(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 805 613 A1

(12) EUROPEAN PATENT APPLICATION

(43) Date of publication:
05.11.1997 Bulletin 1997/45

(51) Int. Cl.[6]: H04S 1/00

(21) Application number: 96600010.1

(22) Date of filing: 23.12.1996

(84) Designated Contracting States:
DE FR GB IT

(30) Priority: 21.12.1995 GR 95100467

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(54) Digital system for the equalisation of frequency and time response of audio systems and devices and of listening spaces

(57) The invention consists in a digital electronic system and a method removing the distortions of the audio signal of audio reproduction systems, by completely equalising not only the distortions generated by these very systems, but also the distortions generated by listening spaces, as described in Figure 3. This is a stand-alone lowest system, connected, as shown in Figure 2, between digital audio sources (eg. CD player) and analog amplifier, which realises, by a Least Squares method, finite impulse response digital filters, with more than 800 coefficients per channel of stereo digital audio signal at the sampling frequency of 44,100 Hz, capable of realising up to 2,200 coefficients per channel by the method described in Figures 4 and 5, and which is easily programmable by the user through a system of PCMCIA card reading, screen and operation control microprocessor or/and through a communication gate with a PC, as shown in Figures 7, 10 and 11.

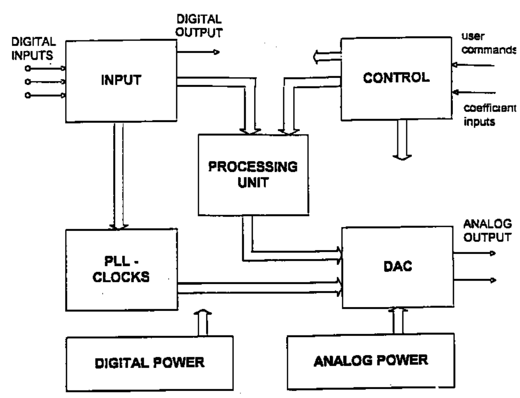

FIGURE 1

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Description

The invention refers to electronic systems improving the performance of audio reproduction systems and listening environments. In particular, the invention refers to systems eliminating the distortions on the source audio signal generated after its transmission through existing audio reproduction systems.

It is known that most audio reproduction systems, and mainly the loudspeakers, generate significant distortions, which deteriorate the source audio signal. The same problem also occurs during the transmission of the signal from the source (eg loudspeaker(s)) to the listener, due to the acoustic response of the listening environment. The above distortions are described mathematically by the frequency response (transfer function) of the specific system or also by its time (impulse) response.

The existing loudspeaker manufacturing technology and all of the usual acoustic spaces (except for specially constructed anechoic studios) appear to have significant deviations from the optimal flat response, which at the frequency response can reach +20dB, and at the time response appear as dispersion and response delays, which can last several seconds (for acoustic transmission channels).

Audio systems and devices which allow some correction of the above distortions, are commonly called equalisers. As it is known, an equaliser is a system or device which forms an appropriate corrective response for an audio system in order to remove some of the existing linear distortions from components of the system (in this case loudspeakers or acoustic environment). The combined result of the use of an equaliser and an audio system achieves an improved response.

However, the state of the art of such equalisers is mainly based on analog electronic technology or digital implementation of existing analog designs. Analog equalisation systems accomplish the desired response by the use of a parallel band-pass electronic filter arrangement (filter banks), which by the appropriate increase or decrease of their response gain, in the specific frequency range in which they function, approximate the required overall frequency response. These arrangements are called graphic equalisers, since the form of their spectral response is represented by the gain position of each one of the filters.

The accuracy of the correction achieved by the above systems is determined by the number and the characteristics of the band-pass filters, which, however, in existing realisations do not allow a better resolution than 1/6 of the octave, which is far lower than the requirements for the exact equilisation of an audio/acoustic system. Therefore, these systems, in the best cases, achieve only an approximate correction of the amplitude of the frequency response of a system.

Furthermore, a problem of these analog filters is that, due to their finite cut-off in neighbouring frequency regions, they generate secondary distortions, and, while they cannot correct the time domain response of a device, they often increase such distortions which degenerate the initial time response of the system.

From the aforementioned problems of existing equilisers, the one of accurate frequency and time-domain correction was sought to be solved by the use of fully digital equilisation filters. In consequence, there have appeared devices, which can achieve a relatively more accurate correction of the amplitude response of a system, whilst their accuracy mainly depends on the number of digital filter coefficients, which they can implement during their operation. Nevertheless, neither of these digital devices have in essense overcome completely the problems referred in the preceding paragraph.

It has been proved that for a complete and accurate equalisation of loudspeakers there are required approximately 600 coeficients (per audio channel, and double for stereo reproduction), whereas for equilisation of listening spaces up to 30,000 coeficients per channel are required (depending to the size and the acoustic characteristics of each space). The state of the art of the above digital equilisation realisations, are mainly based on existing Digital Signal Processing (DSP chips) microprocessors, which, by virtue of the existing technology, allow digital filter realisation of up to 300 coeficients per channel. Hence, although these systems can attain a more accurate loudspeaker equilisation, they are absolutely inappropriate for the equilisation of the acoustics of listening spaces. Moreover, these systems have pre-programmed characteristics and allow the equilisation of only one or maybe a limited number of loudspeaker types.

In addition to the above digital systems, there has been presented a special-purpose device, which by use of high-cost processors allows realisation of 2,200 coeficients per channel, with a potential for partial equilisation of listening space. This device, however, requires the use of special measuring system and computer, as well as the labour of an expert technician for its adjustment. Therefore, this is rather a case of incidental technical work in the equilisation by use of highest special-purpose equipment, than an integrated equalisation system or device, intended for consumer use.

Nevertheless, another problem inherent to every audio equilisation system is the requirement for using an initial measurement or estimate of the response of the system which will be corrected. The common equilisation systems, which are directed to non-expert users, do not allow the use of such measurements and their adjustment is done empirically by altering the gain or every filter. This method, in addition to the limited accuracy and resolution of such devices, discussed above, usually leads to a marginal and approximate improvement of the response of the system under correction, and quite often to erroneous and incorrect effects.

In equilisation systems intended for expert users, this possibility is often provided by a measuring system, accompanying the equalising device, which consists of sub-systems such as acoustic stimulus generator, often amplifier and loudspeaker, microphone, pre-amplifier and data analysis device. Recently, part of the above sub-systems have been integrated into hardware and software, adaptable to Personal Computers. Although the above arrangements, for measuring audio response prior to applying equilisation, are advisable, they constitute high cost devices and require to be used by expert technicians. Furthermore, accurate measurements are time comsuming, demand almost special laboratory conditions and are, therefore, imposible to be achieved during the everyday operation of the audio device.

An additional important drawback of the above measurement techniques for the evaluation of the equalisation data is related to the variability of each measurement with regard to changes in the position of source and receiver as well as to every specific listening environment. For these reasons, any audio measurement and, by extension, any realisation of an equaliser is effective for only one specific combination of the following parameters: loudspeaker type, listening angle, distance form the source and listening position in one specific acoustic space. Operation of the equilised system under different conditions (from the ones used during measurement) introduces significant errors into its audio performance.

This invention consists of a digital system for the equalisation and correction of frequency and time response of audio systems, devices and listening environments, comprising of : (a) an input sub-system, (b) a digital filter realisation sub-system for processing signals derived from the input sub-system, (c) a timing sub-system, (d) a user interface sub-system (e) an output sub-system delivering processed analog audio signal, and (f) a power supply sub-system. Each of the above sub-systems is realised by electronic circuit, as described below. The general diagram of the invention and of each of its sub-systems is shown in Figure 1. In this Figure it can be seen that the input sub-system receives as inputs digital audio signals (DIGITAL INPUTS) and supplies signals to the sub-system realising the digital filter (PROCESSING UNIT). Processed data are subsequently transmitted to the output sub-system (DAC), which then reproduces the analog output audio signal (ANALOG OUTPUT). In the same Figure it can be observed the timing sub-system (PLL-CLOCKS), the user interface sub-system (CONTROL), which receives on the one hand commands through buttons or remote control (user commands) and on the other, coefficient data (coefficient inputs) through external data cards or/and connection to a Personal Computer (PC). The power supply of the digital circuits of the device is effected through the DIGITAL POWER sub-system, whereas for the analog circuits of the device through the ANALOG POWER sub-system.

The above system constitutes realisation of an equalisation and correction algorithm for audio systems and acoustic response. This algorithm allows complete and exact frequency and time-domain correction of audio devices and listening environment acoustic response, without the introduction of secondary or other distortions. The realisation method of this algorithm on the above device allows the implementation of digital filters with more coefficients than allowed by the common processors (real-time DSP chips), in a stand-alone, relatively low-cost device, which is suitable for the correction of numerous audio devices, and allows flexible programming by non-expert users.

The equalisation and correction system according to this invention is stand-alone, relatively low-cost unit, which allows the realisation of Finite Impulse Response (F.I.R.) filters of up to 832 coefficients for stereo audio reproduction at the sampling frequency of 44,100 Hz (as defined by the Compact Disc standard, CD), a higher value than that allowed by the current Digital Signal Processing chip technology. This performance can raise up to 2,200 coefficients per channel by use of an auxiliary circuit.

Moreover, this invention allows flexible programming of each corrective function of the device by the use of suitable filter libraries. Given that up to 1,000 such adjustments can be immediately accessible by the user, there is provision of operating the system with a multitude of audio devices and listening conditions. Furthermore, additional elaborate adjustments can be realised through an external connection to a PC.

From the above it can be deduced, that this invention overcomes the problems of prior art, which, as previously analysed, in its common analog form does not allow accurate and complete response correction while in its current digital realisations does not allow any flexible and complete control over comparative applications.

The system of this invention, a general block diagram of which is shown in Figure 1, is connected to other audio systems by the way illustrated in Figure 2. In this Figure, it is shown that the system of this invention (DIGITAL COMPENSATOR) is connected between any digital audio source (DIGITAL INPUTS), for example a compact disk system (CD), a digital tape system (DAT), a digital sound processor (DIGITAL PROCESSOR) and/or any analog sound source (ANALOG SOUND SOURCE) the electrical signal of which has been converted into digital through an analog to digital converter unit (ADC), and through its analog electronic outputs of line level (ANALOG OUTPUTS) to the amplifier (AUDIO AMPLIFIER) and the loudspeakers (LOUDSPEAKERS) of the audio system, which may operate in an open or closed listening environment (ACOUSTIC SPACE), in the usual manner.

More specifically, the digital output of the above sources (according to SPDIF or AES/EBU format) is connected with the digital inputs of the above device (which will be called digital filter). Analog outputs of the digital filter, transmitting the corrected audio signal (2 channels according to common stereo reproduction mode) are connected to the input of the audio amplifier unit (eg pre-amplifier or intergrated amplifier or pre-amplifier/mixer or self-amplified loudspeaker), which subsequently supplies the loudspeakers to be corrected in the usual manner. Moreover, digital outputs (DIGITAL OUTPUTS) of the digital filter, transmitting the corrected audio signal, can also be connected with digital sound storage systems (such as digital recorders, properly equipped PCs) or processing systems (eg. digital mixers or signal processors). In this manner, the above digital filter also functions, as an integrated digital centre for distribution and processing of sound signals.

The above digital filter is realised, according to a prefered realisation of the invention, with coefficients which have been pre-calculated and are stored in the memory of the device, while the calculation of the coefficients is effected by prior measurements of the device to be corrected, based on the algorithm described below. For the analysis below, the description of the discrete time functions will be adopted, where *n* discribes the time variable (sample number, while, for simplification, the constant of sampling period is omitted). From the initial measurement of the impulse response to be corrected h(n), the finite-time response of the inverse filter hi(n) is calculated. The calculation is based on a minimisation, according to the Least Squares, of the difference e(n) between the optimal output (Delta function δ(n)) and output under calculation y(n), which is:

$$y(n)=h(n)^{*}hi(n) = \sum_{i=0}^{M-1} hi(n)h(n-i)$$

where M is the filter length.
The error minimisation is done according to the equation:

$$I = \sum_{n=0}^{L-1} e^{2}(n) = \sum_{n=0}^{L-1} [\delta(n-k)-y(n)]^{2}$$

where κ is an appropriate delay in the output, which allows the minimisation in case that the input function does not have the minimum-phase properties, and L is the length of the response to be corrected. To evaluate the above inverse filter, a system of equations should be solved, of the form:

$$R^{T} hi = g_{k}$$

where R is the auto-correlation matrix of the function h(n), g_k is the cross-relation matrix of the input with the desired output, so that:

$$g_{k}^{T} = [h(k), ..., h(0), ..., 0]$$

The system is solved preferably using the Simpson Sideways Recursion algorithm and provides as an output the required function hi(n), of the inverse filter. According to the above, this filter will have the property to remove the distortions of a system (eg loudspeaker) with impulse response h(n), according to the equation:

$$h(n)^{*}hi(n) =\delta(n-k)$$

where κ is a (constant for all frequencies) delay, which does not audibly affect the processed signal. By the use of this inverse filter, the device implements a digital filter with response hi(n), so that any input sound signal s(n) (after proper conversion into digital domain by the Digital to Analog Converter), is filtered and produces a new signal in the filter output, the s'(n), according to the equation:

$$s(n) * hi(n) = s'(n)$$

If r(n) is the signal reproduced by the audio device to be corrected, which without the correction of the filter would be:

$$r(n) = s(n) * h(n)$$

after the use of the digital filter the input in the device to be corrected would be the pre-filtered (equalised) signal s'(n), and therefore the corrected reproduced signal will be r'(n), which will be given by the equation:

$$r'(n) = s'(n) * h(n) = s(n) * hi(n) * h(n) = s(n) * \delta(n-k) = s(n-k)$$

which constitutes an accurate delayed version of the input signal.

The preferred digital filter of response hi(n) implements a convolution (described in the above equations) under the form of a difference equation:

$$s'(n) = \sum_{i=0}^{M-1} hi(n)s(n-i)$$

where M is the length of the corrective filter and the number of its coefficients. The above preferred realisation allows the frequency and time correction of the response of audio devices (eg loudspeakers) as shown also by the results given in Figure 3. In this Figure the initial measurements of the impulse response of a loudspeaker is given by the diagram 3(a), whereas the measurement of the impulse response of the same loudspeaker after equalisation of the audio signal through the device of this invention, is given by the diagram 3(b). It is apparent that the method achieves complete correction of all time distortions introduced in the signal by the original audio system. In Figure 3(c) the amplitude and the phase of the frequency response of the above loudspeaker are shown, while in diagram 3(d) the response of the same loudspeaker is shown when the source signal has being equalised by the use of the above method. It appears that the above method achieves complete correction of the amplitude of frequency response of the particular audio device, in its frequency range of operation, with simultaneous correction of phase distortion, into that of a fully linear-phase response. In Figures 3(e) and 3(f) there are presented the three-dimentional time/frequency spectra, resulting from Figures 3(a) and 3(b) respectively, from which it is shown that the above method achieves complete and combined correction of the time/frequency response of a loudspeaker.

This system has the capacity to allow realisation of digital FIR filters of long impulse response, according to a preferred realization of the above difference equation describing such filters. As yet, in realisations of similar expressions there was required a sequential (per sample) access to a Multiplier/Accumulator (M/A), which performes all the numerical operations defined by expression of such a kind, in which cases the overall length of the filter under realization was being limited by the realization speed of such a numerical operation by the processor. In this invention a completely different from the known architectures is used, which is shown in Figure 4. According to this method there is achieved a simultaneous (per sample) access to K Multipliers/Accumulators (indicated by the responses h(0) - h(K-1)) and therefore a significant increase of the overall length of the realised digital filter is possible. If the desired filter must contain M coefficients, where M>K, then a special architecture is realised by this invention, which allows the division of the overall filter of M points, into blocks of K points, which are processed sequentially and combined with the stored partial results of the prior cycle (input from previous block) through a delay element (K cycle delay) at the final output of the filter, as shown in Figure 4. By this way the device has the ability of realising filters with M=832 coefficients per audio channel (at indicative sampling frequency of 44,100 Hz). By the addition of a further auxiliary circuit (of the processor discribed in the following paragraph) this performance can be doubled to 2,200 coefficients and/or their multiples by further additions of the same kind.

According to a preferred realization of the invention the above system can be based on the use of a special gate array-microprocessor, which operates by a special peripheral circuit consisting of additional field-programmable gate-arrays, which realise the recycling of the partial processing results.

The above invention allows a flexible and fast programming by non-expert users, since it uses pre-programmed libraries containing the coefficients of the appropriate corrective filters for every application. By the use of visual information appearing on the screen in the front of the digital filter unit, the user selects the type of the device to be corrected and by simple commands he activates the process of the transfer of coefficients to the digital filter, which in turn operates with the specific corrective response, until another selection is called by the user.

According to a preferred realisation of this invention, the library with the corrective responses can be realised by special data cards (PCMCIA cards) which at a capacity of 2Mbytes allow storing of up to 1,000 different filter types. This great data capacity, which will be increasing with the increase of card capacity by future technology, also allows storing of certain variations of the basic corrective responses, as for example corrective filters with appropriately boosted or lowered frequency response regions, which may be chosen by the user as more desirable. Another noteworthy innovation of this approach is that the library may also contain, on top of the basic corrective responses, other, allowing a further differentiation of the final correction of every device. A further extension of the above principle applies to the method for correcting the acoustic response of listening environments, as is described below.

The above system allows also the correction of the acoustics of listening environments, by a preferred approach based on correction of typical characteristics of listening spaces, wherein the audio devices to be corrected operate. By the use of the libraries described above, the user has the ability to select between correction of a device (eg loudspeaker) functioning without additional acoustic distortions (anechoic correction) and/or correction with distortions of typical listening environments. Calculation of corrective responses for such spaces is based upon a preferred special measurements and grouping methodology by the use of Vector Quantisation. This technique allows an approximating correction which achieves significant improvement (in the order of 5dB improvement of the spectral deviation of the response) in most cases. In application where a more accurate correction of the acoustic response is required, as for example in permanent audio installations (eg in recording studios), the present invention can allow additional storing of corrective responses measured and calculated for the specific acoustic environment, by the use of a PC communication port (RS 232).

The above system has been measured and it has been confirmed that it performes in the desirable and predicted manner. Typical measurement results, acquired by this system, are described by Figure 3. In this Figure the initial measurement of a loudspeaker impulse response is given by the diagram 3(a), while a measurement of the impulse response of the same loudspeaker after equalisation of the audio signal through the device of this invention is given by diagram 3(b). As shown, the method achieves complete correction of all time distortions introduced to the signal by the initial audio system. In Figure 3(c) the amplitude and the phase of the frequency response of the above loudspeaker is presented, while in diagram 3(d) the response of the same loudspeaker is shown when the source signal is being equalised by the use of the above method. As is shown, the above method achieves complete correction of the amplitude frequency response of the audio device under examination, in the frequency range of its operation, with simultaneous correction of its phase distortion into that of a completely linear-phase response. In Figure 3(e) and 3(f) the three dimensional time/frequency spectra resulting from Figures 3(a) and 3(b) respectively are presented, from which it is shown that the above method achieves complete and combined correction of the time/frequency response of a loudspeaker.

From the above examples it is obvious that the present invention solves the problems of prior art as it allows the complete time and spectral correction of the response of electroacoustic devices, by the use of a simple and cost-effective digital filter unit, which can also allow correction of the acoustics of listening spaces and which is easily programmed, so that it achieve the desired by the user - who may now be the average consumer - response.

The description of the Figures that follows concerns one of the realisations of the invention by way of example, which does not in anyway limit the invention and its scope.

Figure 1 presents the general diagram of the digital filter, which consists of the following sub-systems: (a) Input circuit for the digital audio signal (as shown in detail in Figure 6), (b) circuit realising the main digital filter (as shown in detail in Figure 5), (c) user interface control circuit (as shown in detail in Figure 7), (d) clock circuit (as shown in detail in Figure 8), (e) circuits of analog and digital audio output (as shown in detail in Figure 9), (f) power supply circuit. In this Figure it is shown that the input sub-system (INPUT) receives as inputs digital audio signals (DIGITAL INPUTS) and transmitts the signal to the sub-system realising the digital filter (PROCESSING UNIT). The processed data are then transmitted to the output sub-system (DAC), which reproduces the analog output sound signal (ANALOG OUTPUT). In the same Figure there are shown the clock sub-system (PLL-CLOCKS), the user control sub-system (CONTROL), which receives on the one hand comands through buttons or remote control (user commands) and on the other hand coefficients data (coefficient inputs) through external data cards or/and connection with a PC. The electrical power of the digital circuits of the device is achieved through the DIGITAL POWER sub-system, while of the analog circuits through the ANALOG POWER sub-system.

Figure 2 presents the general operating mode of the digital filter device during its corrective function. In this Figure it is shown that the digital filter device (DIGITAL COMPENSATOR) is connected between any digital audio source (DIGITAL INPUTS), for example compact disk system (CD), digital tape system (DAT), digital sound processing system (DIGITAL PROCESSOR) and/or any analog audio source (ANALOG SOUND SOURCE) the electrical signal of which has been converted into digital through an Analog to Digital Converter unit, and through its analog electronic outputs of line level (ANALOG OUTPUTS) and the amplifier (AUDIO AMPLIFIER) and the loudspeakers (LOUDSPEAKERS) of the audio system, which may operate in an open or closed listening environment (ACOUSTIC SPACE), in the usual manner. More specifically, the digital output of the above sources (according to SPDIF or AES/EBU format) is connected with the digital inputs of the above device (which from now on will be called digital filter). Analog outputs of the digital filter, transmitting the corrected audio signal (2 channels according to common stereo reproduction mode) are connected to the input of the audio amplifier unit (eg pre-amplifier or intergrated amplifier or pre-amplifier/mixer or self-amplified loudspeaker), which subseqently supplies the loudspeakers to be corrected in the usual manner. Moreover, digital outputs (DIGITAL OUTPUTS) of the digital filter, transmitting the corrected audio signal, can also be connnected with digital sound storage systems (such as digital recorders, appropriately equipped PCs) or processing systems (eg. digital mixers or signal processors).

Figures 6 presents the general diagram of the input circuit. This circuit consists of a specialised interface CMOS (AES/EBU & SPDIF receiver), 3 discrete digital audio outputs according to the AES/EBU format (XLR), and SPDIF (RCA) with the appropriate receiver circuits (Receivers) and special switch selector (Selector) allowing the user to select the desirable input. Furthermore, there is provided a possibility of signal output through special driver circuit (Drivers). The input signal (serial data) is driven for further processing (to processing unit), while from these data a clock information is extracted, which is sent to the clock sub-system (to PLL).

Figure 5 presents the general diagram of the main digital filter circuit. This circuit consists of the coeficient register, which receives coeficients from the control and user interface sub-system, the gate-array processor and the peripheral circuits realising the delays of the data parts of K points. The array of Multipliers/Accumulators of K coeficients, as described in Figure 4, processes the input audio data (Data in) originating from the input sub-system (Figure 6) and combines them cumulatively with the stored pratial results of the previous process (Cascade input) by the use of appropriate delay (K cycle delay), producing the final processed output data (Data out) which then are driven to the output sub-system of the device.

Figure 7 presents the general diagram of the control and user interface circuit. This circuit allows the interconnection of the main filter unit with an external programming unit through PCMCIA cards, the interconnection with an external programming gate through PC and/or remote control (RS232 & telecontrol), the interface with screen (LCD) and the overall operations control by microcontroller (Controller).

Figure 8 presents the general diagram of the clock circuit of the unit, which achieves removal of any fluctuations in the flow of digital sound samples (jitter) as received by the input circuit (Clock in) and consists of a special phase-locked loop (PLL) containing the Crystal Oscillators (VCXO), Dividers and Phase comparator circuit, Sampling logic filter and special selector of the corrected clock signal (Clock out) used by the other circuits of the device. This sub-system is operated by the logic control of the control circuit (Power down & select logic).

Figure 9 presents the general diagram of the audio data output circuit, which provides stereo analog audio output, through appropriate driver circuits (drivers), either through balanced output line of 3 terminals (Balanced), or through unbalanced output line of 2 terminals (Unbalanced). This sub-system contains an integrated Digital to Analog Converter circuit (DAC) which receives the processed audio data (serial data) from the processing circuit (from processing unit) and functions under the clock of the clock sub-system (from PLL), a data De-emphasis circuit and an output signal Volume control circuit.

On the basis of the above diagrams, an example of electronic circuit describing the invention is implemented, shown in Figures 10 and 11, whereby a device is constructed, which also constitutes an application example of the invention. In Figure 10 it is shown that the input of digital audio data is applied at the inputs RCA1 - RCA3 (unbalanced connection according to SPDIF) and XLR1 - XLR3 (balanced connection according to AES/EBU). These signals are converted into digital voltages by the integrated circuits UNUXO, UMUXIN, which select the signal from the desired input, the signal then driven either to the decoder CS8412 or (without undergoing further processing) to digital outputs XLROUT and RCAOUT through the integrated UXLRT and the auxilliary electronic elements.

The main digital decoding of the input signal is performed by the receiver CS8412 from Crystal (AES/EBU & SPDIF receiver), which reproduces the input data for further processing. The significant output signals from this circuit, as shown in Figure 10, are the SDATA (serial digital sound data), SCK (serial clock data), FSYNC (separating data of the 2 stereo channels) and MCK (sampling frequency). In Figure 10 the remaining parts of the circuit are also shown, which read the coeficients of the inverse filter, perform the programming of the device by the user and the control of the device. The integrated circuit U8751 is the general microcontroller of the device, wherein all the remaining expander circuits (expanders) are connected through the control lines. More specifically, data lines D0 - D7 are connected to 2 integrated expanding circuits (expanders), the U8753 and U8255. U8753 is mainly used for the reading control of the PCMCIA card data, while the other is mainly used for the control of digital input sections, described previously.

Integrated circuit (CARD) of reading PCMCIA card data is controlled by the neighbouring U8757 and C4040, which determine the addresses of the card (CAADA0 - CAADA23) for the retrieval of the appropriate coefficient groups for the inverse corrective filters. The interface function between the device and the LCD screen, in the lower left part of Figure 10 is achieved by the intergated U8754 (tranceiver). Moreover, the circuit includes also the integrated MV601 for decoding user signals through telecontrol, which are received by the receiver SX486. Finally the gates URST are used for the operation of reseting the device.

Figure 11 includes the remaining functions of the device, appearing in general diagrams (Figure 5). The corrective filter coefficients read by the circuits of Figure 10, by the microcontroller and the data lines D0 - D7, through the integrated H245, are introduced into memory CRAM as data groups (low bytes, high bytes). The reading and transmission of the coeficients to the main unit of the digital filter A, as well as the state of its function are determined by the control systems CNTRL0 - CNTRL3, AD12 - AD15, originating from the microcontroller, which are transmitted to all the integrated programmed gate arrays in the diagram (IFG, Field Programmable Gate Array). Of these circuits, IFX3 performs the data inputting operations, while IFXO with memory CRAM the operations of assignment of the appropriate coefficients in the filter. Data recycling operation, as described by Figure 4, is performed by the Recycle Buffer, consisting of the integrated IFX1 and FIFO1, FIFO2. Finally, the integrated H373 is used for coefficient data reading control.

The main operation of the digital filter (for the realisation of the difference equation), described in previous paragraphs, is performed by the integrated gate-array circuit A, which requires for its operation the data of the input signal (DA0 - DA15), after the IFX3, and also the coefficients of the inverse filter (DIN0 - DIN15) after the H373. Partial results of the processing (as described by Figures 4 and 5), are stored in memories FIFO1, FIFO2 (constituting the partial results), or they are sent to IFX2, which stores them in memory OUTRAM, so that they are sent as serial mode digital audio data to the output circuit (Figure 9) which consists of a conventional implementation of the converter CS4328 from Crystal.

According to a typical case of using the device for correction of the response of a specific audio system, consisting of an audio source (eg compact disk system CD), integrated amplifier and loudspeakers, the present device is connected between source (eg the digital output SPDIF to the digital filter input) and amplifier (from the analog digital filter output to the line level input of the amplifier), as is shown in Figure 2. Source, filter and amplifier are set to operation and the user, by the assistance of 4 buttons-indicators on the front of the filter selects the indications in the screen in order to find the loudspeaker type with which the device operates and the audio response which has to be corrected. This operation becomes possible by the circuit of Figures 7 and 10. By pressing the activation command key of the filter for this type of loudspeakers, the appropriate coefficients are transferred from the memory (eg PCMCIA card) to the main filter unit (Figures 7, 10, 11), and then any audio signal passing through the unit (through the signal input circuit of Figures 6 and 10) undrgoes the processing of digital filtering, described by the difference equation (Figure 11). The corrected signal of the output of the device (Figure 9) is transmitted to amplifier and loudspeakers in the usual manner. Similarly, the user is possible to select alternative corrective functions, as for example in order to include certain characteristics for the correction of the acoustics of listening space. By this way, the reproduced audio signal is cleared from audio distortions of the loudspeakers and/or of the listening environment.

Claims

1. An audio signal processing system, consisting of electronic circuits of (a) input of digital signals, (b) finite impulse response digital filter, (c) digital screen, (d) data inputting from a computer, (e) digital microprocessor for operation control, (f) PCMCIA card digital data reader, (g) digital clock, (h) power supply for digital circuits, (i) power supply for analog circuits, characterised in that the digital filter (b), finite impulse response, processes the input signal by, on the one hand, coefficients pre-calculated according to a Least Squares algorithm with use of optimal time delay, where error minimisation is done according to the equation

$$I = \sum_{n=0}^{L-1} e^2(n) = \sum_{n=0}^{L-1} [\delta(n-k)-y(n)]^2,$$

and is realised by a gate-array, where at the filter output a digital electronic circuit is connected for data storage and delay, allowing the use of partial results of the digital filter and combining them with subsequent partial results, and on the other hand by coefficients, programmed through one or more of the circuits for data inputting from a computer (d), digital microprocessor for operation control (e) and PCMCIA card digital data reader (f).

2. A method for the realisation of electronic circuits of a finite impulse response digital filter for audio signal processing, characterised in that the processing of an input signal is done by coefficients pre-calculated according to a Least Squares algorithm with use of optimal time delay, where error minimisation is done according to the equation

$$I = \sum_{n=0}^{L-1} e^2(n) = \sum_{n=0}^{L-1} [\delta(n-k)-y(n)]^2.$$

3. The method of claim (2), characterised in that the filter is realised by a gate-array, where at the filter output a digital electronic circuit is connected for data storage and delay, allowing the use of partial results of the digital filter and combining them with subsequent partial results.

4. The method of claim (3), characterised in that the filter allows an input for coefficients regarding the known devices and systems for audio reproduction and the basic characteristics of various listening spaces, through an electronic circuit for PCMCIA cards digital data reader, appropriately programmed, the reading circuit being connected with a digital screen electronic circuit and a digital microprocessor electronic circuit for operation control, which provide a user with a possibility of selecting the desirable set of coefficients.

5. The method of claim (3), characterised in that the filter allows coefficient input through an electronic circuit for data inputting from a computer.

6. An audio signal equilisation device, consisting of electronic circuits of (a) input of digital signals, (b) finite impulse response digital filter, (c) digital screen, (d) data inputting from a computer, (e) digital microprocessor for operation control, (f) PCMCIA card digital data reader, (g) digital clock, (h) power supply for digital circuits, (i) power supply for analog circuits, characterised in that the digital filter (b), finite impulse response, processes the input signal by, on the one hand, coefficients pre-calculated according to a Least Squares algorithm with use of optimal time delay, where error minimisation is done according to the equation

$$I = \sum_{n=0}^{L-1} e^2(n) = \sum_{n=0}^{L-1} [\delta(n-k)-y(n)]^2,$$

and is realised by a gate-array, where at the filter output a digital electronic circuit is connected for data storage and delay, allowing the use of partial results of the digital filter and combining them with following partial results, and on the other hand by coefficients, programmed through one or more of the circuits for data inputting from a computer (d), digital microprocessor for operation control (e) and PCMCIA card digital data reader (f).

7. The device of claim (6), not including the electronic circuits for reading digital data from PCMCIA card (f), power supply for digital circuits (h) and power supply for analog circuits (i), characterised in that the operations of these circuits are provided by a permanent connection of the other circuits with a computer, under the form of a Personal Computer internal card.

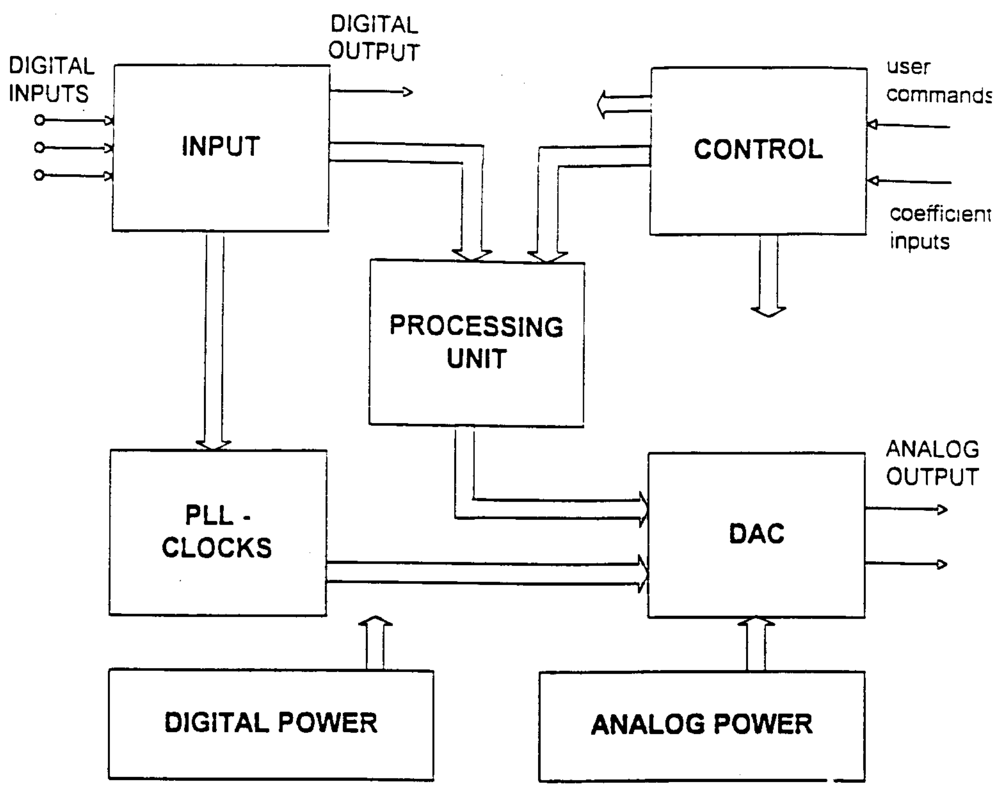

DIGITAL
INPUTS
DIGITAL
OUTPUT
INPUT
CONTROL
user
commands
coefficient
inputs
PROCESSING
UNIT
PLL -
CLOCKS
DAC
ANALOG
OUTPUT
DIGITAL POWER
ANALOG POWER

FIGURE 1

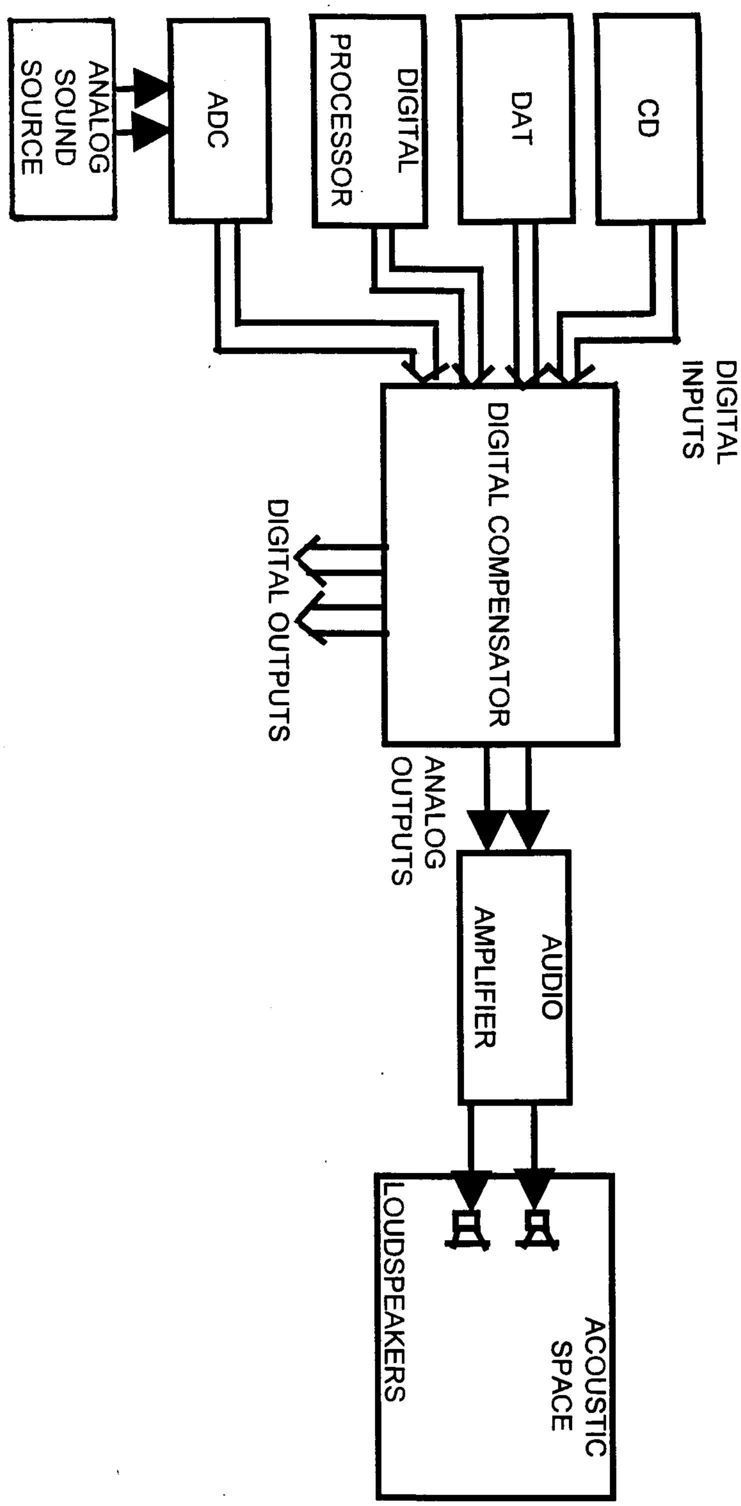

ANALOG
SOUND
SOURCE
ADC
DIGITAL
PROCESSOR
DAT
CD
DIGITAL
INPUTS
DIGITAL COMPENSATOR
DIGITAL OUTPUTS
ANALOG
OUTPUTS
AUDIO
AMPLIFIER
LOUDSPEAKERS
ACOUSTIC
SPACE

FIGURE 2

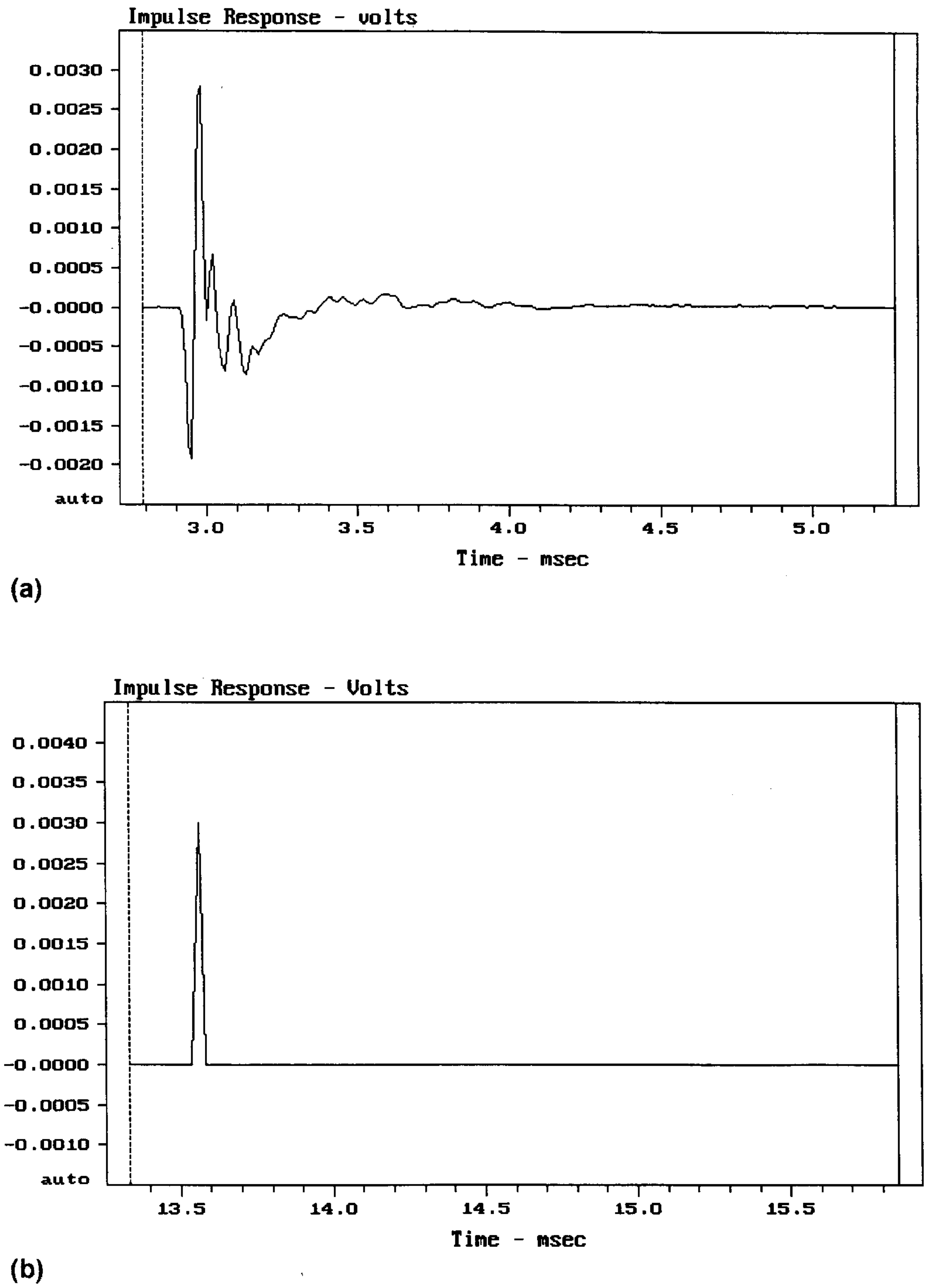
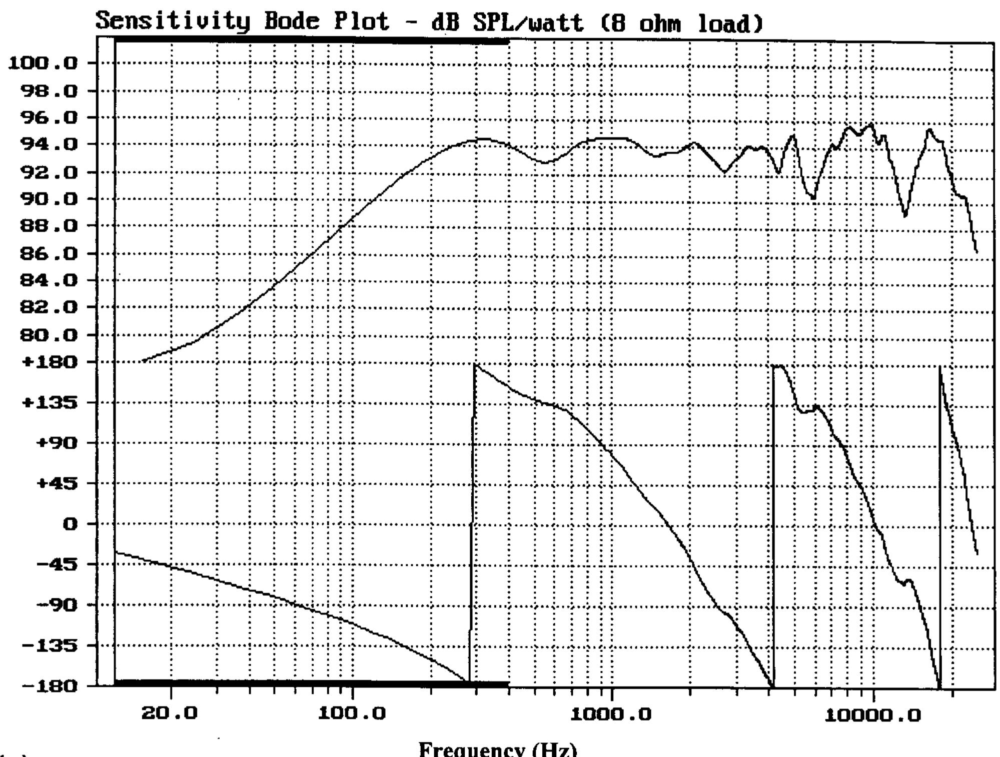
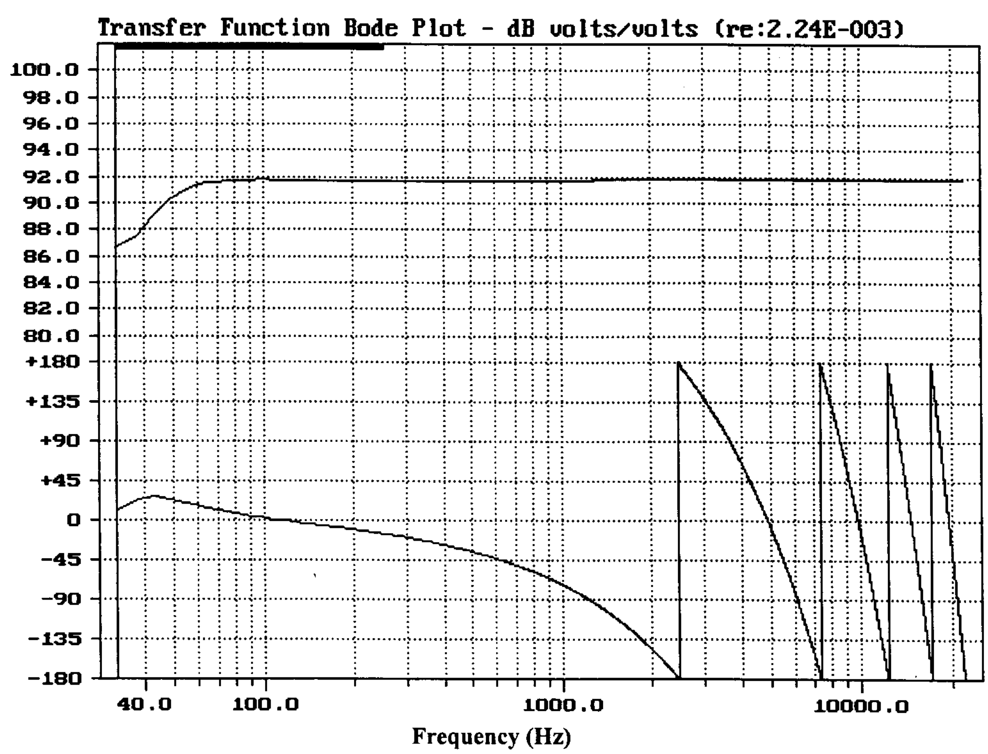

Impulse Response - volts
0.0030
0.0025
0.0020
0.0015
0.0010
0.0005
-0.0000
-0.0005
-0.0010
-0.0015
-0.0020
auto
3.0
3.5
4.0
4.5
5.0
Time - msec
(a)
Impulse Response - Volts
0.0040
0.0035
0.0030
0.0025
0.0020
0.0015
0.0010
0.0005
-0.0000
-0.0005
-0.0010
auto
13.5
14.0
14.5
15.0
15.5
Time - msec
(b)

FIGURE 3

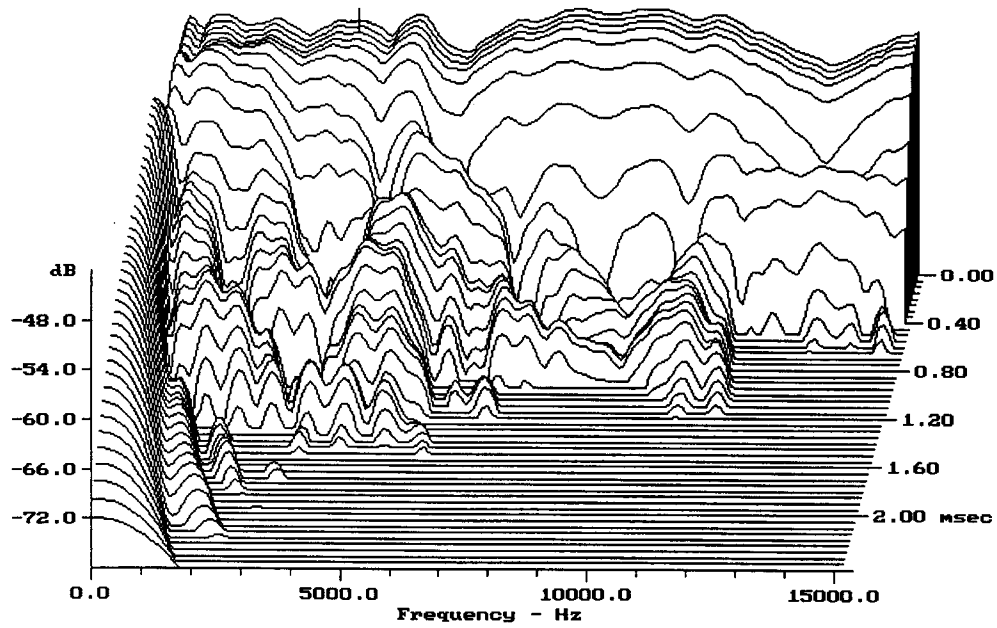

Sensitivity Bode Plot - dB SPL/watt (8 ohm load)
100.0
98.0
96.0
94.0
92.0
90.0
88.0
86.0
84.0
82.0
80.0
+180
+135
+90
+45
0
-45
-90
-135
-180
20.0
100.0
1000.0
10000.0
Frequency (Hz)

(c)

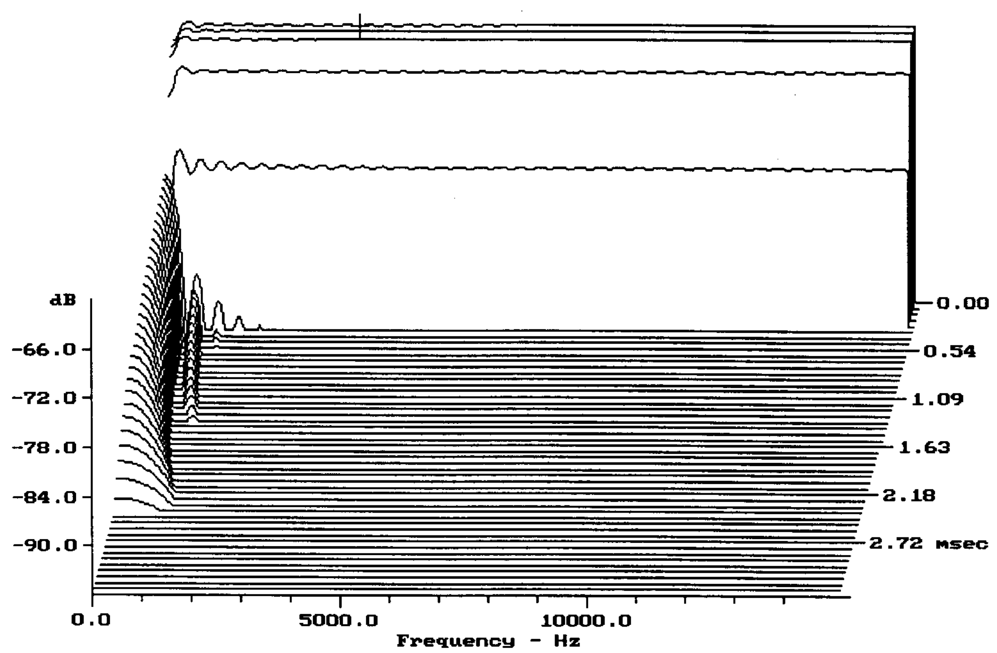

Transfer Function Bode Plot - dB volts/volts (re:2.24E-003)
100.0
98.0
96.0
94.0
92.0
90.0
88.0
86.0
84.0
82.0
80.0
+180
+135
+90
+45
0
-45
-90
-135
-180
40.0
100.0
1000.0
10000.0
Frequency (Hz)

(d)

FIGURE 3


dB
-48.0
-54.0
-60.0
-66.0
-72.0
0.00
0.40
0.80
1.20
1.60
2.00 msec
0.0
5000.0
10000.0
15000.0
Frequency - Hz

(e)


dB
-66.0
-72.0
-78.0
-84.0
-90.0
0.00
0.54
1.09
1.63
2.18
2.72 msec
0.0
5000.0
10000.0
Frequency - Hz

(f)

FIGURE 3

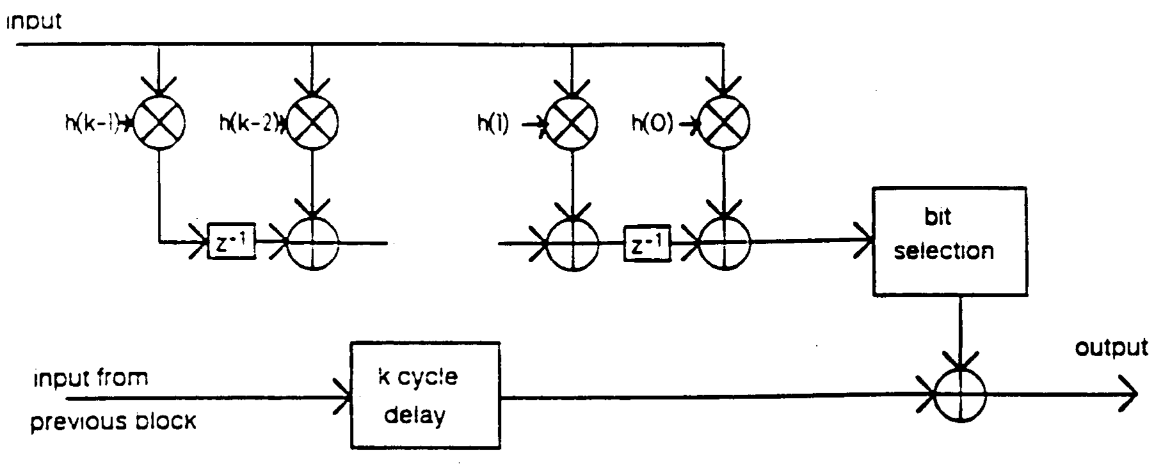

input
h(k-1)
h(k-2)
h(1)
h(0)
z^{-1}
z^{-1}
bit selection
output
input from previous block
k cycle delay

FIGURE 4

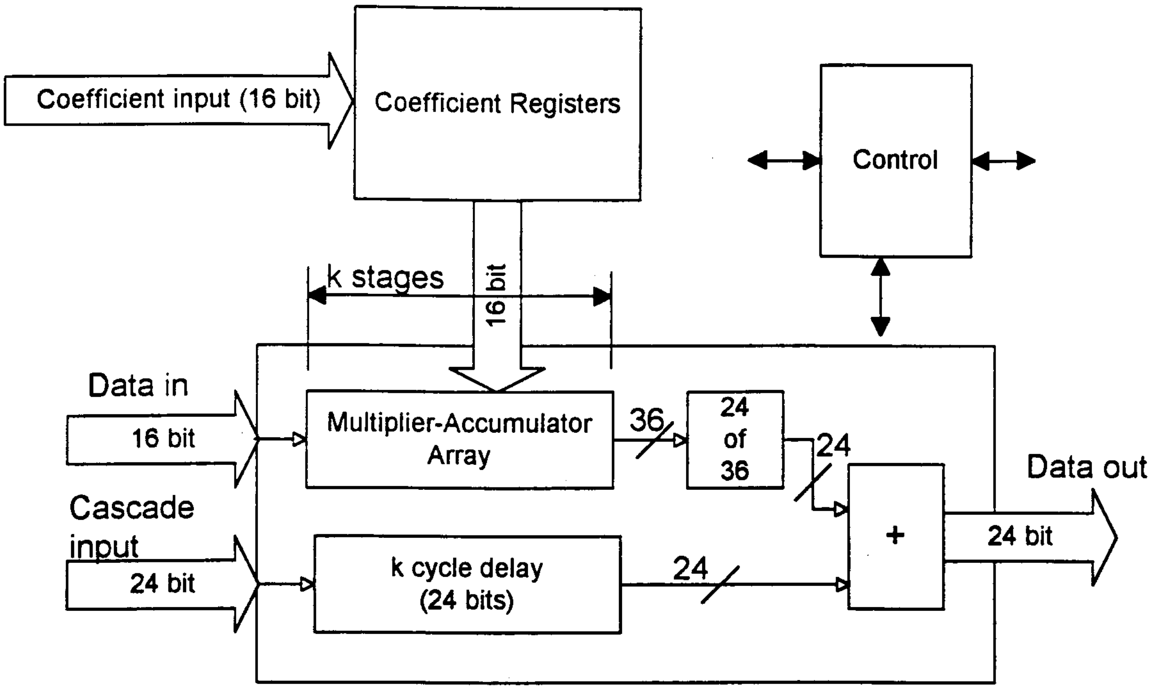

Coefficient input (16 bit)
Coefficient Registers
Control
k stages
16 bit
Data in
16 bit
Multiplier-Accumulator Array
36
24 of 36
24
Data out
Cascade input
24 bit
k cycle delay (24 bits)
24
+
24 bit

FIGURE 5

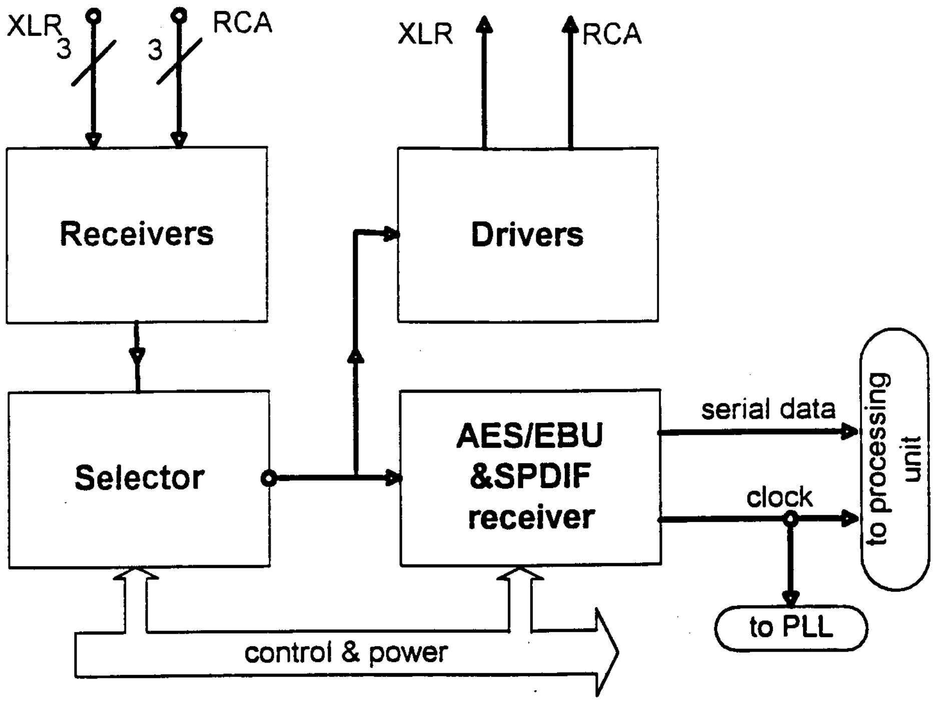

XLR
3
3
RCA
XLR
RCA
Receivers
Drivers
Selector
AES/EBU
&SPDIF
receiver
serial data
clock
to processing
unit
to PLL
control & power

FIGURE 6

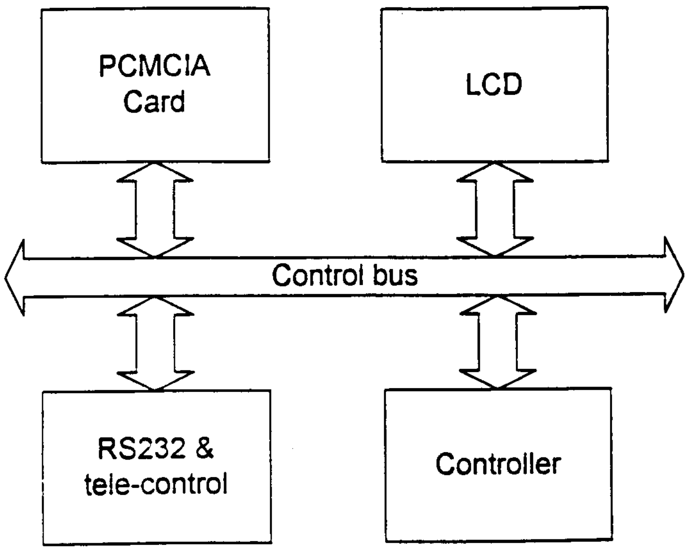


FIGURE 7

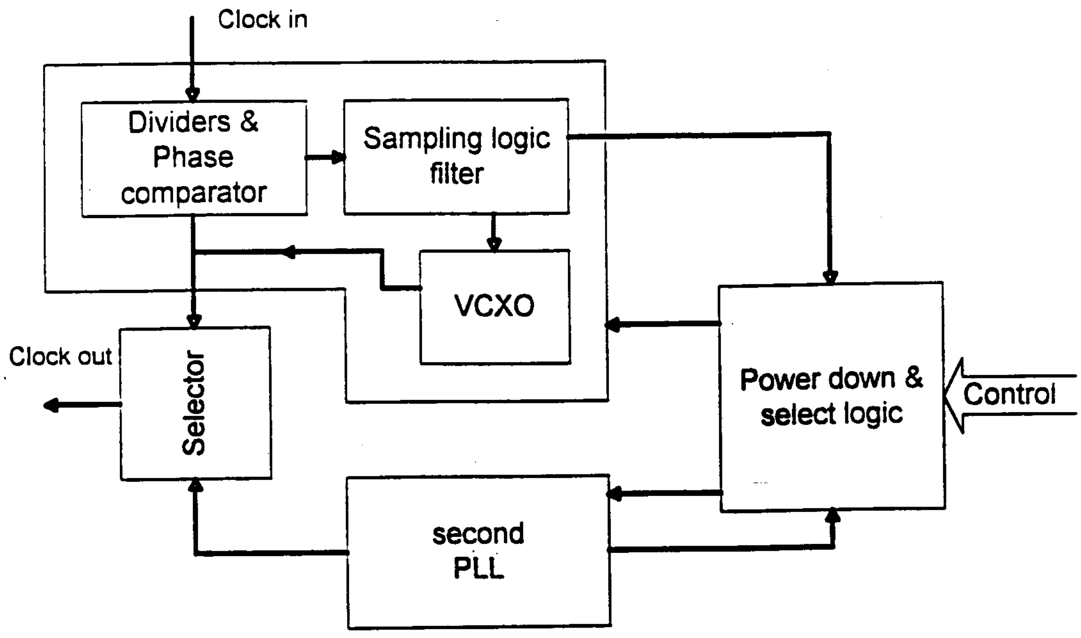

Clock in
Dividers &
Phase
comparator
Sampling logic
filter
VCXO
Clock out
Selector
Power down &
select logic
Control
second
PLL

FIGURE 8

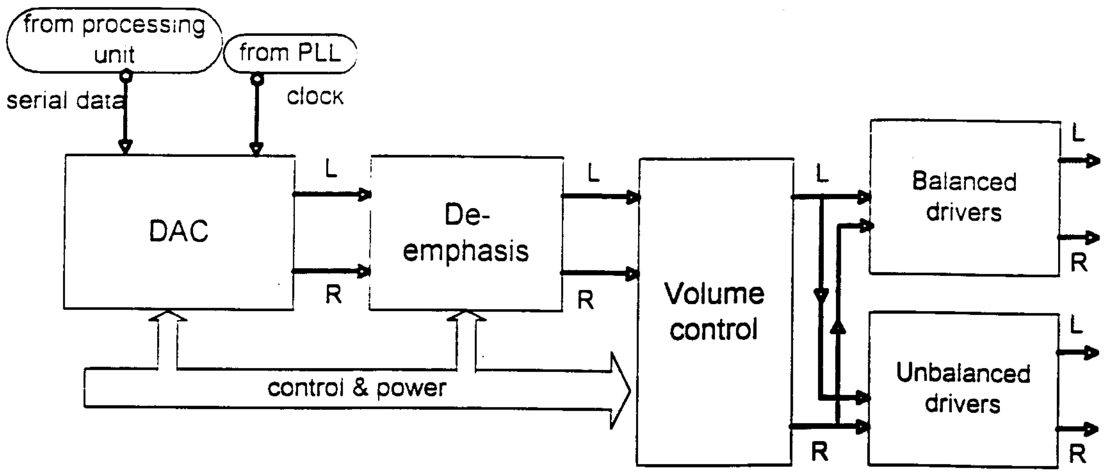

from processing unit
from PLL
serial data
clock
DAC
L
R
De-emphasis
L
R
Volume control
L
R
Balanced drivers
L
R
Unbalanced drivers
L
R
control & power

FIGURE 9

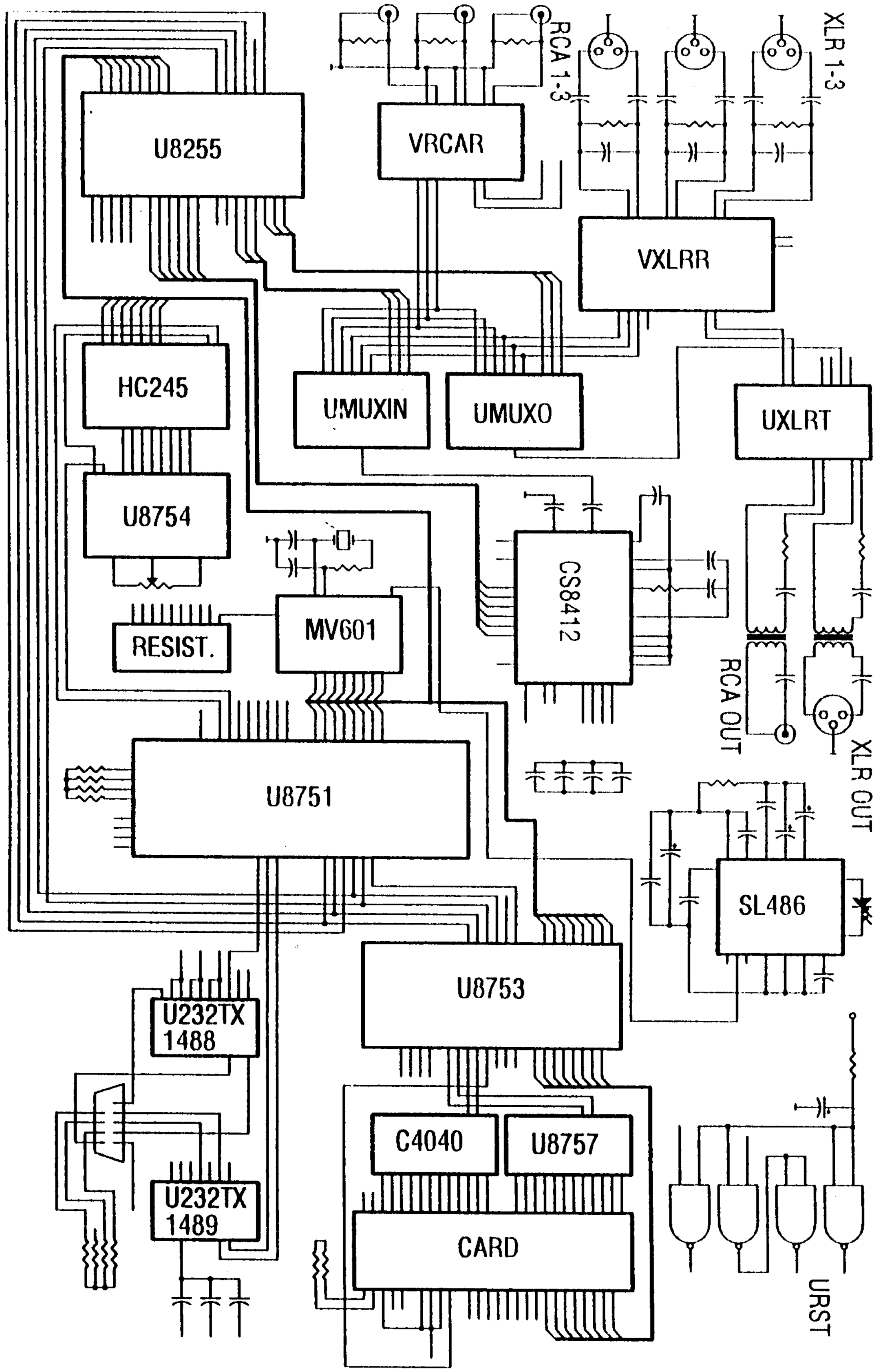

U8255
VRCAR
RCA 1-3
XLR 1-3
VXLRR
HC245
UMUXIN
UMUXO
UXLRT
U8754
CS8412
RESIST.
MV601
RCA OUT
XLR OUT
U8751
SL486
U232TX
1488
U8753
C4040
U8757
U232TX
1489
CARD
URST

FIGURE 10

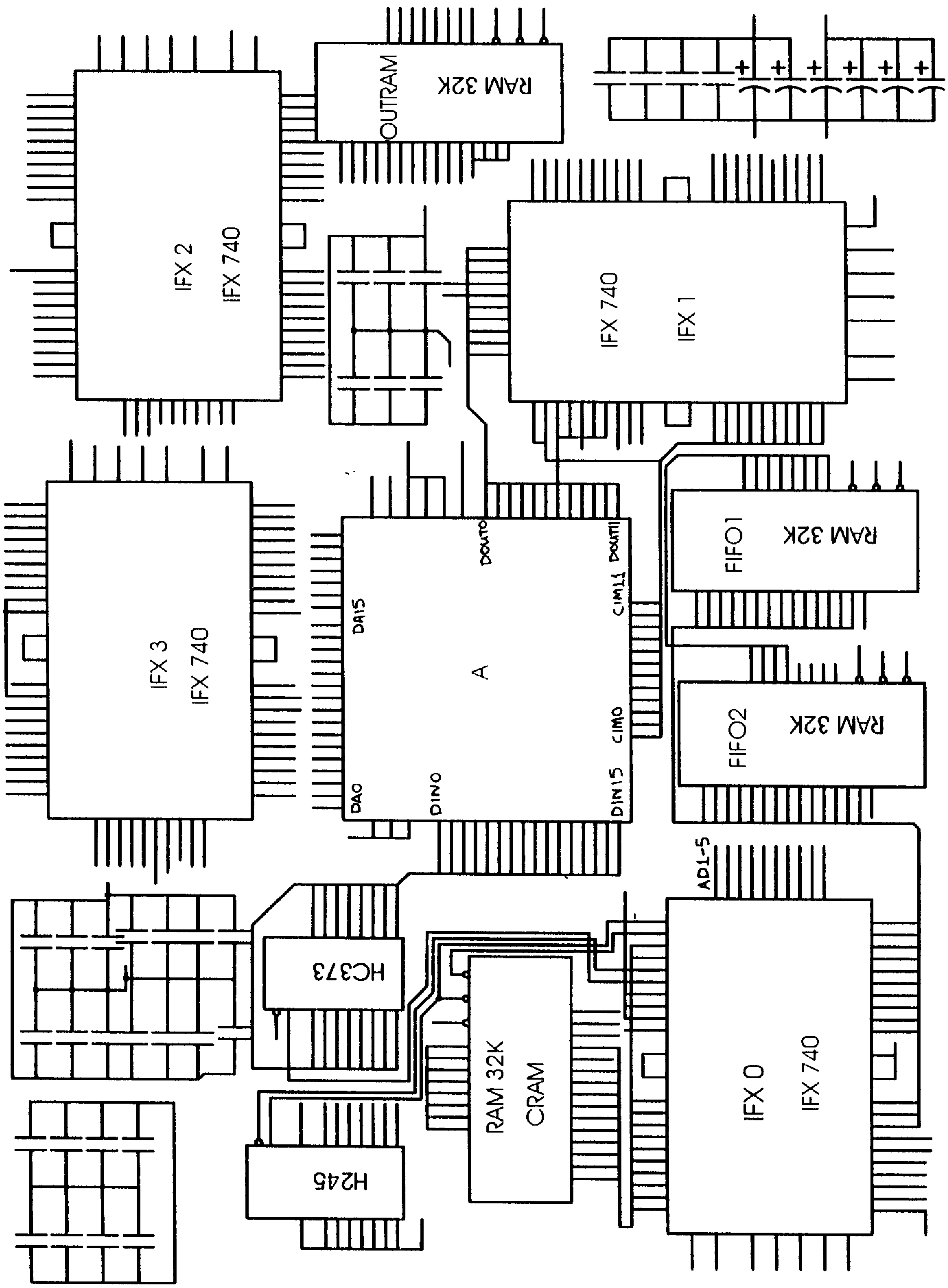

IFX 2
IFX 740
OUTRAM
RAM 32K
IFX 740
IFX 1
IFX 3
IFX 740
A
DA15
DOUT0
DOUT11
CIM11
CIM0
DA0
DIN0
DIN15
FIFO1
RAM 32K
FIFO2
RAM 32K
AD1-5
HC373
RAM 32K
CRAM
IFX 0
IFX 740
H245

FIGURE 11

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 96 60 0010

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
A	US 4 947 362 A (BUI) 7 August 1990 * column 1, line 6-12 * * column 1, line 57 - column 5, line 4 * * column 5, line 17-23 * * column 6, line 28 - column 10, line 56 * ---	1-3,6	H04S1/00
A	GB 2 199 216 A (BRITISH TELECOM) 29 June 1988 * page 2, line 4 - page 3, line 9 * * page 3, line 28 - page 5, line 5 * * page 11, line 1 - page 13, line 13 * ---	1-3,6	
A	SMPTE JOURNAL, vol. 103, no. 11, November 1994, WHITE PLAINS N.Y. US, pages 734-740, XP000475179 STANOJEVIC ET AL.: "THE TOTAL SURROUND SOUND (TSS) PROCESSOR." * page 735, line 77 - page 738, line 180 * ---	1,6	
A	WO 95 13688 A (SPARKOMATIC) 18 May 1995 * page 2, line 13 - page 3, line 8 * * page 5, line 13 - page 10, line 2 * -----	1,4-7	TECHNICAL FIELDS SEARCHED (Int.Cl.6) H04S H04R H03H G06F H04L

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	24 March 1997	Zanti, P

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
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P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

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