



Europäisches Patentamt
European Patent Office
Office européen des brevets



(11) **EP 0 852 052 B1**

(12) **EUROPEAN PATENT SPECIFICATION**

(45) Date of publication and mention
of the grant of the patent:
13.06.2001 Bulletin 2001/24

(21) Application number: **96931552.2**

(22) Date of filing: **13.09.1996**

(51) Int Cl.7: **G10L 21/02**

(86) International application number:
PCT/US96/14665

(87) International publication number:
WO 97/10586 (20.03.1997 Gazette 1997/13)

(54) **SYSTEM FOR ADAPTIVELY FILTERING AUDIO SIGNALS TO ENHANCE SPEECH
INTELLIGIBILITY IN NOISY ENVIRONMENTAL CONDITIONS**

SYSTEM ZUR ADAPTIVEN FILTERUNG VON TONSIGNALEN ZUR VERBESSERUNG DER
SPRACHVERSTÄNDLICHKEIT BEI UMGEBUNGSGERÄUSCHEN

SYSTEME DE FILTRAGE ADAPTATIF DE SIGNAUX AUDIO DESTINE A AMELIORER
L'INTELLIGIBILITE DE LA PAROLE DANS DES ENVIRONNEMENTS BRUYANTS

(84) Designated Contracting States:
BE DE DK ES FI FR GB GR IT NL PT SE

(30) Priority: **14.09.1995 US 528005**

(43) Date of publication of application:
08.07.1998 Bulletin 1998/28

(73) Proprietor: **ERICSSON INC.**
Research Triangle Park, NC 27709 (US)

(72) Inventor: **SÖLVE, Torbjörn, W.**
Cary, NC 27513 (US)

(74) Representative: **Dahnér, Christer**
Ericsson Mobile Communications AB,
Patent Unit
164 80 Stockholm (SE)

(56) References cited:

EP-A- 0 558 312	EP-A- 0 645 756
EP-A- 0 665 530	DE-A- 4 012 349
US-A- 4 461 025	US-A- 4 630 305
US-A- 4 811 404	US-A- 5 251 263

Note: Within nine months from the publication of the mention of the grant of the European patent, any person may give notice to the European Patent Office of opposition to the European patent granted. Notice of opposition shall be filed in a written reasoned statement. It shall not be deemed to have been filed until the opposition fee has been paid. (Art. 99(1) European Patent Convention).

Description

[0001] The present invention relates to noise reduction systems, and in particular, to an adaptive speech intelligibility enhancement system for use in portable digital radio telephones.

[0002] The cellular telephone industry has made phenomenal strides in commercial operations in the United States as well as the rest of the world. Demand for cellular services in major metropolitan areas is outstripping current system capacity. Assuming this trend continues, cellular telecommunications will reach even the smallest rural markets. Consequently, cellular capacity must be increased while maintaining high quality service at a reasonable cost. One important step towards increasing capacity is the conversion of cellular systems from analog to digital transmission. This conversion is also important because the first generation of personal communication networks (PCNs), employing low cost, pocket-size, cordless telephones that can be easily carried and used to make or receive calls in the home, office, street, car, etc., will likely be provided by cellular carriers using the next generation digital cellular infrastructure.

[0003] Digital communication systems take advantage of powerful digital signal processing techniques. Digital signal processing refers generally to mathematical and other manipulation of digitized signals. For example, after converting (digitizing) an analog signal into digital form, that digital signal may be filtered, amplified, and attenuated using simple mathematical routines in a digital signal processor (DSP). Typically, DSPs are manufactured as high speed integrated circuits so that data processing operations can be performed essentially in real time. DSPs may also be used to reduce the bit transmission rate of digitized speech which translates into reduced spectral occupancy of the transmitted radio signals and increased system capacity. For example, if speech signals are digitized using 14-bit linear Pulse Code Modulation (PCM) and sampled at an 8 KHz rate, a serial bit rate of 112 Kbits/sec is produced. Moreover, by taking mathematical advantage of redundancies and other predictable characteristics of human speech, voice coding techniques can be used to compress the serial bit rate from 112 Kbits/sec to 7.95 Kbits/sec to achieve a 14:1 reduction in bit transmission rate. Reduced transmission rates translate into more available bandwidth.

[0004] One popular speech compression technique adopted in the United States by the TIA for use as the digital standard for the second generation of cellular telephone systems (i.e., IS-54) is vector sourcebook excited linear predictive coding (VSELP). Unfortunately, when audio signals including speech, mixed with high levels of ambient noise (particularly "colored noise"), are coded/compressed using VSELP, undesirable audio signal characteristics may be part of the result. For example, if a digital mobile telephone is used in a noisy environment (e.g. inside a moving automobile), both ambient noise and desired speech are compressed using the VSELP encoding algorithm and transmitted to a base station where the compressed signal is decoded and reconstituted into audible speech. When the background noise is reconstituted into an analog format, undesirable, audible distortion of the noise, and occasionally in the speech, is introduced. This distortion is very annoying to the average listener.

[0005] The distortion is caused in large part by the environment in which the mobile telephones are used. Mobile telephones are typically used in a vehicle's interior where there is often ambient noise produced by the vehicle's engine and surrounding vehicular traffic. This ambient noise in the vehicle's interior is typically concentrated in the low audible frequency range and the magnitude of the noise can vary due to such factors as the speed and acceleration of the vehicle and the extent of the surrounding vehicular traffic. This type of low frequency noise also has the tendency of significantly decreasing the intelligibility of the speech coming from the speaking person in the car environment. The decrease in speech intelligibility caused by low frequency noise can be particularly significant in communication systems deploying a VSELP vocoder, but can also occur in communication systems that do not include a VSELP vocoder.

[0006] The influence of the ambient noise on the mobile telephone can also be affected by the manner in which the mobile telephone is used. In particular, the mobile telephone may be used in a hands-free mode where the telephone user talks on the telephone while the mobile telephone is in a cradle. This frees the telephone user's hands to drive but also increases the distance that the telephone user's audible words must travel before reaching the microphone input of the mobile telephone. This increased distance between the user and the mobile telephone, along with the varying ambient noise, can result in noise being a significant portion of the total power spectral energy of the audio signal inputted into the mobile telephone.

[0007] Prior art disclosure contained in EP 0 645 756, EP 0 558 312, EP 0 665 530, DE 4 012 349, U.S. Patent Nos. 4,811,404, 4,461,025, and 5,251,263 all disclose manners by which to filter unwanted signal components.

[0008] In theory, various signal processing algorithms could be implemented using digital signal processors to filter the VSELP encoded background noise. These solutions, however, often require significant digital signal processing overhead, measured in terms of millions of instructions executed per second (MIPS), which consumes valuable processing time, memory space, and power consumption. Each of these signal processing resources, however, is limited in portable radiotelephones. Hence, simply increasing the processing burden of the DSP is not an optimal solution for minimizing VSELP encoded and other types of background noise.

[0009] The present invention provides a method and an apparatus for selectively altering a frame of a digital signal according to claims 1 and 9.

[0010] The present invention provides an adaptive noise reduction system that reduces the undesirable contributions

of encoded background noise while both minimizing any negative impact on the quality of the encoded speech and minimizing any increased drain on digital signal processor resources. The method and system of the present invention increases the intelligibility of the speech in a digitized audio signal by passing frames of the digitized audio signal through a filter circuit. The filter circuit functions as an adjustable, high-pass filter which filters a portion of the digitized signal in a low audible frequency range and passes the portion of the digitized signal falling in higher frequency ranges. Because the noise in a vehicle tends to be concentrated in a low audible frequency range and only a relatively small portion of the intelligibility content of speech falls within this low frequency range, the filter circuit filters a large segment of the noise in the digitized audio signal while only filtering less important segments of the speech. This results in a relatively larger portion of the noise energy being removed compared to the portion of the speech energy removed. By adaptively adjusting and selecting the frequency response curve of the filter circuit, the amount of speech filtered is limited and has a minimal affect on the intelligibility of the speech outputted by the radio.

[0011] A filter control circuit is used to adjust the filter circuit to exhibit different frequency response curves as a function of a noise estimate and/or a spectral profile result corresponding to the noise in the audio signal. The noise estimate and/or the spectral profile result are adjusted on a frame-by-frame basis for the digital signal and as a function of speech detection. If speech is not detected, the noise estimate and/or spectral profile result is updated for the current frame. If speech is detected, the noise estimate and/or spectral profile result is left unadjusted.

[0012] In a first embodiment, the filter circuit calculates noise estimates for the frames of the digitized audio signals. The noise estimates correspond to the amount of background noise in the frames of the digitized audio signals. As the relative amount of background noise to speech in a low frequency range of speech increases, the noise estimates increase. The filter control circuit uses the noise estimates to adjust the filter circuit to filter larger portions of the low frequency range of speech as the relative amount of background noise to speech in a low frequency range of speech increases. When no background noise is present, no portion of the speech signal is filtered. Larger portions of noise and speech information are extracted when there is a higher level of background noise. Because noise tends to be concentrated in a low frequency range and only a relatively small portion of the intelligibility content of speech falls within this low frequency range, the overall intelligibility of the audio signal can be increased by increasing the portion of low frequency energy being filtered as the noise estimates increase.

[0013] In a second embodiment, a modified filter control circuit is used to adjust the filter circuit to exhibit different frequency response curves as a function of a noise profile of the noise estimate over a selected frequency range in the audio signal. The filter control circuit includes a spectral analyzer for determining a noise profile estimate as a function of the detection speech. A noise profile estimate is determined for a current frame and compared to a reference noise profile. Based on this comparison, the filter circuit is adaptively adjusted to extract varying amounts of low frequency energy from the current frame.

[0014] The adaptive noise reduction system according to the present invention may be advantageously applied to telecommunication systems in which portable/mobile radio transceivers communicate over RF channels with each other or with fixed telephone line subscribers. Each transceiver includes an antenna, a receiver for converting radio signals received over an RF channel via the antenna into analog audio signals, and a transmitter. The transmitter includes a coder-decoder (codec) for digitizing analog audio signals to be transmitted into frames of digitized speech information, the speech information including both speech and background noise. A digital signal processor processes a current frame based on an estimate of the background noise and the detection of speech in the current frame to minimize background noise. A modulator modulates an RF carrier with the processed frame of digitized speech information for subsequent transmission via the antenna.

[0015] These and other features and advantages of the present invention will be readily apparent to one of ordinary skill in the art from the following written description, read in conjunction with the drawings, in which:

FIGURE 1 is a general functional block diagram of the present invention;
 FIGURE 2 illustrates the frame and slot structure of the U.S. digital standard IS-54 for cellular radio communications;
 FIGURE 3 is a block diagram of a first preferred embodiment of the present invention implemented using a digital signal processor;
 FIGURE 4 is a functional block diagram of an exemplary embodiment of the present invention in one of plural portable radio transceivers in a telecommunication system;
 FIGURES 5A and 5B is a flow chart which illustrates functions/operations performed by the digital signal processor in implementing the first preferred embodiment of the present invention;
 FIGURE 6A is a graph illustrating a first example of an attenuation vs. frequency characteristic of a filter circuit according to the first preferred embodiment of the present invention;
 FIGURE 6B is a graph illustrating a second example of an attenuation vs. frequency characteristic of a filter circuit according to the first preferred embodiment of the present invention;
 FIGURE 7 is an example look-up table accessible by the filter control circuit of the first preferred embodiment of

the present invention;

FIGURES 8A and 8B are graphs illustrating the amplitude vs. frequency characteristics of example input audio signals;

FIGURES 9A and 9B are graphs illustrating the amplitude vs. frequency characteristics of the input audio signals of Figures 8A and 8B, respectively, after having been filtered by the filter circuit of the present invention;

FIGURE 10 is a block diagram of a second preferred embodiment of the present invention implemented using a digital signal processor;

FIGURE 11 is a flow chart, corresponding to the flow chart of Figure 5B, which illustrates functions/operations performed by the digital signal processor in implementing the second preferred embodiment of the present invention; and

FIGURE 12 is an example look-up table accessible by the filter control circuit of the second preferred embodiment of the present invention.

[0016] In the following description, for purposes of explanation and not limitation, specific details are set forth, such as particular circuits, circuit components, techniques, flow charts, etc. in order to provide a thorough understanding of the invention. However, it will be apparent to one skilled in the art that the present invention may be practiced in other embodiments that depart from these specific details. In other instances, detailed descriptions of well known methods, devices, and circuits are omitted so as not to obscure the description of the present invention with unnecessary details.

[0017] Figure 1 is a general block diagram of the adaptive noise reduction system 100 according to the present invention. Adaptive noise reduction system 100 includes a filter control circuit 105 connected to a filter circuit 115. Filter control circuit 105 generates a filter control signal for a current frame of a digitized audio signal. The filter control signal is outputted to the filter circuit 115, and the filter circuit 115 adjusts in response to the filter control signal to exhibit a high-pass frequency response curve selected based on the filter control signal. The adjusted filter circuit 115 filters the current frame of the digitized audio signal. The filtering signal is processed by a voice coder 120 to produce a coded signal representing the digitized audio signal.

[0018] In an exemplary embodiment of the invention applied to portable/mobile radio telephone transceivers in a cellular telecommunications system, Figure 2 illustrates the time division multiple access (TDMA) frame structure employed by the IS-54 standard for digital cellular telecommunications. A "frame" is a twenty millisecond time period which includes one transmit block TX, one receive block RX, and a signal strength measurement block used for mobile-assisted hand-off (MAHO). The two consecutive frames shown in Figure 2 are transmitted in a forty millisecond time period. Digitized speech and background noise information is processed and filtered on a frame-by-frame basis as further described below.

[0019] Preferably, the functions of the filter control circuit 105, filter circuit 115, and voice coder 120 shown in Figure 1 are implemented with a high speed digital signal processor. One suitable digital signal processor is the TMS320C53 DSP available from Texas Instruments. The TMS320C53 DSP includes on a single integrated chip a sixteen-bit micro-processor, on-chip RAM for storing data such as speech frames to be processed, ROM for storing various data processing algorithms including the VSELP speech compression algorithm, and other algorithms to be described below for implementing the functions performed by the filter control circuit 105 and the filter circuit 115.

[0020] A first embodiment of the present invention is shown in Figure 3. In the first embodiment, the filter circuit 115 is adjusted as a function of background noise estimates determined by the filter control circuit. Frames of pulse code modulated (PCM) audio information are sequentially stored in the DSP's on-chip RAM. The audio information could be digitized using other digitization techniques. Each PCM frame is retrieved from a DSP on-chip RAM and processed by frame energy estimator 210, and stored temporarily in temporary frame store 220. The energy of the current frame determined by frame energy estimator 210 is provided to noise estimator 230 and speech detector 240 function blocks. Speech detector 240 indicates that speech is present in the current frame when the frame energy estimate exceeds the sum of the previous noise estimate and a speech threshold. If the speech detector 240 determines that no speech is present, the digital signal processor 200 calculates an updated noise estimate as a function of the previous noise estimate and the current frame energy (block 230).

[0021] The updated noise estimate is outputted to a filter selector 235. Filter selector 235 generates a filter control signal based on the noise estimate. In the preferred embodiment, the filter selector 235 accesses a look-up table in generating the filter control signal. The look-up table includes a series of filter control values that are each matched with a noise estimate or range of noise estimates. A filter control value from a look-up table is selected based on the updated noise estimate and this filter control value is represented by a filter control signal outputted to a filter bank 265 for the filter circuit 115. To stabilize the process and avoid accessive switching between different filters a hangover time of N frames is set upon the selection of a new filter. A new filter can only be selected every N frames, where N is an integer greater than one and preferably greater than 10.

[0022] The filter circuit 115 is adjusted in response to the filter control signal to exhibit a high-pass frequency response curve that corresponds with the inputted filter control signal and noise estimate. Various different types of filter circuits

well known in prior art can be utilized to exhibit selected frequency response curves in response to the filter control signal. These prior art filters include IIR filters such as Butterworth, Chebyshev (Tschebyscheff) or elliptic filters. IIR filters are preferable to FIR filters, which also can be used, due to lower processing requirements.

[0023] The filtered signal is processed by a voice coder 120 which is used to compress the bit rate of the filtered signal. In the preferred embodiments, the voice coder 120 uses vector sourcebook excited linear predictive coding (VSELP) to code the audio signal. Other voice coding techniques and algorithms such as code excited linear predictive (CELP) codings, residual pulse excited linear predictive (RPE-LTP) coding, improved multiband excited (IMBE) coding can be used. By filtering the frames of audio signals in accordance with the present invention before voice coding, background noise is minimized which substantially reduces any undesired noise effects in the speech when it is re-constituted. It also prevents the speech from being "drowned" in low frequent noise.

[0024] The digital signal processor 200 described in conjunction with Figure 3 can be used, for example, in the transceiver of a digital portable/mobile radiotelephone used in a radio telecommunications system. Figure 4 illustrates one such digital radio transceiver which may be used in a cellular telecommunications network.

[0025] Audio signals including speech and background noise are input in a microphone 400 to a coder-decoder (codec) 402 which preferably is an application specific integrated circuit (ASIC). The band limited audio signals detected at microphone 400 are sampled by the codec 402 at a rate of 8,000 samples per second and blocked into frames. Accordingly, each twenty millisecond frame includes 160 speech samples. These samples are quantized and converted into a coded digital format such as 14-bit linear PCM. Once 160 samples of digitized speech for a current frame are stored in a transmit DSP 200 in on-chip RAM 202, the transmit DSP 200 performs channel encoding functions, the frame energy estimation, noise estimation, speech detection, FFT, filter functions and digital speech coding/compression in accordance with the VSELP algorithm, as described above in conjunction with Figure 3.

[0026] A supervisory microprocessor 432 controls the overall operation of all of the components in the transceiver shown in Figure 4. The filtered PCM data stream generated by transmit DSP 200 is provided for quadrature modulation and transmission. To this end, an ASIC gate array 404 generates in-phase (I) and quadrature (Q) channels of information based upon the filtered PCM data stream from DSP 200. The I and Q bit streams are processed by matched, low pass filters 406 and 408 and passed onto IQ mixers in balanced modulator 410. A reference oscillator 412 and a multiplier 414 provide a transmit intermediate frequency (IF). The I signal is mixed with in-phase IF, and the Q signal is mixed with quadrature IF (i.e., the in-phase IF delayed by 90 degrees by phase shifter 416). The mixed I and Q signals are summed, converted "up" to an RF channel frequency selected by channel synthesizer 430, and transmitted via duplexer 420 and antenna 422 over the selected radio frequency channel.

[0027] On the receive side, signals received via antenna 422 and duplexer 420 are down converted from the selected receive channel frequency in a mixer 424 to a first IF frequency using a local oscillator signal synthesized by channel synthesizer 430 based on the output of reference oscillator 428. The output of the first IF mixer 424 is filtered and down converted in frequency to a second IF frequency based on another output from channel synthesizer 430 and demodulator 426. A receive gate array 434 then converts the second IF signal into a series of phase samples and a series of frequency samples. The receive DSP 436 performs demodulation, filtering, gain/attenuation, channel decoding, and speech expansion on the received signals. The processed speech data are then sent to codec 402 and converted to baseband audio signals for driving loudspeaker 438.

[0028] The operations performed by the digital signal processor 200 for implementing the functions of filter control circuit 105, filter circuit 115, and voice coder 120 will now be described in conjunction with the flow chart illustrated in Figures 5A and 5B. Frame energy estimator 210 determines the energy in each frame of audio signals. Frame energy estimator 210 determines the energy of the current frame by calculating the sum of the squared values of each PCM sample in the frame (step 505). Since there are 160 samples per twenty millisecond frame for an 8000 samples per second sampling rate, 160 squared PCM samples are summed. Expressed mathematically, the frame energy estimate is determined according to equation 1 below:

$$\text{Frame energy} = \sum_{i=1}^{160} \{\text{Samp}(i)\}^2 \quad (\text{equation 1})$$

The frame energy value calculated for the current frame is stored in the on-chip RAM 202 of DSP 200 (step 510).

[0029] The functions of speech detector 240 include fetching a noise estimate previously determined by noise estimator 230 from the on-chip RAM of DSP 200 (step 515). Of course, when the transceiver is initially powered up, no noise estimate will exist. Decision block 520 anticipates this situation and assigns a noise estimate in step 525. Preferably, an arbitrarily high value, e.g. 20 dB above normal speech levels, is assigned as the noise estimate in order to force an update of the noise estimate value as will be described below. The frame energy determined by frame energy

estimator 210 is retrieved from the on-chip RAM 202 of DSP 200 (block 530). A decision is made in block 535 as to whether the frame energy estimate exceeds the sum of the retrieved noise estimate plus a predetermined speech threshold value, as shown in equation 2 below:

$$\text{frame energy estimate} > (\text{noise estimate} + \text{speech threshold}) \quad (\text{equation 2})$$

[0030] The speech threshold value may be a fixed value determined empirically to be larger than short term energy variations of typical background noise and may, for example, be set to 9 dB. In addition, the speech threshold value may be adaptively modified to reflect changing speech conditions such as when the speaker enters a noisier or quieter environment. If the frame energy estimate exceeds the sum in equation 2, a flag is set in block 570 that speech exists. If speech detector 240 detects that speech exists, then noise estimator 230 is bypassed and the noise estimate calculated for the previous frame in the digitized audio is retrieved and used as the current noise estimate. Conversely, if the frame energy estimate is less than the sum in equation 2, the speech flag is reset in block 540.

[0031] Other systems for detecting speech in a current frame can also be used. For example, the European Telecommunications Standards Institute (ETSI) has developed a standard for voice activity detection (VAD) in the Global System for Mobile communications (GSM) system and is described in the ETSI Reference: RE/SMG-020632P which is incorporated by reference. This standard could be used for speech detection in the present invention and is incorporated by reference.

[0032] If speech does not exist, the noise estimation update routine of noise estimator 230 is executed. In essence, the noise estimate is a running average of the frame energy during periods of no speech. As described above, if the initial start-up noise estimate is chosen sufficiently high, speech is not detected, and the speech flag will be reset thereby forcing an update of the noise estimate.

[0033] In the noise estimation routine followed by noise estimator 230, a difference/error delta (Δ) is determined in block 545 between the frame noise energy generated by frame energy estimator 210 and a noise estimate previously calculated by noise estimator 230 in accordance with the following equation:

$$\Delta = \text{current frame energy} - \text{previous noise estimate} \quad (\text{equation 3})$$

A determination is made in decision block 550 whether Δ exceeds zero. If Δ is negative, as occurs for high values of the noise estimate, then the noise estimate is recalculated in block 560 in accordance with the following equation:

$$\text{noise estimate} = \text{previous noise estimate} + \Delta/2 \quad (\text{equation 4})$$

Since Δ is negative, this results in a downward correction of the noise estimate. The relatively large step size of $\Delta/2$ is chosen to rapidly correct for decreasing noise levels. However, if the frame energy exceeds the noise estimate, providing a Δ greater than zero, the noise is updated in block 555 in accordance with the following equation:

$$\text{noise estimate} = \text{previous noise estimate} + \Delta/256 \quad (\text{equation 5})$$

Since Δ is positive, the noise estimate must be increased. However, a smaller step size of $\Delta/256$ (as compared to $\Delta/2$) is chosen to gradually increase the noise estimate and provide substantial immunity to transient noise.

[0034] The noise estimate calculated for the current frame is outputted to the filter selector 235. In the first preferred embodiment, filter selector 235 accesses a look-up table and uses the current noise estimate to select a filter control value (Step 572). The filter circuit 115 (in Step 574) is then adjusted as a function of the selected filter control value to exhibit a frequency response curve intended to increase the amount of noise filtered as the noise estimate and background noise increases. The PCM samples stored in DSP RAM are then passed through the adjusted filter circuit 265 to filter the PCM samples in order to remove noise (Step 576). The filtered PCM samples are then processed by voice coder 120 (step 578), and the coded samples are then outputted to RF transmit circuits (Step 580).

[0035] Figures 6A and 6B show examples of how the filter circuit 115 adjusts to exhibit different frequency response curves F1-F4 for different filter control signals inputted to the filter circuit 115. As shown in Figure 6A, the filter circuit 115 can be selected to exhibit a series of different frequency response curves with the frequency response curves F1-F4 having cut-off frequencies F1c-F4c, respectively. The cut-off frequencies of filter circuit 115 may range in the preferred embodiment from 300 Hz to 800 Hz. As the noise estimates increase, the filter circuit 115 is designed to exhibit frequency response curves having higher cut-off frequencies. The higher cut-off frequencies result in a larger

portion of frame energy falling within the lower frequency range of speech being extracted by the filter circuit 115.

[0036] Likewise, as shown in Figure 6B, the filter circuit 115 can be selected to exhibit a series of different frequency response curves F1-F4 with each frequency response curve having a different slope and the same cut-off frequencies. The cut-off frequency for frequency response curves F1-F4 is in the above-mentioned range. As the noise estimate increases, the filter circuit 115 is adjusted to exhibit frequency response curves having steeper slopes. The steeper slopes result in a larger portion of frame energy falling within the lower frequency range of speech being extracted by the filter circuit 115.

[0037] The filter circuit 115 filters the current frames as a function of the noise estimate calculated for the current frame. The current frame is filtered so that the noise is reduced and a major portion of the speech is passed. The major portion of speech which is passed unfiltered provides for recognizable speech output with only a minimal reduction in the quality of the speech signal. A combination of different cutoff frequencies and different slopes could be used for adaptively extracting selected portions of frame energy falling within a low frequency range of speech.

[0038] Figure 7 depicts an example look-up table accessed by filter selector 235 in order to select one of the filter response curves F1-F4 for filter circuit 115. The look-up table includes a series of potential noise estimates N1-Nn and filter control values F1-Fn that correspond with potential response curves that are exhibitable by the filter circuit 115. Noise estimates N1-Nn can each represent a range of noise estimates and are each matched with a particular filter control value F1-F4. The filter control circuit 105 generates a filter control signal by calculating a noise estimate and retrieving from the look-up table the filter control value associated therewith.

[0039] Figures 8A & B and 9A & B show how the audio signal for two frames are each adaptively filtered to provide an improved audio signal outputted to the RF transmitter. Figures 8A and 8B show a first frame and a second frame of an audio signal containing speech components s1 and s2 and noise components n1 and n2, respectively. As shown, the noise energy n1 and n2 in both frames is concentrated in a low audible frequency range, while the speech energy s1 and s2 is concentrated in a higher audible frequency range. Figure 9A shows the noise signal n1 and speech signal s1 for the first frame after filtering. Figure 9B shows the noise signal n2 and speech signal s2 for the second frame after filtering.

[0040] The adaptive audio noise reduction system 100, as discussed, is designed to account for the difference in noise level between the first frame and the second frame by adjusting the filter control circuit 105 based on a calculated noise estimate for the current frame. For example, a noise estimate N1 and a spectral profile S1 is calculated by filter control circuit 105 and a filter control value of F1 is selected for the first frame. In the preferred embodiment, the filter circuit 115 is adjusted based on filter control value F1 and exhibits a frequency response curve F1 having a cut-off frequency F1c, as shown in Figure 6A. The first frame is passed through this adjusted filter circuit 115. The filter circuit 115 is selected so that a large portion of the noise n1 and only a small portion of speech s1 falls below the cut-off frequency F1c of the frequency response curve F1. This results in noise n1 being effectively filtered and only a relatively insignificant portion of speech s1 being filtered. The filtered audio signal of the first frame is shown in Figure 9A.

[0041] In the second frame shown in Figure 8b, a higher background noise is present, and assuming speech is not detected, a higher noise estimate n2 is calculated by filter control circuit 105. A higher corresponding filter control value F2 is determined for the second frame based on the higher noise estimate. In the first preferred embodiment, the filter circuit 115 is adjusted in response to the higher filter control value F2 to exhibit a frequency response curve having a higher cut-off frequency F2c, as shown in Figure 6A. The subsequent frame of audio signal is passed through the adjusted filter circuit 115. Because the cut-off frequency F2c of the frequency response curve F2 is higher for the subsequent frame, a larger portion of both the noise n2 and speech s2 is filtered. The portion of speech s2 filtered is still relatively insignificant to the intelligibility information contained by the frame so that there is only minimal affect on the speech. The disadvantage of filtering a larger portion of the speech s2 is offset by the advantage of the increased removal of noise n2 from the second frame. The filtered spectral portion of the speech does not significantly contribute to the intelligibility of the speech. The filtered audio signal of the second frame is shown in Figure 9B.

[0042] A second preferred embodiment of adaptive noise reduction system 100 is shown in Figures 10-12. In the second preferred embodiment, the filter control circuit 105 adjusts the filter circuit 115 as a function of noise profile estimates. A noise profile estimate is calculated for each frame and is compared to a reference noise profile. Based on this comparison, the filter circuit 115 is adaptively adjusted to extract varying amounts of low frequency energy from the current frame.

[0043] Referring to Figure 10, a DSP 200 configured according to the second preferred embodiment is shown. As shown, the filter control circuit 105 includes a spectral analyzer 270, in addition to frame energy estimator 210, noise estimator 230, speech detector 240, and filter selector 235 which are described with respect to the first preferred embodiment. The filter control circuit 105 determines noise estimates and detects speech for the received frames as described for the first embodiment and shown in flow charts 5A and 5B. Upon speech detection for a current frame, the spectral analyzer 270 updates the noise profile estimate and uses the noise profile estimate in adjusting the filter circuit 115.

[0044] Referring to Figure 11, the steps of updating the noise profile estimate and adjusting the filter circuit 115 is

shown. Figure 11 shows the steps performed by spectral analyzer 270 incorporated into the overall process previously described in the flow charts of Figures 5A and 5B for the first preferred embodiment.

[0045] When speech is not detected for the current frame, the spectral analyzer 270 first determines a noise profile for the current frame (step 600). The noise profile determined for the current frame includes energy calculations for different frequencies (i.e., frequency bins) within a selected low frequency range of speech for the current frame. In the preferred embodiment, the selected frequency range is approximately 300 to 800 hertz. The noise profile of the current frame can be determined by processing the current frame using a Fast Fourier Transform (FFT) having N frequency bins. Processing digital signals using an FFT is well-known in the prior art and is advantageous in that very little processing power is required where the FFT is limited to a relatively small number of frequency bins such as 32. An FFT having N frequency bins produces energy calculations at N different frequencies. The energy calculations for the frequency bins falling within the selected frequency range form the noise profile for the current frame.

[0046] To determine the noise profile estimate for the current frame (step 604), the noise profile for the current frame is averaged with a noise profile estimate determined for the previous frame of the audio signal. Where no previous noise profile estimate is available, such as after initialization, a stored, initial noise profile estimate can be used. The noise profile estimate includes noise energy estimates e_i (where $i = 1, 2, \dots, n$) located at successively lower frequencies (i.e., e_1 is the noise energy estimate for the highest frequency and e_n is the noise energy estimate for the lowest frequency in the selected frequency range). In the preferred embodiment, each noise energy estimate e_i corresponds to an average of the energy calculations at a particular frequency in the selected frequency range over a plurality of successive frames in which no speech was detected. By using a plurality of frames in determining the noise profile estimate, the filter circuit 115 is adjusted on a more gradual basis. In alternate embodiments, the noise profile estimate can be equated to the noise profile of the current frame.

[0047] The energy estimates e_i of the noise profile estimate are then compared with a reference noise profile (step 604). The reference noise profile includes reference energy thresholds e_{ri} (where $i = 1, 2, \dots, n$) at frequencies corresponding to the frequencies for noise energy estimates e_i of the noise profile estimate. The reference energy thresholds e_{ri} can be determined empirically. The noise energy estimates e_i are successively compared to corresponding reference energy thresholds e_{ri} from the highest frequency energy estimate e_1 to the lowest frequency energy estimate e_n .

[0048] More specifically, noise energy estimate e_1 is first compared to reference noise threshold e_{r1} . If e_1 is greater than reference noise threshold e_{r1} , then a comparison value c_1 is selected and inputted into filter selector 235. If noise energy estimate e_1 is less than reference noise threshold e_{r1} , then noise energy estimate e_2 (which is a noise energy estimate taken at a lower frequency than e_1) is compared to reference noise threshold e_{r2} . If noise energy estimate e_2 is greater than reference noise threshold e_{r2} , then a comparison value c_2 is selected and inputted to filter selector 235. This comparison process is continued until a comparison value c_i (where $i = 1, 2, \dots, n$) is selected.

[0049] The filter circuit 235 uses the determined comparison value c_i to determine a filter control value. The filter control value is selected from a look-up table such as that shown in Figure 12. The look-up table includes a series of comparison values c_i and corresponding filter control values F_i . The filter circuit 115 is adjusted as a function of the selected filter control value. The filter circuit 115 is adjusted to exhibit a frequency response curve for extracting low frequency energy from the current frame. The filter circuit 115 is adjusted to extract increasing amounts of low frequency energy as noise energy estimates at successively higher frequencies surpass their corresponding reference energy thresholds. Figure 6A and 6B show example frequency response curves for selected filter control values.

[0050] Use of noise profile estimates helps improve the ability to adaptively adjust the filter circuit to extract low frequency energy in a manner to improve the overall quality of speech. Since the car environment is not the only environment where a mobile telecommunications device is used, and therefore the noise profile in certain situations could be tilted more towards higher frequencies, the spectral analyzer 270 can be selectively disabled when noise energy in the low frequencies is small. Also, when a significant portion of the noise frequency spectrum resides in lower frequencies a steeper filtering slope could be applied even though some processing power may be sacrificed. This extra processing requirement is still fairly small.

[0051] As is evident from the description above, the adaptive noise filter system of the present invention is implemented simply and without significant increase in DSP calculations. More complex methods of reducing noise, such as "spectral subtraction," require several calculation-relates MIPS and a large amount of memory for data and program code storage. By comparison, the present invention may be implemented using only a fraction of the MIPS and memory required for the "spectral subtraction" algorithm which also introduces more speech distortion. Reduced memory reduces the size of the DSP integrated circuits; decreased MIPS decreases power consumption. Both of these attributes are desirable for battery-powered portable/mobile radiotelephones.

[0052] While the invention has been particularly shown and described with reference to the preferred embodiments thereof, it is not limited to those embodiments. For example, although a DSP is disclosed as performing the functions of the frame energy estimator 210, noise estimator 230, speech detector 240, filter selector 235 and filter circuit 265, these functions could be implemented using other digital and/or analog components. In addition, an adaptive filtering system 100 could be implemented where the filter circuit 115 is adjusted as a function of both noise estimates and

noise profile estimates.

Claims

1. A method for selectively altering a frame of a digital signal formed of a plurality of successive frames, the digital signal representative of an audio signal received at a transmitter, the audio signal formed alternately of a speech component, a noise component, and the speech component together with the noise component, said method comprising the steps of:

estimating an energy level (505) of a frame of the digital signal;
determining (535), responsive to the estimate made during said step of estimating, whether the frame of the digital signal includes a speech component;
updating a noise estimate as a function of a preceding noise estimate and the energy level estimated during said step of estimating when said step of determining determines that the frame does not contain a speech component;
accessing (572) an entry in a look-up table having filter characteristics indexed against levels of noise estimates, the entry accessed associated with the noise estimate updated during said step of updating;
selecting (574) filter characteristics for a filter such that the filter exhibits a frequency response curve having variable gain over different frequency ranges, the filter characteristic selected responsive to the stored filter characteristics of the entry accessed during said step of accessing; and
filtering (576) the frame of the digital data with the filter which exhibits the filter characteristics, thereby to alter the frame of the digital data responsive to the filter characteristics.

2. The method of claim 1 further characterized by the additional intermediary step of determining (600) a noise profile estimate of the frame of the digital signal if the frame of the digital data is determined not to include the speech component.

3. The method of claim 2 wherein the noise profile estimate determined during said step of determining (600) the noise profile estimate is used during said step of updating to update the noise estimate.

4. The method of claim 1 wherein the look-up table accessed during said step of accessing is characterized by a plurality of entries (C1-CN, F4-FN), each entry of the plurality including a separate filter characteristic.

5. The method of claim 4 wherein the separate filter characteristics of the plurality of entries of the look-up table comprise separate high pass filter characteristics, each high pass characteristic defined by a separate cut-off frequency ($F1_c$, $F2_c$, $F3_c$, $F4_c$).

6. The method of claim 4 wherein the separate filter characteristics of the plurality of entries of the look-up table comprise separate high pass filter characteristics, each high pass filter characteristic defined by a separate frequency response curve slope ($F1$, $F2$, $F3$, $F4$).

7. The method of claim 1 characterized by the further step of incrementing a counter value to count each frame for which an energy level is estimated during said step of estimating.

8. The method of claim 7 wherein said step of selecting the filter-circuit filter characteristics is performed when the counter value is incremented each Nth time, N forming an integer value greater than one.

9. An apparatus (100; 200) for selectively altering a frame of a digital signal formed of a plurality of successive frames, the digital signal representative of an audio signal received at a transmitter, the audio signal formed alternately of a speech component, a noise component, and the speech component together with the noise component, said apparatus comprising:

an energy level estimator (210) coupled to receive indications of a frame of the digital signal, said energy level estimator for estimating an energy level of the frame of the digital signal;
a speech detector (240) coupled to said energy level estimator, said speech detector for determining whether the frame of the digital signal includes a speech component;
a noise estimator (230) operable when said speech detector determines that a frame does not contain a speech

component, said noise estimator for updating a noise estimate as a function of a preceding noise estimate, and the energy level estimated by said estimator;
 a look-up table (FIG. 12) containing a plurality of entries, each entry indexed against levels of noise estimates, an entry of said look-up table accessed responsive to a noise estimate formed by said noise estimator; and
 a filter (265) coupled to receive the frame of the digital data, said filter exhibiting selectable filter characteristics enabling the filter to exhibit a frequency response curve having variable gain over different frequency ranges, selection of the filter characteristics of the filter determined responsive to the entry of the look-up table accessed responsive to the noise estimate updated by said noise estimator.

10. The apparatus (100; 200) of claim 9 further characterized by a noise profile estimator (270) for determining a noise profile estimate of the frame of the digital data if the frame of the digital data is determined by said speech component determiner not to include the speech component.

Patentansprüche

1. Verfahren zur selektiven Änderung eines Rahmens eines Digitalsignals, das aus einer Vielzahl von aufeinanderfolgenden Rahmen besteht, wobei das Digitalsignal repräsentativ ist für ein Audiosignal, das an einem Sender empfangen wird, das Audiosignal abwechselnd aus einer Sprachkomponente, einer Rauschkomponente und der Sprachkomponente zusammen mit der Rauschkomponente besteht, und das Verfahren die Schritte umfasst:

Abschätzen eines Energiepegels (505) eines Rahmens des Digitalsignals;

Bestimmen (535), ansprechend auf die während des Abschätzungsschritts gemachte Abschätzung, ob der Rahmen des Digitalsignals eine Sprachkomponente enthält;

Aktualisieren einer Rauschabschätzung als Funktion einer vorangehenden Rauschabschätzung und des Energiepegels, der während des Abschätzungsschritts abgeschätzt wurde, wenn der Bestimmungsschritt bestimmt, dass der Rahmen keine Sprachkomponente enthält;

Zugreifen (572) auf einen Eintrag in einer Nachschlagtabelle, welche Filtercharakteristiken gegenüber Pegeln von Rauschabschätzungen aufführt, wobei der Eintrag, auf den zugegriffen wird, in Beziehung steht mit der Rauschabschätzung, die während des Aktualisierungsschritts aktualisiert wurde;

Wählen (574) von Filtercharakteristiken für ein Filter, so dass das Filter eine Frequenzantwortkurve aufweist, die eine variable Verstärkung über unterschiedliche Frequenzbereiche hat, die Filtercharakteristik ausgewählt wird ansprechend auf die gespeicherten Filtercharakteristiken des Eintrags auf den während des Zugriffsschritts zugegriffen wurde; und

Filtern (576) des Rahmens der Digitaldaten mit dem Filter, welches die Filtercharakteristiken aufweist, um dadurch den Rahmen der Digitaldaten ansprechend auf die Filtercharakteristiken zu ändern.

2. Verfahren nach Anspruch 1, ferner gekennzeichnet durch den zusätzlichen Zwischenschritt der Bestimmung (600) einer Rauschprofilabschätzung des Rahmens des Digitalsignals, wenn bestimmt wird, dass der Rahmen der Digitaldaten die Sprachkomponente nicht enthält.

3. Verfahren nach Anspruch 2, wobei die Rauschprofilabschätzung, die während des Schritts der Bestimmung (600) der Rauschprofilabschätzung bestimmt wurde, während des Aktualisierungsschritts zur Aktualisierung der Rauschabschätzung verwendet wird.

4. Verfahren nach Anspruch 1, wobei die Nachschlagtabelle, auf welche während des Zugriffsschritts zugegriffen wird, gekennzeichnet ist durch eine Vielzahl von Einträgen (C1-CN, F4-FN), wobei jeder Eintrag der Vielzahl eine getrennte Filtercharakteristik enthält.

5. Verfahren nach Anspruch 4, wobei die getrennten Filtercharakteristiken der Vielzahl von Einträgen der Nachschlagtabelle getrennte Hochpass-Filtercharakteristiken umfassen, wobei jede Hochpass-Filtercharakteristik durch eine getrennte Abschnidefrequenz ($F1_c$, $F2_c$, $F3_c$, $F4_c$) gekennzeichnet ist.

6. Verfahren nach Anspruch 4, wobei die getrennten Filtercharakteristiken der Vielzahl von Einträgen der Nachschlagtabelle getrennte Hochpass-Filtercharakteristiken umfassen, wobei jede Hochpass-Filtercharakteristik definiert ist durch eine getrennte Kurvensteigung (F1, F2, F3, F4) der Frequenzantwort.

7. Verfahren nach Anspruch 1, gekennzeichnet durch den weiteren Schritt der weiteren Inkrementierung eines Zählerwerts, um jeden Rahmen zu zählen, für welchen ein Energiepegel während des Abschätzschritts abgeschätzt wird.

8. Verfahren nach Anspruch 7, wobei der Schritt des Wählens der Filterschaltung-Filtercharakteristiken durchgeführt wird wenn der Zählerwert jedes N-te mal inkrementiert wird, wobei N ein ganzzahliger Wert größer als 1 ist.

9. Vorrichtung (100; 200) zur selektiven Änderung eines Rahmens eines Digitalsignals, das aus einer Vielzahl von aufeinanderfolgenden Rahmen besteht, wobei das Digitalsignal repräsentativ ist für ein Audiosignal, das an einem Sender empfangen wird, das Audiosignal abwechselnd aus einer Sprachkomponente, einer Rauschkomponente und der Sprachkomponente zusammen mit der Rauschkomponente besteht, wobei die Vorrichtung umfasst:

einen Energiepegel-Abschätzer (210), der angeschlossen ist Anzeichen eines Rahmens des Digitalsignals zu empfangen, wobei der Energiepegel-Abschätzer dazu dient, einen Energiepegel des Rahmens des Digitalsignals abzuschätzen;

einen Sprachdetektor (240), der an den Energiepegel-Abschätzer angeschlossen ist, wobei der Sprachdetektor dazu dient zu Bestimmen, ob der Rahmen des Digitalsignals eine Sprachkomponente enthält;

einen Rausch-Abschätzer (230), der betreibbar ist, wenn der Sprachdetektor bestimmt, dass ein Rahmen keine Sprachkomponente enthält, wobei der Rausch-Abschätzer dazu dient, eine Rauschabschätzung als Funktion einer vorangehenden Rauschabschätzung, und des durch den Abschätzer abgeschätzten Energiepegels zu aktualisieren;

eine Nachschlagtabelle (Fig. 12), welche eine Vielzahl von Einträgen enthält, wobei jeder Eintrag Pegel von Rauschabschätzungen anzeigt, und ansprechend auf eine durch den Rausch-Abschätzer gebildete Rauschabschätzung auf einen Eintrag der Nachschlagtabelle zugegriffen wird; und

ein Filter (265), das angeschlossen ist den Rahmen der Digitaldaten zu empfangen, wobei das Filter wählbare Filtercharakteristiken aufweist, die es dem Filter ermöglichen, eine Frequenzantwortkurve aufzuweisen, die eine veränderliche Verstärkung über unterschiedliche Frequenzbereiche hat, und die Auswahl der Filtercharakteristiken des Filters bestimmt wird ansprechend auf den Eintrag der Nachschlagtabelle, auf den ansprechend auf die von dem Rausch-Abschätzer aktualisierte Rauschabschätzung zugegriffen wird.

10. Vorrichtung (100; 200) nach Anspruch 9, ferner gekennzeichnet durch einen Rauschprofil-Abschätzer (270) zur Bestimmung einer Rauschprofil-Abschätzung des Rahmens der Digitaldaten, wenn von der Sprachkomponenten-Bestimmungseinrichtung bestimmt wird, dass der Rahmen der Digitaldaten keine Sprachkomponente enthält.

Revendications

1. Procédé pour modifier sélectivement une trame d'un signal numérique formé par une pluralité de trames successives, le signal numérique étant représentatif d'un signal audio reçu à un émetteur, le signal audio étant formé soit par une composante de parole, soit par une composante de bruit, soit par la composante de parole accompagnée de la composante de bruit, ce procédé comprenant les étapes suivantes :

on estime un niveau d'énergie (505) d'une trame du signal numérique;
en réponse à l'estimation effectuée au cours de l'étape d'estimation, on détermine (535) si la trame du signal numérique contient une composante de parole;
on actualise une estimation de bruit en fonction d'une estimation de bruit précédente et du niveau d'énergie estimé au cours de l'étape d'estimation, lorsque l'étape de détermination détermine que la trame ne contient pas une composante de parole;
on accède (572) à une rubrique dans une table à consulter ayant des caractéristiques de filtre indexées vis-à-vis de niveaux d'estimations de bruit, la rubrique à laquelle on accède étant associée à l'estimation de bruit

actualisée au cours de l'étape d'actualisation;

on sélectionne (574) des caractéristiques de filtre pour un filtre de façon que le filtre présente une courbe de réponse en fréquence ayant un gain variable sur différentes plages de fréquence, la caractéristique de filtre étant sélectionnée en réponse aux caractéristiques de filtre enregistrées de la rubrique à laquelle on a accédé au cours de l'étape d'accès; et

on filtre (576) la trame des données numériques avec le filtre qui présente les caractéristiques de filtre, pour modifier ainsi la trame des données numériques en réponse aux caractéristiques de filtre.

2. Procédé selon la revendication 1, caractérisé en outre par l'étape intermédiaire supplémentaire consistant à déterminer (600) une estimation de profil de bruit de la trame du signal numérique, si la trame des données numériques ne contient pas la composante de parole, d'après la détermination.

3. Procédé selon la revendication 2, dans lequel l'estimation de profil de bruit déterminée au cours de l'étape de détermination (600) de l'estimation de profil de bruit est utilisée au cours de l'étape d'actualisation pour actualiser l'estimation de bruit.

4. Procédé selon la revendication 1, dans lequel la table à consulter à laquelle on accède au cours de l'étape d'accès est caractérisée par une pluralité de rubriques (C1-CN, F4-FN), chaque rubrique de la pluralité contenant une caractéristique de filtre séparée.

5. Procédé selon la revendication 4, dans lequel les caractéristiques de filtre séparées de la pluralité de rubriques de la table à consulter comprennent des caractéristiques de filtre passe-haut séparées, chaque caractéristique passe-haut étant définie par une fréquence de coupure séparée (F_{1c} , F_{2c} , F_{3c} , F_{4c}).

6. Procédé selon la revendication 4, dans lequel les caractéristiques de filtre séparées de la pluralité de rubriques de la table à consulter comprennent des caractéristiques de filtre passe-haut séparées, chaque caractéristique de filtre passe-haut étant définie par une pente de courbe de réponse en fréquence séparée (F_1 , F_2 , F_3 , F_4).

7. Procédé selon la revendication 1, caractérisé par l'étape supplémentaire consistant à incrémenter une valeur de compteur pour compter chaque trame pour laquelle un niveau d'énergie est estimé au cours de l'étape d'estimation.

8. Procédé selon la revendication 7, dans lequel l'étape de sélection des caractéristiques de filtre du circuit de filtre est effectuée chaque fois que la valeur de compteur a été incrémentée N fois, N étant une valeur entière supérieure à un.

9. Appareil (100; 200) pour modifier sélectivement une trame d'un signal numérique formé par une pluralité de trames successives, le signal numérique étant représentatif d'un signal audio reçu à un émetteur, le signal audio étant formé soit par une composante de parole, soit par une composante de bruit, soit par la composante de parole accompagnée de la composante de bruit, cet appareil comprenant :

un estimateur de niveau d'énergie (210) couplé pour recevoir des indications d'une trame du signal numérique, cet estimateur de niveau d'énergie étant destiné à estimer un niveau d'énergie de la trame du signal numérique; un détecteur de parole (240) couplé à l'estimateur de niveau d'énergie, ce détecteur de parole étant destiné à déterminer si la trame du signal numérique contient une composante de parole;

un estimateur de bruit (230) pouvant fonctionner lorsque le détecteur de parole détermine qu'une trame ne contient pas une composante de parole, cet estimateur de bruit étant destiné à actualiser une estimation de bruit en fonction d'une estimation de bruit précédente, et du niveau d'énergie estimé par l'estimateur;

une table à consulter (figure 12) contenant une pluralité de rubriques, chaque rubrique étant indexée vis-à-vis de niveaux d'estimations de bruit, et l'accès à une rubrique de la table à consulter étant effectué en réponse à une estimation de bruit formée par l'estimateur de bruit; et

un filtre (265) couplé pour recevoir la trame des données numériques, ce filtre présentant des caractéristiques de filtre pouvant être sélectionnées, permettant au filtre de présenter une courbe de réponse en fréquence ayant un gain variable sur différentes plages de fréquence, la sélection des caractéristiques de filtre du filtre étant déterminée en réponse à la rubrique de la table à consulter à laquelle on a accédé en réponse à l'estimation de bruit actualisée par l'estimateur de bruit.

10. Appareil (100; 200) selon la revendication 9, caractérisé en outre par un estimateur de profil de bruit (270) pour déterminer une estimation de profil de bruit de la trame des données numériques si le détecteur de composante

EP 0 852 052 B1

de parole détermine que la trame des données numériques ne contient pas la composante de parole.

5

10

15

20

25

30

35

40

45

50

55

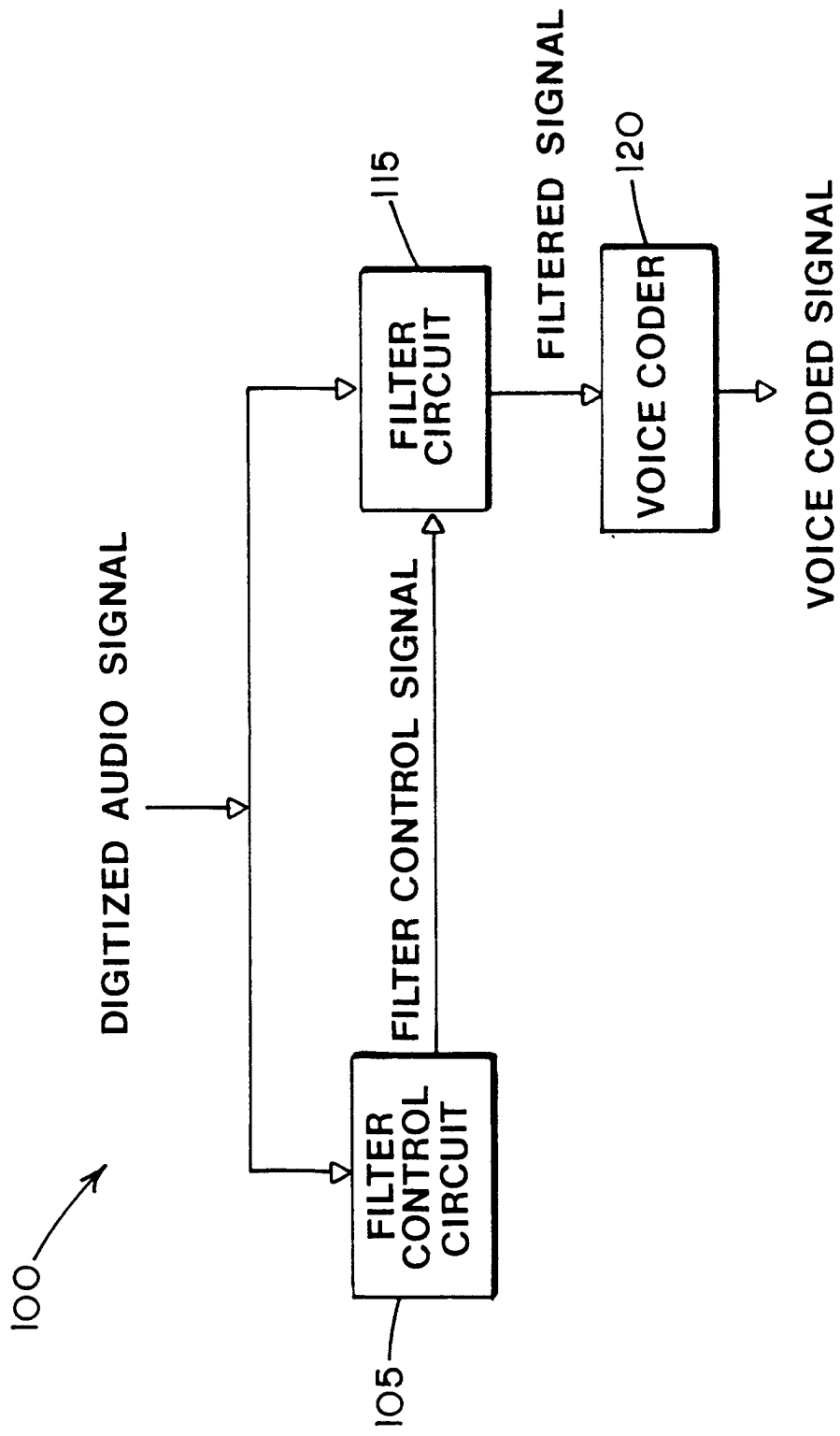


Fig.1

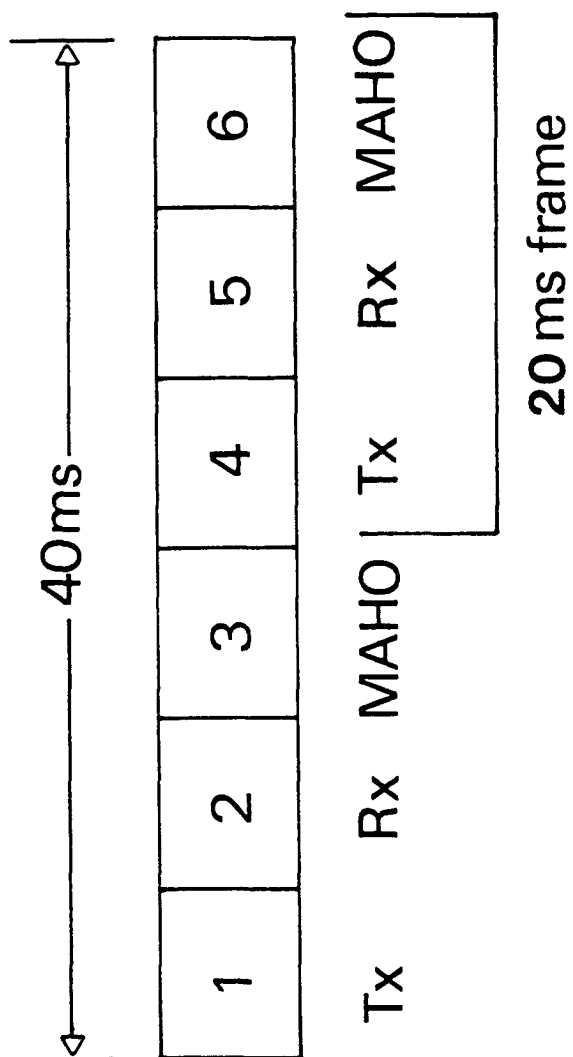


Fig.2

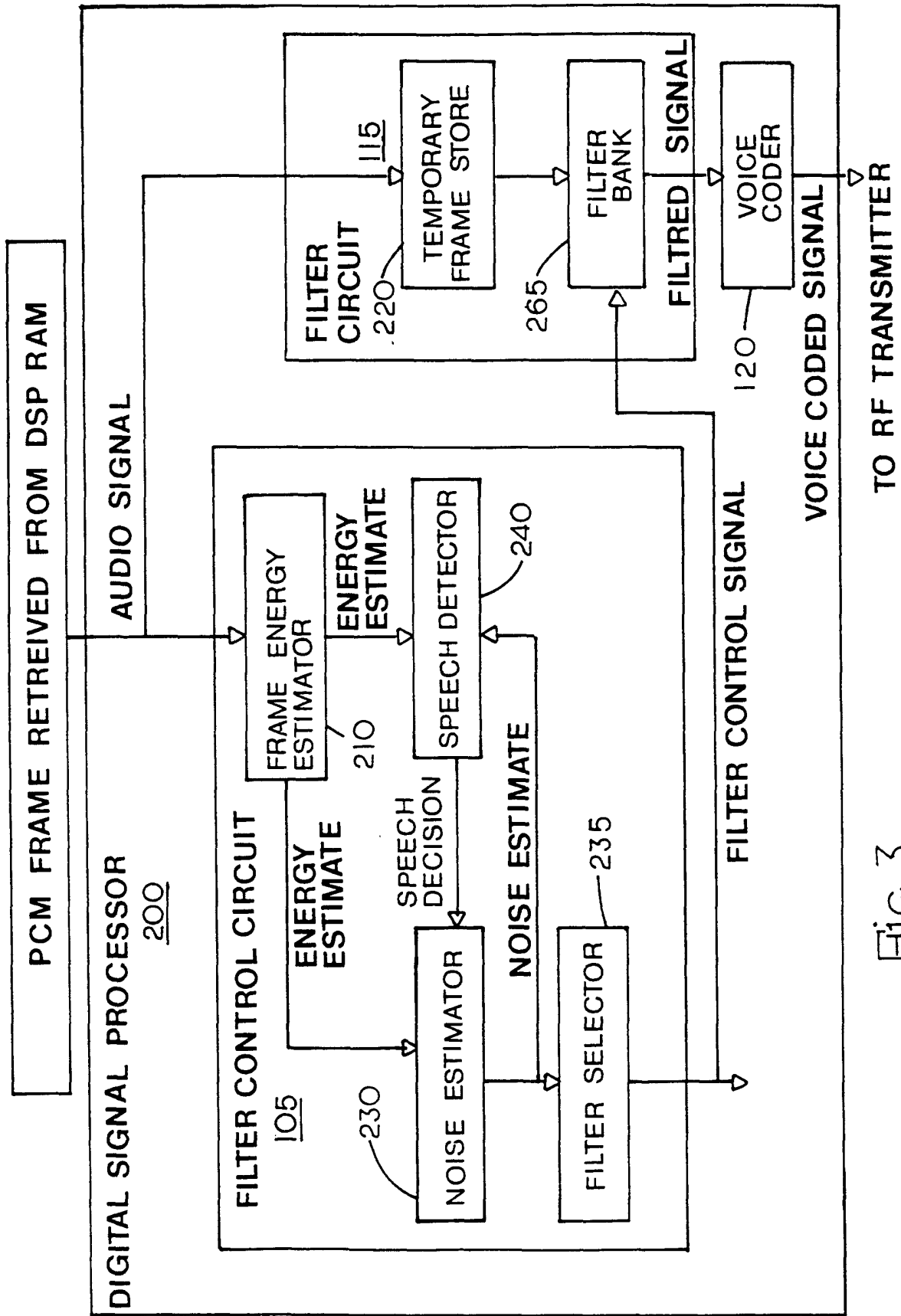


Fig. 3

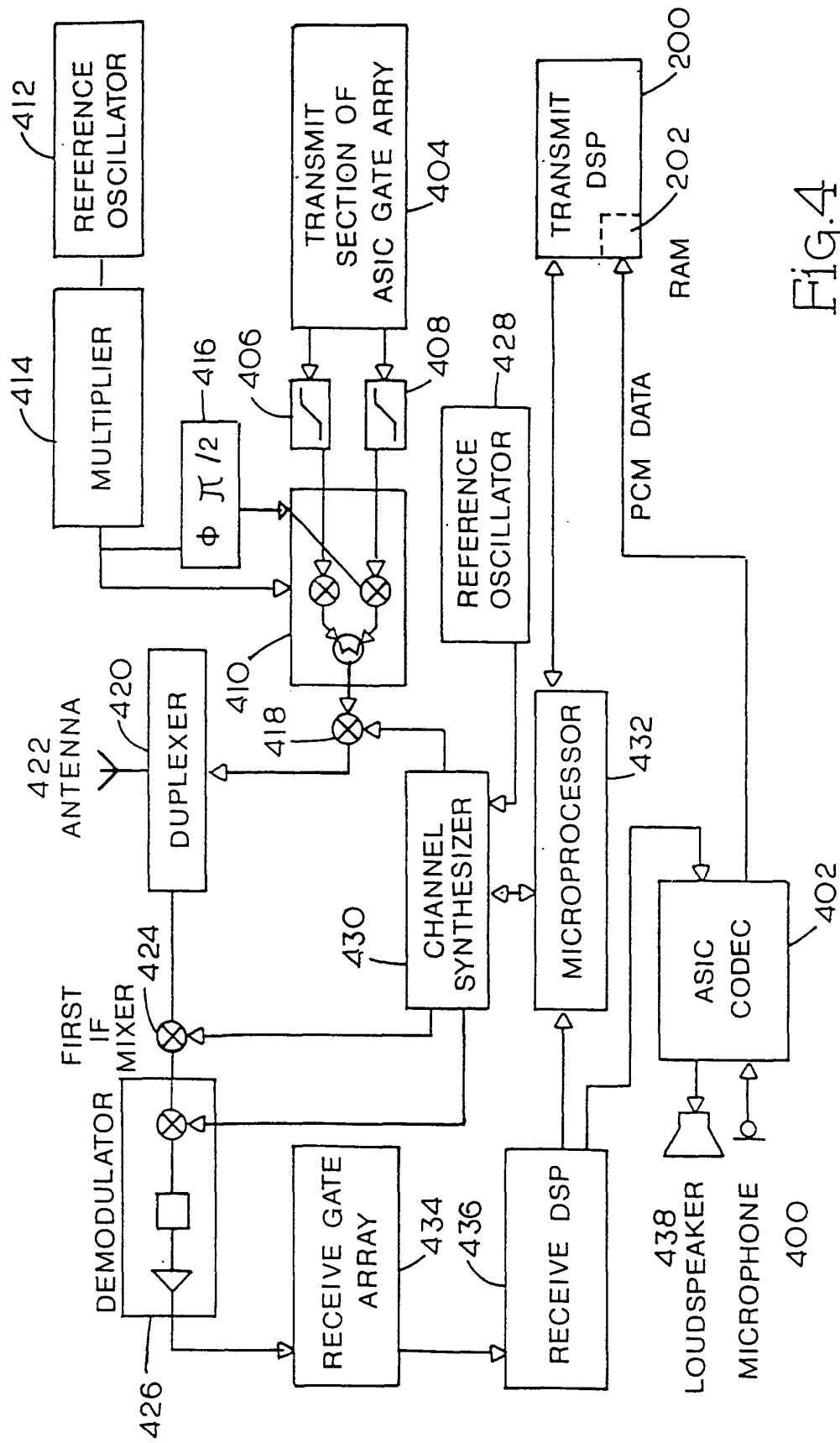


Fig.4

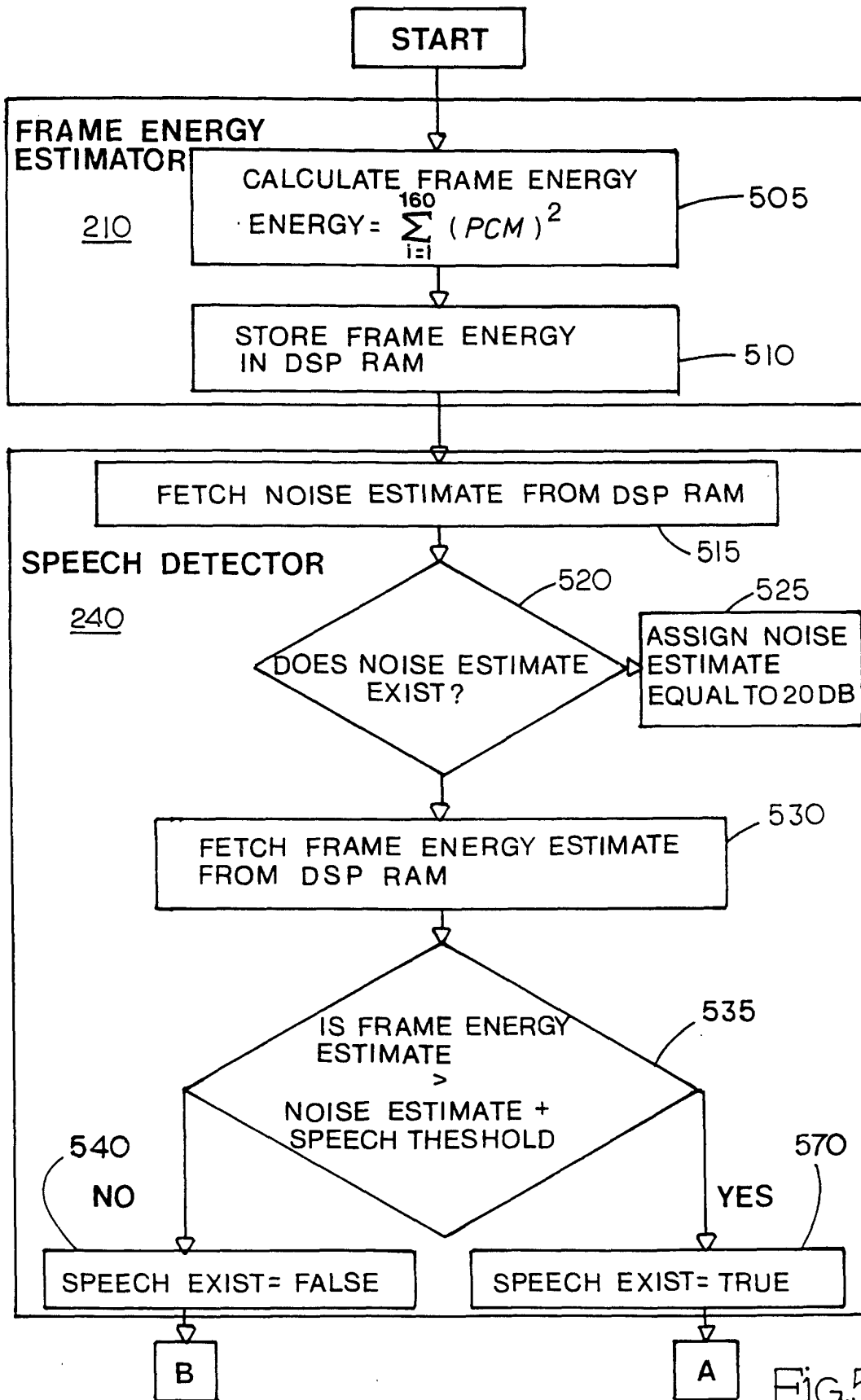


Fig. 5A

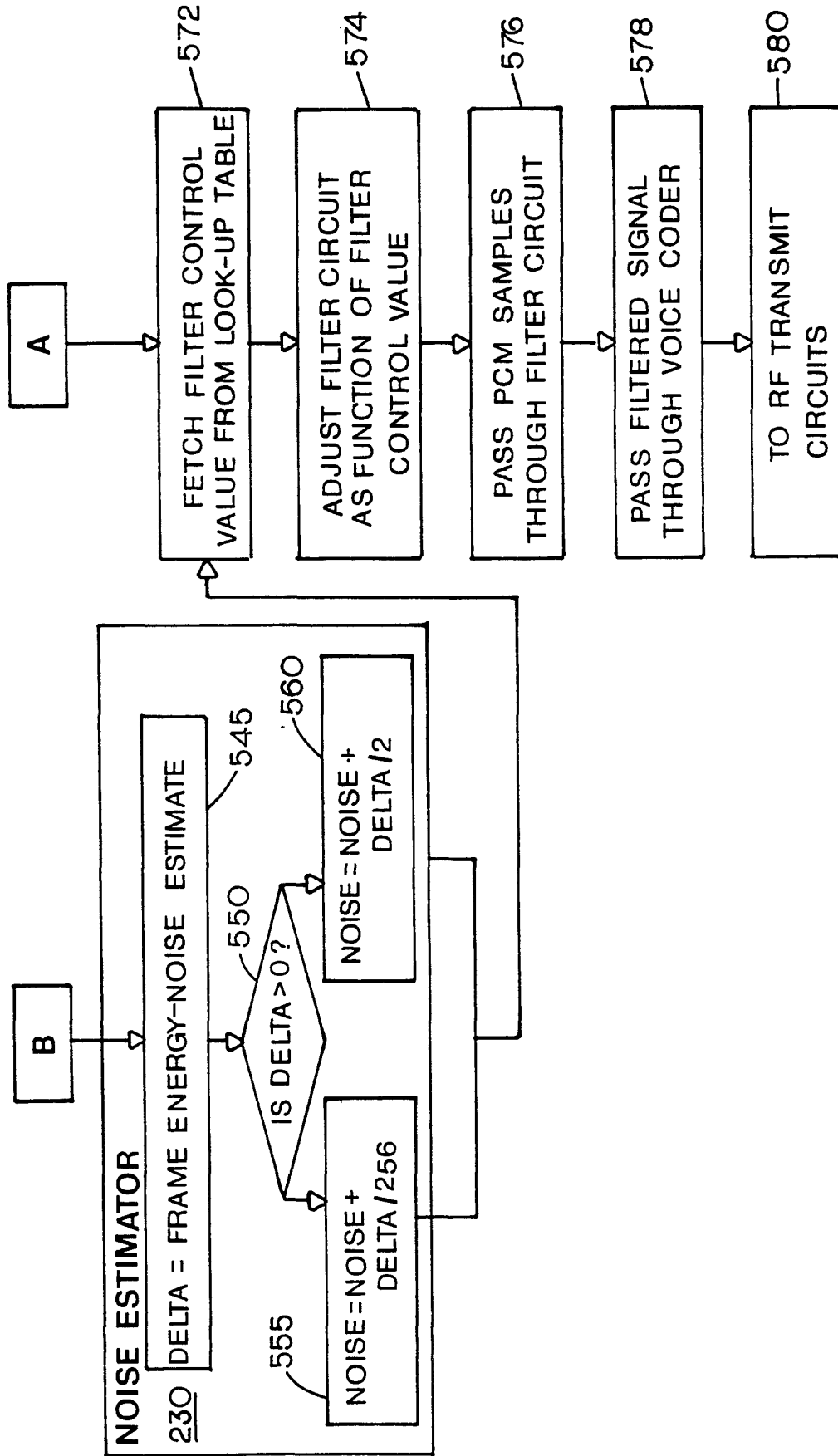


Fig. 5B

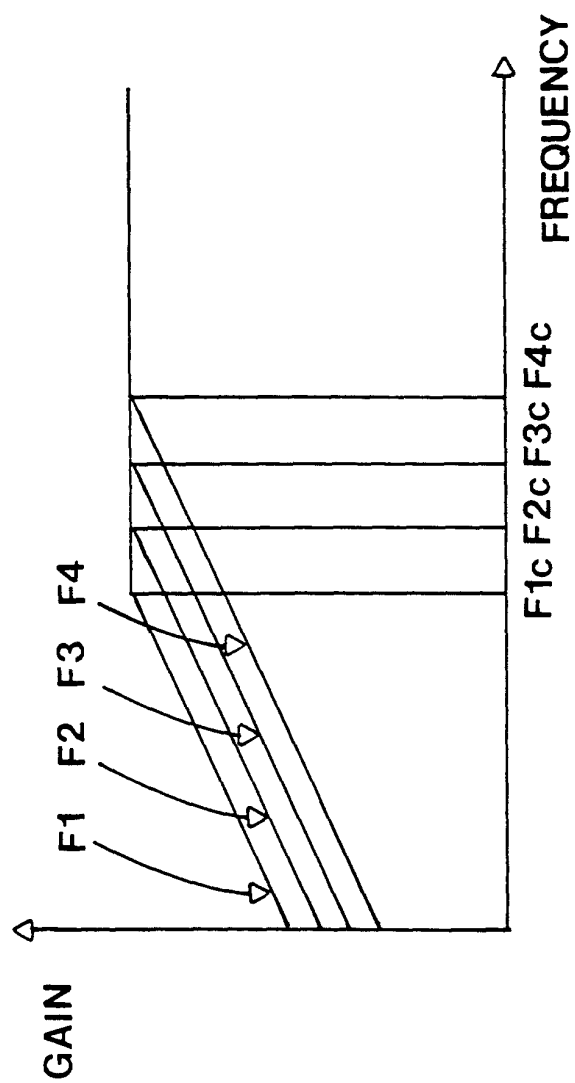


Fig. 6A

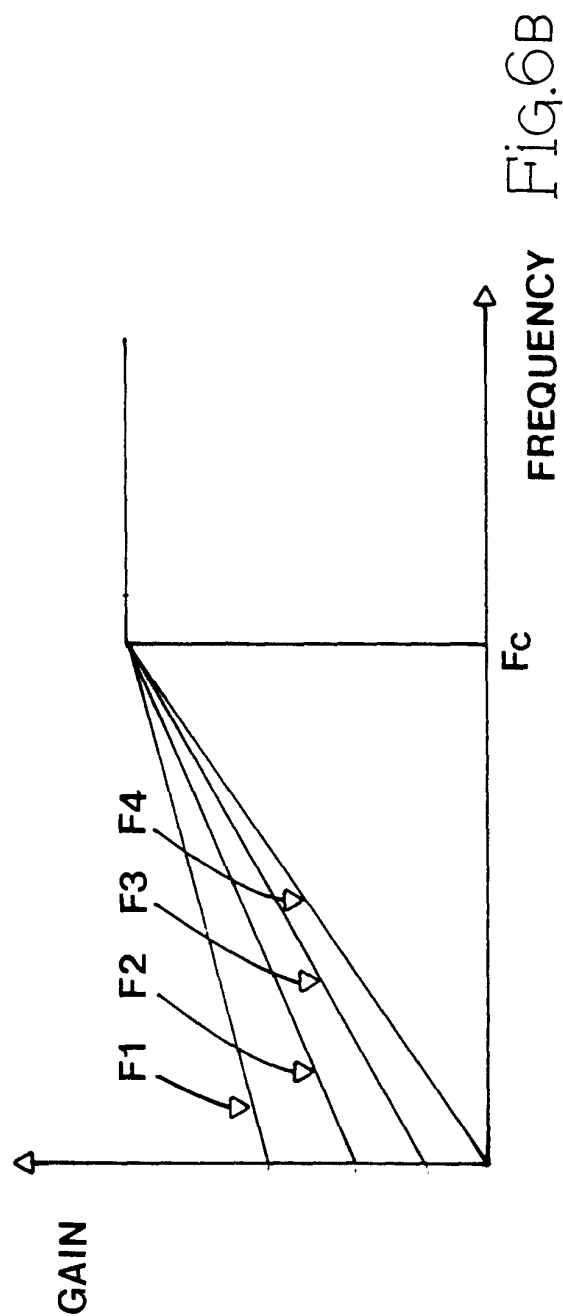
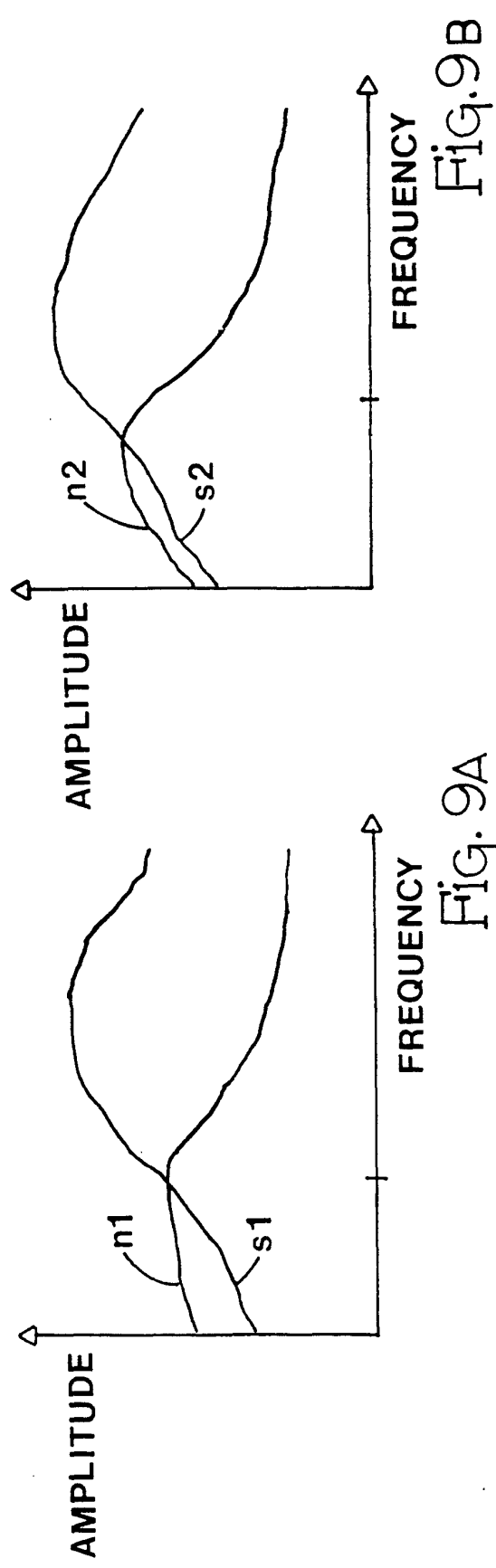
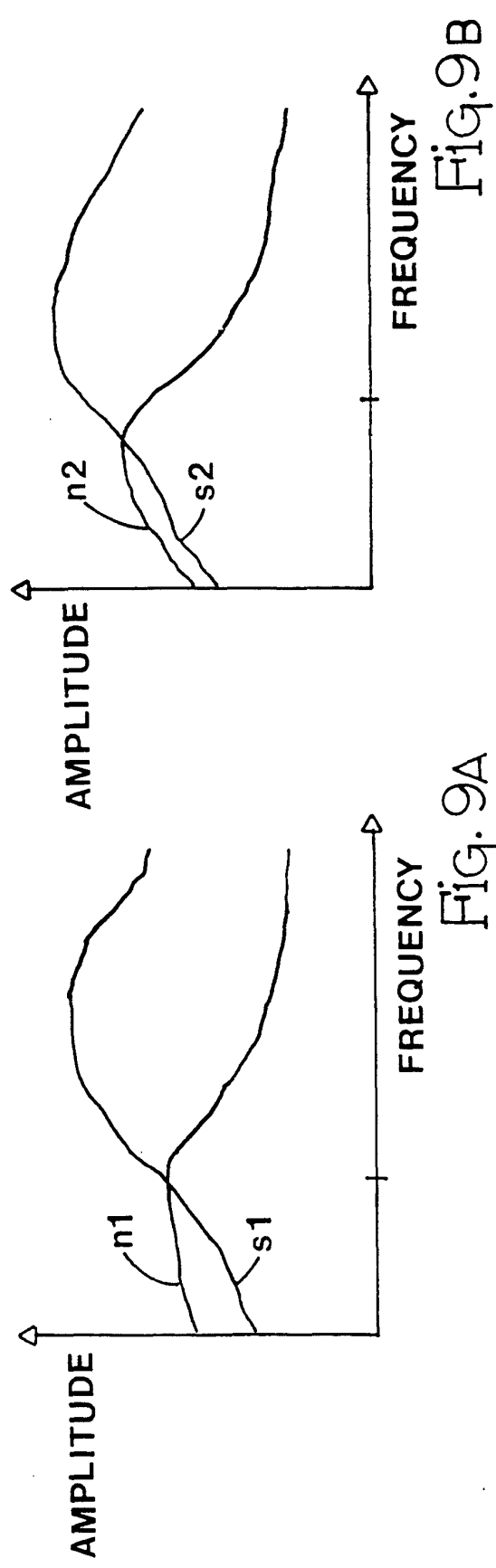
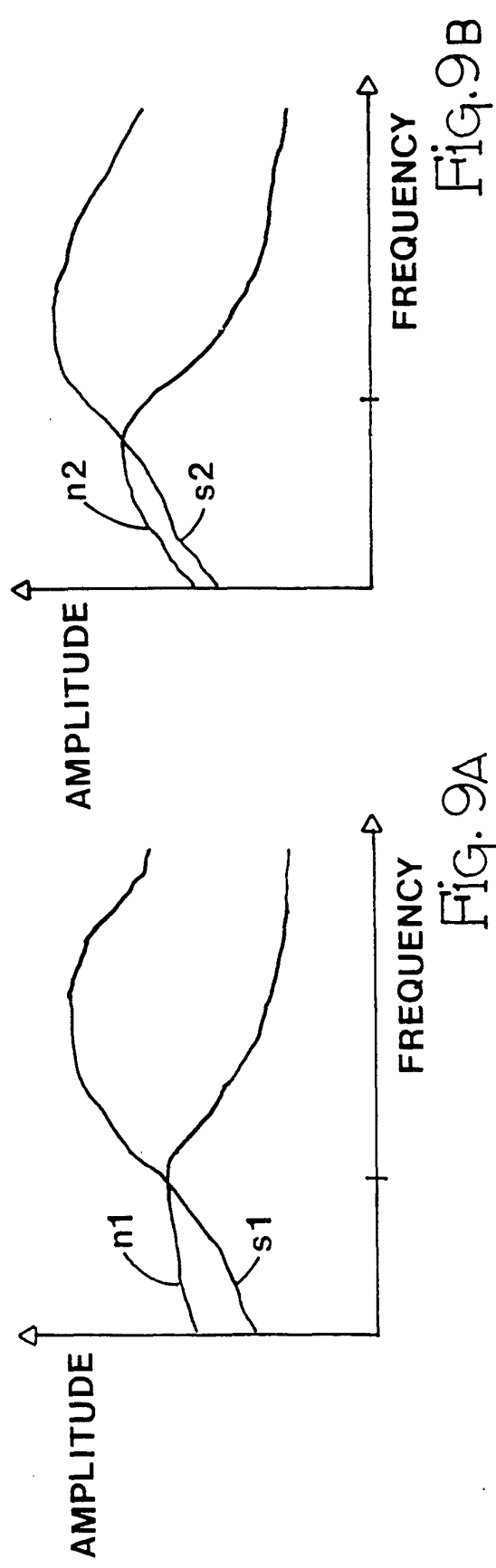
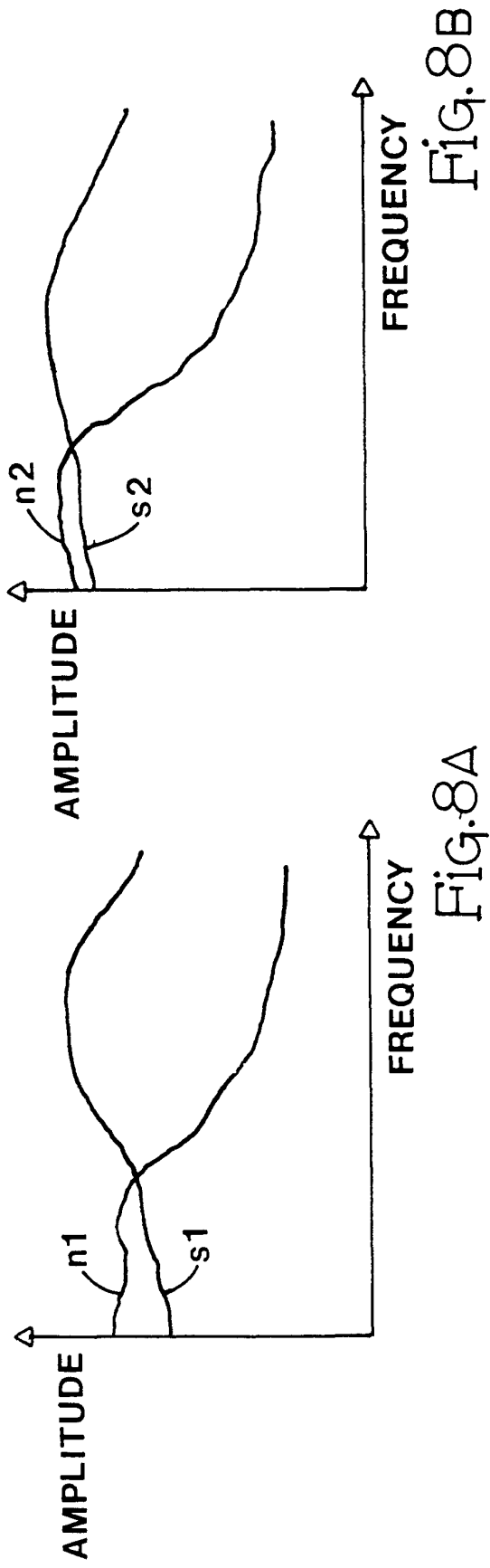


Fig. 6B

LOOK-UP TABLE

NOISE ESTIMAE	FILTER CONTROL VALUE
N1	F1
N2	F2
N3	F3
N4	F4
-	-
Nn	Fn

Fig.7



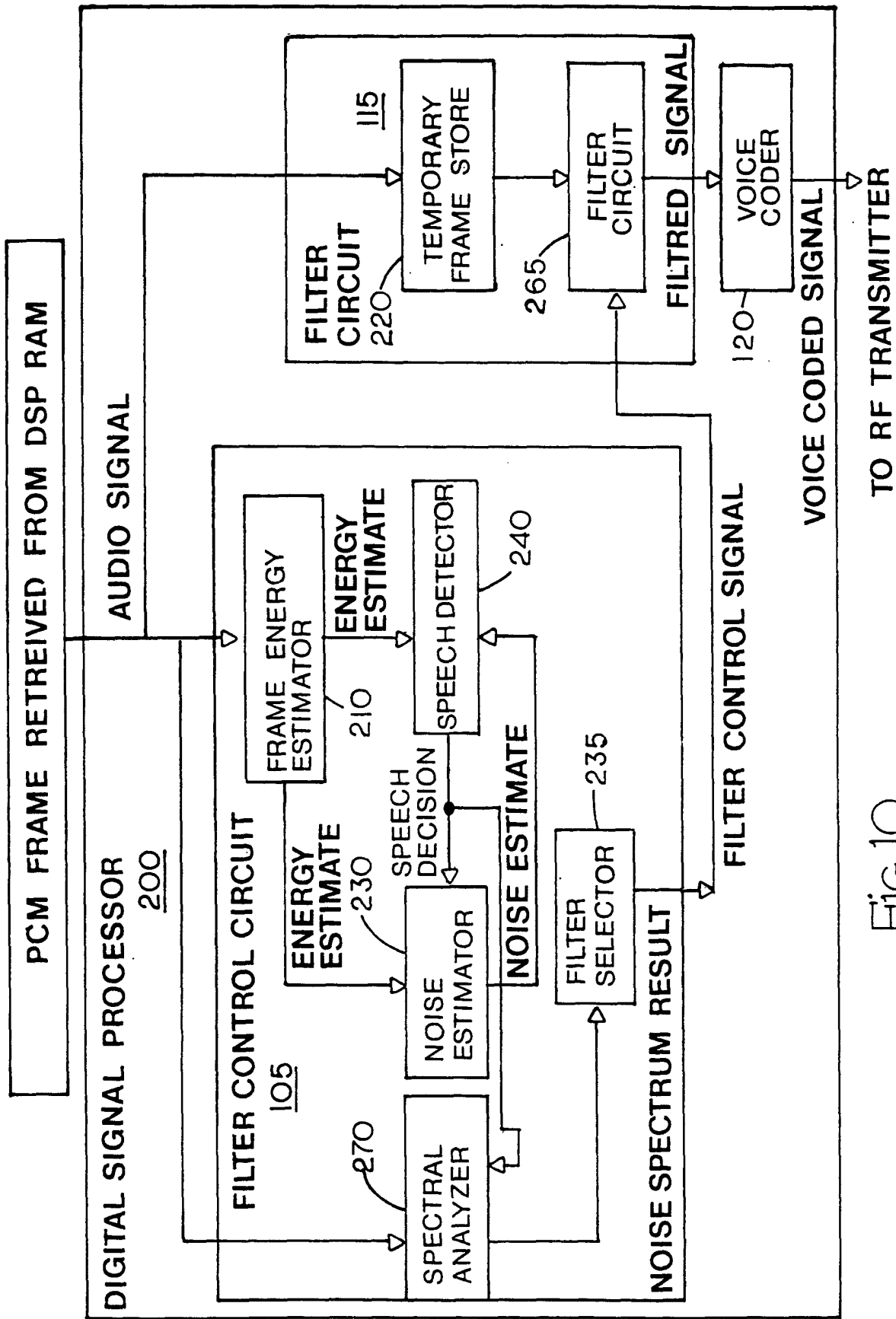
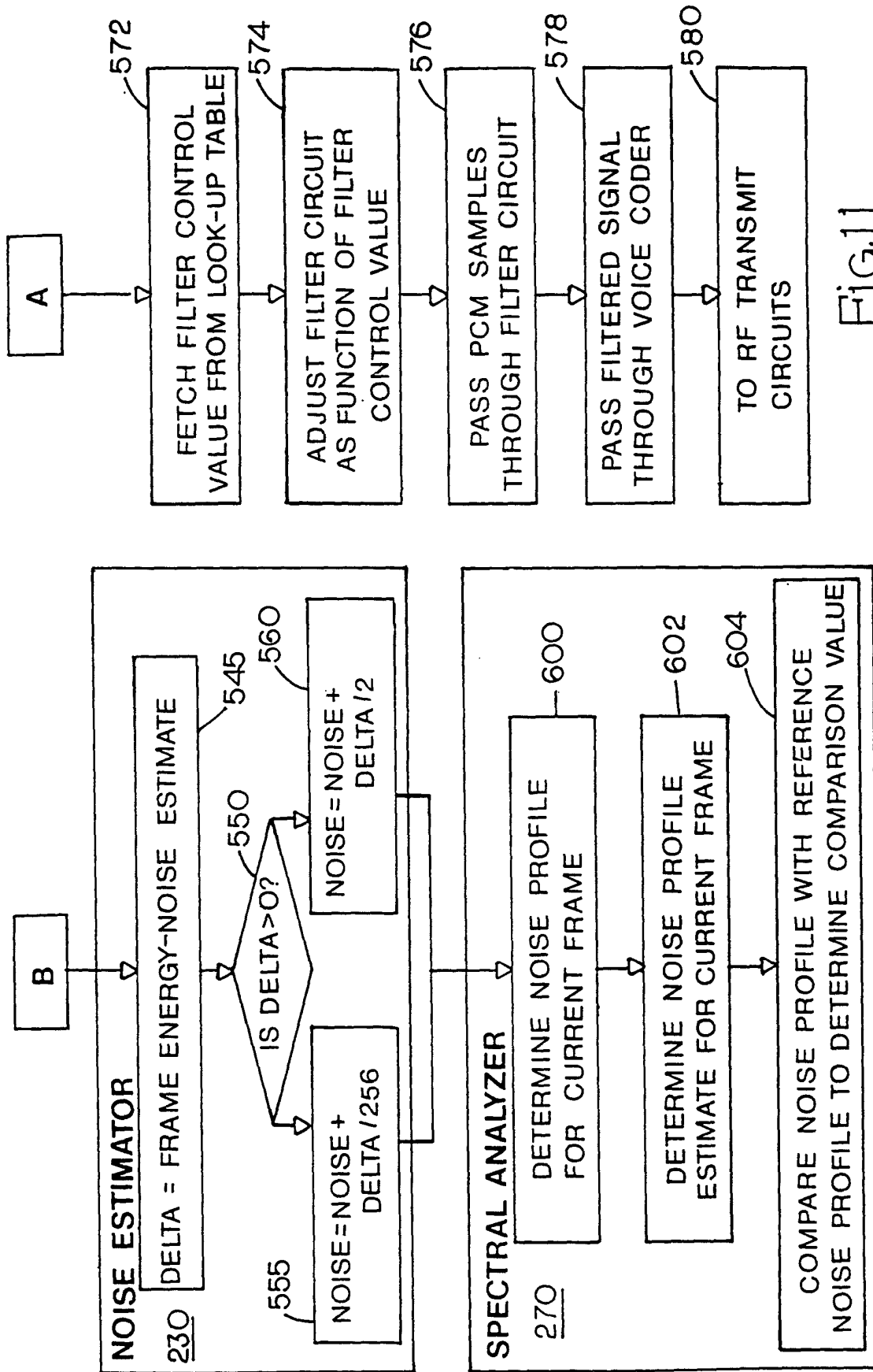


Fig.10



LOOK UP TABLE

NOISE ENERGY ESTIMATE COMPARISON VALUES	FILTER CONTROL VALUES
C1	F4
C2	F3
C3	..
Cn	Fn

Fig.12