(12) INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

(19) World Intellectual Property Organization
International Bureau

(43) International Publication Date
28 April 2005 (28.04.2005)

PCT

(10) International Publication Number
WO 2005/038470 A2

(51) International Patent Classification[7]: G01R

(21) International Application Number:
PCT/US2004/032594

(22) International Filing Date: 4 October 2004 (04.10.2004)

(25) Filing Language: English

(26) Publication Language: English

(30) Priority Data:
10/678,222 6 October 2003 (06.10.2003) US

(71) Applicant: **HARRIS CORPORATION** [US/US]; 1025 W. Nasa Blvd., Melbourne, FL 32919 (US).

(72) Inventor: **CHAMBERLAIN, Mark, Walter**; 740 H.F. Five Points Road, Honeoye Falls, NY 14472 (US).

(74) Agents: **YATSKO, Michael, S.** et al.; Harris Corporation, 1025 W. Nasa Blvd, Melbourne, FL 32919 (US).

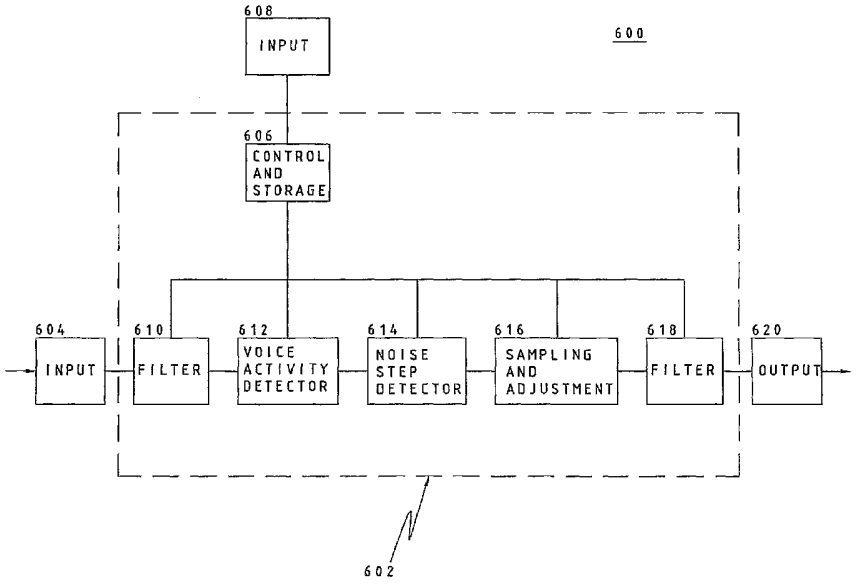
(81) Designated States *(unless otherwise indicated, for every kind of national protection available)*: AE, AG, AL, AM, AT, AU, AZ, BA, BB, BG, BR, BW, BY, BZ, CA, CH, CN, CO, CR, CU, CZ, DE, DK, DM, DZ, EC, EE, EG, ES, FI, GB, GD, GE, GH, GM, HR, HU, ID, IL, IN, IS, JP, KE, KG, KP, KR, KZ, LC, LK, LR, LS, LT, LU, LV, MA, MD, MG, MK, MN, MW, MX, MZ, NA, NI, NO, NZ, OM, PG, PH, PL, PT, RO, RU, SC, SD, SE, SG, SK, SL, SY, TJ, TM, TN, TR, TT, TZ, UA, UG, US, UZ, VC, VN, YU, ZA, ZM, ZW.

(84) Designated States *(unless otherwise indicated, for every kind of regional protection available)*: ARIPO (BW, GH, GM, KE, LS, MW, MZ, NA, SD, SL, SZ, TZ, UG, ZM, ZW), Eurasian (AM, AZ, BY, KG, KZ, MD, RU, TJ, TM), European (AT, BE, BG, CH, CY, CZ, DE, DK, EE, ES, FI, FR, GB, GR, HU, IE, IT, LU, MC, NL, PL, PT, RO, SE, SI, SK, TR), OAPI (BF, BJ, CF, CG, CI, CM, GA, GN, GQ, GW, ML, MR, NE, SN, TD, TG).

Published:
— *without international search report and to be republished upon receipt of that report*

[Continued on next page]

(54) Title: A SYSTEM AND METHOD FOR NOISE CANCELLATION WITH NOISE RAMP TRACKING



(57) Abstract: A system and method for noise cancellation with noise ramp tracking in the presence of severe or ramping acoustic noise. The system conducts an estimation of the noise level in the input signal and modifies the signal based upon this noise estimate. A windowed Fourier transform is performed upon the input speech signal and an estimation of a histogram of the frequency magnitudes of the noise level and other related parameters is generated and used to compute a spectral gain function that is applied to components of the Fourier transform of the input speech signal. The enhanced components of the Fourier transform are processed by an inverse Fourier transform in order to reconstruct a noise reduced speech signal.

WO 2005/038470 A2

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

A SYSTEM AND METHOD FOR NOISE CANCELLATION WITH NOISE RAMP TRACKING

The use of higher order statistics for noise suppression and estimation is well known. With higher order statistics it has been possible to derive more information from a received signal than with second order statistics which have commonly been used in telecommunications. For example, the phase of the transmission channel may be derived from the stationary received signal using higher order statistics. Another benefit of higher order statistic noise suppression is the suppression of Gaussian noise.

One such higher order statistic noise suppression method is disclosed by Steven F. Boll in "Suppression of Acoustic Noise in Speech Using Spectral Subtraction", IEEE Transactions on Acoustics, Speech, and Signal Processing, VOL. ASSP-27, No. 2, April 1979. This spectral subtraction method comprises the systematic computation of the average spectra of a signal and a noise in some time interval and afterwards through the subtraction of both spectral representations. Spectral subtraction assumes (i) a signal is contaminated by a broadband additive noise, (ii) a considered noise is locally stationary or slowly varying in short intervals of time, (iii) the expected value of a noise estimate during an analysis is equal to the value of the noise estimate during a noise reduction process, and (iv) the phase of a noisy, pre-processed and noise reduced, post-processed signal remains the same. Spectral subtraction and known higher order statistic noise suppression methods encounter difficulties when tracking a ramping noise source and do little to reduce the noise contamination in a ramping, severe or non-stationary acoustic noise environment.

For example, Figure 1 illustrates speech of a male speaker ("Tom's birthday is in June", "Frank's neighbor mowed

his lawn", "Clip the pens on the books") in the presence of Gaussian acoustic noise. The illustrated Gaussian noise source contains an amplitude increased ramp at a one dB/second rate. Many noise cancellation algorithms have difficulty tracking a moving noise source of this type. A real world example of this condition is speech that is recorded in a stationary noise level environment such as in recording speech outdoors with a car passing at a distance to the recording device. The noise increases with a relatively constant level and then decreases back down to a fixed stationary level.

Figure 2 illustrates speech of a female speaker ("Do not drink the Coke fast", "Please rent the car to him", "Invest your money now") recorded in the presence of CH-47 helicopter noise. The helicopter noise source is characterized by non-stationary noise and loud volumes resulting in poor Signal-to-Noise-Ratio (SNR) conditions. Typically, digital voice systems are often completely unusable for communications in the presence of such non-stationary noise. Hence, there exists a need in the art for a system and method to improve the intelligibility and quality of speech in the presence of ramping, severe or non-stationary acoustic noise environments.

Therefore, it is an object of the disclosed subject matter to overcome these and other problems in the art and present a novel method and system for noise cancellation with noise ramp tracking in the presence of ramping, severe or non-stationary acoustic noise environments.

It is an object of the disclosed subject matter to present a novel method to reduce the noise source of an input speech signal in a telecommunications system using minimal computational complexity. It is a further object to estimate the noise level present in an input speech signal when the noise source is ramping up or down in amplitude (at least 2-3

dB/second), to correctly identify speech segments from noise only segments so that speech may not degrade when noise levels are varied in amplitude, and to automatically adapt the resulting post-processed noise level to a suitable level even when noise is not present in the input speech.

It is also an object of the disclosed subject matter to present a novel method to filter the noise source of an input speech signal by estimating the noise level present, modify the input speech signal based on the noise estimate, identify and separate speech segments from noise only segments, and adapt post-processed noise levels to an acceptable level.

It is a further object of the disclosed subject matter to present a novel method of noise cancellation by applying a windowed Fourier transform to an input speech signal, estimating the noise level present in an input speech signal, modifying the input speech signal based on the noise estimate, identifying speech segments from the noise only segments, and adapting post-processed noise levels to acceptable levels.

It is an object of the disclosed subject matter to present a novel system for noise cancellation in a severe acoustic environment comprising an input device operably connected to a processor, a processor operably connected to an electronic memory and storage device wherein the processor conducts a noise cancellation technique, a filter for adapting post-processed noise levels to acceptable levels, a storage device operably connected to the processor for storing and applying noise histograms for further noise processing, and an output device operably connected to the processor for communicating the output speech signal.

These and many other objects and advantages of the present invention will be readily apparent to one skilled in

the art to which the invention pertains from a perusal of the claims, the appended drawings, and the following detailed description of the preferred embodiments.

BRIEF DESCRIPTION OF THE DRAWINGS

The subject matter of the disclosure will be described with reference to the following drawings:

FIGURE 1 illustrates input speech in the presence of Gaussian acoustic noise with a ramping noise level increase of 1 dB per second (male speaker - "Tom's birthday is in June", "Frank's neighbor mowed his lawn", "Clip the pens on the books");

FIGURE 2 illustrates input speech in the presence of CH47 Helicopter noise (female speaker - "Do not drink the Coke fast", "Please rent the car to him", "Invest your money now");

FIGURE 3 illustrates the noise reduced output speech for the input speech shown by Figure 1;

FIGURE 4 illustrates the noise reduced output speech for the input speech shown by Figure 2;

FIGURE 5 illustrates the flowchart of the noise cancellation algorithm according to the invention;

FIGURE 6 illustrates a schematic block diagram of a noise cancellation system according to the invention.

DETAILED DESCRIPTION OF DRAWINGS

Embodiments of the disclosed subject matter enhance a speech input signal through an estimation of the noise level in the input signal and a modification based upon this noise estimate. The estimation of the noise level is made in the frequency domain after performing a windowed Fourier transform on the input speech signal. A histogram of the frequency magnitudes of the noise level and other related parameters is generated, estimated and used to compute a spectral gain

function that is multiplied with the real and imaginary components of the Fourier transform of the input speech signal. The enhanced components of the Fourier transform may then be processed by an inverse Fourier transform to reconstruct the noise reduced speech signal.

Figure 3 illustrates the enhanced speech output for the input speech signal shown by Figure 1. An embodiment of the disclosed subject matter tracks the Gaussian noise source containing an amplitude ramp increased at a 1 dB/second rate and effectively reduces the noise to acceptable noise levels. A voice activity detector 507, as illustrated by Figure 5, detects and compensates the ramping noise.

Figure 4 illustrates the enhanced speech output for the input speech signal shown by Figure 2. The resultant speech output has been noise compensated and may be perceived as noise-free. As exemplified by Figure 4, noise in unvoiced speech segments has been reduced by approximately 20 dB. It is also shown that noise levels in voiced segments have been reduced to a level that provides a Signal-to-Noise Ratio ("SNR") improvement and perceived quality enhancement. Though this example of non-stationary noise may be considered a difficult noise type to reduce or compensate, an embodiment of the disclosed subject matter provides speech that may be suitable for communications.

An embodiment for enhancing speech output for an input noise source is illustrated by Figure 5. Figure 5 represents a specific embodiment in which an input speech signal is enhanced by an estimation of a noise level in the input speech signal. A windowed Fourier transform may then be applied to the input speech signal. The windowed Fourier transform controls the spectral leakage between frequency bins of the Fourier transform by controlling the bandwidth of each frequency bin. An application and modification of a histogram

is used to compute a gain function of the input signal which may be applied to the components of the input signal after the application of the windowed Fourier transform. Processing of this modified signal may be conducted using an inverse Fourier transform to produce a noise reduced speech output signal.

As shown in Block 501, an encoded input speech signal may be overlapped and added with previous input signals. The input speech signal may be assigned a frame size respective to its overlapped state. As shown in Blocks 502 and 503, a windowed Fourier transform is applied to the real and imaginary components of the input speech signal. The magnitude of the input speech signal may be approximated through an absolute value estimation in the frequency domain after the performance of the windowed Fourier transform as shown in Block 504.

Block 505 represents a computation of the input speech signal Signal-to-Noise Ratio ("SNR") magnitudes. As shown, a magnitude approximation of the input speech signal may be multiplied by an arbitrary value and divided by the noise level of the input speech signal. An SNR maximum value may be assigned according to the magnitude approximation and forwarded to a filter as exemplified in Block 506. The filter computes an average SNR magnitude through a total summation of SNR magnitude extremes and of a doubling of the sum of all intermediate SNR magnitudes. The total summation is divided by an arbitrary value to compute an average SNR magnitude. The filter further computes an average input speech signal magnitude through a total summation of signal magnitude extremes and of a doubling of the sum of all intermediate signal magnitudes. The total summation is divided by an arbitrary real value to compute an average input speech signal magnitude.

As depicted by Block 507, a voice activity detector may detect and attack a ramping, Gaussian or non-stationary noise signal through conditional comparisons between maximum SNR magnitudes and a maximum SNR threshold, the SNR average magnitude and an average SNR threshold, and a weighted average signal magnitude and an average noise magnitude multiplied by an average SNR threshold. As exemplified by Block 508, a noise step detector detects and attacks a large noise step increase or decrease in amplitude or magnitude and generates a histogram of the frequency magnitudes of the noise level and other related parameters through a conditional comparison and assignment of historical voice activity detection values, historical signal values and noise step values. As represented by Block 509, a spectral gain function is estimated in the input speech signal through conditional comparisons of the input speech signal's noise level, signal gain, and other related parameters.

As depicted by Block 510, the spectral gain function computed and estimated in Block 509 is utilized to reduce noise in the input speech signal through a multiplicative application applied to the real and imaginary components of the Fourier transform of the input speech signal. The input speech signal may then be processed by an inverse Fourier transform, as illustrated by Block 511, to reconstruct a noise reduced speech signal prior to a slew rate adjustment. As depicted by Block 512, a sample of the slew rate from the noise reduced speech signal is taken and an error count is applied to the slew rate dependent upon the signal magnitude.

As illustrated in Block 514, the slew rate is adjusted in the frequency domain through conditional comparisons and computations of error periods, error counts, histograms and peak indices of the input speech signal and other parameters. If the histogram of the sample is greater

than a peak of the sample, then the value of the peak is assigned the histogram value and a peak index is assigned an arbitrary value. However, if the peak value of the sample is greater than zero and the peak index is greater or less than zero and greater or less than an arbitrary value, then the histogram values may be adjusted higher if an error function is greater than an upper slew value or the histogram values may be adjusted lower if the error function is lower than a lower slew value. After slew rate adjustment, the sample may be encoded as represented in Block 515 by indexing the signal magnitude. Further, a noise histogram may be updated as a function of an encoded noise sample as depicted in Block 516, and the noise histogram may be normalized as exemplified in Block 517 through further conditional comparisons and computations of the updated noise histogram value and a maximum historical value. If the updated noise histogram is greater than the maximum historical value, the histogram may be scaled down or normalized as a function of the difference between the updated noise histogram and the maximum historical value. As represented by Block 518, a weighted histogram bin is computed through a summation of the normalized histogram and indexed by a weighted mean. A noise estimate may then be decoded according to the weighted histogram computation and index as illustrated in Block 519. Further slew rate adjustment may be conducted depending upon the frequency domain of the reconstructed noise reduced speech signal.

After slew rate adjustment is complete, a windowed Fourier transform is multiplicately applied to the components of an output speech signal as depicted by Block 522. The output speech signal may be overlapped and added with previous output signals after a performance of the windowed Fourier transform as illustrated in Block 523. Further, the output

speech signal may be assigned a frame size respective to its overlapped state.

A noise filter, as exemplified by Block 524, may filter any average remaining noise component of the output speech signal through a total summation of noise magnitude extremes and of a doubling of the sum of all intermediate noise magnitudes. The summation is divided by a predetermined value to compute an average noise magnitude. The noise cancellation process may be continued if further input speech signals or if new speech frames are present.

A representative algorithm of an embodiment of the noise cancellation process exemplified in Figure 5 is shown below for illustrative purposes only and is not intended to limit the scope of the described method.

Generic algorithm

Magnitude Approximation

```
MagApproximation (x,y)
{x = abs(x)
  y = abs(y)
  if (x<y)
      {temp = x
       x = y
       y = temp}
 if (x>8*y) temp = x
      else {temp = (15*x+7*y)/16}
 return(temp)}
EncodeSample(x)
{index = 0
 big = MAX_POS_VAL
 for j = 0 to 127
      {temp = abs(ENCODE_TABLE[j] - x)
       if (temp<big)
            {big = temp
```

```
            index = j}}
 return(index)}
Block 501 Overlap and add with previous input
SpeechInput[0,…,OVERLAP-1] = SpeechInput[FRAMESIZE,…,FFTSIZE-1]
SpeechInput[OVERLAP,…,FFTSIZE-1] = AudioInput[0,…,FRAMESIZE-1]
Block 502 Apply windowed Fourier Transform
Sig[0,…,FFTSIZE-1] = WINDOW[0,…,FFTSIZE-
1]*SpeechInput[0,…,FFTSIZE-1]
Block 503 Apply Fourier Transform
Sig[0,…,FFTSIZE-1] = FFT(Sig[0,…,FFTSIZE-1]) {256 point real
value FFT}
Block 504 Magnitude Approximation
SigMag[0] = abs(Sig[0])
SigMag[1,…,FFTBINLEN-2] = MagApproximation(Sig[1,…,FFTBINLEN-
2],   Sig[FFTLEN-1,…,FFTLEN-FFTBINLEN+2])
SigMag[FFTBINLEN-1] = abs(Sig[FFTBINLEN-1])
Block 505 Compute SNR magnitudes
Snr[0,…,FFTBINLEN-1] = 256*SigMag[0,…,FFTBINLEN-
     1]/Noise[0,…,FFTBNILEN-1]
SnrMax = MAX(Snr[0,…,FFTBINLEN-1])
Block 506 Filter SNR and signal magnitudes
SnrAvg = (Snr[0] + Snr[128] + 2*SUM(Snr[1,…,127]))/256
AvgSignalMag = (SigMag[0] + SigMag[128] +
2*SUM(SigMag[1,…,127]))/256
Block 507 Voice Activity Detector
NoiseFlag = 0
If (SnrMax < MAX_SNR_THRESHOLD && SnrAvg < AVG_SNR_THRESHOLD)
     NoiseFlag = 1
If (256*AvgSignalMag > AVG_SNR2_THRESHOLD*AvgNoiseMag)
NoiseFlag = 0
Block 508 Noise Step Detector
AllVoice = 1
If (VADHist[0,…,31] = = 0) AllVoice = 0
```

```
Max = 0
Min = MAX_POS_VAL
If (SignalHist[0,…,31] > Max) Max = SignalHist[0,…,31]
If (SignalHist[0,…,31] < Min) Min = SignalHist[0,…,31]
If (ALLVoice && Max < 2*Min && NoiseStep = = 0)
{NoiseStep = 32
 Histogram[0,…,FFTBINLEN-1][0,…,127] = 0
 Noise[0,…,FFTBINLEN-1] = SigMag[0,…,FFTBINLEN-1]}
else if (NoiseStep > 0) NoiseStep = NoiseStep-1
SignalHist[31,…,1] = SignalHist[30,…,0]
VADHist[31,…,1] = VADHist[30,…,0]
SignalHist[0] = AvgSignalMag
VADHist[0] = NoiseFlag XOR 1
Block 509 Estimate gain
for j = 0 to FFTBINLEN-1
{acc = 256*MAX_GAIN
 if (Snr[j] <> 0) acc = acc/Snr[j]
 if (acc> MAX_GAIN) acc = MAX_GAIN
 Nsr = acc
 Temp = (Nsr*SCALE1 + OldNsr[j]*SCALE2)
 Hgain[j] = MAX_GAIN - temp
 If (NoiseFlag) Hgain[j] = MINGAIN
 Else
      {if (Snr[j] > SNR3_THRESHOLD)Hgain[j] = MAXGAIN}
 if (Hgain[j] < MINGAIN) Hgain[j] = MINGAIN
 OldNsr[j] = Nsr}
Block 510 Noise Reduction
Sig[0] = Hgain[0]*Sig[0]
Sig[1,…,FFTBINLEN-2] = Hgain[1,…,FFTBINLEN-
2]*Sig[1,…,FFTBINLEN-2]
Sig[FFTLEN-1,…,FFTLEN-FFTBINLEN+2] = Hgain[FFTLEN-1,…,FFTLEN-
      FFTBINLEN+2] * Sig[FFTLEN-1,…,FFTLEN-FFTBINLEN+2]
Sig[FFTBINLEN-1] = Hgain[FFTBINLEN-1]*Sig[FFTBINLEN-1]
```

```
Block 511 Inverse Fourier Transform
Sig[0,…,FFTSIZE-1] = IFFT(Sig[0,…,FFTSIZE-1])  {real value 256
point Inverse FFT}
SigMag[0,…,FFTBINLEN-1] = (SigMag[0,…,FFTBINLEN-1] +
      OldSigMag[0,…,FFTBINLEN-1])/2
OldSigMag[0,…,FTBINLEN-1] = SigMag[0,…,FFTBINLEN-1]
Block 512 Slew rate sample
If (NoiseStep > 0) AttackRate = FAST_ATTACK_INC
Else  AttackRate = SLOW_ATTACK_INC
If (NoiseFlag)
{Error[0,…,128] = Error[0,…,128] + SigMag[0,…,128]*NOISE_BIAS
 ErrorCount = ErrorCount + 1}
ErrorPeriod = ErrorPeriod + 1
for i = 0 to FFTBINLEN-1
Block 513 Start Counter
LOOPCOUNT = 0
{
      Block 514 Slew rate adjustment
      if (ErrorPeriod = = 16 && ErrorCount <> 0)
      {acc = Error[i]/ErrorCount
       acc = 256*acc/Noise[i]
       Peak = PeakIndex = 0
       For j = 0 to 127
            {if (histogram[i][j] > Peak)
                 {Peak = histogram[i][j]
                  PeakIndex = j}}
       if (Peak > 0 && PeakIndex <> 0 && PeakIndex <> 127)
            {if (acc > SLEW_UPPER)
                 {histogram[i][127,…,1] = histogram[i][126,…,0]
                  histogram[0] = 0}
             else if (acc < SLEW_LOWER)
                 {histogram[i][0,…,126] = histogram[i][1,…,127]
                  histogram[i][127] = 0}}}
```

```
        Block 515 Encode Noise Sample
        stuffindex = EncodeSample(SigMag[i])
        Block 516 Update noise histogram
        temp = histogram[i][stuffindex]
        temp = temp + AttackRate
        histogram[stuffindex] = temp
        Block 517 Normalize histogram
        if (temp >MAX_HIST_VALUE)
        {ScaleDownHist = temp - MAX_HIST_VALUE
         for j = 0 to 127
              {histogram[i][j] = histogram[i][j] - ScaledDownHist
               if (histogram[i][j] < 0) histogram[i][j]}}
        Block 518 Compute weighted histogram bin
        sum = 0
        for j = 0 to 127
        {sum = sum + histogram[i][j]}
        acc = 0
        for j = 0 to 127
        {acc = acc + j*histogram[i][j]}
        mean = 256*acc/sum
        index3 = mean/256
        Block 519 Decode noise estimate
        Noise[i] = ENCODE_TABLE[index3] }
    if (ErrorPeriod = = 16)
    {ErrorPeriod = ErrorCount = 0
     Error[0,…,128] = 0}
    Block 520 Update Counter
    LOOPCOUNT = LOOPCOUNT + 1
    Block 521
    If LOOPCOUNT = FFTBINLEN, continue
    else, GOTO Slew Rate Adjustment
    Block 522 Apply window
```

```
SpeechOutput[0,…,FFTSIZE-1] = WINDOW[0,…,FFTSIZE-
1]*Sig[0,…,FFTSIZE-1]
Block 523 Overlap and add to previous output
SpeechOutput[0,…,OVERLAP-1] = SpeechOutput[0,…,OVERLAP-1] +
     Overlap[0,…,OVERLAP-1]
Overlap[0,…,OVERLAP-1] = SpeechOutput[FRAMESIZE,…,FRAMESIZE +
     OVERLAP-1]
AudioOut[0,…,FRAMESIZE-1] = SpeechOutput[0,…,FRAMESIZE-1]
Block 524 Noise Filter
AvgNoiseMag = (Noise[0] + Noise[128] +
2*SUM(Noise[1,…,127))/256
Block 525
If more speech, continue process
if new FRAME, GOTO step 1
else STOP
```

An embodiment of the disclosed subject matter in which the previously described process may be implemented is illustrated in Figure 6 as system 600. The system 600 includes a processor 602 operably connected to a first input means 604, a second input means 608, and an output means 620. The processor 602 comprises a control and storage means 606, a first filtering means 610, a voice activity detector 612, a noise step detector 614, a sampling and adjustment means 616, and a second filtering means 618. The control and storage means 606 may be used to store a control program which carries out computational aspects of the noise cancellation process previously described and to control the computations of the aforementioned components. Such a control and storage means 606 may comprise of but is not limited to any various known storage devices such as a CD-ROM drive, a hard disk, etc. upon which an embodiment of the algorithm depicted in Figure 5 may be stored. The first input means 604 may comprise of but is

not limited to a communications receiver, audio receiver, or like device that may receive electromagnetic signals. The second input means 608 may comprise a keyboard or similar input device in which historical data may be entered into the control and storage means 606 for access by the processor 602 and other components.

An input speech signal is received by the first input means 604 and relayed to the processor 602 wherein an estimation of the noise level is conducted and a windowed Fourier transform may be applied to the input speech signal within the processor 602. The signal magnitude and SNR may be filtered by a filtering means 610 within the processor 602 and delivered to a voice activity detector 612 wherein several noise types such as but not limited to ramping, non-stationary, and Gaussian may be detected and attacked. The filtering means may comprise of but is not limited to known filters such as low pass filters, band pass filters, or other known filters utilized in the filtering of electromagnetic signals and designed for specific electromagnetic signal parameters of an embodiment of the disclosed subject matter. The signal may then be relayed to a noise step detector 614 wherein a large noise step increase or decrease in amplitude or magnitude may be detected and attacked.

The input speech signal is further processed and a spectral gain function is computed and applied to the real and imaginary components of the Fourier transform of the input speech signal in the processor 602. These components are then processed by an inverse Fourier transform for reconstruction of the signal. The signal may be relayed for further processing, slew rate sampling and adjusting, noise histogram updating and noise histogram normalizing in a sampling and adjustment means 616. The sampling and adjustment means may comprise but is not limited to an electronic circuit or the

like designed to sample an input signal wherein adjustments to specific parameters of the input signal may be made according to comparisons of the sampled parameters. If this processing is complete, a windowed Fourier transform may be applied to the signal and the signal may be overlapped and added with other previous outputs. If the slew rate adjustment and noise histogram updating and normalizing has not been fully completed, further iterations may be performed. Upon processing of the signal, the signal may be relayed to a filtering means 618 in which remaining noise components are filtered out. The signal is then passed to any number of output means 620 comprising of but not limited to an audio or visual output device, a storage medium or the like.

CLAIMS

1. A method of reducing a noise component of an input speech signal comprised of signal frames on a channel comprising the steps of:

(a) applying a windowed Fourier transformation to said signal frames;

(b) approximating signal magnitudes of said signal frames;

(c) computing Signal-to-Noise Ratio magnitudes of said signal frames;

(d) detecting voice activity in said channel;

(e) detecting noise activity in said channel;

(f) estimating gain in said signal frames;

(g) applying an estimated noise history to said signal frames to compute a spectral gain function;

(h) applying said spectral gain function to the components of said windowed Fourier transformation; and,

(i) applying an inverse Fourier transform to said signal frames thereby reconstructing a noise reduced output signal frame.

2. The method of Claim 1 comprising the step of filtering said Signal-to-Noise Ratio magnitude and signal magnitude prior to detecting voice activity in said channel.

3. The method of Claim 1 wherein said detecting voice activity comprises conditional comparisons of received Signal-to-Noise Ratios and average Signal-to-Noise Ratio thresholds.

4. The method of Claim 1 comprising the step of sampling a slew rate of said noise reduced output signal frame.

5. The method of Claim 4 wherein the step of sampling a slew rate comprises the steps of:

(a) starting a counter;
(b) adjusting the sampled slew rate;
(c) encoding a noise sample;
(d) updating a noise histogram;
(e) normalizing said noise histogram;
(f) computing a weighted histogram bin;
(g) decoding a noise estimate;
(h) updating said counter; and,
(i) deciding to continue said sampling.

6. A system for noise cancellation comprising:

(a) a first input means operably connected to a processor said first input means receiving a speech signal;
(b) a second input means operably connected to said processor wherein historical speech and noise data may be entered into a control and storage means for access by said processor;
(c) an output means operably connected to said processor said output means expressing an output speech signal; and,
(d) a processing means operably connected to said first and second input means and said output means, said processing means comprising a control and storage means, a first filtering means, a second filtering means, a voice

activity detector, a noise step detector, and a sampling and adjustment means.

7. The system of Claim 6 wherein said first filtering means filters Signal-to-Noise Ratio magnitudes and signal magnitudes.

8. The system of Claim 6 wherein said sampling and adjustment means samples and adjusts a slew rate and a histogram of said output speech signal.

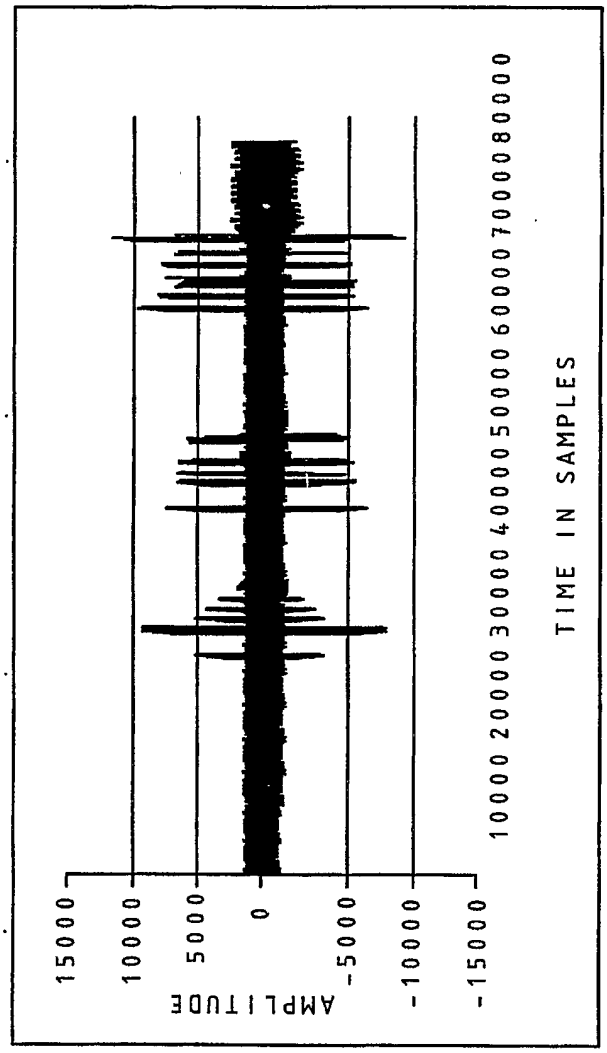

AMPLITUDE
15000
10000
5000
0
-5000
-10000
-15000
10000 20000 30000 40000 50000 60000 70000 80000
TIME IN SAMPLES

FIGURE 1

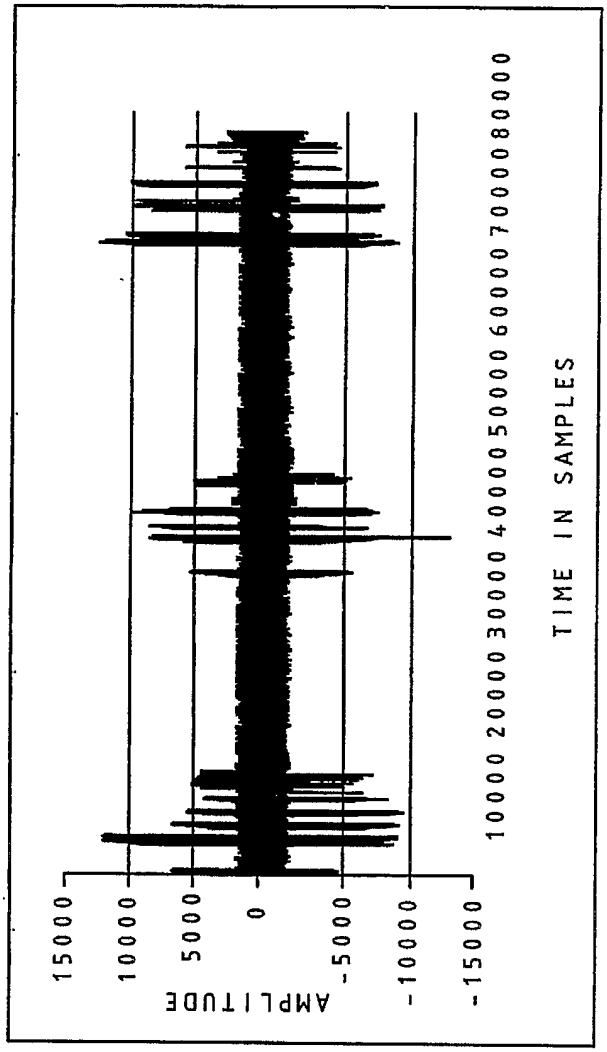

AMPLITUDE
15000
10000
5000
0
-5000
-10000
-15000
10000 20000 30000 40000 50000 60000 70000 80000
TIME IN SAMPLES

FIGURE 2

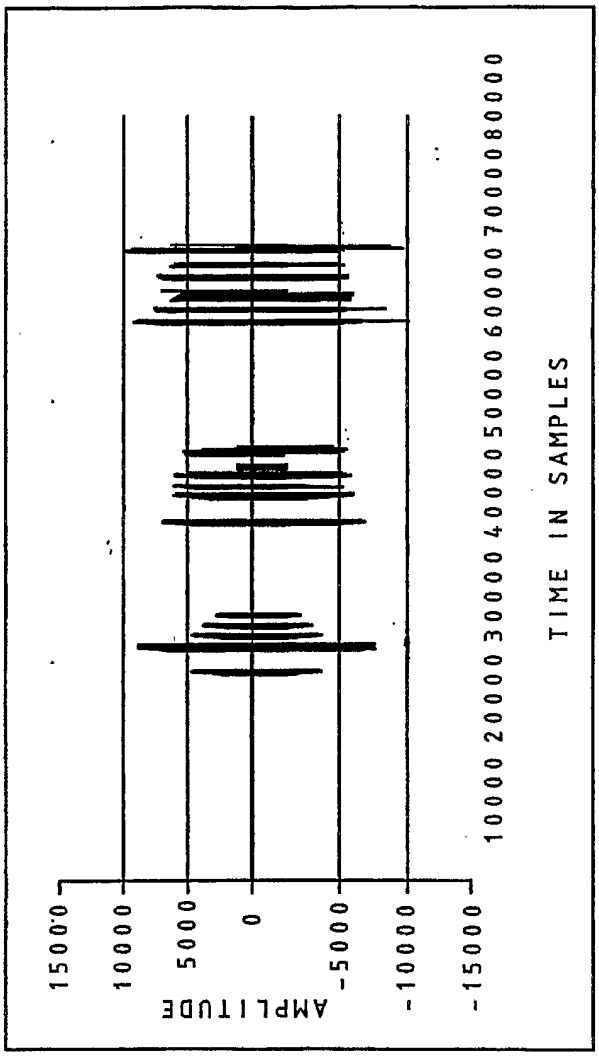

AMPLITUDE
15000
10000
5000
0
-5000
-10000
-15000
10000 20000 30000 40000 50000 60000 70000 80000
TIME IN SAMPLES

FIGURE 3

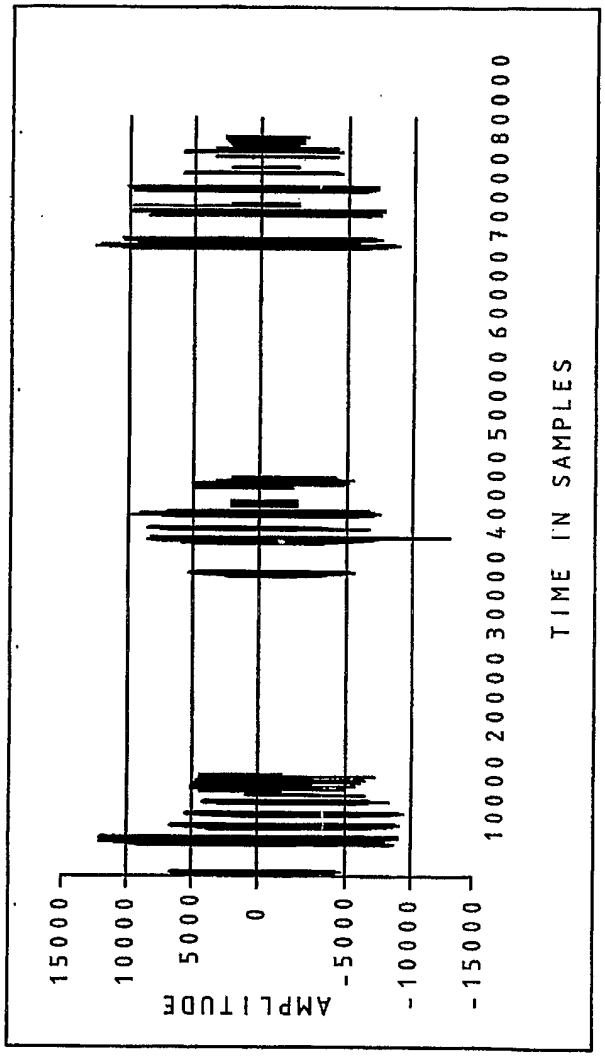

AMPLITUDE
15000
10000
5000
0
-5000
-10000
-15000
10000 20000 30000 40000 50000 60000 70000 80000
TIME IN SAMPLES

FIGURE 4

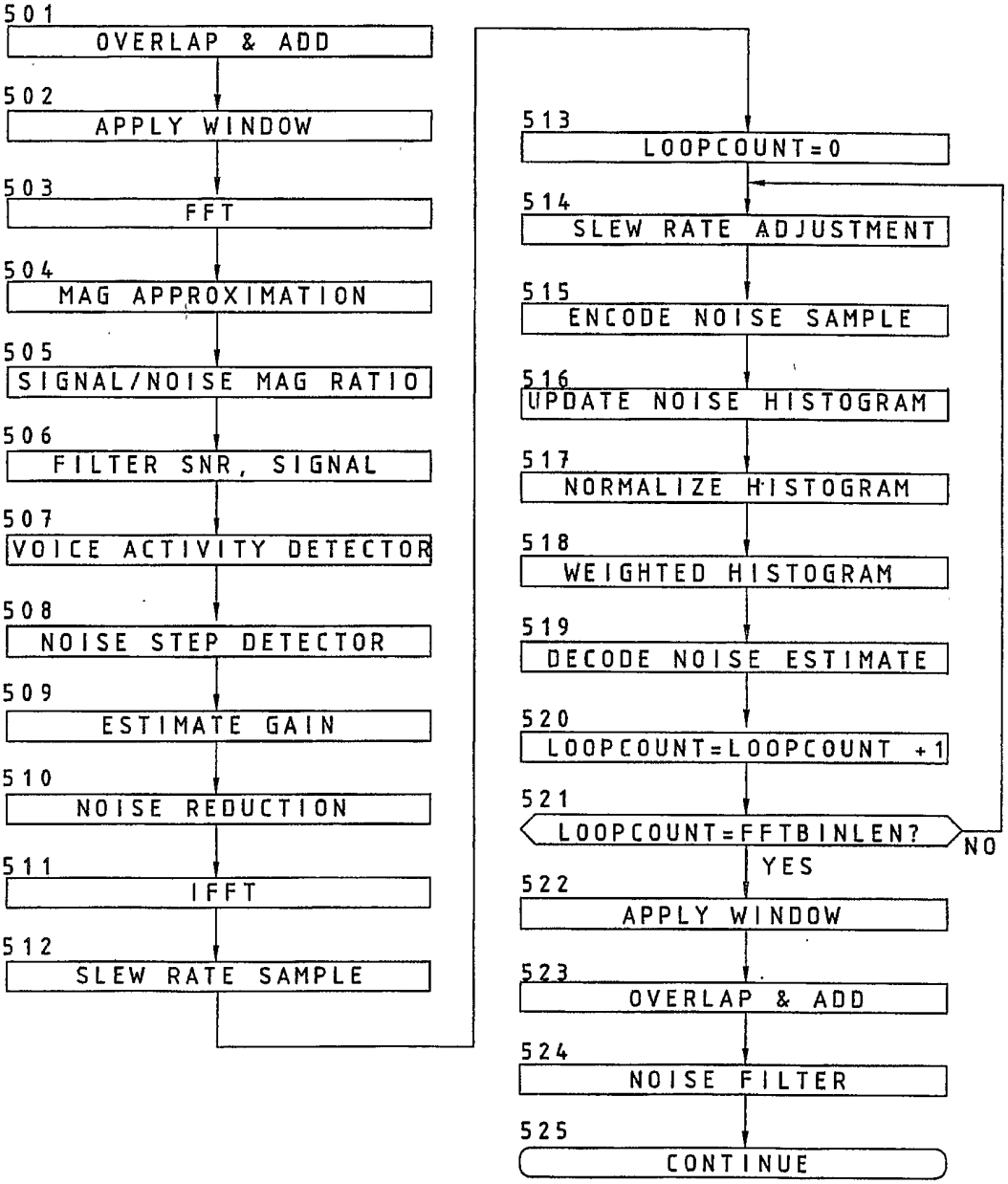

501
OVERLAP & ADD
502
APPLY WINDOW
503
FFT
504
MAG APPROXIMATION
505
SIGNAL/NOISE MAG RATIO
506
FILTER SNR, SIGNAL
507
VOICE ACTIVITY DETECTOR
508
NOISE STEP DETECTOR
509
ESTIMATE GAIN
510
NOISE REDUCTION
511
IFFT
512
SLEW RATE SAMPLE
513
LOOPCOUNT=0
514
SLEW RATE ADJUSTMENT
515
ENCODE NOISE SAMPLE
516
UPDATE NOISE HISTOGRAM
517
NORMALIZE HISTOGRAM
518
WEIGHTED HISTOGRAM
519
DECODE NOISE ESTIMATE
520
LOOPCOUNT=LOOPCOUNT +1
521
LOOPCOUNT=FFTBINLEN?
NO
YES
522
APPLY WINDOW
523
OVERLAP & ADD
524
NOISE FILTER
525
CONTINUE

FIGURE 5

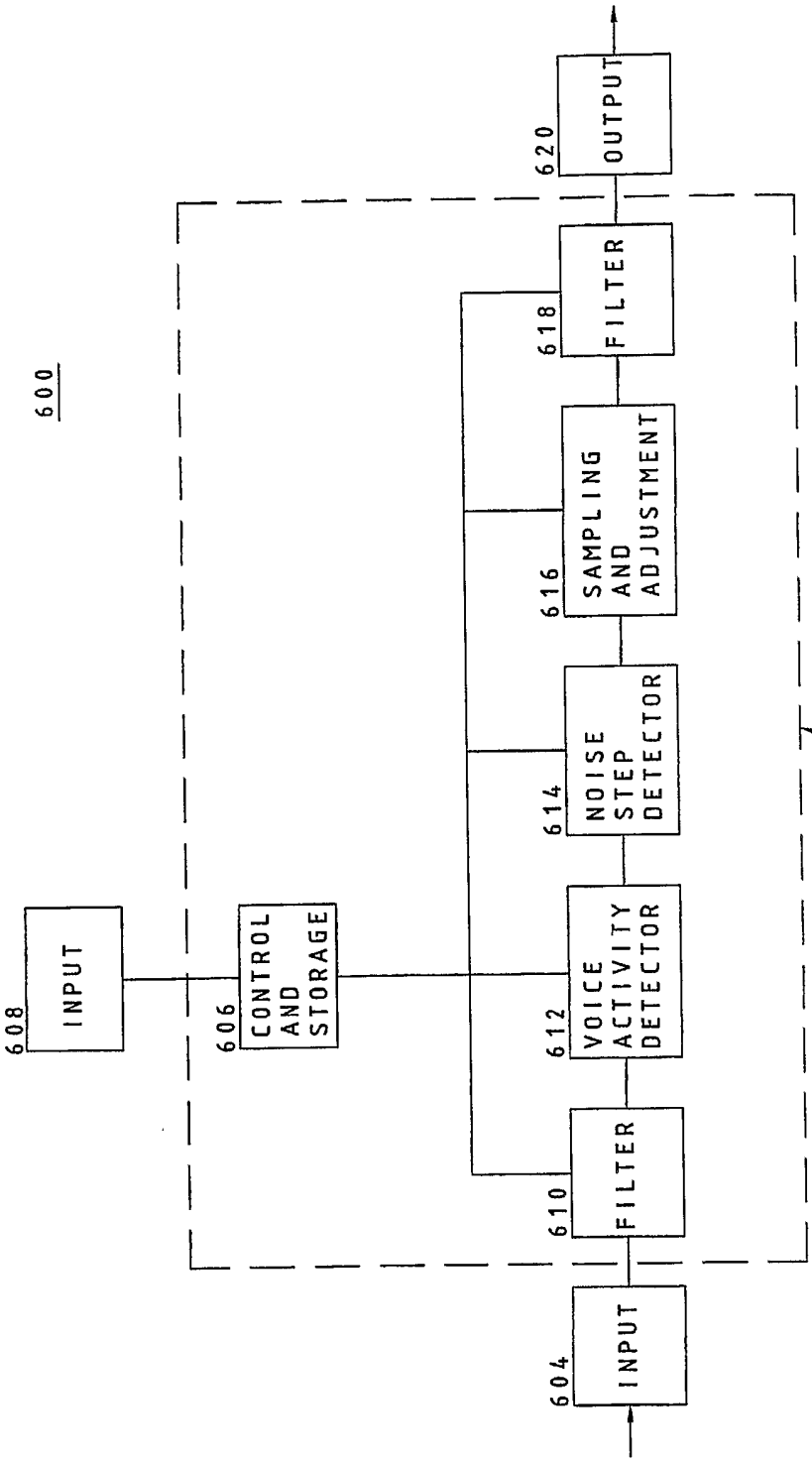

600
602
604
INPUT
606
CONTROL AND STORAGE
608
INPUT
610
FILTER
612
VOICE ACTIVITY DETECTOR
614
NOISE STEP DETECTOR
616
SAMPLING AND ADJUSTMENT
618
FILTER
620
OUTPUT

FIGURE 6