# THE TECHNIQUE USED FOR PRE-PROCESSOR FOR NOISE REMOVAL FROM THE INCOMING SOUND FOR THE COCHLEAR IMPLANT USER

# Abstract

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The proposal aims to enhance speech quality for Cochlear Implant (CI) users by developing an intelligent pre-processor that integrates existing models with new techniques to mitigate background noise interference. The CI process aims to restore hearing for individuals facing various hearing impairments, but existing techniques have limitations, particularly regarding speech intelligibility and noise interference.

**Keywords:** Cochlear Implant, Processor, Hearing.

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## I. 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 are 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 (Figure 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 (Figure 2). 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.



Figure 1: Structure of the ear [2]

Sound travels through a remarkable journey, undergoing a series of fascinating transformations. It begins it's voyage in the outer ear, where it encounters acoustic pressure waves eagerly waiting to be explored. These waves, ever so eager, venture into the middle ear, where they encounter a group of tiny bones that work harmoniously to convert them into delightful mechanical vibrations.

But the journey doesn't end there. The sound waves continue their adventure into the inner ear, a mystical realm known as the cochlea. This snail-shaped sanctuary is filled with fluid, ready to receive and transform mechanical vibrations into fluidic vibrations. As the sound waves dance within the cochlear fluids, pressure variations emerge, causing a flexible membrane known as the basilar membrane to sway and respond to the frequency of the captivating acoustic signals.

After that, the basilar membrane is adorned with magnificent hair cells, like delicate tendrils awaiting their moment of glory. These remarkable hairs are attached to the basilar membrane, and as it sways and dances to the rhythm of the sound waves, the hairs gracefully bend and sway in unison. And as they do, a wondrous electrochemical substance is released, causing a cascade of excitement within the neurons.

These neurons, ever diligent, fire and transmit signals to the magnificent central nervous system. They communicate the news, the presence of exhilarating excitation at a particular location within the inner ear. And with great zeal, the neurons convey this vital information to the brain, allowing it to unravel the mysteries of the acoustic signal and comprehend the beauty of sound.

Thus, the journey of sound, from the outer ear to the brain, unfolds with elegance and grace, revealing the intricate mechanisms by which we perceive the world of auditory delights [3,4, 5, 6, 7, 8]



Figure 2: Normal Cochlea



Figure 3: 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 (Figure 3) 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].

2. 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 positioned behind the ear. This device is responsible for capturing the incoming sounds and transforming them 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. After the sound is converted into a digital code by the sound processor, it is further transformed into electrical impulses by the cochlear implant. These electrical signals are then transmitted through the electrode array, which is positioned in the cochlea, the inner ear. The purpose of the electrode array is to stimulate the hearing nerve within the cochlea. As a result, the stimulated nerve sends these impulses to the brain. Finally, the brain interprets these signals as sound, allowing the individual to perceive and understand the auditory information.CI is placed in the ear as shown in Figure 4 [15, 16].



Figure 4: Cochlear Implant Placement in Ear

**3.** 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:

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

The block diagram of the proposed Intelligent Preprocessor model is shown below in Figure 5



Figure 5: 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.

4. Frequency Amplitude Modulation Encoding (FAME): In the realm of sound, each tone possesses its own rhythmic vibration. A simple tone gracefully resonates with a singular frequency, while a complex tone unveils the beauty of multiple frequencies intertwined harmoniously. As these tones emerge, their intensity dances to the tune of the mood, shaping the emotional landscape they traverse. Within the vast tapestry of voices, every speaker possesses a unique tone, an auditory fingerprint that sets them apart. It is through this melodic signature, carried within their voice, that individuals with keen hearing can unravel the identity of the speaker. The tone becomes a captivating thread that weaves connections, allowing recognition and familiarity to flourish.

The FAME (Frequency Amplitude Modulation Encoding), is a magical technique employed to preserve the energy of the signal. It embraces the elegant dance of amplitude and frequency modulation, bringing forth a symphony of benefits. Through this harmonious interplay, precious frequencies emerge, revealing secrets such as speaker identification, speaker formants, and the very essence of the speaker's tone. Thus, within the symphony of sound, the vibrations of tones paint a vivid canvas of sensory delight. Their intricacies and variations intertwine with human perception, creating a tapestry of voices that echoes the essence of our shared existence.

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Figure 6: Block Diagram of FAME (Frequency Amplitude Modulation Encoding)

Frequency modulation (FM) in cochlear implants offers enhanced sound clarity to the user. It codes the temporal fine structure of speech signals. The functioning of Frequency Allocation for Multiple Electrodes (FAME) is illustrated in the block diagram (Figure 6). What sets FAME apart from other coding techniques is its distinctive approach of utilizing both amplitude and frequency modulation to generate output signals. These signals are then delivered to the electrodes as 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).

5. 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 Figure 7 [30, 31, 32, 33, 34, 35].

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Figure 7: 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 Figure 8 below:



Frequency

Figure 8: 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 (Figure 9): Futuristic Trends in Medical Sciences e-ISBN: 978-93-6252-593-2 IIP Series, Volume 3, Book 21, Part 1, Chapter 1 THE TECHNIQUE USED FOR PRE-PROCESSOR FOR NOISE REMOVAL FROM THE INCOMING SOUND FOR THE COCHLEAR IMPLANT USER



Figure 9: Flow chart of ICA for Noise removal from speech

# II. ICA USED AS A NOISE REMOVAL TECHNIQUE IN PREPROCESSOR

In this study, the primary focus was on two important parameters: quality and intelligibility. It was observed that sometimes when the quality was high, the intelligibility was reduced. Therefore, to prevent the distortion of intelligibility, it was deemed advisable to introduce a certain level of noise. This consideration was considered during the experiments.

The experiments were conducted in eight different noisy environments, including a bus, train, babble, airport, car, traffic, restaurant, and submarine. In each of these noise environments, the Hearing in Noise Test (HINT) was utilized. Various existing tests are available for assessing speech communication in noise. These tests are specifically designed to measure the percentage of intelligibility at fixed speech and/or noise levels. To ensure consistency in speech presentation, recorded sentences were used instead of live presentations. The HINT sentence test, which includes a minimum of 4 to 5 syllables in each sentence, was determined to be the most effective among all the available test types [36].

All the sentences were recorded by a male speaker in the English language. The recordings were conducted digitally using a data acquisition system with a sampling

frequency of 44.1 kHz and a 16-bit A/D converter. The recordings took place in a sound-treated room using Avid Pro Tools, specifically the Pro Tools11 model. For each participant with a Cochlear Implant (CI), the experiment duration was approximately two hours. However, for individuals with normal hearing, one hour was found to be sufficient for the experiment.

- 1. Training Protocol: For subjects with a Cochlear Implant (CI), training was necessary to acquaint them with the apparatus and the experimental protocol. Since the sentences used in the experiments contained new words, it was essential to expose the CI subjects to these unfamiliar words. Not all CI subjects were of an age where they could comprehend all the words present in the sentences. Therefore, it was important to ensure that all subjects were familiarized with every word used in the experiment to obtain accurate results.
- 2. Creation of Test Material: A set of sentences was carefully chosen for recording, ensuring that they had a uniform length and maintained a sense of naturalness. The sentence recordings were adjusted digitally to an appropriate level for the listeners. As a result, the sentences sounded natural, and their intelligibility was preserved even when noise was added to the speech.

To conduct the adaptive measurement of the Hearing in Noise Test (HINT), a total of nine lists were created. Each list consisted of ten sentences that were phonetically balanced. These lists were designed to facilitate an accurate assessment of speech intelligibility in the presence of background noise.

**3. Development of Sentence Materials:** All the sentences used in the HINT recordings were carefully adjusted to have equal Mean Square Amplitude (MSA) and intelligibility when used as noise test material. Although the noise levels may vary across the sentences, measures were taken to ensure phonetic balance, word familiarity, and variation in notation. This was done to maintain intelligibility in the presence of noise, allowing users to identify all the words at a fixed signal-to-noise ratio.

After recording the HINT sentences, eight different types of noise, such as bus, train, babble, airport, car, traffic, restaurant, and submarine, were added to the sentences at various decibel levels ranging from -6dB to 10dB Sound Pressure Level (SPL). Audacity software was used to make these modifications.

An experiment was conducted using subjects with normal hearing to determine the optimal output from the pre-processor. The Independent Component Analysis (ICA) technique was employed for all types of noisy environments. The output of the preprocessor was used as input with both CIS and FAME coding techniques. The study aimed to compare the performance of CIS and FAME with the pre-processor in different noise conditions using HINT sentences. The evaluation aimed to understand the differences between CIS and FAME and determine the best technique for Cochlear Implant (CI) users A total of 30 participants with normal hearing took part in the experiment, consisting of 18 males and 12 females. All participants underwent a screening process using a questionnaire test, confirming their normal hearing abilities. The age range of the subjects was between 20 and 35 years, and they were required to have a minimum of 5 years of exposure to the English language. All participants were either graduate or postgraduate students, and they provided signed consent forms before participating in the study.

Additionally, seven Cochlear Implant (CI) users also took part in the experiment. Among them, two participants were post-lingual CI users, while the remaining five were pre-lingual CI users. The age of the CI participants ranged from 8 to 34 years, and they had a minimum of 2 years of experience with a cochlear implant and unilateral sensorineural hearing loss. CI users below the age of 8 were excluded from the study since their feedback would not serve the research purpose. All adult CI participants, or their parents in the case of minors, signed consent forms. A post-lingual CI user refers to someone who had normal hearing during language development, while a pre-lingual CI user refers to someone with a hearing impairment from birth.

All subjects were tested at a fixed signal-to-noise ratio using various types of noise. A PC with a digital dual-core processor was utilized, and the output was delivered through two channels. SONY headphones of the MDR-ZX110A model were used as transducers for participants with normal hearing, while a loudspeaker of the Sony Mega Bass XS-FB162E 6.5-inch Speakers model was used for CI participants.

There was a total of nine sets of HINT sentences, with each set containing ten sentences, resulting in a total of 90 sentences. The sentences were presented randomly during testing, and measures were taken to ensure the minimum and maximum threshold levels for CI participants, ensuring their comfort and preventing discomfort during the experiment. Participants were encouraged to provide accurate responses when they felt comfortable doing so.

Initially, participants were required to listen to the sounds presented to them. They were instructed to write down what they heard on paper if they were adults. For participants who were not adults, they were asked to speak and their parents were responsible for writing down what they heard. The sentences were played at different sound pressure levels (dB SPL) including -6dB, -5dB, -3dB, 0dB, 3dB, 5dB, 6dB, and 10dB, with various types of noise such as bus, train, babble, airport, car, traffic, restaurant, and submarine.

If a participant had difficulty understanding a sentence, the presentation level was increased by 2dB. The majority of sentences were played at levels between 60 to 70dB SPL for normal hearing participants and 70 to 80dB SPL for CI users. Both coding techniques, Continuous Interleaved Sampling (CIS) and Frequency Amplitude Modulation Encoding (FAME), were used to play the sentences for normal-hearing participants, allowing for a comparison between the two techniques.

The percentage of words clearly understood by the listener in each sentence was calculated, which served as the primary measurement. For example, if a sentence had 7 words and the listener understood only 5 words, the percentage would be calculated as

71%. By analyzing the results, it becomes possible to determine which technique provides better speech perception for CI participants in different types of noise and at various decibel levels. However, for CI users, only the output of the Independent Component Analysis (ICA) technique was played, as they already had a coding technique implanted in their CI.

To assess the accuracy of sentence understanding, the participant's written responses at different decibel levels and noise types were compared. Tables 1 and 2 were used to calculate the mean and present the data. The ICA algorithm was utilized as a pre-processor algorithm.

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: Performance of CIS in percentage with ICA noise removal technique

Table 2: Performance of FAME in percentage with ICA noise removal tech	nnique
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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



Figure 10: Graph based on the mean values of CIS







Figure 12: Linear Regression for CIS values



Figure 13: Linear Regression for FAME values

## **III. 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 and Table 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 and Table 2. Based on these values graphs have been plotted. The graphs which are shown in Figure 10 and Figure 11 regarding CIS and FAME respectively gave more transparency to satisfy the decision.

For more simplicity linear regression was also calculated for CIS Figure 12 and FAME Figure 13 individually on the basis of Table 1 and Table 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 3 and Table 4 for CIS and FAME respectively.

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.

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 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	20.60	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

Table 4: Error calculation through Standard deviation for FAME

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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