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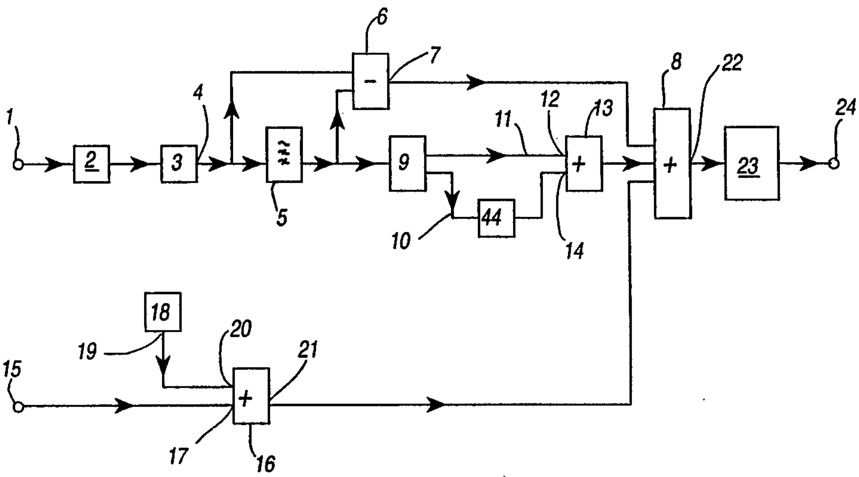

(57) Abstract

A sound signal applied to an input (1) is digitized in a sampler (2) and floating-point-compressed in a compressor (3). The result is divided into a first signal comprising the spectral component up to a quarter of the sampling frequency and a second signal comprising the spectral component between a quarter and a half the sampling frequency by means of a filter (5) and a subtracter (6), the first signal being applied directly to an adder (8). The second signal is applied to the adder (8) after having the least significant bit of each of its constituent samples stripped in a bit stripper (9). A data signal applied to a second input (15) is encoded by means of an adder (16) and a carrier signal generator (18) into a bit stream having the form of a carrier signal modulated by the data, the carrier signal having half the sample frequency. This bit stream is also applied to the adder so that these bits effectively replace the least significant bits stripped from the second signal. More than one of the least significant bits of the second signal may be replaced in this way. The output signal of the adder constitutes the signal for transmission, the data component thereof being inaudible when the sound signal component is reproduced.

SIMULTANEOUS TRANSMISSION OF AUDIO AND DATA SIGNALS

This invention relates to a method of transmitting an audio signal and data simultaneously over a single transmission channel, in which method the data is encoded to form an auxiliary signal occupying a predetermined portion of the audible spectrum and the audio and auxiliary signals are combined for transmission. The invention also relates to a transmitter for implementing such a method and to a receiver for data transmitted by such a transmitter.

Such a method is known from US-A-4238849. In this known method the data is used to modulate one or more pilot signals the frequencies of which lie within the audible range but above the frequency range of the transmitted audio signal. The spectral energy of an incoming sound signal is analysed to determine a threshold of audibility for the pilot signals and their amplitudes are continuously adjusted to lie below this threshold. The transmission rate for the data is constant except when the threshold is determined to be too low to permit satisfactory transmission at this rate; when such a situation occurs the data transmission is interrupted.

It is an object of the invention to enable data to be transmitted even when the spectral energy of the incoming sound signal is substantially zero.

To this end, according to one aspect of the invention a method as defined in the first paragraph is characterized in that the audio signal is derived from an incoming sound signal by digitization into a series of multi-bit samples which are compressed according to a floating point technique, in that the auxiliary signal is a bit stream having the form of a carrier signal modulated by the data, the frequency of the carrier signal being one half the sample frequency, and in that the successive bits of the bit stream are substituted for the least significant bits of the frequency component of the audio signal which lies within a frequency range extending from a frequency equal to one half the sample frequency to a given lower frequency.

Because the audio signal is derived from an incoming sound signal by digitization into a series of multibit samples which are compressed according to a floating point technique, for example according to the NICAM standard (see, for example, "Television" March 1988 pages 368 - 369, particularly Fig. 4 on page 368), and the successive bits of the bit stream are substituted for the least significant bits of a frequency component of this audio signal, the effective level of the auxiliary signal is automatically continuously adapted to the level of the sound signal represented by the audio signal. Furthermore, choosing the said frequency component to be that which lies within a frequency range extending from a frequency equal to one half the sample frequency to a given lower frequency, i.e. to be at the top of the audible frequency range, and arranging that the auxiliary signal is a bit stream having the form of a carrier signal modulated by the data, the frequency of the carrier signal being one half the sample frequency, allows use to be made of the fact that the threshold of audibility rises steeply at the top of the audible frequency range; it can be arranged that the effective level of the auxiliary signal remains below the threshold of audibility, allowing data to continue to be transmitted, even when the spectral energy of the incoming sound signal is substantially zero (silence).

More than one of the least significant bits of the (samples of) said frequency component may be substituted by successive bits of respective bit streams if desired. Thus, for example, further data may be transmitted simultaneously with the audio signal over the transmission path, the further data being encoded to form a second auxiliary signal, the second auxiliary signal being a second bit stream having the form of a second carrier signal modulated by the further data, the frequency of the second carrier signal being one half the sample frequency, the successive bits of the second bit stream being substituted for the second-to-least-significant bits of said frequency component of the audio signal.

Similarly, still further data may be also transmitted simultaneously with the audio signal over the transmission path, the still further data being encoded to form a third auxiliary signal, the third auxiliary signal being a third bit stream having the form of a third carrier signal modulated by the still further data, the frequency of the third carrier signal being one half the sample frequency, the successive bits of the third bit stream being substituted for the third-to-least-significant bits of said frequency component of the audio signal.

When the incoming sound signal has been coded according to the NICAM standard it has been found that up to six of the least significant bits of each sample of the said frequency component may be substituted in this way without any substantial audible effect, provided that the data rate carried by those of the least significant bits which have the greatest significance is not to high (which would result in the sidebands of the relatively high level relevant auxiliary signal extending downwards to such an extent as to cross the audible threshold at these lower frequencies).

Preferably said frequency component is removed from said series of multibit samples by filtering, the successive bits of the or each bit stream are then substituted for the bits of the relevant significance of said frequency components, and the result is added back to the series of multibit samples from which said frequency component has been removed.

According to another aspect the invention provides a transmitter for transmitting an audio signal and data simultaneously over a single transmission channel, comprising an encoder for encoding the data to form an auxiliary signal occupying a predetermined portion of the audible spectrum and a signal combiner for combining the audio and auxiliary signals for transmission, characterized in that compression means are provided for compressing an incoming digitally sampled sound signal according to a floating point technique to produce the audio signal, the encoder is arranged to generate the auxiliary signal as a bit stream having the form of a carrier signal modulated by the data, the frequency of the carrier signal being one half the sample frequency, and substitution means are provided for substituting the successive bits of the bit stream for the least significant bits of the frequency component of the audio signal which lies within a frequency range extending from a frequency equal to one half the sample frequency to a given lower frequency.

According to yet another aspect the invention provides a receiver for data transmitted simultaneously with an audio signal over a single transmission path by such a transmitter, comprising a synchronous demodulator for said auxiliary signal, and means for deriving from the transmitted combined audio and auxiliary signals that frequency component thereof which lies within said frequency range and applying the least significant bits of the derived frequency component to said synchronous demodulator.

An embodiment of the invention will now be described, by way of example, with reference to the accompanying diagrammatic drawings in which

Fig. 1 is a block diagram of part of a transmitter for transmitting an audio signal and data simultaneously over a single transmission channel, and

Fig. 2 is a block diagram of part of a receiver for a signal transmitted by the transmitter of Fig. 1.

In Fig. 1 an incoming sound signal, for example the left or right channel of a stereo pair, is received at an input 1. This signal is digitized in a sampler 2 into a series of multibit samples which are compressed in a compressor 3 according to a floating point technique, in the present case in accordance with the NICAM standard. The effect of the compression at 3 is that large values of the samples are produced at the output 4 of compressor 3 with relatively coarse resolution whereas small values of the samples are produced with relatively fine resolution.

The audio signal comprising the series of samples produced at the output 4 is applied both to a high-pass filter 5 and to a digital subtractor 6. The high-pass filter 5 has a cut-off frequency equal to one quarter the frequency of the samples occurring at the output 4, the latter frequency being in the present example 32KHz. Thus the output signal of the filter 5 consists of the frequency component of the audio signal appearing at the output 4 which lies within the frequency range 8KHz to 16KHz (the sampler 2 incorporating an anti-alias low-pass filter having a cut-off frequency equal to half the sample frequency, i.e. to 16KHz).

The output signal of filter 5 is applied to the subtractor 6 which therefore produces at its output 7 the frequency component of the audio signal which lies below 8KHz. This frequency component is applied to one input of a digital adder 8.

The samples appearing at the output of filter 5, which sample, comprise the frequency component of the audio signal which lies above 8KHz, are also applied to a bit stripper 9, which diverts the least significant bits thereof onto a path 10 and the remaining bits of these samples onto a path 11 which leads to one input 12 of a digital adder 13. The path 10 leads to an accumulator 44 which accumulates the successive stripped least significant bits and, each time the accumulated value becomes equal to the value of the least significant bit of the samples carried on the path 11, applies this value to a second input 14 of adder 13. This value is therefore added to the sample currently applied to the input 12 of adder 13 to compensate for d.c. level changes which occur in the samples carried by the path 11 due to the stripping of the least significant bits in stripper 9. The resulting compensated samples appearing at the output of adder 13 are applied to a second input of adder 8.

Data to be transmitted simultaneously with the audio signal is applied serially to an input 15 and thence to a first input 17 of an adder 16 which may take the form of an Exclusive-OR gate. A signal generator 18 has its output 19 connected to a second input 20 of the adder 16.

Generator 18 generates a binary signal the fundamental frequency of which is equal to half the sample frequency, i.e. to 16KHz in the present example. In the simplest case generator 18 may comprise a simple clock signal generator having a frequency equal to 16KHz although in practice some spectral spreading of the output signal thereof is preferred, in which case generator 18 may comprise a maximum length sequence generator (known per se) in the form of a shift register having its output ExORed with the output of another one of its stages and the result fed back to its input, the shift register being clocked at the sample frequency, i.e. at 32KHz in the present example.

The result of the addition in the adder 16 is that an auxiliary signal appears on the adder output 21, this auxiliary signal being a bit stream having the form of a carrier signal modulated by the data applied to the input 15, this carrier signal having a frequency equal to one half the sample frequency, i.e. to 16KHz in the present example. The bit stream appearing at the output 21 is applied to a third input of the adder 8.

The signal appearing at the output 22 of adder 8 therefore consists of a succession of multibit samples, these samples carrying at full resolution the frequency component of the audio signal appearing at the output 4 of compressor 3 which lies below 8KHz, (derived from the output 7 of subtractor 6) and, with one bit less resolution, the frequency component which lies above 8KHz (derived from the adder 13), the least significant bits of the latter component being substituted by the successive bits of the auxiliary signal produced at the output 21 of adder 16.

The signal appearing at the output 22 may be further processed at 23 for transmission, for example scrambled according to another aspect of the NICAM standard, the result appearing at an output 24.

Many of the operations occurring in the receiver of Fig. 2 are similar to, or the inverse of, operations occurring in the transmitter of Fig. 1.

The signal transmitted from the output 24 of the transmitter of Fig. 1 is received at 25 and processed at 26 in a manner which is the inverse of that occurring in the processor 23 of Fig. 1. The resulting samples are applied both to one input of a subtractor 27 and to a high pass filter 28 which has the same characteristic as the filter 5 of Fig. 1. The output signal of filter 28 is applied both to a second input of subtractor 27 and to a bit stripper 29. The output signal of the subtractor 27 therefore consists of a sequence of multibit samples representing the frequency component of the audio signal lying below 8KHz. These samples are applied to one input of an adder 30.

The bit stripper 29 diverts the least significant bits of the samples appearing at the output of filter 28 onto a path 31 and supplies the remaining bits of these samples to a path 32. The samples on the path 32 are relatively coarsely quantized representations of the frequency component of the audio signal lying above 8KHz and are added in the adder 30 to the samples produced by subtractor 27, i.e. to the samples representing the frequency component below 8KHz. The output samples of adder 30 are expanded in an expander 33 which operates in an inverse manner to the compressor 3 of Fig.1. The expanded samples produced by expander 33 are converted to analogue form in a D/A converter 34 and applied to an output 35 for reproduction as sound.

The least significant bits diverted onto the path 31 are applied to one input of an adder 36 which operates in the same way as the adder 16 of Fig. 1. A second input of the adder 36 is fed from the output of a signal generator 37 which is arranged to generate an output signal which is identical to that generated by the generator 18 of Fig. 1 and is synchronised therewith; synchronism may be achieved by means of a correlator fed with the output of the generator 37 and the bit stream occurring on the path 31 in a manner which is conventional where spread spectrum techniques are employed. Adder 36 in consequence therefore operates as a synchronous demodulator for the auxiliary signal carried by the path 31, supplying a succession of data bits identical to those applied to input 15 of Fig. 1 to an output 38,

As mentioned previously, more than one of the least significant bits of each of the output samples of filter 5 may be replaced by a bit of a respective auxiliary signal if desired, in which case the stripper 9 of Fig. 1 will have to be arranged to strip the appropriate number of bits and the adder 16 together with its data input be duplicated an appropriate number of times, each of the resulting adders then being arranged to generate a stream of bits of the respective significance. If this is the case the data rates carried by the bits of the various significances are preferably lower the greater the significance is, in order to shape the spectrum of the auxiliary signal in such a way that it has a relatively wide bandwidth for auxiliary signals having a relatively low effective amplitude and a relatively narrow bandwidth for auxiliary signals having a relatively high effective amplitude. Of course, similar modifications will then also have to be made to the receiver of Fig. 2, i.e. the stripper 29 will have to be arranged to strip the appropriate number of bits and the adder 36 will have to be duplicated an appropriate number of times, each of the resulting adders being supplied with bits of a respective significance by the stripper 29.

As so far described the data rate transmitted is constant. However, this is not necessarily the case; if desired the data rate can be adaptively adjusted in accordance with what will be audibly masked by the spectral energy in the audio signal. Such adaptation is known per se, for example from conference publication Proc. ICASSP90, Alberquerque, New Mexico, April 3 - 6 1990, pages 1097 - 1100, and may be achieved in the transmitter of Fig. 1 by analysing the spectrum of the output signals of compressor 3 and adjusting both the number of bits which are stripped by stripper 9 and the number of adders similar to 16 which are employed in accordance with the result so that more bits are stripped and more adders are employed the greater the spectral energy which is currently present to mask the auxiliary signals. Similar spectral analysis means will then have to be provided in the receiver of Fig. 2 operating on the output signal of processor 28 and controlling the number of bits stripped by stripper 29 and the number of adders similar to adder 36 employed.

It will be evident that many modifications may be made to the embodiment described within the scope of the invention as defined by the claims. For example the generators 18 and 37 may be constituted by look-up tables which store the required bit sequence and which are read out as required. If these generators are simply clock pulse generators operating at half the sample frequency and the adders 16 and 36 are EXOR gates the data applied to the input 15 is preferably Manchester bi-phase coded to give an ancillary signal having an appropriate spectral characteristic.

As an alternative to adding the data stream to the output signal of generator 18 one or more data bits may simply be arithmetically added to alternate ones of the output samples of adder 13 and arithmetically subtracted from the other alternate ones of these samples.

If desired the accumulator 44 and adder 13 of Fig. 1 may be omitted although, as mentioned previously, this may give rise to problems in respect of the average d.c. level of the bit-stripped samples.

CLAIMS

1. A method of transmitting an audio signal and data simultaneously over a single transmission channel, in which method the data is encoded to form an auxiliary signal occupying a predetermined portion of the audible spectrum and the audio and auxiliary signals are combined for transmission, characterized in that the audio signal is derived from an incoming sound signal by digitization into a series of multibit samples which are compressed according to a floating point technique, in that the auxiliary signal is a bit stream having the form of a carrier signal modulated by the data, the frequency of the carrier signal being one half the sample frequency, and in that the successive bits of the bit stream are substituted for the least significant bits of the frequency component of the audio signal which lies within a frequency range extending from a frequency equal to one half the sample frequency to a given lower frequency.

2. A method as claimed in Claim 1, characterized in that further data is also transmitted simultaneously with the audio signal over the transmission path, the further data being encoded to form a second auxiliary signal, the second auxiliary signal being a second bit stream having the form of a second carrier signal modulated by the further data, the frequency of the second carrier signal being one half the sample frequency, the successive bits of the second bit stream being substituted for the second-to-least-significant bits of said frequency component of the audio signal.

3. A method as claimed in Claim 2, characterized in that still further data is also transmitted simultaneously with the audio signal over the transmission path, the still further data being encoded to form a third auxiliary signal, the third auxiliary signal being a third bit stream having the form of a third carrier signal modulated by the still further data, the frequency of the third carrier signal being one half the sample frequency, the successive bits of the third bit stream

being substituted for the third-to-least-significant bits of said frequency component of the audio signal.

4. A method as claimed in any preceding claim, characterized in that said frequency component is removed from said series of multibit samples by filtering, the successive bits of the or each bit stream are then substituted for the bits of the relevant significance of said frequency components, and the result is added back to the series of multibit samples from which said frequency component has been removed.

5. A method as claimed in Claim 1, characterized in that the given lower frequency is equal to one quarter the sample frequency.

6. A transmitter for transmitting an audio signal and data simultaneously over a single transmission channel, comprising an encoder for encoding the data to form an auxiliary signal occupying a predetermined portion of the audible spectrum and a signal combiner for combining the audio and auxiliary signals for transmission, characterized in that compression means are provided for compressing an incoming digitally sampled sound signal according to a floating point technique to produce the audio signal, the encoder is arranged to generate the auxiliary signal as a bit stream having the form of a carrier signal modulated by the data, the frequency of the carrier signal being one half the sample frequency, and substitution means are provided for substituting the successive bits of the bit stream for the least significant bits of the frequency component of the audio signal which lies within a frequency range extending from a frequency equal to one half the sample frequency to a given lower frequency.

7. A receiver for data transmitted simultaneously with an audio signal over a single transmission channel by a transmitter as claimed in Claim 6, comprising a synchronous demodulator for said auxiliary signal, and means for deriving from the transmitted combined audio and auxiliary signals that frequency component thereof which lies within said frequency

range and applying the least significant bits of the derived frequency component to said synchronous demodulator.

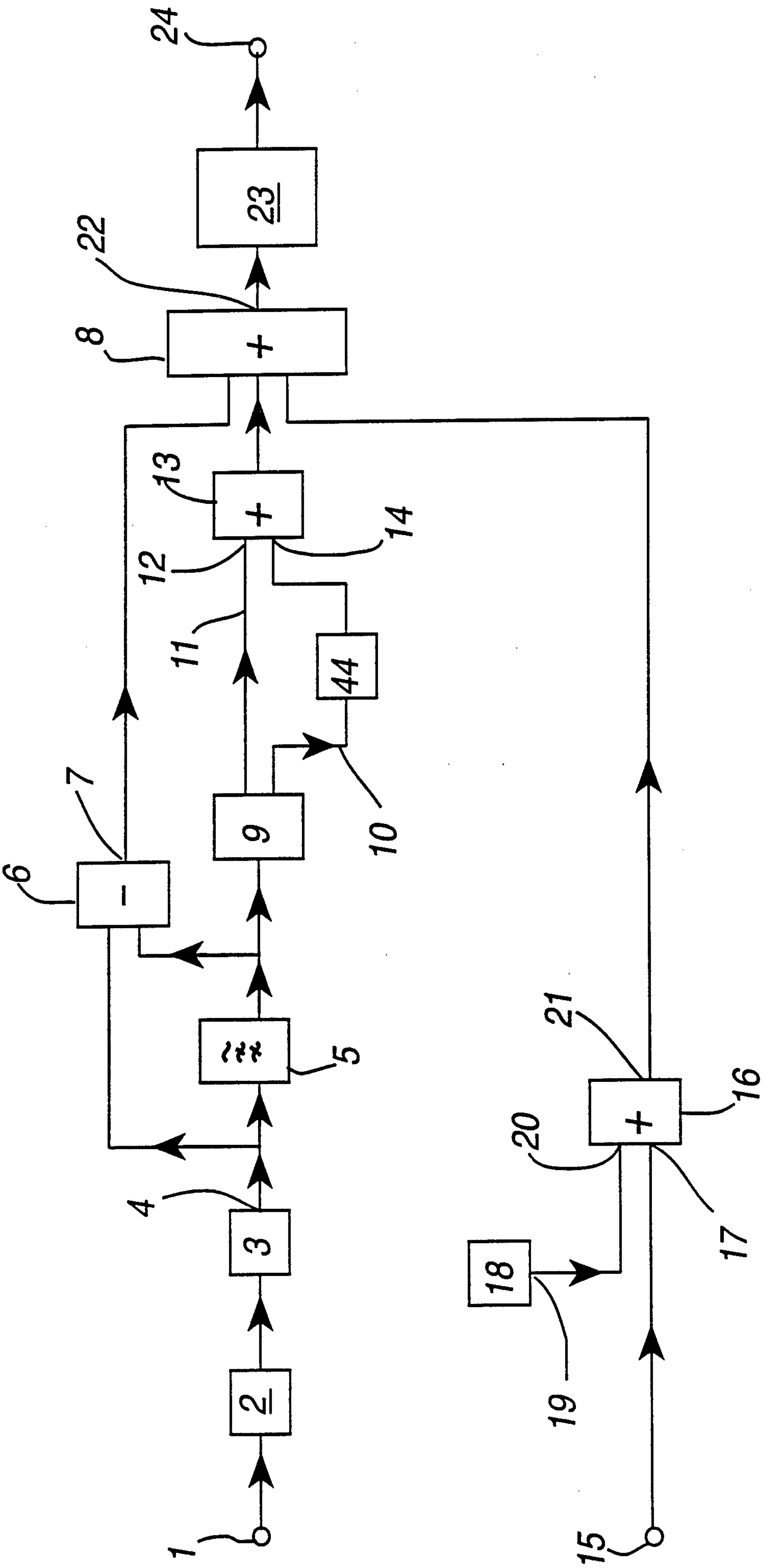

Fig.1.

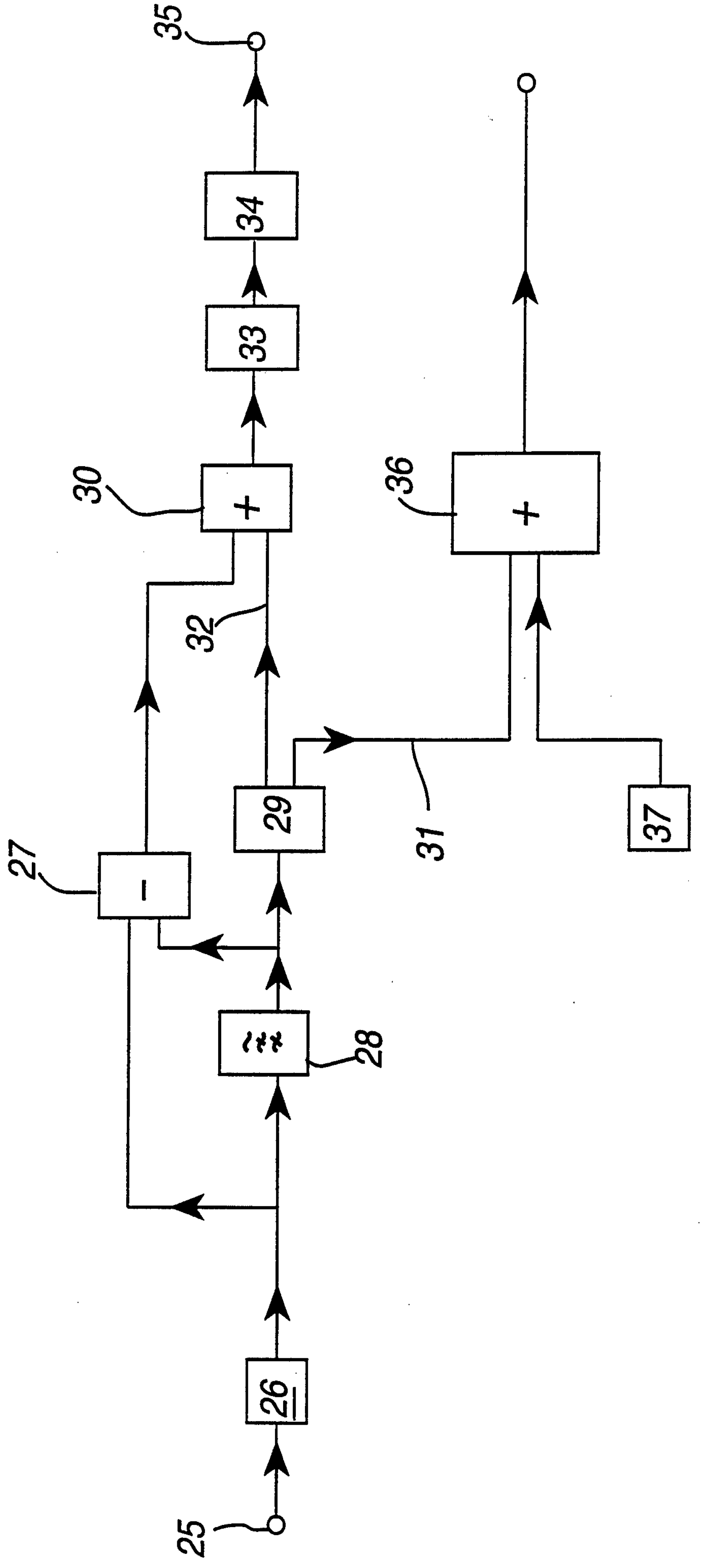

Fig.2.

INTERNATIONAL SEARCH REPORT

International Application No
PCT/GB 93/02261

A. CLASSIFICATION OF SUBJECT MATTER
IPC 5 H04H1/00 H04J3/12 H03G7/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 5 H04H H04J H03G

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	RUNDFUNKTECHNISCHE MITTEILUNGEN vol. 30, no. 4 , July 1986 , NORDERSTEDT DE pages 158 - 167 PLENGE, SPIKOFSKI, THEILE 'VARIABELE DYNAMIK- EIN KONZEPT FÜR VERBESSERTE VERSORGUNG IM HöRFUNK UND FERNSEHEN' see paragraph 2 ---	1,6
A	WO,A,92 00637 (TELEFUNKEN FERNSEH UND RUNDFUNK G.M.B.H.) 9 January 1992 see page 1, line 1 - page 4, line 5; claim 1; figures 1-4 --- -/--	1,6

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance
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"&" document member of the same patent family

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Date of the actual completion of the international search	Date of mailing of the international search report
24 February 1994	17. 03. 94

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	De Haan, A

Inter... ...onal Application No
PCT/GB 93/02261

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP,A,0 359 325 (N.V. PHILIPS GLOEILAMPENFABRIEKEN) 21 March 1990 see column 1, line 1 - column 2, line 43; claims 1,3,4,8,9 see column 3, line 39 - column 4, line 29 ---	1,6
A	DE,A,35 23 809 (POLYGRAM G.M.B.H.) 27 November 1986 see page 4, line 10 - page 6, line 29; claims 1,2,6 ---	1,6
A	EP,A,0 102 608 (HITACHI) 14 March 1984 see page 1, line 1 - page 5, line 9; claims 1-4 -----	1,6

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INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/GB 93/02261

Patent document cited in search report	Publication date	Patent family member(s)		Publication date
WO-A-9200637	09-01-92	DE-A-	4020932	09-01-92
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		CA-A-	2084791	31-12-91
		EP-A-	0538289	28-04-93
EP-A-0359325	21-03-90	NL-A-	8802291	17-04-90
		EP-A-	0545915	09-06-93
		JP-A-	2119446	07-05-90
		US-A-	5146457	08-09-92
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		EP-A,B	0205200	17-12-86
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		JP-A-	62026672	04-02-87
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