**The technique used for pre-processor for noise removal from the incoming sound for the cochlear implant user**

**Introduction:**

The cochlear Implant process helps to give solutions to different regions of disability in hearing. The principal goal is to make hearing easier and more meaningful. There is a significant advancement in both hardware as well as software used for implants. After intensive research in this field, it was noticed that existing Cochlear Implant techniques can’t significantly satisfy major types of deficiency in hearing. Hearing-impaired people are still facing lots of problems with regard to noise that interfered with the speech signal. There are a few challenges that need to be overcome so that CI users can listen as well as understand speech and voice just like people with natural hearing. Perfect pitch identification is one criterion and speech intelligibility is another important term. Most of the works have been done taking care of speech quality and not speech intelligibility which could have given better results in terms of speech identification, feature extraction, and noise removal. Researchers are still trying to fulfill these gaps in making a robust Cochlear Implant System. Proceeding in this direction, an intelligent pre-processor for speech quality enhancement in Cochlear Implants to aid hearing impairment is being proposed. Now biological and other important terms used in relevant chapters have been further described as per the basic requirements.

1. **Physiology of hearing**

The ear carries out two functions - hearing and balancing. It has three parts external, middle and inner (Fig 1.1).

The external and middle ear are meant for hearing while the inner ear is responsible for both hearing and balance. The ear canal is about 25mm long. Fine hairs and glands are found in the outer part (Fig 1.2(a)). The external ear collects airborne sounds and also protects the eardrum from mechanical damage. Humans can perceive sound in the range of 20 to 20000 Hz [1]. Anything that does not lie within this range is not considered audible and hears loss becomes apparent.



**Fig-1.1 Structure of the ear [2]**

“Sound travels through the outer ear, middle ear, inner ear, auditory nerve and then into the brain as a series of transformations [3, 4]. The outer ear picks up acoustic pressure waves that are converted to mechanical vibrations by a series of small bones in the middle ear. In the inner ear, the cochlea, a snail-shaped cavity filled with fluid, transforms the mechanical vibrations into vibrations in the fluid. Pressure variations within the fluids of the cochlea lead to displacements of a flexible membrane, called the basilar membrane, in the cochlea. These displacements contain information about the frequency of the acoustic signal.

Hair cells are used to bend according to the displacements of the basilar membrane since they are attached to the basilar membrane. The bending of these hairs releases an electrochemical substance that causes neurons to fire, signalling the presence of excitation at a particular site in the inner ear. These neurons communicate with the central nervous system and transmit information about the acoustic signal to the brain [5, 6, 7, 8].”

|  |  |
| --- | --- |
|  |  |
| **(a) Normal Cochlea with hair cells** | **(b) Cochlea after insertion of an electrode** |
|  |  |

**Fig-1.2 (a) Normal Cochlea (b) Cochlea after implant placement**

Hearing loss has become a common problem, to reason like, noise, aging, disease, and heredity.

Nowadays hearing loss caused by noise pollution is a worldwide concern. Efforts are being done seriously to cut it as much as possible. Hearing loss can be of three types conductive, sensorineural, and mixed [9, 10]. To recover from such kinds of problems, a Cochlear implant is the only solution. The hearing aid is of no use because hearing aids only amplify the sound while CI is provided the sound. Cochlear Implant implantation happened in the inner ear to restore hearing.

However, implantation is not possible in every case. Implantation can only be done when hair cells of the inner ear are damaged (Fig 1.2(b)) or the auditory nerves are partially damaged. But in cases where all the auditory nerves are damaged CI is not useful to the user, because the CI is essentially meant to transfer the sound to the auditory nerves, which in turn transmit the sensation of hearing to the brain. The aim of Cochlear implant developers is to stimulate remaining normal neurons through electrical signals [11, 12]. The immense challenge is to find out useful information according to the user by stimulating auditory neurons as speech which will be understood by the brain [13, 14].

1. **Cochlear Implant**

Cochlear Implant is an electronic device, that is a boon to people who are hearing impaired. It is the only implant, that can help people with 100% hearing impaired to listen and understand voice and speech. “In a Cochlear implant, a sound processor is placed behind the ear, which captures the incoming sound and converts it into audible digital code. There is no internal battery due to biocompatibility reasons and only the sound processor has a battery that provides power for the entire system. The internal part of the CI gets digitally-coded sound by the processor through the coil. Then digitally-coded sound is converted into electrical impulses by the implant and sends them along with the electrode array placed in the cochlea (the inner ear) to the brain. The electrodes stimulate the cochlea's hearing nerve, which then sends the impulses to the brain where they are interpreted as sound.” CI is placed in the ear as shown in Fig 1.3 [15, 16].



**Fig-1.3 Cochlear Implant Placement in Ear**

1. **Methodology**

Keeping all the things in mind in terms of sound and speech there is a need to design a model which can give better results to the CI users in the midst of background noise. Thus, a formulation of a research proposal is required:

The main phases of our research proposal are-

1. Implementation of existing FAME-based speech synthesis model.
2. Design and implementation of the intelligent pre-processor model.
3. Validation of both models.

The block diagram of the proposed Intelligent Preprocessor model is shown below in Fig 1.4.

 speech signal synthesized speech

Intelligent

Pre-processor

**Fig-1.4 Block diagram of the proposed Intelligent Preprocessor model**

Here (Frequency Amplitude Modulation Encoding) FAME is a basic coding technique of the cochlear implant through which the sound signal sends to the electrodes of the cochlear implant. In the same way, there is another coding technique which is called (Continuous Interleaved Sampling) CIS, which is currently used by all the manufacturers of cochlear implants.

1. **Frequency Amplitude Modulation Encoding (FAME)**

Every tone has regular vibration. Simple tone has only one frequency while complex tone has two or more. Its intensity may vary, which depends upon mood. Every speaker has a unique tone which actually helps a person with normal hearing to identify the speaker through his/her voice. FAME, does amplitude and frequency modulation which have been used here, because of which signal energy is conserved leading to many benefits in the form of frequencies like speaker identification, speaker formants, and speaker tone.

BPF 1

BPF N

AM Extraction

Demodulator

AM Extraction

Demodulator

FM Extraction

FM Extraction

+fc1

integration

integration

+fcn

Sound

 **.**

 **.**

 **.**

Summation

Synthesized sound

**Fig-1.5 Block Diagram of FAME (Frequency Amplitude Modulation Encoding)**

These characteristics of frequency modulation provide more clarity in sound to the user. FM codes temporal fine structure from the speech signal. The functioning of FAME is shown in the block diagram (Fig 1.5).

The unique feature of FAME which makes it different from other coding techniques is the combined functioning of amplitude and frequency modulation for getting the output in the form of signals. These signals reached the electrodes in the form of biphasic pulses [17, 18, 19, 20, 21, 22, 23, 24, 25, 26, 27, 28, 29].

An intelligent pre-processor for speech quality enhancement in Cochlear Implants to aid hearing impairment is being proposed. In this pre-processor, one additional noise removal technique has been used with FAME basic coding technique. This noise removal technique is Independent component analysis (ICA).

1. **Independent component analysis (ICA)**

Independent component analysis is used for learning a linear transform of a random vector. With ICA one can find out independent and non-Gaussian components. The independent component analysis theory was mainly developed in the 1990s. It was developed for analyzing multivariate data. ICA is able to work in that manner where classical methods fail to find the underlying components from the mixed sources. With the help of statistical properties, ICA tries to find the original components or sources. In ICA the underlying processes are supposed to be independent of each other. ICA works on non-Gaussian components, which are used for the recovery of the underlying components from the data. ICA is the unsupervised method because it takes a single matrix in the form of data. In ICA division of data for calculation or desired output should be known and is not required.

If there is a room where a number of people are speaking simultaneously just like a party. In that room if a speaker is placed near every person according to location. In that situation, if the calculation of all original signals can be done by these microphones then it is called the cocktail party problem Fig 1.6 [30, 31, 32, 33, 34, 35].



**Fig-1.6 ICA Functioning**

Here ICA was used for the identification of the different speakers. It was used in a quiet environment like an office meeting where the CI user is unable to understand the speech because multiple speakers speak simultaneously. By this method, users will be able to listen to each person’s voice individually. Output is given in Fig 1.7 below:

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**Amplitude**

**Frequency**

**Fig-1.7 Different speech filtration by ICA**

In research work, ICA is used for the identification of speech with different kinds of noises in different decibels like -6, -5, -3, 0, 3, 5, 6, and 10dB. It was used in a noisy environment like a restaurant, bus stand, etc. where CI user is unable to understand the speech because different kinds of noises are being transmitted simultaneously. By this method, a user is able to listen to voice better in terms of clarity. Method steps can be seen in the flow chart below (Fig 1.8):

Transpose of signal

Inverse Covariance

Matrix square root

Calculate mean and size of signal

Repeat copies of array and remove from original signal

Multiplication

Sum of array and repeat copies of array

Decompose singular values of matrix

Multiply singular value with original signal

Band-pass filtering

**Fig-1.8 Flow chart of ICA for Noise removal from speech**

**ICA used as a noise removal technique in preprocessor**

In this study, the main focus was on two important parameters like quality and intelligibility. It was noticed that sometimes when the quality was good the intelligibility gets reduced. Therefore, it is advisable to have some noise level to avoid distortion of intelligibility. This point has been taken into consideration while carrying out the experiments. The experiments have been conducted in 8 noisy environments such as bus, train, babble, airport, car, traffic, restaurant, and submarine. Under each of these noise environments, the Hearing in Noise Test (HINT) has been used.

Several existing tests can be used to assess speech communication in noise. These tests have been designed to measure percent intelligibility at fixed speech and/or noise levels. Live presentation lost consistency of speech. Thus, recorded sentences are required. In HINT sentences minimum of 4 to 5 syllables are present and out of all types of tests, HINT sentence test is the best [36].

All sentences recording has been done by a male speaker in the English language. The sentences were recorded digitally on a data acquisition system at 44.1 kHz sampling frequency and using a 16-bit A/D converter in a sound-treated room with Avid Pro Tools, Pro Tools11 model. Each CI subject needs two hours for an experiment. In the case of a normal user, one hour is sufficient.

**Training Protocol:**

For CI subject training was required to make them familiar with the apparatus and protocol of the experiments. New words need to be exposed to CI subjects. All CI subjects are not old enough to understand all words present in the sentences, so it was required. Familiarization of all words by all subjects was necessary for the correct output of the experiment.

**Creation of Test Material:**

A number of sentences have been taken for recording in which length was uniform and naturalness was also present in all sentences. The digitally recorded sentence level was adjusted for the listeners. Thus, it seems natural and intelligibility also remains when noise is added to speech. A total of 9 phonetically balanced with 10 sentences each list has been made for adaptive measurement of Hearing in noise test (HINT).

**Development of sentence materials:**

All the sentences have equal Mean square amplitude and intelligibility when used as a noise test material. In these sentences, noise may not be equal. However, phonetic balance, word familiarity, and variation in notation have been taken care of so that intelligibility remains present in the sentences along with noise and the user is able to identify all words in noise at a fixed signal-to-noise ratio.

After the recording of HINT sentences, 8 kinds of noises like bus, train, babble, airport, car, traffic, restaurant, and submarine were added in the sentences with different decibels like -6dB, -5dB, -3dB, 0dB, 3dB, 5dB, 6dB, 10dB (Sound Pressure Level) SPL. To do all these changes Audacity software has been used.

An experiment was conducted using normal hearing subjects to find out the best output from the pre-processor. The ICA technique was used for all kinds of noisy environments. The pre-processor output was used as an input with CIS and FAME coding techniques. The study was done to check the difference between CIS and FAME with a pre-processor for different kinds of noises using HINT sentences. Thus, in this experiment evaluation was carried out between CIS and FAME to understand the difference and to use the best technique for the CI users.

Total 30 normal-hearing participants 18 males and 12 females participated. All subject’s screening has been done with the questionnaire test and all were having normal hearing. All subjects age ranged from 20 to 35 year and who were exposed to the English language at least for 5 years was selected. All were Graduate and postgraduate students. Consent forms have been signed by all subjects.

Cochlear Implant subjects also participated in the experiment. A total of 7 participants participated. 2 participants were post-lingual and 5 were pre-lingual. Participants’ age range from 8 to 34 years with a minimum of 2-year cochlear implant experience and unilateral sensorineural hearing loss. The feedback of CI users below 8 years was of no use for the research purpose. That is the reason they were excluded from the experiment. All CI participants if adults otherwise their parents have signed a consent form. Post-lingual CI user is a user who was having normal hearing capability at the time of language development while a pre-lingual CI user is a user who is having a hearing deficiency from birth.

All subjects were tested at a fixed signal-to-noise ratio with different noise types. PC with a digital dual-core processor has been used. The output was two channels. SONY headphones of the MDR-ZX110A model were used as transducers. In the case of CI user’s loudspeaker of Sony Mega Bass XS-FB162E 6.5-inch Speakers model, was used.

There were 9 sets of HINT sentences and each set contain 10 sentences. Hence a total number of 90 sentences have been used. Sentences were managed with 8 kinds of noise with different SNR levels. In testing a random list of sentences has been played. All other things have been taken care like the minimum threshold and maximum threshold level for the CI participant so that the CI participant did not feel discomfort during the experiment. If the participant feels comfortable then will give the correct output.

First participants need to hear the sounds. After that whatever listeners were listening, have to write down on paper, if participants were adults otherwise whatever participants were listening they have to speak and their parents have to write. The sentences were played at different dB SPL levels like -6dB, -5dB, -3dB, 0dB, 3dB, 5dB, 6dB, and 10dB, and with different kinds of noises like bus, train, babble, airport, car, traffic, restaurant, and submarine. Presentation levels of sentences were increased by 2dB if the participant was not able to understand the sentence. Most sentences were played between 60 to 70dB SPL for normal users and 70 to 80dB SPL for CI users. Sentences were played for normal hearing participants by both coding techniques Continuous Interleaved Sampling and Frequency Amplitude Modulation Encoding to do the comparison. The percentage calculation has been done on the basis of a number of words clearly understood by the listener in a sentence. Proper understanding of words in a sentence was a primary measurement that was considered. In that way, if a sentence has 7 words then out of 7 words if the listener understands only 5 words then the percentage will be 71%. From the results, individuals can get which one is better for CI participants regarding the good perception of speech in different kinds of noises at different decibels. In the case of CI users, only ICA technique output was played without FAME and CIS. Because every CI user is using any coding technique which is already implanted in their CI.

By the comparison of the response of the participant in the form of written sentences at different decibels with different types of noises, anyone can judge the accuracy of sentence understanding. For this mean in Table 1.1 and Table 1.2 have been calculated. The ICA algorithm is used as a preprocessor algorithm.

**Table 1.1: Performance of CIS in percentage with ICA noise removal technique**

|  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- |
| **Noise Type** | **-6 dB** | **-5 dB** | **-3 dB** | **0 dB** | **3 dB** | **5 dB** | **6 dB** | **10 dB** |
| Airport | 73.93 | 67.43 | 81.83 | 87.00 | 87.00 | 82.50 | 93.20 | 94.80 |
| Bus | 43.83 | 70.67 | 85.14 | 88.72 | 93.39 | 91.58 | 96.33 | 97.50 |
| Car | 79.77 | 73.00 | 75.45 | 82.50 | 92.00 | 88.93 | 61.72 | 96.67 |
| Babble | 63.57 | 59.17 | 79.00 | 71.67 | 96.00 | 80.83 | 75.83 | 96.33 |
| Train | 1.48 | 0.00 | 0.00 | 17.78 | 10.00 | 18.70 | 14.44 | 52.96 |
| Traffic | 1.69 | 14.07 | 20.00 | 36.66 | 55.00 | 61.11 | 52.41 | 62.86 |
| Restaurant | 0.69 | 0.69 | 2.67 | 2.07 | 40.59 | 37.59 | 42.21 | 62.90 |
| Submarine | 0.00 | 0.00 | 14.07 | 26.29 | 17.78 | 43.45 | 8.57 | 91.59 |

**Table 1.2: Performance of FAME in percentage with ICA noise removal technique**

|  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- |
| **Noise Type** | **-6 dB** | **-5 dB** | **-3 dB** | **0 dB** | **3 dB** | **5 dB** | **6 dB** | **10 dB** |
| Airport | 74.50 | 61.43 | 83.50 | 86.33 | 88.83 | 83.17 | 94.33 | 94.23 |
| Bus | 52.17 | 72.73 | 88.20 | 90.77 | 92.07 | 90.70 | 97.80 | 98.50 |
| Car | 79.10 | 72.33 | 77.24 | 88.93 | 93.80 | 84.17 | 65.10 | 96.67 |
| Babble | 65.40 | 60.00 | 80.67 | 69.10 | 93.33 | 82.50 | 79.83 | 96.00 |
| Train | 1.48 | 0.00 | 0.00 | 18.52 | 10.74 | 21.48 | 17.26 | 53.14 |
| Traffic | 2.32 | 14.57 | 27.86 | 36.21 | 63.28 | 63.21 | 56.55 | 64.64 |
| Restaurant | 1.72 | 0.00 | 6.21 | 2.07 | 46.31 | 38.28 | 49.62 | 64.10 |
| Submarine | 0.00 | 0.71 | 20.00 | 31.07 | 25.19 | 42.07 | 11.79 | 91.59 |

**Noise Type**

**Fig-1.9 Graph based on the mean values of CIS**

**Noise Type**

**Fig-1.10 Graph based on the mean values of FAME**

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**Fig-1.11 Linear Regression for CIS values**

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## Fig-1.12 Linear Regression for FAME values

## Result:

This experiment has been done in two ways. Four types of noise are two-dimensional like buses, babble, airports, and cars, and four types of noise are dimensional like traffic, restaurant, train, and submarine. This was done to show that ICA was useful in all noisy situations when signals were two-dimensional or more but its results get degraded when signals were one-dimensional. Thus, for good results in all kinds of noises minimum, two-dimensional signals are required. This can be achieved by using a minimum of two microphones in a cochlear implant. It shows that ICA is best with FAME for all kinds of noises with different decibels.

From the results which were shown in Table 1.1 and Table 1.2, it is clear that FAME is better than CIS in all kinds of noises with a pre-processor. The mean values which were shown in Table 1.1 and Table 1.2. Based on these values graphs have been plotted. The graphs which are shown in Fig 1.9 and Fig 1.10 regarding CIS and FAME respectively gave more transparency to satisfy the decision.

For more simplicity linear regression was also calculated for CIS Fig 1.11 and FAME Fig 1.12 individually on the basis of Table 1.1 and Table 1.2 values.

For more precision of the experiment result, the calculation of errors has been done at data based on standard deviation which was shown in Table 1.3 and Table 1.4 for CIS and FAME respectively.

**Table 1.3: Error calculation through Standard deviation for CIS**

|  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- |
| **Noise Type** | **-6 dB** | **-5 dB** | **-3 dB** | **0 dB** | **3 dB** | **5 dB** | **6 dB** | **10 dB** |
| Airport | 20.10 | 30.44 | 14.53 | 21.84 | 18.03 | 20.88 | 8.47 | 9.25 |
| Bus | 25.18 | 26.64 | 17.94 | 18.93 | 11.57 | 15.34 | 7.40 | 4.69 |
| Car | 18.67 | 33.54 | 19.48 | 11.84 | 13.90 | 17.56 | 24.51 | 20.42 |
| Babble | 34.00 | 34.42 | 36.92 | 18.29 | 15.22 | 29.86 | 26.65 | 30.53 |
| Train | 5.08 | 0.00 | 0.00 | 21.35 | 14.50 | 22.42 | 26.40 | 32.93 |
| Traffic | 6.60 | 24.45 | 26.52 | 33.15 | 40.95 | 36.08 | 23.40 | 40.12 |
| Restaurant | 3.65 | 3.65 | 8.68 | 6.10 | 36.21 | 36.81 | 37.54 | 35.33 |
| Submarine | 0.00 | 0.00 | 24.95 | 22.93 | 24.92 | 27.60 | 18.65 | 22.45 |

**Table 1.4: Error calculation through Standard deviation for FAME**

|  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- |
| **Noise Type** | **-6 dB** | **-5 dB** | **-3 dB** | **0 dB** | **3 dB** | **5 dB** | **6 dB** | **10 dB** |
| Airport | 20.60 | 33.25 | 11.90 | 21.73 | 16.90 | 20.45 | 8.15 | 9.43 |
| Bus | 21.72 | 23.80 | 16.97 | 17.94 | 12.42 | 14.47 | 7.25 | 4.58 |
| Car | 19.01 | 34.31 | 19.10 | 11.84 | 12.11 | 16.72 | 22.22 | 20.42 |
| Babble | 31.13 | 34.49 | 34.26 | 25.37 | 15.22 | 28.73 | 24.44 | 29.91 |
| Train | 5.08 | 0.00 | 0.00 | 21.15 | 15.46 | 24.93 | 28.00 | 32.37 |
| Traffic | 7.10 | 23.31 | 29.37 | 33.10 | 39.83 | 34.29 | 22.93 | 38.44 |
| Restaurant | 5.31 | 0.00 | 10.70 | 6.10 | 36.94 | 38.16 | 37.98 | 33.61 |
| Submarine | 0.00 | 3.65 | 28.53 | 31.38 | 28.27 | 27.03 | 19.73 | 22.45 |

In this chapter, the experiments have been done with different kinds of noises with different decibels to do the comparison between CIS and FAME coding techniques. As with the experiment it was proved here that the FAME coding technique is better than the CIS coding technique for CI users in all respect.

Due to social inhibitions and other practical issues, people hesitate to come and be part of the experiments. Further, it is also a big problem if a repeat of the experiment is required. Despite multiple efforts in various cities in the research domain, very limited sources were available. A total of 40 normal hearing subjects and 7 CI subjects’ feedback have been used in our experiments. Although more than 15 CI subjects’ feedback has been taken in the experiments. Due to the age of most CI users, which is below 10 years. The feedback taken from those CI subjects was of no use. So finally, a total of 7 CI user’s feedback was used, which gave meaningful results to the experiments.

There is one more implant that is related to hearing. This is called a Bone anchored hearing system. This system is useful when a person is having a problem in the middle ear (3 bones) or in the ear canal. This technology sends the sound vibration to the inner ear directly. This technology is useful when the middle ear or ear canal is not working properly but the inner ear is working properly [37].

This is different from a hearing aid. This is an implant that fits in the middle part of the ear. This also works like the middle part of the ear. This technique helps the person who is having conductive or mixed hearing loss or single side deafness or chronically draining ears [38].

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