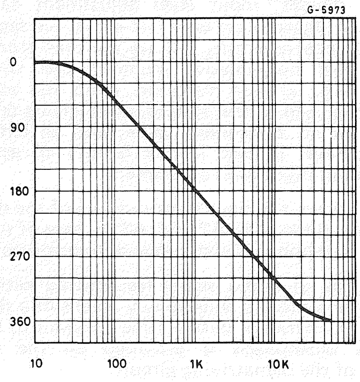
Deemphasis of audio signal transmitted with the two carrier system is achieved by means of a simple onepole SC network with a time constant of 50μs. Separate deemphasis is provided on the three outputs of the dematrixing circuit.

For a TV receiver of the new generation, input connections for different audio sources have to be provided: TV (from two-carrier system transmission), stereo VCR and an additional demodulated not-preemphasized audio signal (e.g. from an AM transmission). Likewise, outputs for different audio lines have to be provided: loudspeakers, earphones, stereo VCR and "Hi-Fi".
Selection and distribution of audio signals from inputs to outputs are performed according to the functional requirements of the peritelevision or "SCART" plug. In case of selection of TV-audio inputs (A1, A2 and A3), operating audio-mode (monophonic, stereophonic or bilingual) is selected under software control. The microprocessor takes into account both the TV-user's command and the transmission-mode information decoded by the pilot-tone decoding section, and loads the proper operating-mode code into the integrated microprocessor interface through the serial bus.

Audio lines for loudspeakers need volume and tone control and special stereo effects, while lines for earphones need only volume control.
The other audio output lines (VCR and Hi-Fi) need no control so selected audio signals are directly fed to the output drivers. Only mute function is provided in these lines.

Pseudo-stereo effect is required in stereophonic sound equipments to give stereo impression when monophonic sound is processed.
A good stereophonic image is obtained by introducing increasing phase shift (0° to 360°) of the right channel in the medium frequency range. This is achieved by means of a SC circuit. Fig. 9 shows the typical phase-shift response obtained with this circuit.
In case of stereophonic transmission, to provide a better stereo image with commercial TV-sets, enlargement of speaker basewidth is generally required.
To simulate enlarged basewidth, antiphase crosstalk of middle-frequency components from the left to the right channel and viceversa is needed. Crosstalk of about 50% is introduced to obtain good stereo basewidth impression with common commercial stereo TV-sets (actual speaker basewidth equal to about 70 cm.).

Fig. 9 - Phase shift for pseudo-stereo effect



Special stereo-effects circuit are placed before the stage which perform volume and tone control since they need balanced signals to work properly.

Volume control is achieved by means of software programmable active capacitive dividers, which allow fully-integrated control with minimum power dissipation and harmonic distortion.

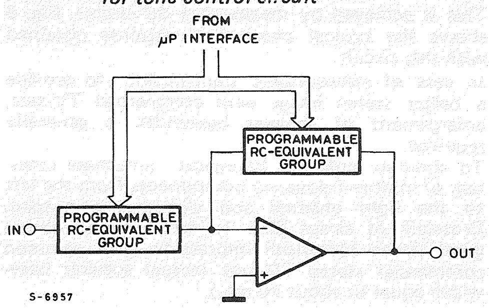
Volume control on loudspeakers lines is divided in two parts. The first provides 0 to -20dB attenuation, to allow the user to boost tone control up to +20dB even in the presence of the maximum input signal level. In this case, in fact, attenuation of processed-signal level up to a value corresponding to the tone boost is required before the tone control stages to avoid distortion.
Total volume control on loudspeakers lines is 0 to -80dB with resolution of 1dB per step, plus mute.

Earphones lines are also volume controlled (0 to -60dB, 2dB per step, plus mute). No other control is required for these lines.

Volume balance is easily obtained under microprocessor control since volume control is separate on left and right channel, both on loudspeakers and on earphones lines.

Separate bass and treble control is achieved by means of two software-programmable first-order sections implemented with DC technique (fig. 11).

Fig. 11 - Diagram of the first-order section used for tone control circuit



The transfer function of the equivalent time-continuous first-order section is:

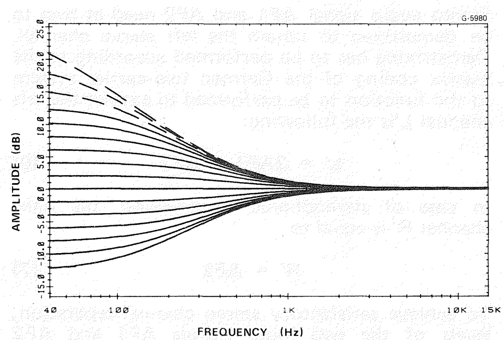
$$H(s) = K \frac{1 + sT_1}{1 + sT_2} \tag{9}$$

By varying the values of the constants K, T_1 and T_2 in eqn.9, one can change low-frequency gain, pole and zero (hence high-frequency gain, too) of each section, thus achieving the required tone control.

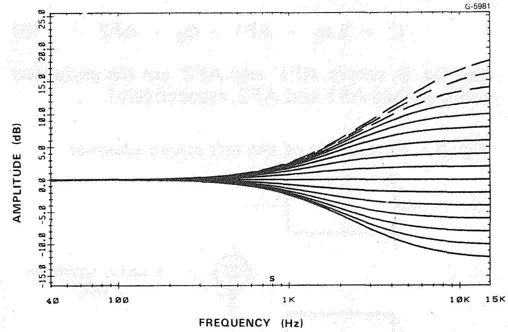
Simulated frequency response of SC tone control sections are shown in fig. 12.
Both bass and treble are controlled in the range -12 to +12dB, with regulation of 2dB per step.
Three more steps of boost control are provided to allow loudness correction under microprocessor control (dashed lines in fig. 12).

Fig. 13 shows the characteristics of tone control circuit when extreme values of both controls are programmed without loudness correction.

Fig. 12 - Frequency response of tone control sections (computer simulation)
a) bass control; b) treble control



a)



b)

Fig. 13 - Characteristics of extreme tone control without loudness correction (computer simulation)
a) bass +12dB, treble +12dB
b) bass -12dB, treble -12dB
c) bass +12dB, treble -12dB
d) bass -12dB, treble +12dB

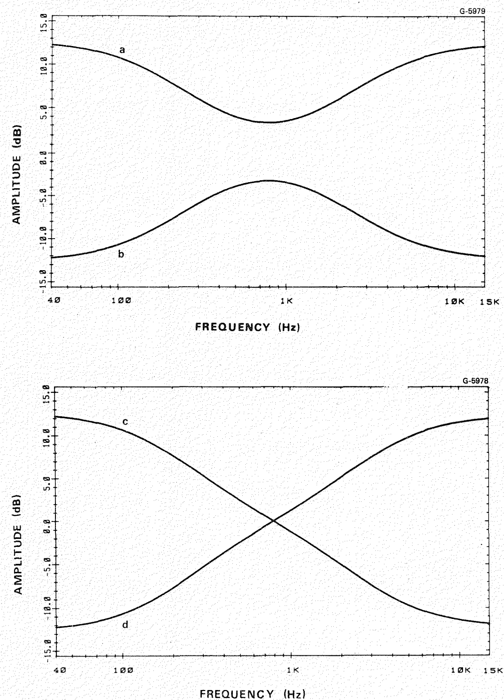
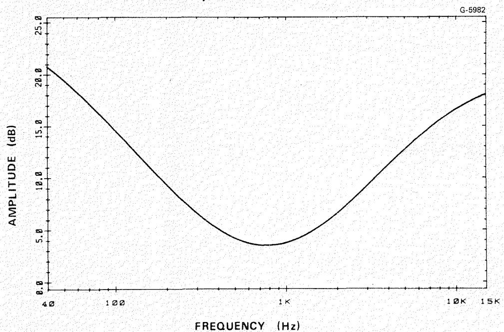


Fig. 14 shows frequency response of tone control circuit when full loudness correction is inserted together with maximum tone boost.
Tone control is common to both channels.

Fig. 14 - Frequency response of tone control circuit with maximum loudness correction and tone boost (computer simulation)



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