TITLE

Medial Olivocochlear Reflex Sound Coding with Bandwidth Normalization

[0001] This application claims priority from U.S. Provisional Patent Application 62/525,902, filed June 28, 2017, which is incorporated herein by reference in its entirety.

FIELD OF THE INVENTION

[0002] The present invention relates to hearing implant systems, and more specifically, to techniques for producing electrical stimulation signals in such systems.

BACKGROUND ART

[0003] A normal ear transmits sounds as shown in Figure 1 through the outer ear **101** to the tympanic membrane **102**, which moves the bones of the middle ear **103** (malleus, incus, and stapes) that vibrate the oval window and round window openings of the cochlea **104**. The cochlea **104** is a long narrow duct wound spirally about its axis for approximately two and a half turns. It includes an upper channel known as the scala vestibuli and a lower channel known as the scala tympani, which are connected by the cochlear duct. The cochlea **104** forms an upright spiraling cone with a center called the modiolus where the spiral ganglion cells of the acoustic nerve **113** reside. In response to received sounds transmitted by the middle ear **103**, the fluid-filled cochlea **104** functions as a transducer to generate electric pulses which are transmitted to the cochlear nerve **113**, and ultimately to the brain.

[0004] Hearing is impaired when there are problems in the ability to transduce external sounds into meaningful action potentials along the neural substrate of the cochlea **104**. To improve impaired hearing, hearing prostheses have been developed. For example, when the impairment is related to operation of the middle ear **103,** a conventional hearing aid may be used to provide mechanical stimulation to the auditory system in the form of amplified sound. Or when the impairment is associated with the cochlea **104**, a cochlear implant with an implanted stimulation electrode can electrically stimulate auditory nerve tissue with small currents delivered by multiple electrode contacts distributed along the electrode.

[0005] Figure 1 also shows some components of a typical cochlear implant system, including an external microphone that provides an audio signal input to an external signal processor **111** where various signal processing schemes can be implemented. The processed signal is then converted into a digital data format, such as a sequence of data frames, for transmission into the implant **108**. Besides receiving the processed audio information, the implant **108** also performs additional signal processing such as error correction, pulse formation, etc., and produces a stimulation pattern (based on the extracted audio information) that is sent through an electrode lead **109** to an implanted electrode array **110**.

[0006] Typically, the electrode array **110** includes multiple electrode contacts **112** on its surface that provide selective stimulation of the cochlea **104**. Depending on context, the electrode contacts **112** are also referred to as electrode channels. In cochlear implants today, a relatively small number of electrode channels are each associated with relatively broad frequency bands, with each electrode contact **112** addressing a group of neurons with an electric stimulation pulse having a charge that is derived from the instantaneous amplitude of the signal envelope within that frequency band.

[0007] In some coding strategies, stimulation pulses are applied at a constant rate across all electrode channels, whereas in other coding strategies, stimulation pulses are applied at a channel-specific rate. Various specific signal processing schemes can be implemented to produce the electrical stimulation signals. Signal processing approaches that are well-known in the field of cochlear implants include continuous interleaved sampling (CIS), channel specific sampling sequences (CSSS) (as described in U.S. Patent No. 6,348,070, incorporated herein by reference), spectral peak (SPEAK), and compressed analog (CA) processing.

[0008] Figure 2 shows the major functional blocks in a typical cochlear implant signal processing system wherein band pass signals are processed and coded to generate electrode stimulation signals to stimulation electrodes in an implanted cochlear implant electrode array. For example, commercially available Digital Signal Processors (DSP) can be used to perform speech processing according to a 12-channel CIS approach. The initial acoustic audio signal input is produced by one or more sensing microphones, which may be omnidirectional and/or directional. Filter Bank **201** pre-processes the initial acoustic audio signal with a bank of multiple band pass filters, each of which is associated with a specific band of audio frequencies—for example, a digital filter bank having 12 digital Butterworth band pass filters of 6th order, Infinite Impulse Response (IIR) type—so that the acoustic audio signal is filtered into some M band pass signals, B_1 to B_M where each signal corresponds to the band of frequencies for one of the band pass filters. Each output of the CIS band pass filters can roughly be regarded as a sinusoid at the center frequency of the band pass filter which is modulated by the signal envelope. This is due to the quality factor ($Q \approx 3$) of the filters. In case of a voiced speech segment, this envelope is approximately periodic, and the repetition rate is equal to the pitch frequency. Alternatively and without limitation, the Filter Bank **201** may be implemented based on use of a fast Fourier transform (FFT) or a short-time Fourier transform (STFT). Based on the tonotopic organization of the cochlea, each electrode contact in the scala tympani often is associated with a specific band pass filter of the external filter bank.

[0009] Figure 3 shows an example of a short time period of an audio speech signal from a microphone, and Figure 4 shows an acoustic microphone signal decomposed by band-pass filtering by a bank of filters into a set of signals. An example of pseudocode for an infinite impulse response (IIR) filter bank based on a direct form II transposed structure is given by Fontaine et al., *Brian Hears: Online Auditory Processing Using Vectorization Over Channels*, Frontiers in Neuroinformatics, 2011; incorporated herein by reference in its entirety:

```
for j = 0 to number of channels - 1 do
   for s = 0 to number of samples - 1 do
         Yj(s) = B0j *Xj(s) + Z0j
         for i = 0 to order- 3 do
               Zij = Bi+1,j *Xj(s) + Zi+1,j - Ai+1,j * Yj(s)
         end for
         Zorder-2,J = Border-1,,j * Xj(s) -Aorder-1,j * Yj(s)
   end for
end for
```

[0010] The band pass signals B_1 to B_M (which can also be thought of as frequency channels) are input to a Signal Processor **202** which extracts signal specific stimulation information—e.g., envelope information, phase information, timing of requested stimulation events, etc.—into a set of N stimulation channel signals S_1 to S_N that represent electrode specific requested stimulation events. For example, channel specific sampling sequences (CSSS) may be used as described in U.S. Patent 6,594,525, which is incorporated herein by reference in its entirety. For example, the envelope extraction may be performed using 12 rectifiers and 12 digital Butterworth low pass filters of 2nd order, IIR-type.

[0011] A Pulse Timing and Coding Module **203** applies a non-linear mapping function (typically logarithmic) to the amplitude of each band-pass envelope. This mapping function—for example, using instantaneous nonlinear compression of the envelope signal (map law)—typically is adapted to the needs of the individual cochlear implant user during fitting of the implant in order to achieve natural loudness growth. This may be in the specific form of functions that are applied to each requested stimulation event signal S_1 to S_N that reflect patient-specific perceptual characteristics to produce a set of electrode stimulation signals A_1 to A_M that provide an optimal electric representation of the acoustic signal. A logarithmic function with a form-factor c typically may be applied as a loudness mapping function, which typically is identical across all the band pass analysis channels. In different systems, different specific loudness mapping functions other than a logarithmic function may be used, with just one identical function applied to all channels or one individual function for each channel to produce the electrode stimulation signals A_1 to A_M outputs from the Pulse Timing and Coding Module **203**.

[0012] A Pulse Generation Module **204** develops the set of electrode stimulation signals A_1 to A_M into a set of output electrode pulses E_1 to E_M for the electrode contacts in the implanted electrode array which stimulate the adjacent nerve tissue. The output electrode pulses E_1 to E_M may be symmetrical biphasic current pulses with amplitudes that are directly obtained from the compressed envelope signals.

[0013] In the specific case of a CIS system, the stimulation pulses are applied in a strictly non-overlapping sequence. Thus, as a typical CIS-feature, only one electrode channel is active at a time and the overall stimulation rate is comparatively high. For example, assuming an overall stimulation rate of 18 kpps, where kpps stands for thousand stimulation pulses per second, and a 12 channel filter bank, the stimulation rate per channel is 1.5 kpps. Such a stimulation rate per channel usually is sufficient for adequate temporal representation of the envelope signal. The maximum overall stimulation rate is limited by the minimum phase duration per pulse. The phase duration cannot be arbitrarily short because, the shorter the pulses, the higher the current amplitudes have to be to elicit action potentials in neurons, and current amplitudes are limited for various practical reasons. For an overall stimulation rate of 18 kpps, the phase duration is 27 μs, which is near the lower limit.

[0014] In the CIS strategy, the signal processor only uses the band pass signal envelopes for further processing, i.e., they contain the entire stimulation information. For each electrode channel, the signal envelope is represented as a sequence of biphasic pulses at a constant repetition rate. A characteristic feature of CIS is that the stimulation rate is equal for all electrode channels and there is no relation to the center frequencies of the individual channels. It is typical that the pulse repetition rate is not a temporal cue for the patient (i.e., it should be sufficiently high so that the patient does not perceive tones with a frequency equal to the pulse repetition rate). The pulse repetition rate is usually chosen at greater than twice the bandwidth of the envelope signals (based on the Nyquist theorem).

[0015] Another cochlear implant stimulation strategy that does transmit fine time structure information is the Fine Structure Processing (FSP) strategy by Med-El. Zero crossings of the band pass filtered time signals are tracked, and at each negative to positive zero crossing, a Channel Specific Sampling Sequence (CSSS) is started. Typically, CSSS sequences are only applied on the first one or two most apical electrode channels, covering the frequency range up to 200 or 330 Hz. The FSP arrangement is described further in Hochmair I, Nopp P, Jolly C, Schmidt M, Schößer H, Garnham C, Anderson I, *MED-EL Cochlear Implants: State of the Art and a Glimpse into the Future,* Trends in Amplification, vol. 10, 201-219, 2006, which is incorporated herein by reference.

[0016] In addition to the specific processing and coding approaches discussed above, different specific pulse stimulation modes are possible to deliver the stimulation pulses with specific electrodes —*i.e.* mono-polar, bi-polar, tri-polar, multi-polar, and phased-array stimulation. And there also are different stimulation pulse shapes—*i.e.* biphasic, symmetric triphasic, asymmetric triphasic pulses, or asymmetric pulse shapes. These various pulse stimulation modes and pulse shapes each provide different benefits; for example, higher tonotopic selectivity, smaller electrical thresholds, higher electric dynamic range, less unwanted side-effects such as facial nerve stimulation, etc. But some stimulation arrangements are quite power consuming, especially when neighboring electrodes are used as current sinks. Up to 10 dB more charge might be required than with simple mono-polar stimulation concepts (if the power-consuming pulse shapes or stimulation modes are used continuously).

[0017] Bilateral stimulation has long been used in hearing aids, but it has only recently become common in hearing implants such as cochlear implants (CI). For cochlear implants, binaural stimulation requires a bilateral implant system with two implanted electrode arrays, one in each ear. The incoming left and right side acoustic signals are similar to those in hearing aids and may simply be the output signals of microphones located in the vicinity of the left and right ear, respectively. Bilateral cochlear implants provide the benefits of two-sided hearing which can allow a listener to localize sources of sound in the horizontal plane. Two-sided hearing also is known to make speech easier to understand in noise. This is explained more fully, for example, in Bronkhorst, A. W., and Plomp, R., *The Effect Of Head-Induced Interaural Time And Level Differences On Speech Intelligibility In Noise*, J. Acoust. Soc. Am. 83, 1508-1516, 1988, which is incorporated herein by reference.

[0018] It is also known that for natural hearing, the central nervous system controls sound coding in the cochlea via medial olivocochlear (MOC) efferents. MOC efferents innervate outer hair cells in the cochlea, the activation of which inhibits the motion of the basilar membrane for low- and moderate-level sounds. This inhibitory effect restores the dynamic range of individual auditory nerve fibers in noise and possibly increases the proportion of nerve fibers that discharge within their dynamic range. Presumably this improves the neural representation of transient sound features and thus facilitates speech recognition and sound localization in noise. For further details, refer to Guinan JJ., *Cochlear efferent innervation and function.* Curr. Opin. Otolaryngol. Head & Neck Surgery, 18(2010), pp. 447-453; which is incorporated herein by reference. It might also magnify spatial release from masking, see for example Kim S. H., Frisina R. D., Frisina D.

R. *Effects of age on speech understanding in normal hearing listeners: Relationship between the auditory efferent system and speech intelligibility in noise.* Speech Communication (2006) 48, 862.

[0019] CI users have greater difficulty understanding speech in noise than listeners with normal hearing. This may be partly due to their lacking the antimasking benefits of the medial olivocochlear (MOC) reflex, and the hearing-in-noise capacity of CI users may be improved with sound processing strategies that mimic MOC effects on compression. *See* Patent Cooperation Treaty Publication WO 2015/169649; Lopez-Poveda et al., *Roles of the contralateral efferent reflex in hearing demonstrated with cochlear implants*, Adv. Exp. Med. Biol. 894, 105-114, 2016; and Lopez-Poveda et al., *A bilateral sound coding strategy inspired by the contralateral medial olivocochlear reflex*, Ear Hear. 37, e138-e148, 2016; all of which are incorporated herein by reference in their entireties. On the other hand, the ability of CI users to recognize speech in noise can be thought to concomitantly depend on the effective speech-to-noise ratio (SNR) at the output of the CI audio processor *and* on the sensitivity of the CI user to the corresponding electrical stimulation.

SUMMARY OF THE INVENTION

[0020] Embodiments of the present invention are directed to a signal processing arrangement and corresponding method for a bilateral hearing implant system having left side and right side hearing implants. Each hearing implant includes at least one sensing microphone configured for sensing a sound environment to develop a corresponding microphone signal output. A filter bank is configured for processing the microphone signal to generate multiple band pass signals, wherein each band pass signal represents an associated band of audio frequencies. A channel compression module is configured to develop an inhibition-adjusted band pass signal for each band pass signal using a channel-specific dynamic inhibition based on a channel-normalized medial olivocochlear reflex model reflecting bandwidth energy for a corresponding contralateral band pass signal and bandwidth energy for a selected reference contralateral band pass signal. A pulse timing and coding module is configured for processing the inhibition-adjusted band pass signals to develop stimulation timing signals. And a pulse generation module is configured for processing the stimulation timing signals to develop electrode stimulation signals for the hearing implant for perception as sound.

[0021] In further specific embodiments, the medial olivocochlear reflex model may specifically be configured to produce larger channel-specific dynamic inhibition adjustments for lower frequency band pass signals. The channel inhibition module may be configured to use a channel-specific dynamic inhibition adjustment function = $\frac{\ln(1+c\cdot x)}{\ln(1+c)}$, where x represents amplitude of the input band pass signal, y represents amplitude of the adjusted band pass signal, and c is a dynamically determined inhibition factor that determines amount of the inhibition adjustment. In such embodiments, x and y may range within an interval [0, 1], and/or the dynamically determined inhibition factor c and the bandwidth energy for the corresponding contralateral band pass signal may be inversely related such that the greater the bandwidth energy for the corresponding contralateral band pass signal, the smaller the value of the dynamically determined inhibition factor c. The bandwidth energy for the corresponding contralateral band pass signal may be based on root mean square output amplitude integrated over a preceding exponentially decaying time window with two time constants, τ_a and τ_b. And the pulse timing and coding module may be configured to use a continuous interleaved sampling (CIS) or Channel Specific Sampling Sequences (CSSS) based or Fine Structure Processing (FSP) coding strategy or a combination thereof to develop the stimulation timing signals.

BRIEF DESCRIPTION OF THE DRAWINGS

[0022] Figure 1 shows a section view of a human ear with a typical cochlear implant system designed to deliver electrical stimulation to the inner ear.

[0023] Figure 2 shows various functional blocks in a continuous interleaved sampling (CIS) processing system.

[0024] Figure 3 shows an example of a short time period of an audio speech signal from a microphone.

[0025] Figure 4 shows an acoustic microphone signal decomposed by band-pass filtering by a bank of filters into a set of signals.

[0026] Figure 5 shows various functional blocks in a system for signal processing according to an embodiment of the present invention.

[0027] Figure 6 shows various logical blocks in a method for signal processing according to an embodiment of the present invention.

DETAILED DESCRIPTION OF SPECIFIC EMBODIMENTS

[0028] Previous signal processing arrangements that used a medial olivocochlear reflex (MOCR) model, were based on a inhibition parameter c for each band pass channel that depended on output energy E from the corresponding contralateral band pass channel. For broadband signals, the resulting contralateral inhibition could have been greater for higher frequency channels than for lower frequency channels because high-frequency channels were broader in frequency and may have contained more energy than lower frequency channels. Embodiments of the present invention address that problem for binaural CI sound coding using a channel-specific dynamic inhibition adjustment based on a channel-normalized MOCR model that reflects bandwidth energy for a corresponding contralateral band pass signal and bandwidth energy for a selected reference contralateral band pass signal.

[0029] Figure 5 shows various functional blocks in a system, and Figure 6 shows various logical blocks in a method, for signal processing according to embodiments of the present invention for a bilateral hearing implant system having a left side hearing implant **500** and a right side hearing implant **501**. The system shown in Figure 5 is based on the system discussed above with respect to Figure 2 for CIS-based sound coding. So on each side there initially is at least one sensing microphone configured for sensing a sound environment to develop a corresponding microphone signal output, step **601**, that is processed by a Filter Bank **201**, step **602**, to generate multiple band pass signals B_1 to B_M (which can also be thought of as frequency channels), each representing an associated band of audio frequencies. For example, the Filter Bank **201** might specifically include a high-pass pre-emphasis filter such as a first-order Butterworth filter with a 3-dB cutoff frequency of 1.2 kHz, followed by a bank of 12 sixth-order Butterworth band-pass filters with 3-dB cutoff frequencies that follow a modified logarithmic distribution between 100 and 8500 Hz.

[0030] A Signal Processor-Channel Compression Module **502** then processes the band pass signals B_1 to B_M as described above with respect to Fig. 2 (e.g., performing envelope extraction via full-wave rectification and low-pass filtering using a fourth-order Butterworth low-pass filter with a 3-dB cutoff frequency of 400 Hz) and additionally performing a channel-specific dynamic logarithmic inhibition adjustment, step **603**, as discussed more fully below that is based on a channel-normalized medial olivocochlear reflex model to produce adjusted band pass signals S_1 to S_N.

[0031] Pulse Timing and Coding Module **203** then applies pulse coding (e.g., using a continuous interleaved sampling (CIS) coding strategy) and a non-linear mapping function as described above, step **604**, to the adjusted band pass signals S_1 to S_N to produce a set of electrode stimulation signals A_1 to A_M that the Pulse Generation Module **204** develops into the output electrode pulses E_1 to E_M, step **605**, for the electrode contacts in the implanted electrode array.

[0032] MOC Module **503** applies the channel-normalized medial olivocochlear reflex model to the band pass signals B_1 to B_M from each Filter Bank **201** for the Signal Processor-Channel Compression Module **502** to perform the channel-specific dynamic inhibition adjustment in step **603**, which may specifically be configured to produce larger channel-specific dynamic inhibition adjustments for lower frequency band pass signals. For example, this may be based on a channel-specific dynamic inhibition adjustment function $y = \frac{\ln(1+c\cdot x)}{\ln(1+c)}$, where x represents amplitude of the input band pass signal, y represents amplitude of the adjusted band pass signal, and c is a dynamically determined inhibition factor that determines amount of the inhibition adjustment. For further details, refer to the discussion in Boyd PJ, *Effects of programming threshold and maplaw settings on acoustic thresholds and speech discrimination with the MED-EL COMBI 40+ cochlear implant.* Ear Hear. 27, 608-618, 2006; which is incorporated herein by reference in its entirety. In another example the channel-specific dynamic inhibition adjustment function $y = ax^p + k$, where x represents amplitude of the input band pass signal, y represents amplitude of the adjusted band pass signal, k a constant, and p is a dynamically determined inhibition factor that determines amount of the inhibition adjustment. It is however understood, that the invention is not limited to these two dynamic inhibition adjustment functions, but other suitable functions may be used. More formally, inhibition factor *c* (or p) is controlled using the output energy *E*, normalized to the channel bandwidth *E*':

$$E' = E \cdot \left(\frac{BW_{ref}}{BW}\right)^{0.5}$$

where BW is the channel bandwidth, and BW_{ref} is a reference channel bandwidth.

[0033] In a typical embodiment, *x* and *y* may range within an interval [0, 1]. The dynamically determined inhibition factor *c* and the bandwidth energy for the corresponding contralateral band pass signal may be inversely related such that the greater the bandwidth energy for the corresponding contralateral band pass signal, the smaller the value of the dynamically determined inhibition factor *c*. For example, the relationship between the instantaneous value of *c* and the instantaneous contralateral output energy may be such that the greater the output energy, the smaller the value of *c*. In one example the relationship may be given by

$$c(t) = \frac{c_a - c_b}{1 + exp[-\alpha(E'(t) - \beta)]} + c_b$$

where c_a, c_b and β are constants and E' is the normalized instantaneous output energy *E* for a given time *t*, as described in Patent Cooperation Treaty Publication WO2015/169649 more detailed, incorporated herein by reference. In another example, the relationship may be given by

$$log_{10}\big(c(t)\big) = \frac{log_{10}(c_a) - log_{10}(c_b)}{1 + exp\left[-\alpha\left(log_{10}(E'(t)) - log_{10}(\beta)\right)\right]} + log_{10}(c_b)$$

where c_a, c_b and β are constants and E' is the normalized instantaneous output energy *E* for a given time *t*. It is however understood, that the invention is not limited to these two functions, but other suitable functions may be used.

Specifically, *c* might vary between approximately 30 and 1000 for contralateral output energies of 0 and –20 dB full scale (FS; where 0 dB FS corresponds to peak amplitude at unity), respectively.

[0034] Based on the exponential time-course of activation and deactivation of the MOC effect (*see, e.g.,* Backus and Guinan, *Time-course of the human medial olivocochlear reflex*, J. Acoust. Soc. Am. 119, 2889-2904, 2006; which is incorporated herein by reference in its entirety), the bandwidth energy for the corresponding contralateral band pass signal may be based on root mean square output amplitude integrated over a preceding exponentially decaying time window with two time constants, τ_a and τ_b. For example, to reflect the time course of activation and deactivation of the natural MOCR, time constants might be set to $\tau_a = 2$ ms and $\tau_b = 300$ ms.

[0035] For example, in an embodiment where channels #1 and #12 are respectively the lowest and highest in frequency, overall inhibition will be greatest when BW_{ref} equals the bandwidth of channel #12 ($BW_{\#12}$), and gradually decreases for lower number channels. Bandwidth normalizations that produce greater inhibition can compromise audibility and reduce intelligibility. In addition, for normal hearing listeners, it has been observed that a contralateral broadband noise at 60 dB SPL increases auditory thresholds by about 1 to 9 dB. The inventor has observed, that normalization with $BW_{ref} = BW_{\#6}$, $BW_{\#7}$, or $BW_{\#8}$ produces more reasonable overall inhibition, with effectively equal or greater inhibition in the lower than in the higher frequency channels without compromising audibility significantly.

[0036] Embodiments of the invention may be implemented in part in any conventional computer programming language. For example, preferred embodiments may be implemented in a procedural programming language (*e.g.*, "C") or an object oriented programming language (*e.g.*, "C++", Python). Alternative embodiments of the invention may be implemented as pre-programmed hardware elements, other related components, or as a combination of hardware and software components.

[0037] Embodiments can be implemented in part as a computer program product for use with a computer system. Such implementation may include a series of computer instructions fixed either on a tangible medium, such as a computer readable medium (*e.g.*, a diskette, CD-ROM, ROM, or fixed disk) or transmittable to a computer system, via a modem or other interface device, such as a communications adapter connected to a network over a medium. The medium may be either a tangible medium (*e.g.*, optical or analog communications lines) or a medium implemented with wireless techniques (*e.g.*, microwave, infrared or other transmission techniques). The series of computer instructions embodies all or part of the functionality previously described herein with respect to the system. Those skilled in the art should appreciate that such computer instructions can be written in a number of programming languages for use with many computer architectures or operating systems. Furthermore, such instructions may be stored in any memory device, such as semiconductor, magnetic, optical or other memory devices, and may be transmitted using any communications technology, such as optical, infrared, microwave, or other transmission technologies. It is expected that such a computer program product may be distributed as a removable medium with accompanying printed or electronic documentation (*e.g.,* shrink wrapped software), preloaded with a computer system (*e.g.,* on system ROM or fixed disk), or distributed from a server or electronic bulletin board over the network (*e.g.,* the Internet or World Wide Web). Of course, some embodiments of the invention may be implemented as a combination of both software (*e.g.*, a computer program product) and hardware. Still other embodiments of the invention are implemented as entirely hardware, or entirely software (*e.g.*, a computer program product).

[0038] Although various exemplary embodiments of the invention have been disclosed, it should be apparent to those skilled in the art that various changes and modifications can be made which will achieve some of the advantages of the invention without departing from the true scope of the invention.

CLAIMS

What is claimed is:

1. A signal processing system for signal processing in a bilateral hearing implant system having left side and right side hearing implants, the system for each hearing implant comprising:

at least one sensing microphone configured for sensing a sound environment to develop a corresponding microphone signal output;

a filter bank configured for processing the microphone signal to generate a plurality of band pass signals, wherein each band pass signal represents an associated band of audio frequencies;

a channel compression module configured to develop an inhibition-adjusted band pass signal for each band pass signal using a channel-specific dynamic inhibition adjustment based on a channel-normalized medial olivocochlear reflex model reflecting bandwidth energy for a corresponding contralateral band pass signal and bandwidth energy for a selected reference contralateral band pass signal;

a pulse timing and coding module configured for processing the inhibition-adjusted band pass signals to develop stimulation timing signals; and

a pulse generation module configured for processing the stimulation timing signals to develop electrode stimulation signals for the hearing implant for perception as sound.

2. The system according to claim 1, wherein the medial olivocochlear reflex model is configured to produce equal or larger channel-specific dynamic inhibition adjustments for lower frequency band pass signals.

3. The system according to claim 1, wherein the channel compression module is configured to use a channel-specific dynamic inhibition adjustment function $y = \frac{\ln(1+c\cdot x)}{\ln(1+c)}$, where x represents amplitude of the input band pass signal, y represents amplitude of the adjusted band pass signal, and c is a dynamically determined inhibition factor that determines amount of the inhibition adjustment.

4. The system according to claim 3, wherein x and y range within an interval [0, 1].

5. The system according to claim 3, wherein the dynamically determined inhibition factor *c* and the bandwidth energy for the corresponding contralateral band pass signal are inversely related such that the greater the bandwidth energy for the corresponding contralateral band pass signal, the smaller the value of the dynamically determined inhibition factor *c*.

6. The system according to claim 1, wherein the bandwidth energy for the corresponding contralateral band pass signal is based on root mean square output amplitude integrated over a preceding exponentially decaying time window with two time constants, τ_a and τ_b.

7. The system according to claim 1, wherein the pulse timing and coding module is configured to use a continuous interleaved sampling (CIS) or Channel Specific Sampling Sequences (CSSS) based or Fine Structure Processing (FSP) coding strategy or a combination thereof to develop the stimulation timing signals.

8. A method for signal processing in a bilateral hearing implant system having left side and right side hearing implants, the method for each hearing implant comprising:

sensing a sound environment with at least one sensing microphone to develop a corresponding microphone signal output;

processing the microphone signal with a filter bank to generate a plurality of band pass signals, wherein each band pass signal represents an associated band of audio frequencies;

developing an inhibition-adjusted band pass signal for each band pass signal using a channel-specific dynamic inhibition adjustment based on a channel-normalized medial olivocochlear reflex model reflecting bandwidth energy for a corresponding contralateral band pass signal and bandwidth energy for a selected reference contralateral band pass signal;

processing the inhibition-adjusted band pass signals with a pulse timing and coding module to develop stimulation timing signals; and

processing the stimulation timing signals with a pulse generation module to develop electrode stimulation signals for the hearing implant.

9. The method according to claim 8, wherein the medial olivocochlear reflex model is configured to produce equal or larger channel-specific dynamic inhibition adjustments for lower frequency band pass signals.

10. The method according to claim 8, wherein developing the inhibition-adjusted band pass signals includes using a channel-specific dynamic inhibition adjustment function $y = \frac{\ln(1+c \cdot x)}{\ln(1+c)}$, where x represents amplitude of the input band pass signal, y represents amplitude of the adjusted band pass signal, and c is a dynamically determined inhibition factor that determines amount of the inhibition adjustment.

11. The method according to claim 10, wherein x and y range within an interval [0, 1].

12. The method according to claim 10, wherein the dynamically determined inhibition factor c and the bandwidth energy for the corresponding contralateral band pass signal are inversely related such that the greater the bandwidth energy for the corresponding contralateral band pass signal, the smaller the value of the dynamically determined inhibition factor c.

13. The method according to claim 8, wherein the bandwidth energy for the corresponding contralateral band pass signal is based on root mean square output amplitude integrated over a preceding exponentially decaying time window with two time constants, τ_a and τ_b.

14. The method according to claim 8, wherein a continuous interleaved sampling (CIS) or Channel Specific Sampling Sequences (CSSS) based or Fine Structure Processing (FSP) coding strategy or a combination thereof is used by the pulse timing and coding module to develop the stimulation timing signals.

ABSTRACT

A signal processing arrangement is described for signal processing in a bilateral hearing implant system. A channel compression module develops a inhibition-adjusted band pass signal for each band pass signal using a channel-specific dynamic inhibition adjustment based on a channel-normalized medial olivocochlear reflex model that reflects bandwidth energy for a corresponding contralateral band pass signal and bandwidth energy for a selected reference contralateral band pass signal.

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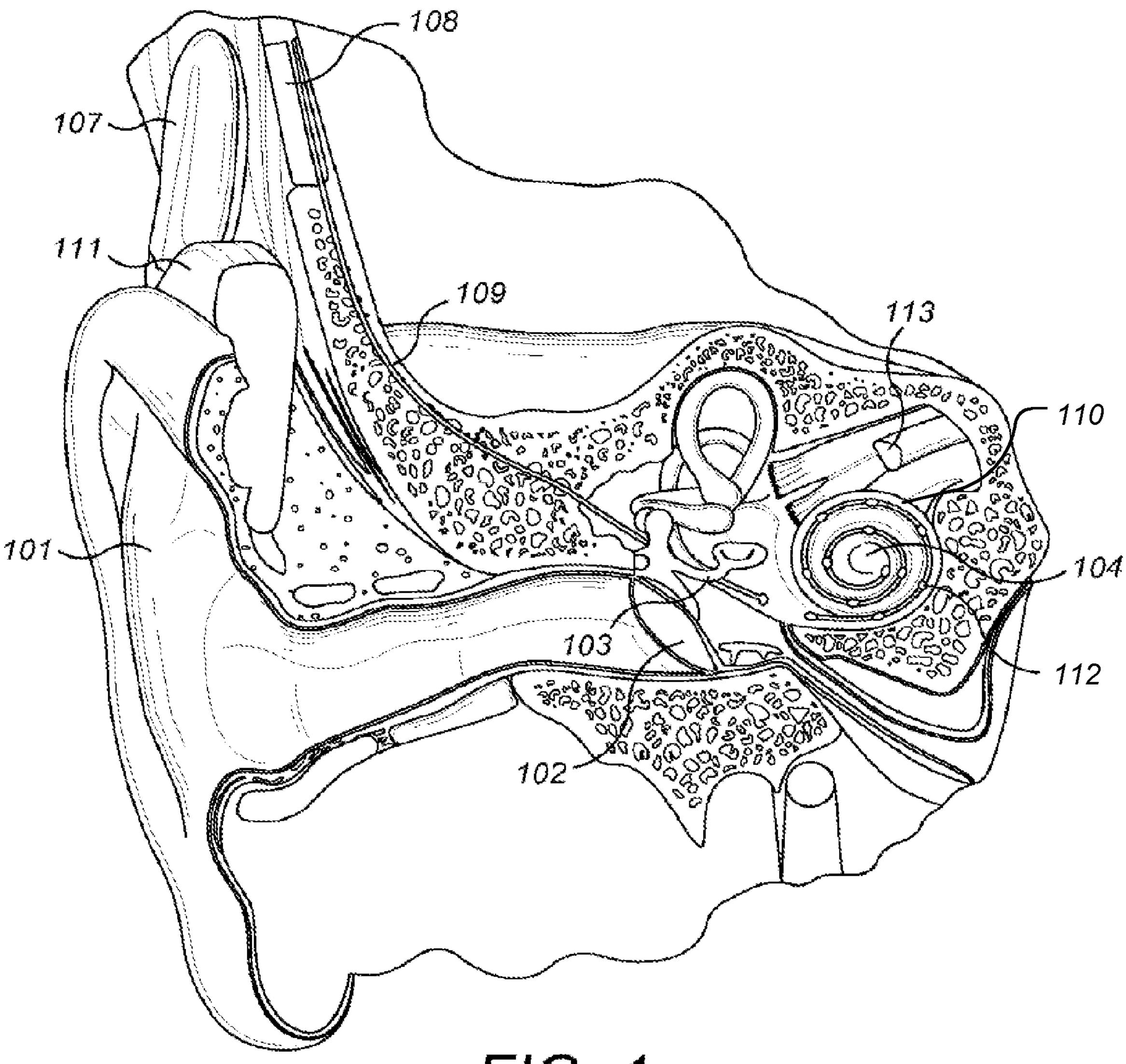
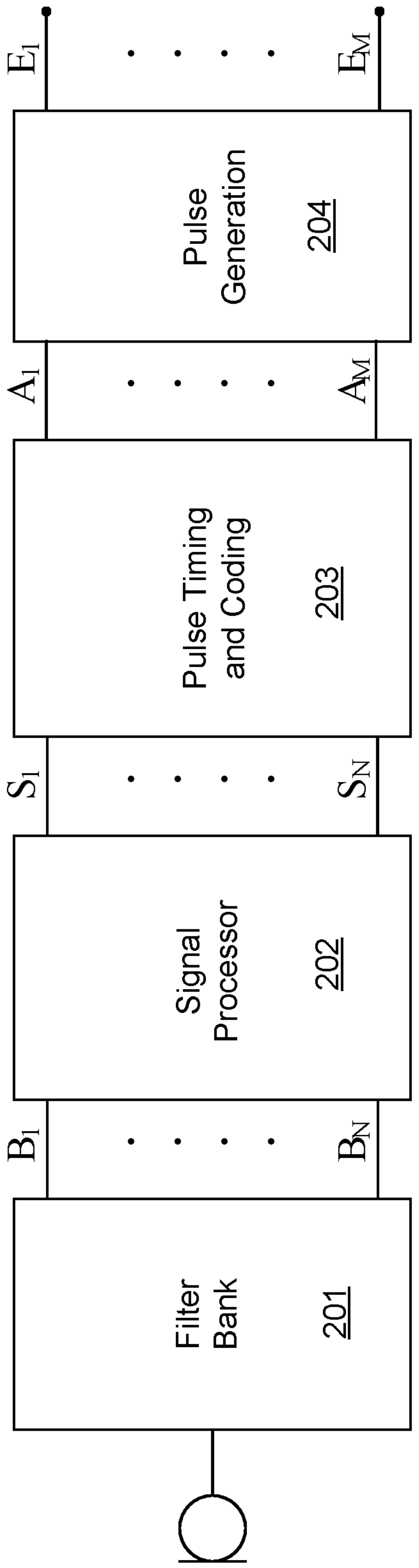
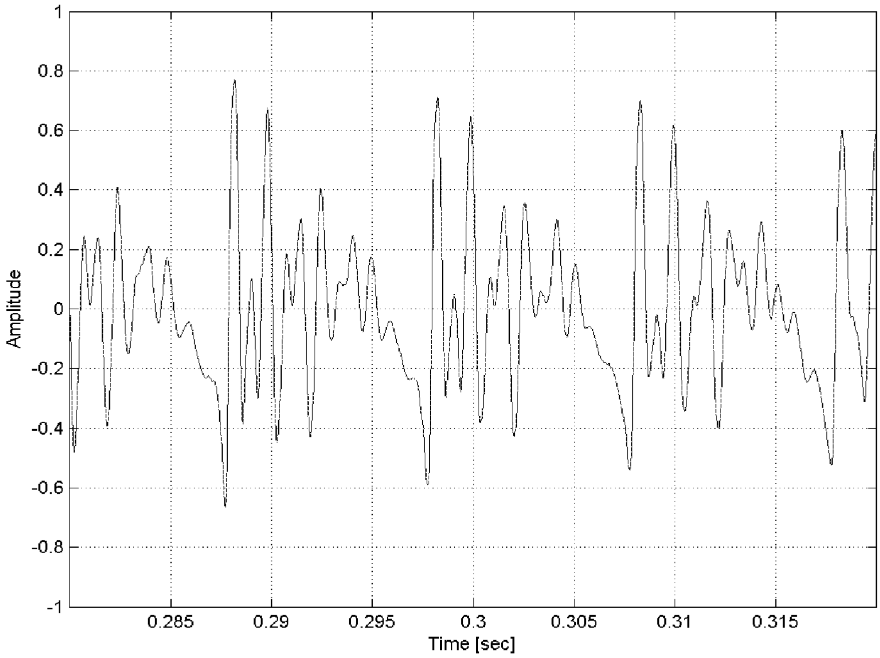


FIG. 1



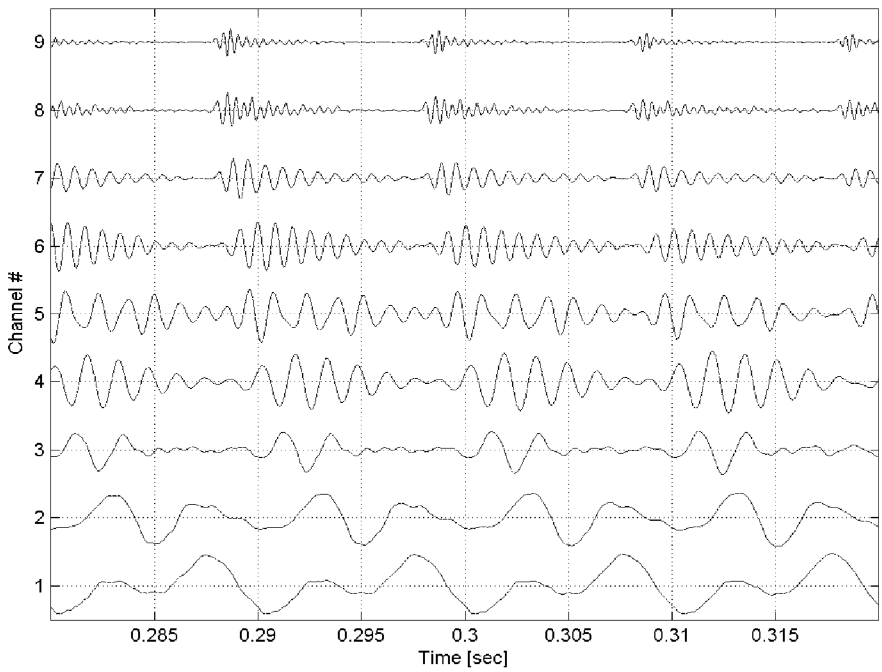
Filter Bank
201
B_1
B_N
Signal Processor
202
S_1
S_N
Pulse Timing and Coding
203
A_1
A_M
Pulse Generation
204
E_1
E_M

Fig. 2



Amplitude
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
0.285
0.29
0.295
0.3
0.305
0.31
0.315
Time [sec]

FIG. 3



Channel #
9
8
7
6
5
4
3
2
1
0.285
0.29
0.295
0.3
0.305
0.31
0.315
Time [sec]

FIG. 4

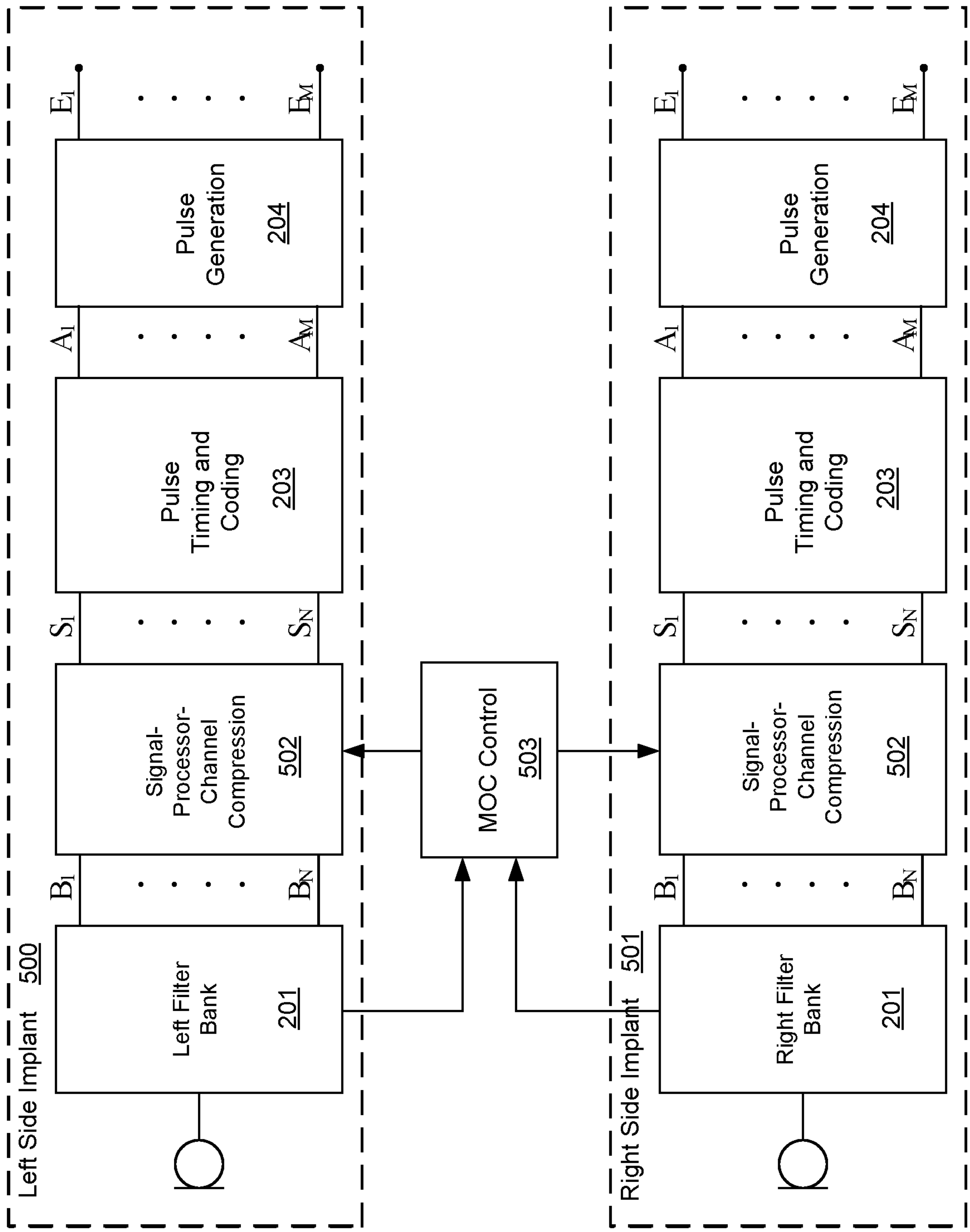

Left Side Implant 500
Left Filter Bank 201
B_1 B_N
Signal-Processor-Channel Compression 502
S_1 S_N
Pulse Timing and Coding 203
A_1 A_M
Pulse Generation 204
E_1 E_M
MOC Control 503
Right Side Implant 501
Right Filter Bank 201
B_1 B_N
Signal-Processor-Channel Compression 502
S_1 S_N
Pulse Timing and Coding 203
A_1 A_M
Pulse Generation 204
E_1 E_M

Fig. 5

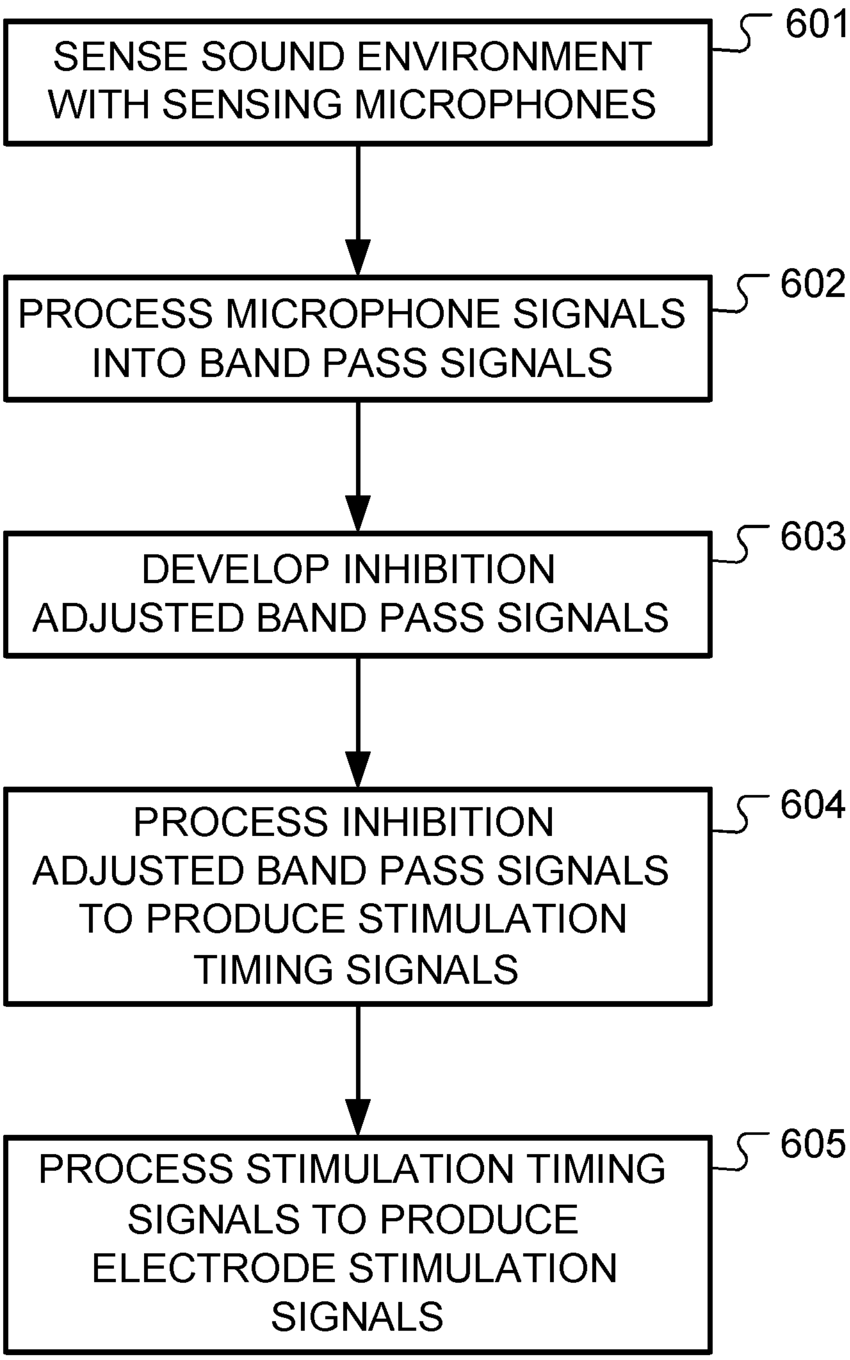

601
SENSE SOUND ENVIRONMENT WITH SENSING MICROPHONES
602
PROCESS MICROPHONE SIGNALS INTO BAND PASS SIGNALS
603
DEVELOP INHIBITION ADJUSTED BAND PASS SIGNALS
604
PROCESS INHIBITION ADJUSTED BAND PASS SIGNALS TO PRODUCE STIMULATION TIMING SIGNALS
605
PROCESS STIMULATION TIMING SIGNALS TO PRODUCE ELECTRODE STIMULATION SIGNALS

Fig. 6