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Analysis of the Influence of Sound Source Position and Electrode Placement on the Stimulation Pattern of the Cochlear Implant under Noise Conditions: Optimization of Auditory Perceptual Effects

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Abstract

Background: Cochlear implants (CI) are widely used to restore hearing in people with severe to profound hearing loss. However, optimizing CI performance, especially in difficult listening environments with background noise, remains a major challenge. Understanding the influence of factors such as sound source position and electrode placement on CI stimulation patterns is critical to improving auditory perception. **Methods:** In this study, an analysis was conducted to investigate the influence of sound source position and electrode placement on CI stimulation patterns under noisy conditions. For this purpose, a special measurement setup with a CI speech processor-microphone test box was used to simulate realistic listening scenarios and measure CI performance. **Results:** The results show that the effectiveness of CI noise reduction systems is influenced by factors such as the position of the sound source and electrode placement. In particular, the beamforming ultra zoom mode showed significantly better noise reduction than the omnidirectional mode, especially under real listening conditions. Furthermore, differences in electrode responses indicate individual variability in the CI user experience, highlighting the importance of personalized fitting algorithms. **Conclusions:** The results demonstrate the importance of considering environmental factors and individual differences when optimizing CI performance. Future research efforts should focus on the development of personalized fitting algorithms and the exploration of innovative strategies, such as the integration of artificial intelligence, to improve CI functionality in different listening environments. This study contributes to our understanding of CI stimulation patterns and lays the foundation for improving auditory perception in CI users.

Keywords

Cochlear Implants, Noise Reduction, Ultra Zoom Beamforming, Anatomical Variations, Personalized Fittings

1. Introduction

A cochlear implant (CI) is a medical device that enables people who are deaf or close to deafness to hear again. Unlike hearing aids, which amplify sound, a CI takes over the function of the damaged inner ear by sending electrical signals directly to the auditory nerve. People opt for a cochlear implant when conventional hearing aids are no longer sufficient. CIs enable many people to participate in daily life again by giving them access to speech and environmental sounds. It can be particularly difficult for CI users to communicate in changing soundscapes [1], and conversational sounds, which vary greatly in frequency compared to the general background noise, can cause additional difficulties [2]. A common technique for noise suppression based on signal processing methods with multiple microphones is beamforming. Beamforming is a technique in which directionality is achieved through the coordinated use of multiple microphones. All types of beamforming technologies are based on the basic principles of time and phase differences in the signals received by two or more spatially separated microphones. In clinical studies, the adaptive beamforming technology with an omnidirectional setting was compared with a fixed cardioid polar pattern. No significant differences were found when the interfering signal was positioned directly behind the test subject. However, the adaptive mode showed clear advantages when the interfering signal arrived laterally at an angle of 90˚ or moved across different loudspeakers [3]. It was shown that adaptive beamforming technologies enable dynamic adjustment of noise reduction and can detect interfering signals from different angles. Transducer shadows and diffraction effects can affect directivity, but complex transducer models can predict and minimize these effects [4]. In their series of measurements, the authors Koehler and Wright (2018) found that noise suppression in hearing aids works before 15 seconds. They also found that the stimulus duration had no influence on their results and they were therefore able to dispense with an extension of the measurement duration that had been considered in advance [5]. In audiology, considerable progress has been made in recent years in the development of measurement systems for the technical verification of non-implanted hearing systems. These measurement instruments enable precise evaluation of the performance of hearing aids and thus provide important insights for improving hearing aid technology. Despite these advances, the technical verification of the function of CI speech processors (CI-SP) has not yet received the same attention and established measurement technology. Currently, the evaluation of CI-SP is mainly limited to a so-called "listening check", which only allows subjective listening to the recorded microphone signals [6] [7]. However, this approach does not ensure that the recorded signals are correctly translated into a corresponding stimulation pattern and sent to the implant. This means that the ability to detect defects in important CI components such as the electronics, the microphones or cable breaks in the coil cable at an early stage remains severely limited or even impossible. At the 25th annual conference of the German Society of Audiology (DGA) in Cologne (Germany), a prototype was presented for the first time that goes far beyond the conventional "Listening Check". This prototype not only enables a listening check to be carried out, but also records the individual stimulation currents at the electrode level and evaluates them with the help of a specially developed measuring box [8]. This also makes it possible for the first time to analyze the stimulation pattern of a cochlear implant.

1.1. RMS Value as a Measure of the Average Stimulation (RMS)

The electrode currents are recorded individually for each channel using the developed measuring box. In order to analyze the stimulation pattern, the effective value (RMS value) of the electrode currents recorded individually for each channel—the average stimulation current—is determined as a compact measure for each electrode. Equation (1) calculates the RMS value (in μA units) as follows:

$$x_{\mathrm{RMS}} = \sqrt{\frac{1}{N}\sum\nolimits_{n=1}^{N}\left|x_n\right|^2} \tag{1}$$

with x_n for the N individual samples. Figure 1 shows an example of the average stimulation for the individual channels (right) calculated from the electrode-individual stimulation currents shown on the left.

The different configurations can then be compared with each other using the channel-specific RMS.

1.2. Measuring Systems for Technical Verification

When fitting hearing systems that can be verified via acoustic signals, it is common practice to check them objectively in a measurement box on the patient's ear before fitting, particularly with regard to excessively loud sound. A measurement system is used to offer the patient the best possible setting in the frequency-dynamic fitting in advance. This setting is then further modified based on the patient's subjective hearing impressions to ensure better acceptance. In contrast, the technical testing of speech processors for cochlear implants has so far been limited to a subjective evaluation of the recorded microphone signals, also known as a "listening check". However, this does not allow any statement to be made as to whether the stimulation currents emitted are sufficient for the

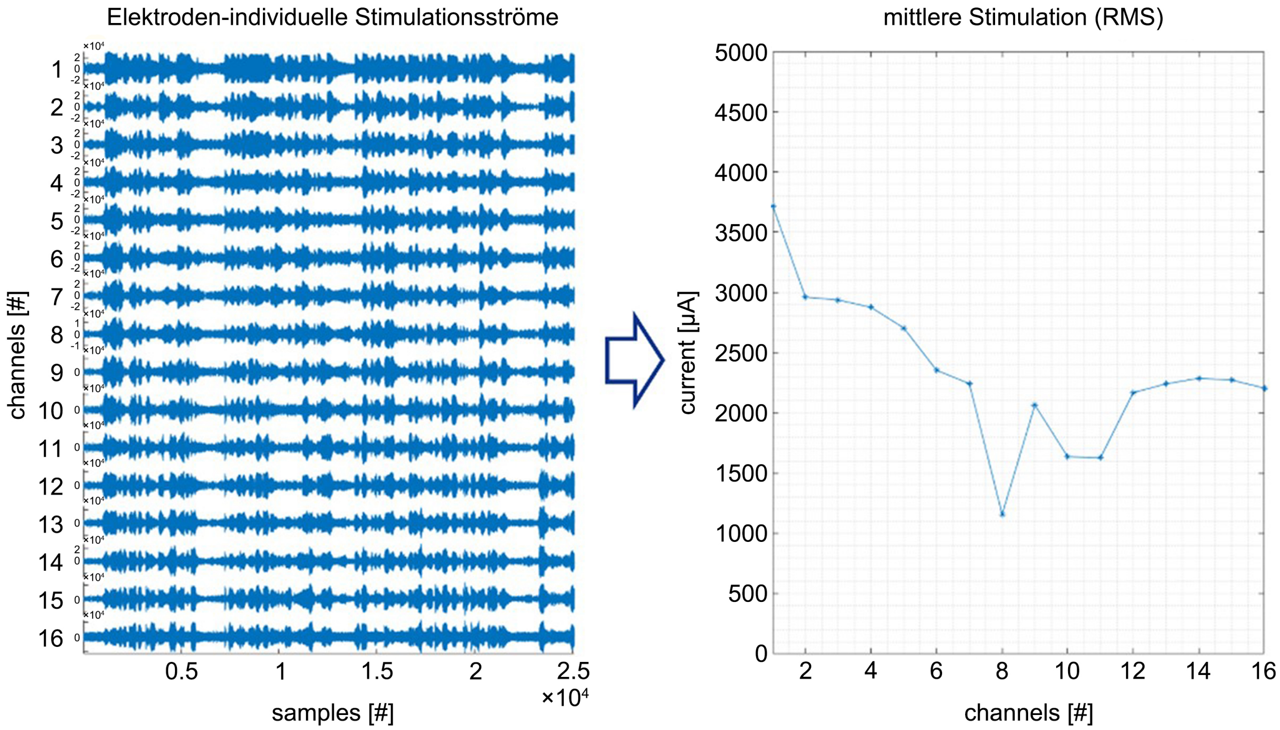

Figure 1. The illustration shows the channel-specific stimulation currents (signal patterns) of the 16 electrodes on the left-hand side. The arrow in the middle is used to combine these into an average stimulation current, the RMS value.

patient, particularly with regard to tonal perception or the basic stimulation threshold. The need for an objective measurement on a measurement box when fitting cochlear implants becomes particularly clear if functions such as beamforming are to be successfully integrated into the everyday life of the CI user, but the technical influencing factors are unknown. In the literature, the behavior of directivity characteristics is sometimes described without objective verification. For example, research results show that the effectiveness of the beamformer decreases when more interfering signals occur and the acoustic environment becomes more diffuse. The results also show that users of cochlear implants can achieve a significant improvement in their performance in noisy environments, especially when the noise level is spatially separated from the target speech [9]. It has been shown that the effectiveness of the beamformer decreases when more noise is present and the acoustic environment becomes more diffuse. The results show that cochlear implant users can achieve a significant improvement in their performance in noisy environments where the noise is spatially separated from the target speech [9]. Stronks *et al.* (2022) found that in a group of hearing aid users who wore a hearing aid in one ear and an implant in the other ear, no improvement in speech intelligibility was achieved in situations with uniform background noise and in conversations with more than one speaker [10]. This was surprising as, based on previous signal-to-noise ratio (SNR) measurements, it was expected that the benefits of beamforming would be more apparent when using a fixed directivity. Although both monaural and binaural adaptive directional microphones as well as fixed directional microphones generally offer clear advantages in difficult listening situations [10]. Herrmann *et al.* (2022) discuss in their publication that the automatic system in the hearing aid can react realistically to different input signals from different listening situations, provided that an authentic spatial representation is taken into account. For this purpose, it is necessary to test the effectiveness of the beamformer under different noise angles [11]. Most hearing impairments affect speech understanding, especially in environments with background noise. There is therefore a significant need to improve the signal-to-noise ratio (S/N) of hearing technologies in general. However, the technical measurement of these improvements presents a challenge. The previous IEC 60118-15 standard has undergone a major revision, with a particular focus on improving the measurement signals for noise measurement (SNR). This becomes particularly important when new technologies such as beamforming are to be integrated into the everyday lives of cochlear implant users, the technical influencing factors of which are still unknown. Holube's research group (2015) modified the ISTS in the IEC 60118-15 and ANSI S3.42 standards as a masking device in speech tests to enable accurate measurement of noise suppression in adaptive directional microphone technologies. The pause duration of the signal was shortened from up to 600 ms to a maximum of 250 ms to avoid possible confusion with the ISTS. The order of the sections within the signal was restructured and the modified signal was given the name "International Female Fluducing Masker" (IFFM). In addition, a stationary noise called International Female Noise (IFnoise) was composed, which has the same spectral properties as ISTS and IFFM, but without temporal fluctuations [12]. An earlier study by Holube *et al.* (2011) has shown that the speech recognition threshold (SRT) for sentences in IFFM is similar to that for natural speech [13]. Hagerman and Olofsson (2004) were the first to develop a precise measurement method for hearing aids under laboratory conditions in a special hearing aid measurement box. This method was later standardized in the industry standards IEC 60118-15 and ANSI S3.42. In this method, two signals, consisting of speech and noise, are presented simultaneously and two measurements are carried out. One of the measurements reverses the phase of the noise. By adding or subtracting the corresponding output signals, the extracted speech or noise can be determined. In this way, the method makes it possible to calculate the gain for each of the two signals, even if they occur simultaneously and affect the signal processing of the hearing aid in the conventional way [14]. The question now arises as to whether this proven methodology can also be applied to the study of the stimulation pattern of a cochlear implant, which is investigated below by analyzing the root mean square (RMS) value of the electrode currents. The specially developed measurement box by Fehling *et al.* (2023) serves as a prototype for the application of this method in CI users. Acoustic perception and the localization of sound sources are essential abilities of the human auditory system that enable us to understand and orient ourselves in our environment. An essential aspect of this ability is the head shadow effect, which is caused by the anatomy of the human head and the associated auditory processing. In the head shadow effect, the sound is attenuated or amplified depending on its position relative to the head, which enables sound sources to be localized precisely. The measurements initially focus on the CI speech processor Advanced Bionics Naída CI Q70 (Naída CI), in which Phonak's ultra zoom beamforming technology is integrated. Ultra zoom is a monaural adaptive beamformer that processes the signals from two omnidirectional microphones along the front and rear axles of the hearing aid in such a way that sound from different directions is dynamically and frequency-specifically attenuated in the rear hemisphere [15]. After the cochlear implant measurement box was developed as a prototype, it has now been successfully used for further measurements similar to the hearing aid measurement box in the laboratory [8]. This opens up the possibility of drawing conclusions about ultra zoom beamforming and better understanding the technical influencing factors.

1.3. Research objectives and hypotheses

The scientific work is based on the following considerations: Hypothesis on sound source position (H1): It is tested whether the position of the sound source has a statistically significant influence on the stimulation pattern of the cochlear implant between the omni-directional and beamforming ultra zoom recording modes. Hypothesis on electrode position (H2): It is hypothesized that there will be a statistically significant difference in the stimulation pattern of the cochlear implant depending on the electrode position. Interaction hypothesis (H3): It will be investigated whether the interaction between sound source position and the omni-directional and beamforming ultra zoom recording modes has a significant effect on the stimulation pattern of the cochlear implant. The results of these investigations are analyzed for significance in order to identify relevant patterns and correlations.

2. Design and Method

2.1. Hardware and Software

Data was collected using the ACAM4CI objective measurement system from Acousticon Hörsysteme GmbH (Reinheim, Germany) with serial number 0002. Electrical stimulation of the reference implant CI HiRes 90k with serial number 7095446-008 from Advanced Bionics (Stäfa, Switzerland) was performed using the AB CPI-3 programming interface with serial number 102081 from the same manufacturer. The AB SoundWave 3.1 software was used to program the AB Naida CI Q70 S/N 1042111 (SP2 blue) CI speech processor. In experimental audiology, artificial head models such as the Head and Torso Simulator (HATS) from Brüel & Kjær or the G.R.A.S. KEMAR HATS from GRAS Sound & Vibration are often used to simulate human anatomy and auditory processing. Although such models cannot perfectly replicate human anatomy and auditory processing, the recorded stimulation patterns of the cochlear implant still allow the recording of head-related, channel-specific electrode currents (RMS values). An artificial head made of polystyrene was available in our research laboratory, which enabled a much more compact measurement setup compared to the head models mentioned above. Even if this does not simulate all aspects of the human anatomy, it provides a sufficiently accurate representation of the head shadow effect caused by the anatomy of the human head. The CI speech processor was mounted on an artificial head made of polystyrene for measurement under real conditions (see **Figure 2**).

The raw data was processed using MathWorks MatLab and the 2015a version of the Signal Processing Toolbox. Both the International Female Fluducing Masker (IFFM) and International Female Noise (IFnoise) were controlled using the same version of MathWorks MATLAB and played back through two M-Audio BX5 active 2-way nearfield studio monitor reference speakers. Both IFFM and IFnoise are available at www.ehima.com under "Documents". The statistical analysis was carried out using Microsoft Excel (Office 365, Microsoft, Redmond, USA).

2.2. Basic Settings on the CI Speech Processor

The universal headpiece (UHP) was connected to the Acam4CI measurement box via an extended coil cable. Two listening programs (P1 and P2) were created with the same basic MAP configuration (MAP #1). The T-level threshold across all electrodes was set to 40 CU and the M-level threshold to 400 CU. Functions such as ClearVoice and SoftVoice were switched off. The HiRes Optima Speech

Figure 2. The picture shows the artificial head to which the CI speech processor (AB Naida CI Q70) is attached by means of a headband. The CI speech processor is connected via the AB CPI-3 programming interface and coupled to the measuring box via a long coil cable with the transmitter coil. The measuring box contains a CI HiRes 90k reference implant, which can be used to measure the stimulation currents.

coding strategy was used, the IDR was set to 60, the sensitivity to 0 dB, the AGC control to Dual Loop, the pulse width to APW II with 44.0 μs and the channel rate to 1515 pps. For the listening program P1, declared as a music program, the microphone mode was set to omni-directional microphones with Mic Only audio mixing. Functions such as Windblock, SoundRelax and ComPilot were deactivated while the processor microphone mode was activated. However, for the P2 listening program, which is intended for speech in noise, the microphone mode was set to ultra zoom with Mic Only audio mixing. The processor microphone mode was also activated, while Windblock, SoundRelax, ComPilot and Echoblock were deactivated. Live mode was activated by not using the basic MAP configuration, but simply selecting the desired listening program (P1 or P2) in the CI speech processor and starting stimulation. This was done using both processor microphones (front mic and rear mic), as shown in Figure 3.

2.3. Reference Measurement without the Influence of Background Noise

In order to obtain an unadulterated reference measurement, referred to here as the "baseline", without the influence of background noise and sound shadow effects caused by the artificial head, the CI speech processor was positioned in the middle between two loudspeakers at a distance of one meter. The position of the CI speech processor was set to 0 degrees as the axis of rotation. This prepared the basic measurement setup with two loudspeakers for the following measurements. The selection of the specific angles for the study was based on several considerations. Firstly, angles were chosen to represent typical hearing ranges of CI users in order to assess the performance of the cochlear implant in realistic

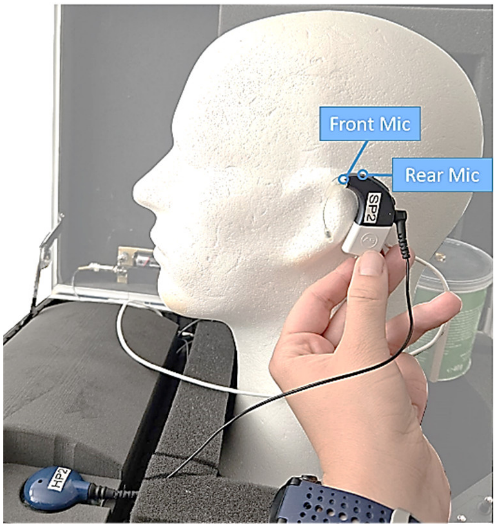

Figure 3. Side view of the artificial head with a rudimentary raised auricle. The Naida CI Q70 speech processor (SP2) is fixed to the head in this position with a headband. The activated processor microphones front mic and rear mic are used for the measurements in the study to compare the effect of ultra zoom beamforming with the effect of the omnidirectional microphone.

hearing scenarios. Secondly, the angles were chosen so that different aspects of sound localization and the head shadow effect could be investigated. By varying the angles, the head shadow effect could be simulated at different angles of sound incidence on the head. The angles were varied according to industry standards IEC 60118-15 and ANSI S3.42 and theoretical considerations to allow a comprehensive investigation of sound localization and cochlear implant performance. The CI speech processor was rotated at eight equidistant angles of ±45 degrees relative to the measurement setup (angles: 0, 45, 90, 135, 180, 225, 270 and 315 degrees). Each position was sonicated for 16 seconds with 70 dB SPL useful sound (IFFM) from a single loudspeaker. Five repeat measurements were carried out in each case. For this measurement setup, the basic setting of the CI speech processor in listening program P1 was used, with the omni-directional microphone mode set. The resulting "baseline" is used in the following section to calculate the relative effect of noise reduction in the form of quotient values for statistical evaluation.

2.4. Measurement Setup of the Loudspeakers under Noise Conditions

In accordance with the IEC 60118-15 and ANSI S3.42 standards, the recommended measurement setup was used in this study to investigate the influence of the position of the sound sources on the stimulation pattern of the cochlear implant. The CI speech processor was positioned in the middle between two loudspeakers at a distance of one meter. The position of the CI speech processor was defined as the axis of rotation at 0 degrees. To clarify: the focus was not on the position of the artificial head, but on the CI speech processor. The artificial head performed a decentered rotation with the speech processor. There were two measurement conditions: (A) Dichotic measurement setup (S0 N180): One speaker emitted the IFFM (useful sound) from the front, while a second speaker emitted the IFnoise (background noise) from the rear. (B) Monotonic measurement setup (S180 N180): One loudspeaker emitted both the IFFM and the IFnoise from behind. In both measurement conditions, the CI speech processor was rotated at eight equidistant angles of ±45 degrees relative to the measurement setup (angles: 0, 45, 90, 135, 180, 225, 270 and 315 degrees). Each position was sounded for 16 seconds at a volume of 70 dB SPL for the useful sound and 50 dB SPL for the background noise (+20 dB SNR), corresponding to the two measurement conditions. Five repeat measurements were carried out in each case. Both speech and noise were included, not only because they are occasionally recommended as preferred stimuli in the IEC 60118-15 and ANSI S3.42 test standards, but also because advanced hearing aid algorithms such as Phonak ultra zoom have designed their signal preprocessing to preprocess speech and noise. The described test setup was measured once under laboratory conditions (see **Figure 4**), *i.e.* with rotation of the CI speech processor alone, and once under real conditions (see **Figure 5**) with an artificial head.

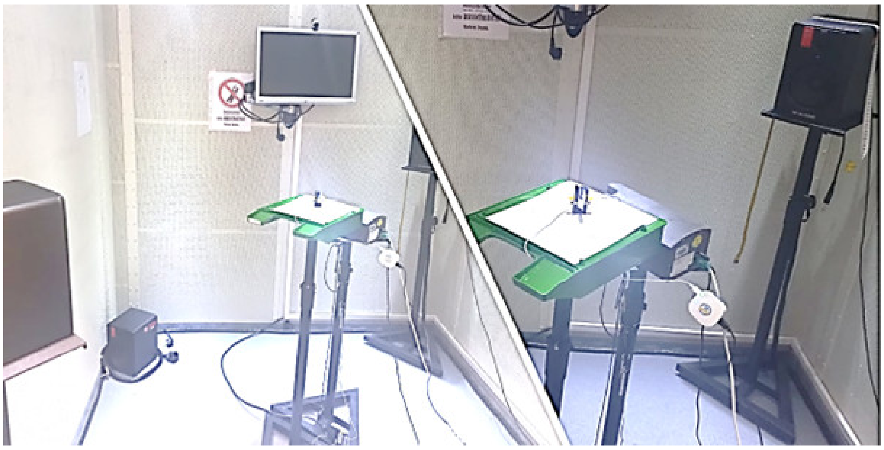

Figure 4. The figure is divided into two parts to better illustrate the measurement setup. The speech processor is exposed to sound via two loudspeakers in an anechoic chamber, while the directional characteristic is measured via the stimulation currents. The speech processor is rotated to simulate different sound angles. The same measurement setup is used to determine the baseline and obtain an interference-free signal pattern, which minimizes acoustic shadow effects.

Figure 5. The figure is divided into two parts to better illustrate the measurement setup. On the left side of the ear, the speech processor is attached to the artificial head with a headband and sound is played through two loudspeakers. The axis of rotation corresponds to the axis of symmetry of the speech processor around which the head is rotated. The head shadow effects are measured via the stimulation currents.

During the measurement under real conditions, a CI speech processor was attached to the left side of an artificial head. The respective measurement setup should allow conclusions to be drawn as to whether the phase of the noise can be rotated as accurately as possible and whether the extracted speech or the extracted noise can be determined by adding or subtracting the corresponding output signals of the stimulation pattern. For the statistical evaluation, the relative effect of the noise suppression was calculated in the form of quotient values. Equation (2) was calculated using the following method:

$$\text{quotient value} = \left(\frac{(\text{dichotisch} - \text{monotisch}) - \text{baseline}}{\text{baseline}} \right) \times 100 \quad (2)$$

Here, "dichotic" stands for the measured effect in the dichotic measurement arrangement, "monotonic" for the measured effect in the monotonic measurement arrangement and "baseline" for the initial state of the stimulation without background noise. The quotient value obtained is expressed as a percentage to quantify the relative change in noise suppression.

2.5. Statistics

With regard to the statistical analysis, the two-factor analysis of variance with repeated measures (ANOVA) was chosen as the preferred method. This decision is based on the ability of ANOVA to capture interaction effects between variables while adequately accounting for repeated measurements. This method of analysis enables a holistic assessment of the influences of different parameters on the electrode currents. The choice of ANOVA is directly related to the research objectives of the study, as it helps to answer the specific questions regarding the influence of sound source position, electrode position and the interaction between different recording modes on the stimulation pattern of the cochlear implant. The significance level was set at 0.05.

3. Results

The measurement setup without the artificial head is shown in Figure 6 and with artificial head in Figure 7, in each case in comparison between omnidirectional microphone mode and ultra zoom beamforming mode. The relative effect (quotient values) of the noise reduction, measured at eight different angles, is shown. A detailed statistical analysis of the data can be found in Table 1 to Table 6.

The two-factor analysis of variance with repeated measures (ANOVA) shown in Table 1 revealed no statistically significant differences between the microphone modes. This applied to both the omnidirectional mode and the beamforming ultra zoom mode with the same measurement setup without an artificial head. This was confirmed by the high F-value ($F = 14.9212$, $p = 0.0001$), while the critical F-value was 3.8833. The F-value is a measure of whether the mean

Table 1. P1 without head (omni-directional) vs. P2 without head (ultra zoom).

Cause of Variation	Sum of Squares (SS)	Degrees of Freedom (df)	Mean Square (MS)	Test Statistic (F)	P-Value	Critical F-Value
Sample	1068.83	1	1068.83	14.9212	0.0001	3.8833
Columns (Electrodes)	3213.66	15	214.24	2.9909	0.00022	1.7112
Interaction	3207.15	15	213.81	2.9849	0.00023	1.7112
Error	16045.43	224	71.63			
Total	23535.08	255				

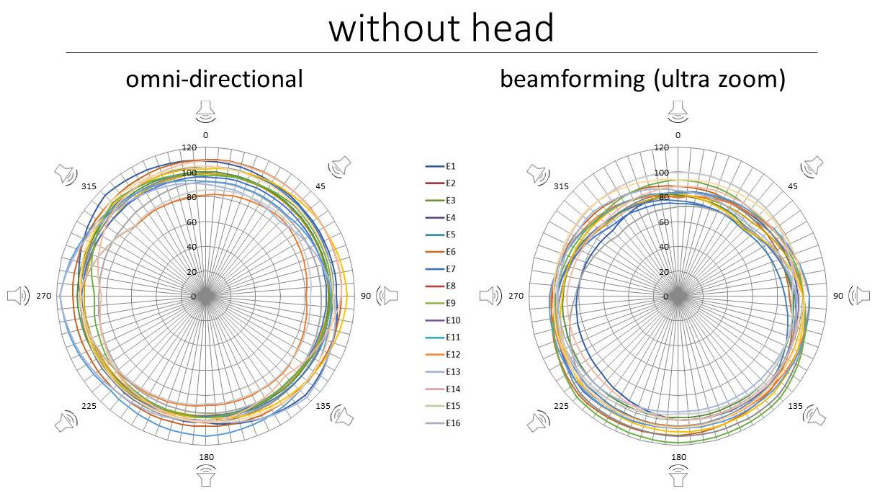

Figure 6. Measurement setup without artificial head. The left polar diagram shows the omni-directional microphone mode compared to the right polar diagram in ultra zoom beamforming mode. All 16 electrodes are shown for each angle. The eight defined interference sound angles are marked by a loudspeaker pictogram at the edge of the polar diagram. The strongest noise suppression can be shown in beamforming mode at an angle of 180 degrees (from behind).

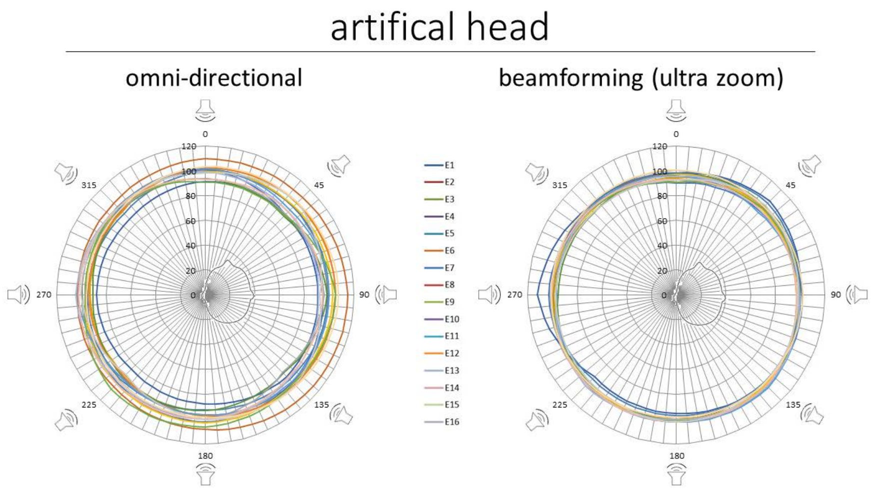

Figure 7. Measurement setup with artificial head. The left polar diagram shows the omni-directional microphone mode compared to the right polar diagram in ultra zoom beamforming mode. All 16 electrodes are shown for each angle. The eight defined interference sound angles are marked by a loudspeaker pictogram at the edge of the polar diagram. The strongest noise suppression can be shown in the ultra-zoom beamforming mode.

values of several groups differ significantly from each other. A high F-value indicates that the mean values of the groups differ significantly from each other. The differences between the electrodes were also not statistically significant, as the test statistic ($F = 2.9909$, $p < 0.0001$) was above the critical F value of 1.7112. The interaction regarding the influences in the interaction between the two microphone modes was also not significant, as shown by the test statistic ($F = 2.9849$, $p < 0.0001$) and the critical F value of 1.7112.

The differences remained statistically insignificant in the measurement setup without the artificial head and only became statistically significant when the artificial head was added (see Figure 7).

The analysis of variance shown in Table 2 revealed no statistically significant difference between the microphone modes (test statistic F = 3.9970, p = 0.0468) with a critical F value of 3.8833. However, significant differences were found in the electrode values (test statistic F = 4.6232, p < 0.0001, critical F value = 1.7112). The interaction of the two microphone modes also showed significant influences in the interaction (test statistic F = 4.9225, p < 0.0001, critical F value = 1.7112), indicating a statistically significant influence of the different conditions.

The following tables contain detailed comparisons between the measurements with and without the artificial head. Significant differences were found in several comparisons, particularly in relation to the presence of the artificial head and the use of the ultra-zoom beamforming mode. The analysis of variance shown in Table 3 and Table 4 revealed no significant difference between the recording modes omni-directional and ultra-zoom in the test setup without artificial head compared to data collection with artificial head. In contrast, the difference between the electrodes was statistically significant (test statistic F = 9.3515, p < 0.0001, critical F value = 1.7112), as were the influences in the interaction between the two measurement setups (test statistic F = 7.4856, p < 0.0001, critical F value = 1.7112). The significance tests underlined that the measured differences were not random but statistically significant.

Table 2. P1 artifical head (omni-directional) vs. P2 artifical head (ultra zoom).

Cause of Variation	Sum of Squares (SS)	Degrees of Freedom (df)	Mean Square (MS)	Test Statistic (F)	P-Value	Critical F-Value
Sample	58.35	1	58.35	3.9970	0.0468	3.8833
Columns (Electrodes)	1012.29	15	67.49	4.6232	<0.0001	1.7112
Interaction	1077.83	15	71.86	4.9225	<0.0001	1.7112
Error	3269.82	224	14.60			
Total	5418.28	255				

Table 3. P1 without head (omni-directional) vs. P1 artifical head (omni-directional).

Cause of Variation	Sum of Squares (SS)	Degrees of Freedom (df)	Mean Square (MS)	Test Statistic (F)	P-Value	Critical F-Value
Sample	75.66	1	75.66	3.0742	0.0809	3.8833
Columns (Electrodes)	3452.40	15	230.16	9.3515	<0.0001	1.7112
Interaction	2763.57	15	184.24	7.4856	<0.0001	1.7112
Error	5513.12	224	24.61			
Total	11804.74	255				

The analysis of variance revealed a significant difference between the measurements with and without artificial head (see Table 5) in beamforming ultra zoom (test statistic F = 16.2399, p < 0.0001, critical F value = 3.8833). In contrast, no significant differences were found between the electrode values (test statistic F = 1.1853, p = 0.2843, critical F value = 1.7112). The analyses also revealed no significant influence in the interaction between the two measurement setups (test statistic F = 1.2977, p = 0.2048, critical F value = 1.7112).

There was a statistically significant difference between the measurements in omni-directional mode with artificial head and the measurements without artificial head in beamforming ultra zoom (see Table 6). This was confirmed by the high test value (F = 9.0450, p = 0.0029), while the critical F value was 3.8833.

Table 4. P1 without head (omni-directional) vs. P2 artifical head (ultra zoom).

Cause of Variation	Sum of Squares (SS)	Degrees of Freedom (df)	Mean Square (MS)	Test Statistic (F)	P-Value	Critical F-Value
Sample	1.12	1	1.12	0.0498	0.8237	3.8833
Columns (Electrodes)	2448.41	15	163.23	7.2302	<0.0001	1.7112
Interaction	1790.20	15	119.35	5.2865	<0.0001	1.7112
Error	5057.01	224	22.58			
Total	9296.75	255				

Table 5. P2 artifical head (ultra zoom) vs. P2 without head (ultra zoom).

Cause of Variation	Sum of Squares (SS)	Degrees of Freedom (df)	Mean Square (MS)	Test Statistic (F)	P-Value	Critical F-Value
Sample	1000.65	1	1000.65	16.2399	<0.0001	3.8833
Columns (Electrodes)	1095.55	15	73.04	1.1853	0.2843	1.7112
Interaction	1199.42	15	79.96	1.2977	0.2048	1.7112
Error	13802.13	224	61.62			
Total	17097.75	255				

Table 6. P1 artifical head (omni-directional) vs. P2 without head (ultra zoom).

Cause of Variation	Sum of Squares (SS)	Degrees of Freedom (df)	Mean Square (MS)	Test Statistic (F)	P-Value	Critical F-Value
Sample	575.74	1	575.74	9.0450	0.0029	3.8833
Columns (Electrodes)	3595.58	15	239.71	3.7658	<0.0001	1.7112
Interaction	676.73	15	45.12	0.7088	0.77456	1.7112
Error	14258.24	224	63.65			
Total	19106.30	255				

A significant difference was found in the electrode values, with the test value ($F = 3.7658$, $p < 0.0001$) being above the critical F value of 1.7112. In contrast, no significant difference could be detected in the influences of the interaction of the two measurement setups, as shown by the test variable ($F = 0.7088$, $p = 0.774560901$) and the critical F value of 1.7112.

4. Discussion

This study investigated the influence of sound source position and electrode placement on the stimulation pattern of cochlear implants (CI) under noise conditions. The findings provide important information for optimizing auditory perception effects. In the discussion, expectations, possible causes and consequences of the results are highlighted. The findings are interpreted and placed in the current state of research. The limitations of the study are explained and possible starting points for future research are identified. When planning the study, certain expectations were formulated, which are considered in the context of the results. It was expected that the electrode placement and the position of the sound source would have a significant influence on the noise suppression of the CI. The results show that electrode placement has no significant effect on noise suppression, which has not previously been published in conjunction with a measurement box for CI. This suggests that CI stimulation provides consistent results regardless of the exact electrode position, possibly due to the robust performance of modern cochlear implants. The improvement in noise suppression in beamforming mode ultra zoom under more realistic conditions, especially with an artificial head, was an expected result. To date, there are no comparative data in the literature that allow a direct comparison of channel-individually recorded electrode currents of a cochlear implant. However, it can be stated that the stimulation patterns exhibit comparable noise suppression, which applies both to the simulation of realistic acoustic scenes and to the evaluation of signal processing algorithms for hearing aids as a function of the sound source position. Overall, the statistical analyses performed show the influence of the different measurement conditions and microphone modes on the stimulation pattern of the cochlear implant. The integration of head and ear shape information was considered a critical factor in the effectiveness of CI systems, and the results confirm this in line with work focusing on hearing aid measurement. Significant differences in electrode readings suggest that individual anatomical differences between CI users can lead to variations in response to noise. This highlights the importance of personalized fittings to further increase the effectiveness of CI systems. The results are interpreted against the background of the current state of research. The present study contributes to the existing literature by relativizing the importance of electrode placement for noise reduction and focusing on the integration of head and ear shape information. There are no publications that can be used to compare the results, which also found no significant differences in noise reduction depending on electrode placement. For this reason, it is assumed that CI users can achieve consistent results in different real-world environments regardless of the exact electrode position. The improvement in noise reduction in beamforming mode ultra zoom under more realistic conditions confirms the importance of head and ear shape information for the performance of CI systems. This is consistent with published work focusing on the measurement of hearing aids. Significant differences in electrode values expand the understanding of individual variations in noise suppression. This underscores the need to explore and implement personalized fittings to maximize the clinical effectiveness of cochlear implants. The limitations of this study are that it focused mainly on technical aspects and used an artificial head as a simplified model. This approach could limit the generalizability to real users. Further studies at the individual patient level would therefore be desirable to capture subjective differences in perception. In addition, the tests were conducted under controlled conditions that may not represent all aspects of the diverse everyday situations of CI users. The inclusion of real-life noise situations was a step towards realism, but a more comprehensive assessment of different environmental influences could provide further insights. Furthermore, this study focused on technical performance without directly considering the individual hearing abilities and needs of CI users. Extending the research to include subjective measurements and user evaluations could provide additional insights into the practical applicability of the results obtained. The present study offers starting points for future research at various levels. One possible extension could be to continue the investigations at patient level in order to identify subjective differences in perception and adaptation. This would be an important step towards a holistic view of CI effectiveness. The inclusion of target curves and percentiles in the adjustment of CI settings was identified as a promising approach to further optimize auditory perceptual effects. A more detailed investigation of this approach could enable the development of personalized fitting algorithms that are more closely aligned with the individual needs of CI users. Extending the investigations to include subjective measurements and user evaluations could provide additional insights into the practical applicability of the results obtained. Use of the AB-Vocoder at the CI Hackathon [16] [17] as a simulation could be used in future studies to determine improved patient performance. The evaluation tool (as of 26.01.2021; Commit: 0eb025a; https://github.com/jabeim/GMT) could not only deepen the results of this study in clinical practice, but also help to further optimize innovative coding strategies for cochlear implants. To facilitate the transition from laboratory conditions to real-world application scenarios, research could also investigate the integration of artificial intelligence (AI) into cochlear implant systems. AI-based algorithms could help to develop adaptive noise reduction systems that dynamically adjust to different environmental conditions, improving the effectiveness of CI systems in a variety of situations. It would also be interesting to focus future studies on how different noises encountered in everyday situations affect the perception and performance of cochlear implants. This could lead to more specific recommendations for fitting CI systems in different environments to ensure an optimal hearing experience for users. It should be noted that future research should pay more attention not only to technical aspects but also to the individual needs and preferences of CI users. Comprehensive research that integrates both objective technical parameters and subjective user evaluations could provide a holistic perspective on the performance of cochlear implants. Overall, the present study highlights the need for further research efforts to optimize the fitting of cochlear implants and ensure the best possible hearing experience for CI users. Collaboration between research institutions, technology developers and clinical users could help to develop innovative solutions and improve the quality of life of CI users in the long term. It is hoped that the knowledge gained in this study will contribute to the continuous improvement of cochlear implants and their fitting procedures and thus have a positive impact on the hearing experience of affected patients.

5. Conclusion

In this study, the influence of sound source position and electrode placement on the stimulation pattern of cochlear implants (CI) under noise conditions was investigated. Surprisingly, the results showed that the exact electrode placement has no significant influence on noise suppression. This is in contrast to previous literature, which often focused on anatomical differences and their impact on CI performance. The robust performance of modern cochlear implants seems to enable consistent results regardless of the exact electrode position. However, the study confirmed the expectation that noise suppression is improved in beamforming mode ultra zoom under realistic conditions, especially with an artificial head. This underlines the importance of integrating head and ear shape information for the efficiency of CI systems. These results are consistent with previous work that has emphasized the importance of considering the wearer's individual anatomy to achieve optimal results. The discussion of the results underlines the relevance of these findings in the context of current research. In particular, the importance of investigating anatomical differences in CI users in relation to noise reduction is emphasized. The lack of significant differences depending on electrode placement suggests that CI users can achieve consistent results in different real-world environments despite individual anatomical variations. Despite the promising results, the study has some limitations, including the focus on technical aspects and the use of a simplified model with an artificial head. Future research could provide greater insight by capturing subjective differences at the individual patient level and considering more complex environmental conditions. The results suggest that a holistic perspective is needed to fully understand the effectiveness of cochlear implants. This includes the integration of technical parameters with subjective user evaluations. Collaboration between research institutions, technology developers and clinical users is seen as crucial to develop innovative solutions and sustainably improve the quality of life of CI users. Overall, this study provides valuable insights for the continuous development of cochlear implants and their fitting procedures. In summary, the results of this study offer promising perspectives for the implementation of speech reconstruction. The results stimulate future projects involving investigations on patient collectives to evaluate possible improvements in cochlear implant fitting. The application of target curves and percentiles as well as the integration of the AB vocoder of the CI Hackathon show promising approaches to realize a more precise and individualized CI fitting. These advances could make a significant contribution to increasing the effectiveness of CI systems and improving the quality of life of affected patients in the long term. It is recommended that these innovative approaches be explored in greater depth in further research and that their practical applicability in a clinical context be tested.

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Authors' Contributions

The study was conducted independently by the author as part of his master's thesis. The data were collected in the rooms of the Hearing Research Group at the Department of Otorhinolaryngology, Head and Neck Surgery, University Hospital Mannheim, Germany.

Conflicts of Interest

The authors declare no conflicts of interest regarding the publication of this paper.

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