METHOD AND APPARATUS FOR ADAPTING SPEECH CODERS TO IMPROVE COCHLEAR IMPLANT PERFORMANCE

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BRIEF DESCRIPTION

The present invention relates generally to speech processing strategies, and more particularly to adapting speech coders to improve the performance of cochlear implants.

Cochlear implant performance is improved by extracting pitch information and encoding such pitch information into the processor of a cochlear implant. One embodiment of the invention is to explicitly extract the pitch and deliver it to the cochlear implant by co-varying the stimulated site and rate. Another embodiment of the invention is to implicitly encode the pitch information via a code book that serves as the carrier of stimulation in the cochlear implant.

FULL DESCRIPTION

When the development of speech processing strategies in cochlear implants is compared to that of speech coding algorithms in modern communication, it is apparent that, except for specific earlier versions which used a feature extraction strategy, all current cochlear implants are based on the “channel vocoder” concept. This concept was first conceived and implemented by Horner Dudley at Bell Labs (Dudley 1939). The “channel vocoder” concept, as illustrated in FIG. 1, involves an initial extraction of band-specific temporal envelopes followed by extraction of voice pitch (if the sound was voiced). The extracted band-specific temporal envelopes are then used to amplitude modulate either a periodic pulse train that corresponds to the voice pitch (if the sound was voiced) or a noise (if the sound was unvoiced). Unfortunately, speech synthesized using this channel vocoder concept may be intelligible but may also have a “machine-like” sound quality due to inaccuracy in pitch extraction and/or other factors.

Until recently, the temporal envelope has been thought to be the major cue contributing to speech intelligibility, while fine structure has been thought to contribute mostly to sound quality and speaker identification. However, it now appears that fine structure is crucial to speech recognition in noise, particularly when noise is another competing voice. As such, encoding temporal fine structure in cochlear implants remains a significant challenge. The problem is that while continuous-interleaved-stimulation (CIS) strategies may improve the temporal envelope representation, they all but totally discard the temporal fine structure. Additionally, recently-proposed strategies using higher filter density at low frequencies than at high frequencies to improve fundamental frequency (F0) encoding have the unfortunate drawback of reduced filter density at high frequencies which degrades speech intelligibility. Therefore, methods and apparatus for adapting speech coders to improve cochlear implant performance are needed.

Disclosed and claimed herein are methods and apparatus for improving sound processing by a cochlear implant. In one embodiment, a method includes receiving sound containing a voiced component, extracting pitch information from said sound for the voiced component, and adding the pitch information into a continuous-interleaved-stimulation processor of the cochlear implant.

PATENT STATUS

Country	Type	Number	Dated	Case

OTHER INFORMATION

CATEGORIZED AS

- » **Medical**
  - » Devices
  - » Other

RELATED CASES

2005-696-0

United States Of America	Issued Patent	8,185,383	05/22/2012	2005-696	

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