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

(11) EP 0 984 433 A2

(12) **EUROPEAN PATENT APPLICATION**

(43) Date of publication:
08.03.2000 Bulletin 2000/10

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

(21) Application number: **99117227.1**

(22) Date of filing: **02.09.1999**

(84) Designated Contracting States:
AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE
Designated Extension States:
AL LT LV MK RO SI

(30) Priority: **04.09.1998 GB 9819224**

(71) Applicant: **MOTOROLA LTD**
Basingstoke, Hampshire RG22 4PD (GB)

(72) Inventor: **Chan, Dominic Sai Fan**
Walthamstow, London E17 6NS (GB)

(74) Representative:
Dunlop, Hugh Christopher
Motorola,
European Intellectual Property Operations
Midpoint
Alencon Link
Basingstoke, Hampshire RG21 7PL (GB)

(54) Noise suppresser speech communications unit and method of operation

(57) A speech communications unit is provided. The speech communications unit includes a speech processor for receiving an input speech signal having a periodic noise interferer (50, 70), the speech processor being operably coupled to a noise determining means (52, 72) for determining an amplitude of the periodic noise interferer a gaussian noise generator (54, 74) for generating a known gaussian noise sequence and combining the speech signal with the known gaussian noise sequence to provide a resultant signal, and inputting the resultant signal into a noise suppression procedure to provide a noise suppressed speech signal. The speech processor determines an amplitude level of the suppressed gaussian noise and subtracts a respective level of suppressed gaussian noise from the resultant signal thereby reducing the periodic noise content in the speech signal.

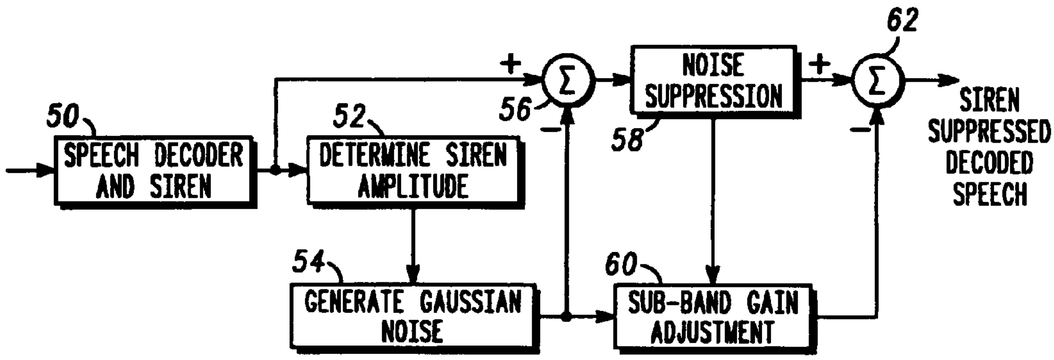

FIG. 3

EP 0 984 433 A2

Description

Field of the Invention

[0001] This invention relates to suppressing noise in communications systems and more particularly to suppressing periodic noise from car engines or police sirens in a mobile communications system.

Background of the Invention

[0002] Many voice communications systems, such as the TErrestrial Trunked RAdio (TETRA) system for private mobile radio users, use speech processing units to encode and decode speech patterns. In such voice communications systems the speech encoder converts the analogue speech pattern into a suitable digital format for transmission and the speech decoder converts a received digital speech signal into an appropriate analog speech pattern.

[0003] As spectrum for such voice communications systems is a valuable resource, it is desirable to limit the channel bandwidth used, to maximise the number of users per frequency band. Hence, the primary objective in the use of speech coding techniques is to reduce the occupied capacity of the speech patterns as much as possible, by use of compression techniques, without losing fidelity of speech signals.

[0004] Speech coding typically uses speech production modelling techniques to compress pulse code modulation (PCM) speech signals into bit-rates that are suitable for different kinds of bandwidth-limited applications such as speech communication systems or voice storage systems.

[0005] The basic speech production model, that is commonly used in speech coding algorithms, is shown in FIG. 1. The model in FIG. 1 was used in early linear predictive coding (LPC) based vocoders. The LPC filter models the combined effect of the glottal pulse model, the vocal tract and the lip radiation. For voiced speech, the voiced excitation, which consists of a pulse train separated by the pitch duration T, is used as an input signal to the LPC filter. Alternatively, for unvoiced speech, a gaussian noise source is used as the LPC filter input excitation.

[0006] The advance of speech coding development led to the introduction of Analysis by Synthesis technique used in CELP (Code Excited Linear Prediction) such as (Algebraic Code Excited Linear Prediction). The improved speech production model or the synthesis model used in the ACELP case is shown in FIG. 2.

[0007] The excitation in the ACELP case is a weighted combination of the innovative codebook vector and the adaptive codebook vector. Typically research papers on the subject matter of CELP-based speech coding techniques refer to two codebooks, namely an ìinnovativeî codebook as the basic codebook for CELP, in order to distinguish the codebook from the ìadaptiveî codebook. The innovative codebook in the ACELP case consists of code-vectors each contains only a small number of pulses and zero value elsewhere. The periodicity of the excitation, which is needed for voiced speech, derives from the last frame total LPC filter input excitation based on the present frame pitch lag value.

[0008] The main customers of TETRA radios are public safety organisations. The noise level in the mobile operating environment is often higher than that in fixed telecommunication systems. There are mainly two kinds of noise that will affect the TETRA speech quality, namely wideband noise and periodic noise. Wideband noise comes from the various operating environments; such as car or wind noise, street noise, babble noise. Periodic noise mainly comes from repetitive motion in car engines, as well as the sirens of public safety vehicles. For car engine noise, the fundamental of the periodic noise is mainly concentrated at low frequencies, typically of the order of less than 250 Hz.

[0009] Siren signal suppression is sometimes necessary especially during interconnect duplex communication. It has been argued by J. R. Deller Jr., J. G. Proakis and J. H. L. Hansen in ìDiscrete-Time Processing of Speech signalsî, published by MacMillan in 1993 that narrow-band noise sources such as a varying sinusoidal signal or an artificial noise component fatigue the auditory system faster than wideband noise.

[0010] One problem with developing algorithms to suppress siren signals is that the siren signal spectrum fulfils the conditions which most noise suppression algorithm uses to decide whether the incoming signal is a speech signal.

[0011] Thus it is desirable to suppress periodic noise in speech codecs, particularly in the mobile communication environment.

Summary of the Invention

[0012] According to a first aspect of the invention, a speech communications unit is provided. The speech communications unit includes a speech processor for receiving an input speech signal having a periodic noise interferer. The speech processor is operably coupled to a noise determining means for determining an amplitude of the periodic noise interferer a gaussian noise generator for generating a known gaussian noise sequence and combining the speech signal with the known gaussian noise sequence to produce a resultant signal, and inputting the resultant signal into a noise suppression procedure to provide a noise suppressed speech signal, wherein the speech processor determines an amplitude level of the suppressed gaussian noise and subtracts a respective level of suppressed gaussian noise from the resultant signal thereby reducing the periodic noise content in the speech signal.

[0013] In this manner, the introduction of a gaussian noise signal to the periodic interferer, into the speech signal reduces the periodic noise content of the signal.

Preferably, the gaussian noise signal generated is of substantially equal amplitude to the periodic interferer. Additionally, the speech communications unit further includes a noise suppresser function, coupled to the output of the summing junction, for further suppressing noise in the speech signal. The speech processor is either a speech post-processing function in a speech decoder or a speech pre-processing function in a speech encoder. The speech communications unit preferably includes a gain adjuster, operably coupled to the gaussian noise generator and the noise suppresser function for receiving the known gaussian noise sequence and the noise suppressed signal, the gain adjuster being operably coupled to a second summing junction for recombining the gain adjusted signal with the noise suppressed signal thereby suppressing the periodic noise interferer.

[0014] In accordance with a second aspect of the preferred embodiment of the invention, a method of reducing a periodic interferer in a speech signal is provided. The method includes the steps of determining an amplitude of the periodic interferer; generating a gaussian noise signal of substantially similar amplitude to the periodic interferer; and introducing the gaussian noise signal into the speech signal having the periodic interferer to produce a resultant signal, inputting the resultant signal into a noise suppression procedure to provide a noise suppressed speech signal, wherein the speech processor determines an amplitude level of the suppressed gaussian noise and subtracts a respective level of suppressed gaussian noise from the resultant signal thereby reducing the periodic noise content in the speech signal.

[0015] A preferred embodiment of the invention will now be described, by way of example only, with reference to the drawings.

Brief Description of the Drawings

[0016]

FIG. 1 shows a block diagram of a synthesis functional model of a basic LPC codec.

FIG. 2 shows a block diagram of a synthesis functional model of a basic ACELP codec.

FIG. 3 shows a block diagram of a decoding siren suppression algorithm according to a preferred embodiment of the invention.

FIG. 4 shows a block diagram of an encoding siren suppression algorithm according to a preferred embodiment of the invention.

FIG. 5 shows experimental results of the siren suppression algorithm of the preferred embodiment of the invention.

Detailed Description of the Drawings

[0017] Referring first to FIG. 1, a block diagram of a synthesis functional model of a basic LPC codec is shown. A voiced excitation source 10 provides a pulse train signal, of pitch duration T into a voiced gain element 12. The amplified pulse train signal from voiced gain element 12 is then selectively input, via a switch 14, to a Linear Predictive Coder (LPC) Filter 16. When no voice signal is present, an unvoiced excitation source 18 provides a gaussian noise signal into an unvoiced gain element 20. The amplified gaussian noise signal from unvoiced gain element 20 is selectively input, via switch 14, to the Linear Predictive Coder (LPC) Filter 16, when no voice is present. The output from the LPC filter 16 is synthetic speech.

[0018] In this manner, a series of amplified pulses from the voiced excitation source 10 are combined with amplified signals from an unvoiced excitation source 18, filtered with the resultant generated signal being representative of synthetic speech.

[0019] Referring next to FIG. 2, a block diagram of a synthesis functional model of a basic ACELP codec is shown. An excitation vector from the ìInnovtiveî codebook 30 is chosen and input to the voice gain element 31. Another excitation vector from the ìAdaptiveî codebook 32 is also chosen according to the present frame pitch lag value T and input the gain element 33. The output of voice gain element 31 and the output of voice gain element 33 are input to a summation device 34. The output of the summation device 34 is input to the Linear Predictive Coder filter 35. The output of the summation device 34 is also used to update the ìAdaptiveî codebook for next frame speech synthesis. The output from the LPC filter 35 is then synthetic speech.

[0020] In this manner, a series of amplified excitation vectors from the ìAdaptiveî codebook 32, incorporating a feedback path from the excitation vector source, are combined with a variety of amplified pulses selected from an ìInnovativeî codebook 30 (unvoiced excitation source). The combined signal is then filtered with the resultant signal from the LPC filter being representative of synthetically generated speech. The particular vectors are chosen to best imitate the speech signal to be transmitted, or being received.

[0021] The arrangements used in FIG. 1 or FIG. 2 are implemented in the encoding functions of a speech codec. The corresponding functions are required in reverse when decoding received speech.

[0022] Referring now to FIG. 3, a block diagram of a decoding siren suppression algorithm, according to a preferred embodiment of the invention, is shown. A speech signal is received and decoded in a speech decoder 50. The decoded speech signal with a siren signal contained within it, is then input to a first summing junction 56. Additionally, the decoded speech signal, together with the siren signal is processed to determine the amplitude of the siren signal in processor 52. The amplitude of the siren signal is then used to generate a gaussian noise signal of that same amplitude in the noise generator 54. The output gaussian noise signal is also fed to the first summing junction 56 and input to a sub-band gain adjustment block 60. The output from the first summing junction 56 is effectively the decoded speech signal, plus the gaussian noise signal which hides the periodic noise from the siren. This speech and gaussian noise signal is then used in a noise suppression algorithm 58 to reduce the level of the gaussian noise. The improved speech signal, i.e. with a reduced gaussian noise level, is then input to the sub-band gain adjustment block 60. The output from the sub-band gain adjustment block 60 is combined with the improved speech signal in a second summing junction 62. The resultant signal is the decoded speech signal with a greatly reduced siren noise effect.

[0023] Under high SNR condition, preliminary experimental results show that by injecting a known gaussian noise sequence into the incoming signal, at a level comparable to the siren, a standard wide-band noise suppression algorithm is able to suppress the siren plus the gaussian noise signal by the level specified in the noise suppression algorithm, whilst maintaining good speech quality. As the gaussian noise sequence is known, it can be extracted from the noise suppression process output.

[0024] This arrangement has the advantage of requiring just a single microphone input and avoids the need for precise estimation of the siren harmonic frequency. The only requirements are the detection of the siren signal and determination of its amplitude.

[0025] Advantageously, suppression of siren signal under high signal to noise conditions is achieved using just a single microphone input. Furthermore, the need for precise estimation of the siren harmonic frequency is avoided, by purely detecting the siren signal and estimating its amplitude.

[0026] Referring now to FIG. 4, a block diagram of an encoding siren suppression algorithm is shown. The encoding process deals with the situation where the siren is close to the transmitting unit, performing the speech encoding function. The encoding function is basically the decoding function in reverse, with the determination of the siren amplitude being calculated and used to generate a gaussian noise signal of similar amplitude.

[0027] A speech signal, having a siren signal contained within it, is input to a first summing junction 76. Additionally, the speech signal, together with the siren signal is processed to determine the amplitude of the siren signal in processor 72. The amplitude of the siren signal is then used to generate a gaussian noise signal of that same amplitude in the noise generator 74. The output gaussian noise signal is also fed to the first summing junction 76 and input to a sub-band gain adjustment block 80. The output from the first summing junction 76 is effectively the speech signal, minus the gaussian noise signal which hides the periodic noise from the siren. This speech and gaussian noise signal is then used in a noise suppression algorithm 78 to reduce the level of the gaussian noise. The improved speech signal, i.e. with a reduced gaussian noise level, is then input to the sub-band gain adjustment block 80. The output from the sub-band gain adjustment block 80 is combined with the improved speech signal in a second summing junction 82. The resultant signal is the encoded speech signal with a greatly reduced siren noise content.

[0028] In such a manner, a speech signal is generated with an interfering periodic noise signal having a greatly reduced effect.

[0029] Referring now to FIG. 5, experimental results of the siren suppression algorithm, according to the preferred embodiment of the invention, is shown. The results are shown with regard to amplitude versus time of the speech waveforms. Three distinct waveforms are provided. The first waveform 90 shows the input speech signal with the effect of the periodic interference (siren signal). The periodic noise content can be clearly seen with the darkly shaded areas indicating a rapidly changing and relatively constant interfering source. The second waveform 92 shows the input speech signal after applying the standard wide-band noise suppression algorithm. It is clearly shown that the periodic interference (siren signal) is not affected by the standard wide-band noise suppression algorithm, with the darkly shaded areas showing little change from the original speech plus siren signal. The third waveform 94 shows the input speech signal after applying the gaussian noise suppression algorithm, together with the standard wide-band noise suppression algorithm. The third waveform clearly shows a significant reduction in the periodic interference (siren signal) content, with the darkly shaded areas showing approximately a 10 dB siren suppression.

[0030] Thus, the present invention transforms a periodic noise contaminated speech signal into a wideband gaussian noise contaminated speech signal such that a standard wideband noise suppression procedure can be used to suppress the periodic noise. The periodic noise is then detected and its average amplitude estimated. A known gaussian noise sequence with a comparable amplitude is then added to the periodic noise contaminated speech signal. The noise (periodic + gaussian) in the resultant signal is then suppressed using a standard wideband noise suppression procedure (for example the Motorola sub-band noise suppression algorithm). At the same time, the underlying periodic noise is also suppressed. Since the added gaussian noise sequence is known and the sub-band gains used by the noise suppression algorithm have already determined, the suppressed gaussian noise at the noise suppression procedure output is then calculated and subtracted. The resultant signal is the speech signal with the periodic noise suppressed.

[0031] Hence, an arrangement for suppressing periodic noise in a contaminated speech signal is provided.

Claims

1. A speech communications unit comprising a speech processor for receiving an input speech signal having a periodic noise interferer,

   the speech communications unit characterised in that the speech processor is operably coupled to noise determining means (52, 72) for determining an amplitude of the periodic noise interferer (50, 70), a gaussian noise generator (54, 74) for generating a known gaussian noise sequence and combining the input speech signal with the known gaussian noise sequence and inputting a resultant signal into a noise suppression procedure to provide a noise suppressed speech signal, wherein the speech processor determines an amplitude level of the suppressed gaussian noise and subtracts a respective level of suppressed gaussian noise from the resultant signal thereby reducing the periodic noise content in the speech signal.

2. The speech communications unit of claim 1 wherein the known gaussian noise sequence is generated at a substantially similar amplitude to the periodic noise interferer (50, 70).

3. The speech communications unit of claim 1 wherein the speech processor is a speech encoder or speech decoder and the speech communications unit further comprises a noise suppresser function coupled to an output of a summing junction operably coupled to the gaussian noise generator (54, 74) for further suppressing noise in the speech signal.

4. The speech communications unit of claim 3, further comprising a gain adjuster, operably coupled to the gaussian noise generator (54, 74) and the noise suppresser function for receiving the known gaussian noise sequence and the noise suppressed signal, the gain adjuster being operably coupled to a second summing junction for recombining the gain adjusted signal with the noise suppressed signal thereby suppressing the periodic noise interferer.

5. The speech communications unit of claim 1, wherein the speech processor is a speech post-processing or a speech pre-processing function.

6. A method of reducing a periodic interferer in a speech signal, the method comprising the steps of:

   determining (52, 72) an amplitude of the periodic interferer;
   generating (54, 74) a gaussian noise signal of substantially similar amplitude to the periodic interferer;
   introducing (56, 76) the gaussian noise signal into the speech signal having the periodic interferer to produce a resultant signal;
   inputting the resultant signal into a noise suppression procedure (58, 78) to provide a noise suppressed speech signal;
   determining an amplitude level of the suppressed gaussian noise; and
   subtracting (62, 82) a respective level of suppressed gaussian noise from the resultant signal thereby reducing the periodic noise content in the speech signal.

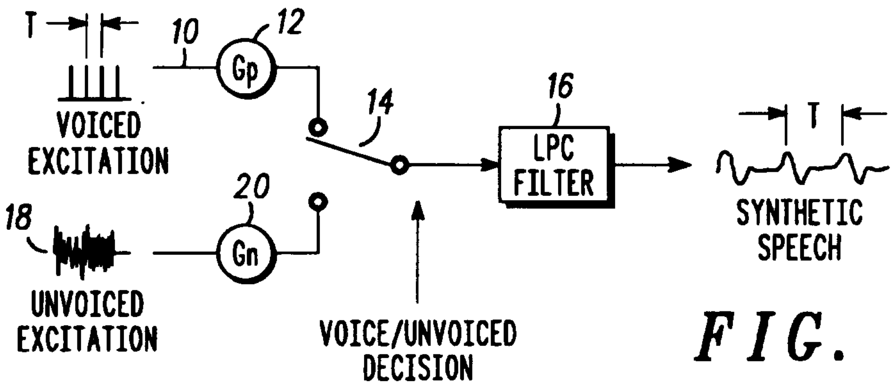

FIG. 1

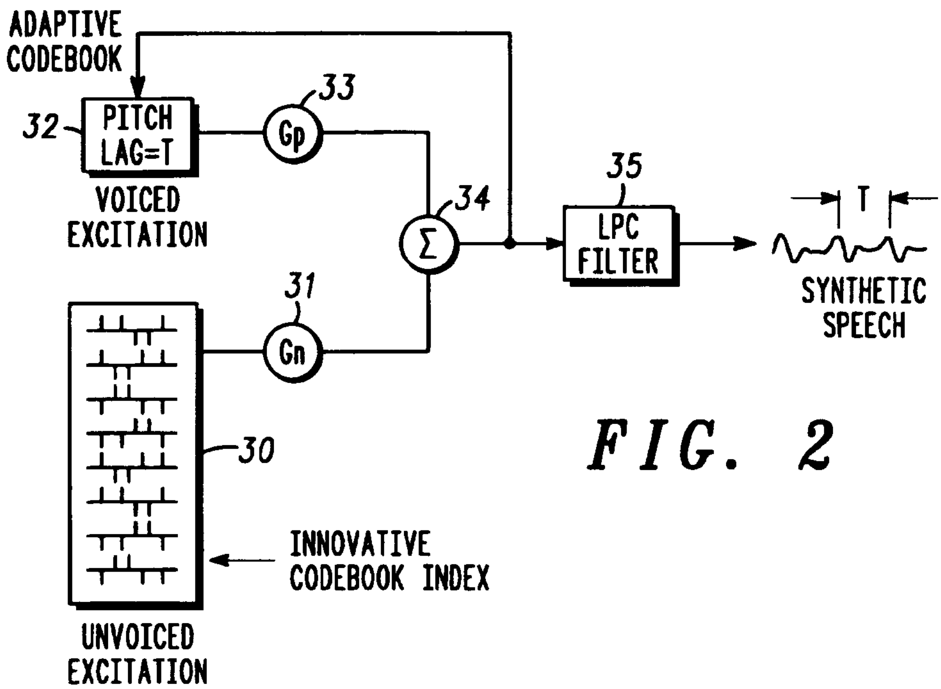

FIG. 2

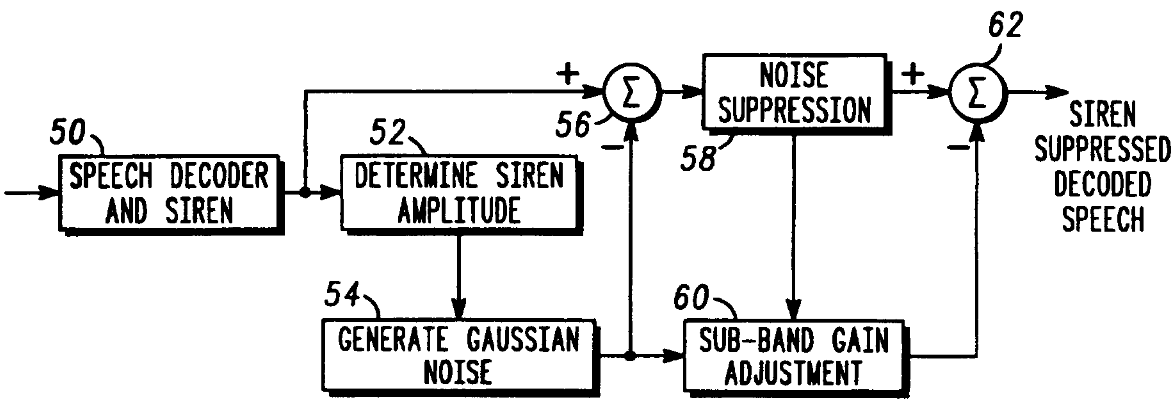

FIG. 3

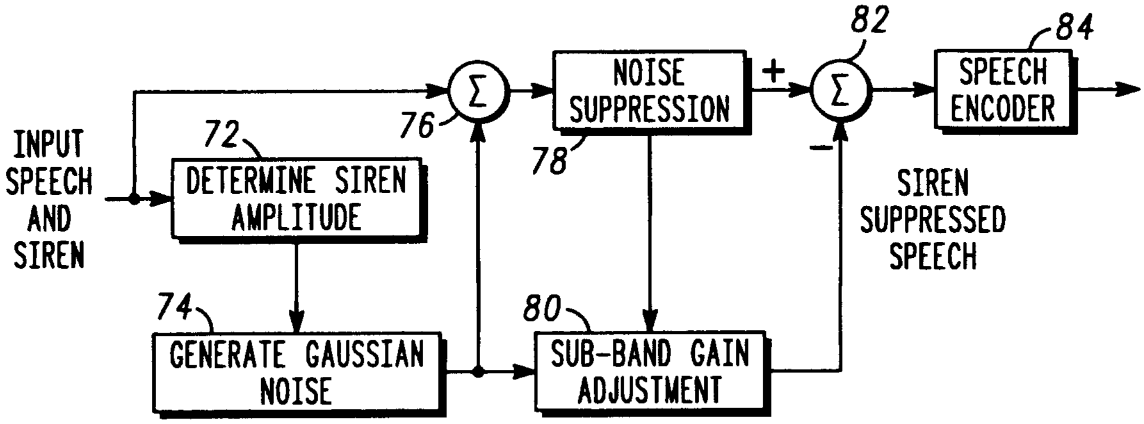

FIG. 4

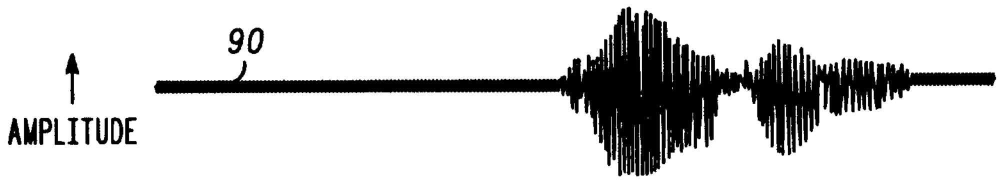

FIG. 5A

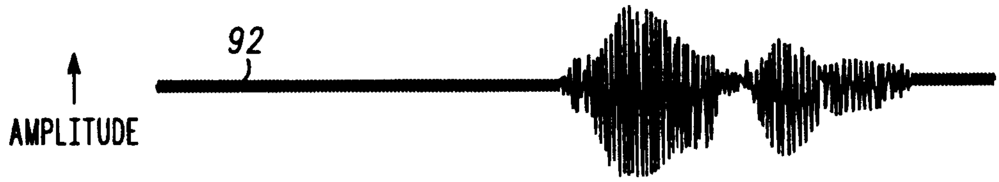

FIG. 5B

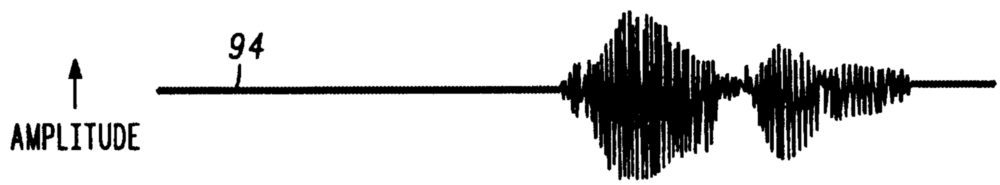

FIG. 5C