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Yang et al.

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[54] NOISE-REDUCTION SYSTEM

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[21] Appl. No.: **47,556**

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[51] **Int. Cl.**[6] **H04B 15/00**

[52] **U.S. Cl.** **381/94**; 381/37; 381/47

[58] **Field of Search** 381/94, 37, 46, 47; 379/406, 392

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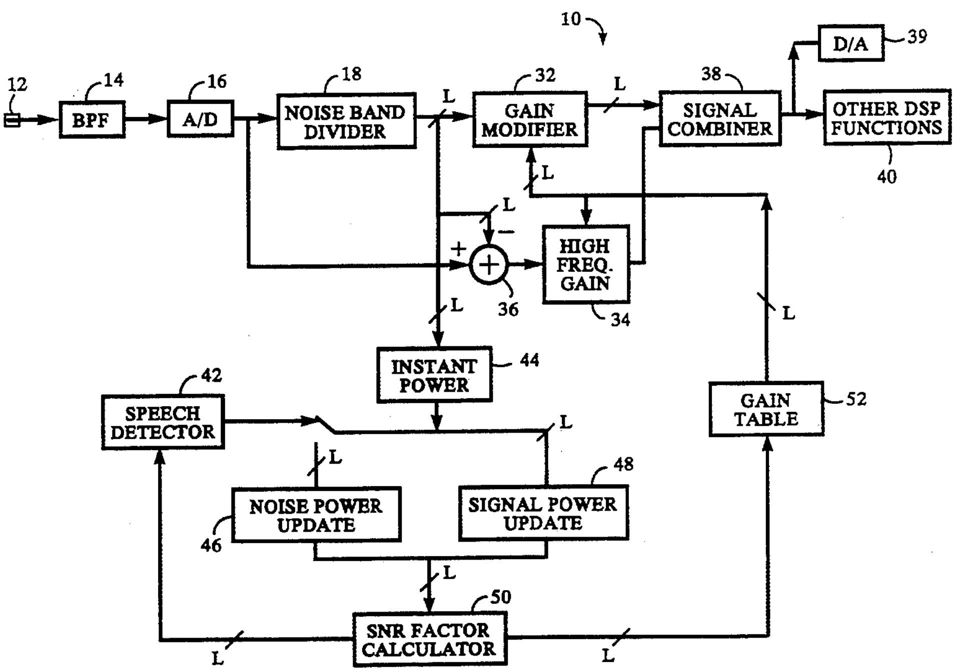
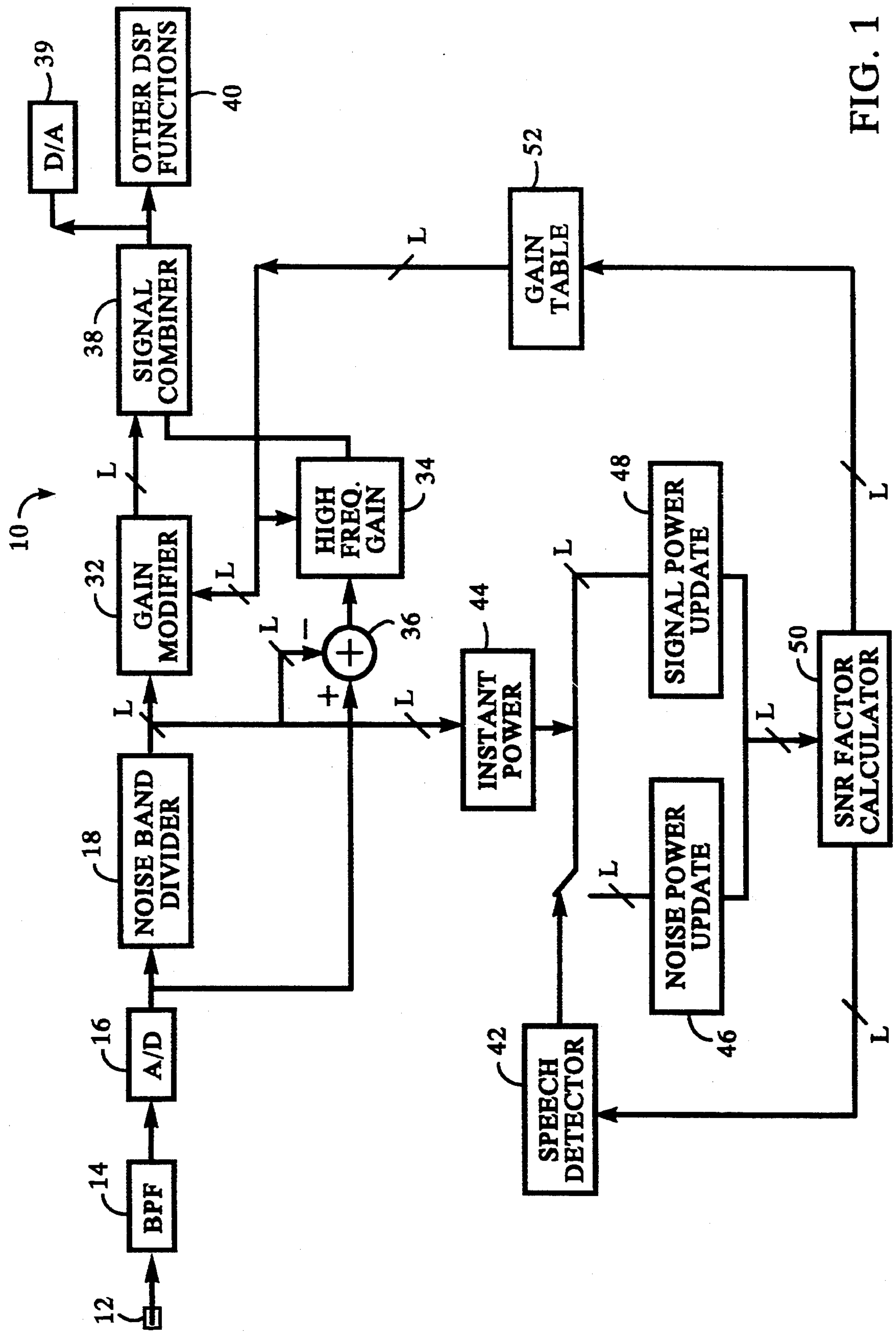
[57] **ABSTRACT**

A noise-suppression circuit (**10**) divides the signal from a microphone (**12**) into a plurality of frequency subbands by means of a noise-band divider (**18**) and a subtraction circuit (**36**). By means of gain circuits (**32**) and (**34**), it applies separate gains to the separate bands and then recombines them in a signal combiner (**38**) to generate an output signal in which the noise has been suppressed. Separate gains are applied only to the lower subbands in the voice spectrum. Accordingly, the noise-band divider (**18**) is required to compute spectral components for only those bands. By employing a sliding-discrete-Fourier-transform method, the noise-band divider (**18**) computes the spectral components on a sample-by-sample basis, and circuitry (**50, 52**) for determining the individual gains can therefore update them on a sample-by-sample basis, too.

16 Claims, 3 Drawing Sheets




10
12
BPF
14
A/D
16
NOISE BAND DIVIDER
18
GAIN MODIFIER
32
SIGNAL COMBINER
38
D/A
39
OTHER DSP FUNCTIONS
40
HIGH FREQ. GAIN
34
36
+
−
INSTANT POWER
44
SPEECH DETECTOR
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NOISE POWER UPDATE
46
SIGNAL POWER UPDATE
48
SNR FACTOR CALCULATOR
50
GAIN TABLE
52
L

FIG. 1

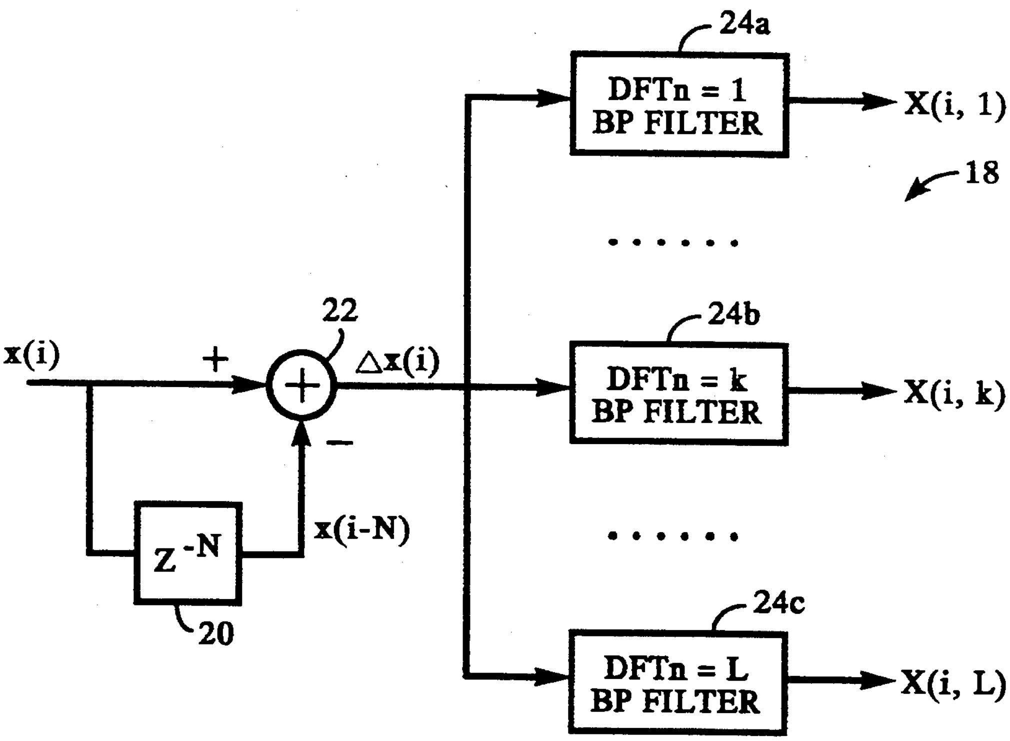
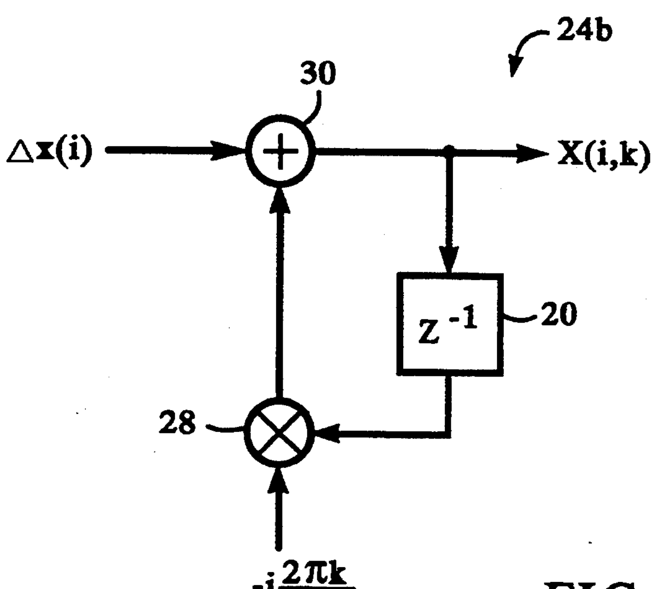

24a
DFTn = 1
BP FILTER
X(i, 1)
18
22
24b
x(i)
+
Δx(i)
DFTn = k
BP FILTER
X(i, k)
−
Z -N
x(i-N)
20
24c
DFTn = L
BP FILTER
X(i, L)

FIG. 2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X(i,k)
Z -1
20
28

FIG. 3

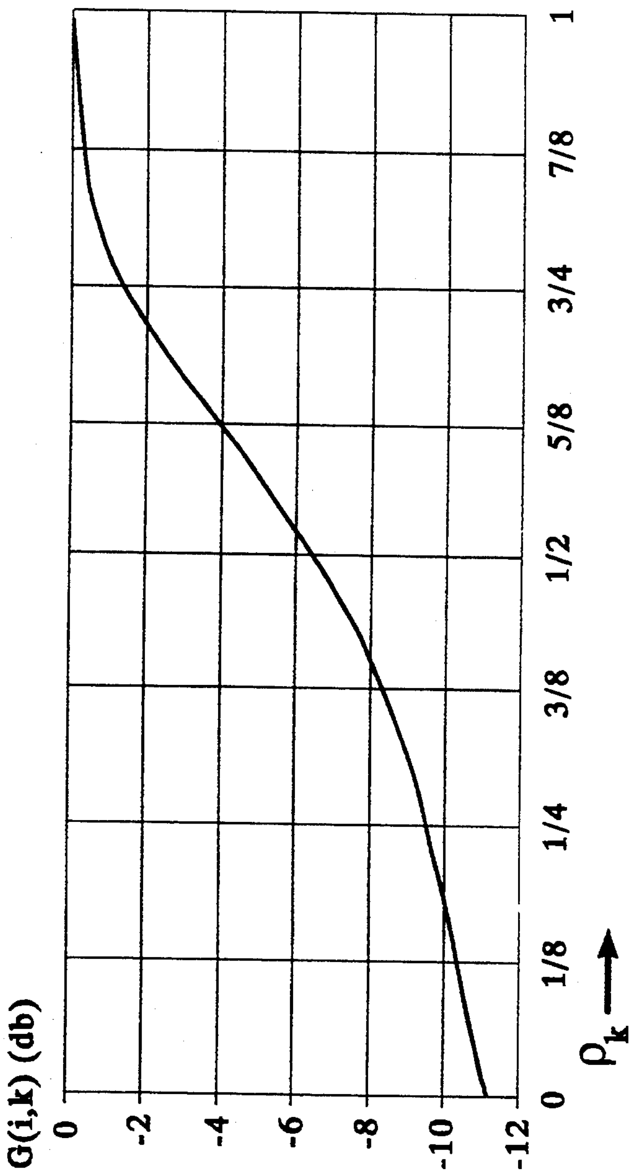

FIG. 4

NOISE-REDUCTION SYSTEM

BACKGROUND OF THE INVENTION

The present invention is directed to electronic devices for suppressing background noise of the type that, for example, occurs when a mobile-telephone user employs a hands-free telephone in an automobile.

A mobile-cellular-telephone user's voice often has to compete with traffic and similar noise, which tends to reduce the intelligibility of the speech that his cellular telephone set transmits from his location. To reduce this noise, a general type of noise-suppression system has been proposed in which the signal picked up by the microphone (i.e., speech plus noise) is divided into frequency bins, which are subjected to different gains before being added back together to produce the transmitted signal. (Of course, this operation can be performed at the receiving end, but for the sake of simplicity we will describe it only as occurring at the transmitter end.) The different gains are chosen by reference to estimates of the relationship between noise and voice content in the various bins: the greater the noise content in a given bin, the lower the gain will be for that bin. In this way, the speech content of the signal is emphasized at the expense of its noise content.

The noise-power level is estimated in any one of a number of ways, most of which involve employing a speech detector to identify intervals during which no speech is present and measuring the spectral content of the signal during those no-speech intervals.

Properly applied, this use of frequency-dependent gains does increase the intelligibility of the received signal. It nonetheless has certain aspects that tend to be disadvantageous. In the first place, many implementations tend to be afflicted with "flutter." A certain minimum record, or frame, of input signal is required in order to divide it into the requisite number of frequency bands, and the abrupt changes in the gain values at the end of each such record during non-speech intervals can cause a fluttering sound, which users find annoying. Methods exist for alleviating this problem, but they tend to have drawbacks of their own. For instance, some systems temporally "smooth" the gain values between input records by incrementally changing the gains, at each sample time during a frame, toward the gain dictated by the computation at the end of the last frame. This approach does largely eliminate the flutter problem, but it also reduces the system's responsiveness to changing noise conditions.

One could solve the frame problem by using a bank of parallel bandpass filters, each of which continually computes the frequency content of its respective band. But most commonly used bandpass-filter implementations would make obtaining the necessary resolution and reconstructing the gain-adjusted signals prohibitively computation-intensive for many applications.

Another drawback of conventional implementations of this general approach is that they distort the speech signal: the relative amplitudes of the frequency components in the transmitted signal are not the same as they were in the signal that the microphone received.

SUMMARY OF THE INVENTION

The present invention reduces these effects while retaining the benefits of the frequency-dependent-gain approach.

One aspect of the present invention, which is particularly applicable to mobile-cellular-telephone installations, takes advantage of the fact that background noise in automobile environments tends to predominate in the lower-frequency part of the speech band, while the information content of the speech falls disproportionately in the higher-frequency part. According to this aspect of the invention, gains are separately determined for different bands in the lower-frequency regions, as is conventional. But in the upper-frequency bins, which carry a significant part of the intelligibility, gains for different bins are kept equal. As a result, fewer Fourier components and fewer gain values need to be computed, but most of the noise-suppression effect remains, since it is the lower bands that ordinarily contain the most noise. Moreover, this approach can avoid most of the distortion that afflicts conventional frequency-dependent-gain approaches.

In employing this approach, we favor use of a gain function that approximates the maximum-likelihood function for high signal-to-noise ratios but approaches a predetermined value between −6 db and −20 db for low signal-to-noise ratios.

In accordance with another aspect of the invention, the gains to be employed for the various frequency bins are re-computed from the current noise contents at each sample time rather than only once each frame. This largely eliminates the flutter problem without detracting from the system's responsiveness to changing conditions. Without the present invention, such an approach might prove computationally prohibitive, because the frames used to compute the contents of the various frequency bins have to be heavily overlapped. In accordance with the present invention, however, the computation is performed by virtue of the "sliding discrete Fourier transform," whereby a Fourier component for a transform of an input record that ends with a given sample is computed from that sample, the corresponding Fourier component computed for the same-length frame that ended with the previous sample, and the sample with which that same-length frame began. That is,

$$X(i,k)=x(i)-x(i-N)+e^{-j2\pi k/N}X(i-1,k), \qquad (1)$$

where X(i,k) is the kth frequency component in an N-point discrete Fourier transformation taken over a record that ends with the ith sample, and x(i) is the ith sample of an input signal x from which the transform X is computed. By employing this "sliding DFT," as it is known in some signal-processing contexts, the computational burden that would otherwise result from re-computing the gains at each sample time is greatly reduced.

In accordance with yet another aspect of the invention, the speech detector determines whether speech is present by comparing with a threshold value an average of a plurality of factors ρ_k associated with respective frequency bins. Each ρ_k factor is the result of computing a first average of the Fourier components associated with that factor's associated frequency bin for samples that include those taken when the speech detector has indicated the presence of speech, computing a second average of Fourier components associated with that frequency bin for samples taken when the speech detector has indicated the absence of speech, and taking ρ_k as the ratio that the difference between the first and second averages bears to the first average.

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