Real Time Simulations for Cochlear Implants

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Abstract. The cochlear implant is electronic device that bypasses a non-functional inner ear by stimulating hearing nerve by patterns of electrical pulses, so that sounds and speech can be medicated to profound hearing loss patients. This paper presents the possibility of real time simulations with normal hearing volunteers.

1 Introduction

The speech processing algorithm which converts speech to information about current pulses is called speech strategy. The verification of new strategies with cochlear implant users is very time-consuming. So the simulations on computer or subjective testing on normal hearing volunteers is used. If these results show, that the new strategy could bring better speech and music performance, tests with cochlear implant users are accomplished.

2 Methods

The cochlear implant system consists of two main parts, the implant and the speech processor [1] The implant consists of an electronic chip in a metal box, a receiver coil with a permanent magnet and two electrodes – the intracochlear electrode array with 22 electrodes and the ball electrode used in case of the monopolar stimulation. The implant is responsible for stimulation of the auditory nerve and it could have another special use for measuring of the neural response of the auditory nerve. The body of this implant is situated under patient's skin and the electrode array is inserted in the cochlea.

The second part of the cochlear implant system is the speech processor converts sound or speech in current pulses. The signal processing algorithm (speech strategy) specifies a set of rules which defines how the speech will be analyzed and which information will be selected and transmitted into the implant.

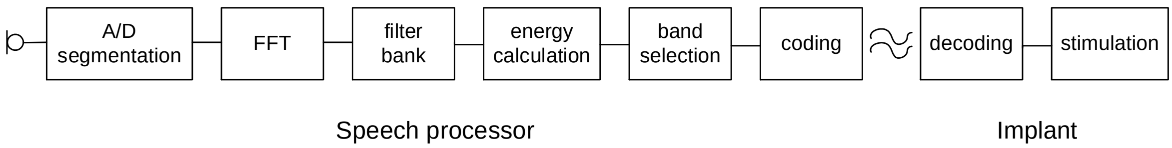

Fig 1. Cochlear implant system

In Figure 1 the block diagram of cochlear implant system is shown. Sound is picked up by a microphone, digitalized and divided in segments. Frequency analysis is applied on each segment and band selection is undertaken. The signal energy in each band is calculated and the most important information is chosen, depending on the used coding strategy and patient's setting. The information is coded and transmitted into the implant via radio frequency. The information is decoded in the implant and the nerve fiber stimulation is carried out.

2.1 ACE Strategy

The Advanced Combination Encoder (ACE) strategy [1] is the newest strategy used in Nucleus® 24 implant systems. The block diagram is depictured in Fig. 2. The input speech is segmented and the spectrum is calculated. After, filter bank with 22 band-pass filters is used. The basal electrode corresponds to the band 22; the apical electrode corresponds to the band 1. The energy calculation block computes energy in each processed band. The band selection block selects N bands with maximal energy. The energy of all non-selected bands is not used for stimulation. The LGF block implements the logarithmic relationship between current amplitude and loudness of auditory perception. The channel mapping block translates information from the LGF block to current pulses used for stimulation and defines the stimulating rate. The NVMem block stores information required for individual setting of cochlear implant user.

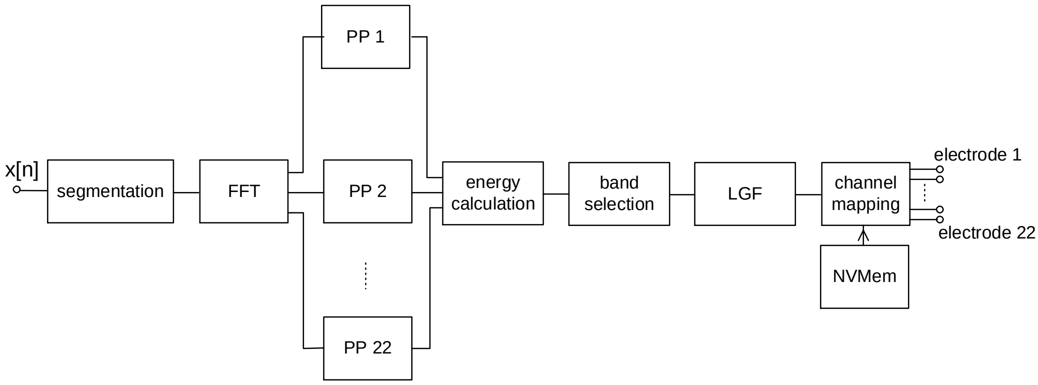

Fig 2. Block diagram of the ACE strategy

2.2 Speech Reconstruction

Backward speech reconstruction (see Fig 3) is a process which translates the stimulation frames normally used for stimulation to the speech. The reconstructed speech could be used for simulations with hearing volunteers. One possibility how to reconstruct speech signal from current pulses is synthesis using superimposed sinusoidal signals. The signal reconstruction using the pure tone generator is based on the assumption that every audio signal can be approximately reconstructed using a finite number of pure tones. Because ACE give no information about phase of processed signal (only absolute value of spectrum is used), the reconstruction using superimposed sinusoidal signals uses no phase information, too. The reconstructed signal *s(n)* [4] is given by the formula:

$$s(n) = \sum_{k=1}^{N} A_k(n) \cdot \sin(2pi \cdot f_k \cdot n) \tag{1}$$

The frequency value f_k is given by the central frequency of *k*-th band of currently used speech strategy. The number of bands *N* is 22 same as in ACE strategy. The $A_k(n)$ represents the amplitudes of the pure tones at a discrete time sample n = 1, 2…*L*. The length of time sample corresponds to stimulating rate. For each time sample *n* the $A_k(n)$ is:

a) equal to amplitude of current pulse stimulated in electrode *k* (if the k-th band is selected as maximum),

b) equal to zero.

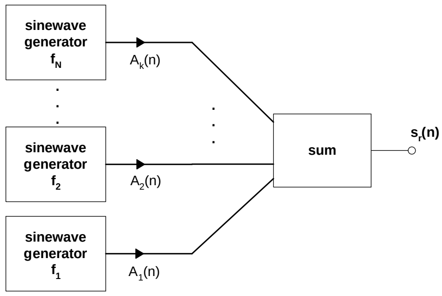

Fig 3. Block diagram of the SSS method

2.3 Implementation

The above described algorithm was implemented using Matlab [3] programming language. For the real-time processing, Data Acquisition Toolbox with multifunctional I/O card MF 624 or soundcard Digigram VX Pocket 440 were used. As an input, the microphone or radio output and as an output headphones were exercised over. The advantage of use soundcard is no extra hardware and possibility of portable system. But there is a limitation of sample rate and several samples of input data could be lost. The multifunctional I/O card is more robust, allows up to eight analog inputs (microphone array) with optional sample rate and the required computing power of PC is lower then in case of soundcard.

3 Results

The real time simulation was used for verification of ACEv (Advanced Combination Encoder with virtual electrodes) strategy [4] developed in CVUT-FEE was compared with ACE strategy which is mostly used by cochlear implant users. Firstly a 30 min record consists of speech and sound was presented to 13 normal hearing voluntaries both using ACE and ACEv strategy. Secondly TV or radio output was used as an input to speech processing algorithm. Finally the microphone in different environments was applied. The tested persons were asked for subjective comparison of both strategies and their answers will be used for ACEv optimalization.

4 Conclusions

The speech processing algorithm Ace and ACEv were used for real-time simulations in Matlab programming language.. As an input, microphone, MP3 player or TV output could be used. The multifunctinal I/O card allows conencton of input and tested algorithm. The output of speech processing algorithm was backward reconstructed in speech. The reconstructed speech was transmited to headphones using multifunctinal I/O card. The real-time simulations allows verification of proposed changes in speech processing algorithm for a very long time, so the tested subjects are able to adapt on the startegy.

Acknowledgement

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