Wireless Transmission of Sound between Speech Processor and Transmitter for Cochlear Implant

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***Abstract*—The latest cochlear Implant Naida CI Q70 with advanced wireless feature has got its speech processor and transmitter in the form of reliable wire communication architecture. This architecture creates problem for deaf in terms of maintenance and cost for its complex structure. Hence we propose a wireless architecture for speech processor to communicate with transmitter.**

A band pass digital auditory filter is introduced in transmitter circuit which separates the frequency from overlapping and reduces unwanted noise. The whole optimized communication architecture design is to reduce the noise and gives a convenient structure for deaf to provide a reliable and improved hearing.

***Index Terms*—Cochlear implant (C.I), digital auditory filter Transmitter, speech processor, finite impulse response filter(FIR), Signal to noise ratio (SNR)**

I. Introduction

A cochlear implant [1] is a surgical treatment for hearing loss that works like an artificial human cochlea in the inner ear, which sends sound from ear to brain. Normally hair cells stimulate the hearing nerve, which transmits sound signals to the brain. When hair cells stop functioning the hearing nerve remain without stimulation and a person cannot hear. Cochlear implant has internal and external parts. Microphone, speech processor and transmitter as external devices. Stimulator and electrodes which are held inside the skull as internal device. Microphone picks up the sound from the environment. Speech processor which selectively filters sound to audible speech, it splits the sound into channels and sends the electrical sound signals through a cable to the transmitter. A transmitter is a coil held in the position by a magnet placed behind external ear. Transmitter sends sound to stimulator inside the skull by electromagnetic induction. Stimulator is connected with electrode cables which are connected to cochlea. Cochlea cell sends electric impulse to brain. [1]

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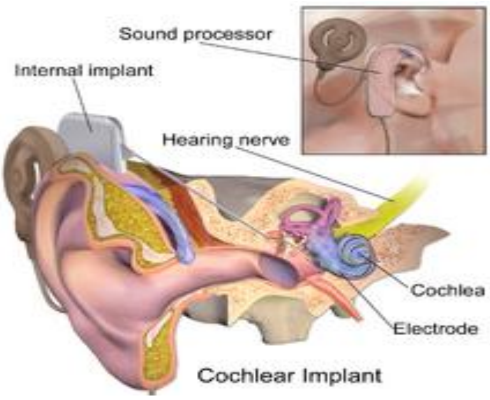

Figure 1. Cochlear device with its components Transmitter, Speech Processor, Electrode and Microphone

The clinical cochlear implant (CI) has good speech recognition under quiet conditions, but noticeably poor recognition under noisy conditions [2]. For 50% sentence understanding [3, 4], the required signal to noise ratio (SNR) is between 5 and 15 dB for CI recipients, but only −10 dB for normal listeners. The SNR in the typical daily environment is about 5–10 dB, which results in <50% sentence recognition for CI users in a normal noise environment. Most previous studies on recognition improvement have focused on the coding strategy, design of the electrode array, and stimulation adjustment of pitch recognition, as well as on the virtual electrode technique [5, 6] and optical CIs [7]. More recent efforts have focused on the microphone array technique [8, 9]. This array beam forming method promises [10] to be more effective for situations in which the desired voice and ambient noise originate from different directions, the usual work environment for CI devices. Speech-enhancement methods include single and multichannel techniques. Spectral estimation methods are the most widely used single-channel techniques. Typical single channel approaches, such as the spectral subtraction [11, 12], Wiener filtering, and subspace approach, are based on estimations of the power spectrum or higher order spectrum, assume the noise to be stationary, and use the noise spectrum in the non speech frame to estimate the

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speech-frame noise spectrum. Algorithm performance sharply weakens when the noise is non-stationary, or under typical situations with music or ambient speech noise. The microphone array technique considers the signal orientation information and focuses on directional speech enhancement. Specifically, the generalized sidelobe canceller and delay beam forming use multiple microphones to record signals for spatial filtering. For CI devices, the generalized sidelobe canceller is overly complicated and requires too many microphones, conditions that exceed the capabilities of current CI devices. Delay beam forming technologies, such as the first-order differential microphone (FDM) and adaptive null-forming method (ANF) , are adopted in hearing aids. These methods need only 2 microphones, which is an appropriate set-up for the CI size constraint and real-time processing.CI devices are similar with the hearing aids in size constraint and the requirement of front-end noise suppression. So, for CI speech enhancement, one simple solution for CI speech enhancement is to directly utilize the microphone-array–based noise reduction methods from the present hearing aids, in which the sensor-array techniques have been more widely used. However, the difference between CI devices and hearing aids is prominent, and a direct application of these algorithms to CI speech processing is not appropriate. Firstly, the principle is very different. CI devices transfer the acoustic signal to electrical stimulation into the cochlea wirelessly, and then the electrical pulses are used to directly stimulate the acoustic nerve to yield the auditory perception. But the hearing aids only need to change the corresponding gains in different sub bands for multi-frequency signal loss. In brief, the hearing aid is only an amplifier with adjustable gain in different frequency band. Secondly, the application of the microphone array technique is different. Many algorithms for speech application were borrowed from the narrowband methods in radar and antenna. Algorithms for front-end enhancement are

Indispensable to match the CI speech strategy. Thirdly, the solution for low frequency roll-off may be different. The hearing aids need to calibrate and preset the sub band gain based on user's hearing loss. Therefore, in the hearing aid, one solution is to directly present the sub band gains in the filter banks in the processor by both taking the hearing loss and signal loss in microphone array algorithm into account. However, for CI devices with the modulated electrical pulse directly stimulate the cochlear nerves, we only need to adjust the algorithm loss. Finally, the signal distortion is different

II. Functional Mechanism of Wired Communication Between Speech Processor And Transmitter.

Cochlear implant replaces the normal inner ear by Transforming acoustic sound signals into electric stimuli and sends to the auditory nerve which is called as cochlea. Microphone picks up the sound and sends to speech processor. Speech processor splits the auditory sound signals into different frequencies. Here speech processor consists of filter bank to split the speech spectrum into signals of Microphone picks up the sound and sends to speech processor. Speech processor splits the auditory sound signals into different frequencies. Here speech processor consists of filter bank to split the speech spectrum into signals of Microphone picks up the sound and sends to speech processor. Speech processor splits the auditory sound signals into different frequencies. Here speech processor consists of filter bank to split the speech spectrum into signals of various band width. Different frequency spectrum reaches transmitter through a wire.

III. Proposed Wireless Archtecture Design between Speech Processor and Transmitter.

Wireless communication of sound signal is implemented between speech processor and transmitter by placing radio frequency transmitter and radio frequency receiver in speech processor and transmitter respectively. Sound signal transmit from speech processor is converted into digital data by using ADC (Analog to Digital Conversion). Radio frequency transmitters can be used to transfer digital data wirelessly to the transmitter. Transmitted signal from speech processor will undergo through a band pass auditory filter [13] to reduce noise and unwanted frequencies. Received signal in the transmitter get transfer to stimulator by electromagnetic induction. See Fig. 2.

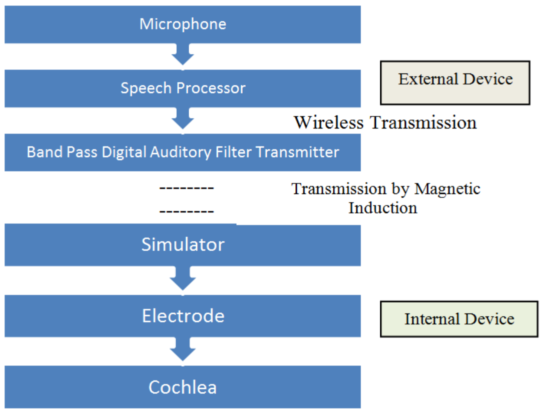

Figure 2. Function of finite impulse response filter.

Filter design and its structure of finite impulse response is discussed in [13]. Filter design and its structure of finite impulse response simulations are proven to be efficient in terms of timing and suitable [15]. F.I.R. filters and avoids interaction of speech signal of different channels. It send stable frequency to transmitter. F.I.R. filter[14] avoids the overlapping of signal. See Fig. 3.

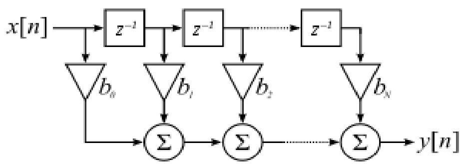

Figure 3. FIR filter of order N (direct form)

A direct form discrete-time FIR filter of order N. The top part is an N-stage delay line with N + 1 taps. Each unit delay is a z−1 operator in Z-transform notation. See Fig. 4.

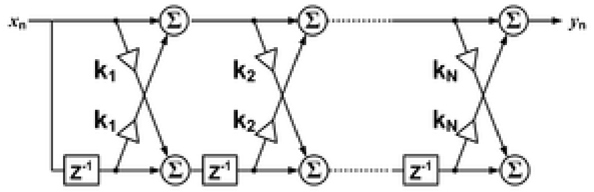

Figure 4. FIR filter of order N (lattice-form)

A lattice-form discrete-time FIR filter of order N. Each unit delay is a z−1 operator in Z-transform notation.

For a causal discrete-time FIR filter of order *N*, each value of the output sequence is a weighted sum of the most recent input values:

$$y[n] = b_0 x[n] + b_1 x[n-1] + \cdots + b_N x[n-N] = \sum_{i=0}^{N} b_i * x[n-i]$$

where:

- $x[n]$ is the input signal,
- $y[n]$ is the output signal,
- N is the filter order; an n^{th} order filter has $(N + 1)$ terms on the right-hand side
- b_i is the value of the impulse response at the i^{th} instant for $0 \leq i \leq N$ of an n^{th} order FIR filter. If the filter is a direct form FIR filter then b_i is also a coefficient of the filter.

This computation is also known as discrete convolution.

The $x[n-1]$ in these terms are commonly referred to as *taps*, based on the structure of a tapped delay line that in many implementations or block diagrams provides the delayed inputs to the multiplication operations. One may speak of a *5th order/6-tap filter*, for instance.

The impulse response of the filter as defined is nonzero over a finite duration. Including zeros, the impulse response is the infinite sequence:

$$h[n] = \sum_{i=0}^{N} b_i \cdot \delta[n-1] = \{ b_n \quad 0 \leq n \leq N, 0 \; otherwise$$

If an FIR filter is non-causal, the range of nonzero values in its impulse response can start before *n* = 0, with the defining formula appropriately generalized.

IV. Results and Discussion

TABLE I. Table Shows the Output Frequencies Through Wireless Transmission. Results Shows the Input Frequencies Will be Same as Input. These Results Confirm No Loss of Frequencies during Wireless Transmission

s.no	Example	Approximate frequency ranges(dB)	Input sound wave frequency(dB)at microphone	Output sound wave frequency at transmitter(dB)
1	Audible sound threshold	3-5 db	5	5
2	Normal breath	10-12 db	11	11
3	Normal conversion	30-35 db	33	33
4	Busy street	60-70 db	64	64
5	Subway or person shouting	80 -85 db	81	81
6	Loud stereo	90-95db	94	94
7	Table saw, auto horn	100-105 db	102	102
8	Elevated train, thunder	120-125 Db	120	120

V. Conclusion and Future work

The proposed wireless architecture for CI devices is based on introduction of band pass digital auditory filter in transmitter circuit which separates the frequency from overlapping and reduces unwanted noise. The whole optimized communication architecture design is to reduce the noise and gives a convenient structure for deaf to provide a reliable and improved hearing. Our results demonstrate the input frequencies and output frequencies through wireless transmission will be same. These results confirm no loss of frequencies during wireless transmission.

The whole exercise is to investigate the technique that aims to suppress the noise and improve the speech recognition of CI devices. In we construct the Algorithms and hardware Circuit which includes the Speech Processor and transmitter in single circuit and an algorithm that reduces the SNR ratio. In future we will carry the experiments to evaluate the Circuit and algorithm performance in a real working environment for CI users.

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