Combined Effect of Noise Reduction, FBS-Based Spectral Splitting, and Dynamic Range Compression of Speech Signal on Source Localization in Binaural Hearing Aids

Jyoti M. Katagi, Pandurangarao N. Kulkarni

***Abstract:** Localizing sound sources in three spatial dimensions (azimuth, elevation, and distance) is critical for human hearing comfort. It relies on two binaural cues: interaural time difference (ITD) and interaural level difference (ILD). Cochlear or auditory nerve injury can result in sensorineural hearing loss (SNHL). Hearing aids enable people with sensorineural hearing loss to converse more effectively and hear better. However, there is an apprehension that the binaural hearing aids may degrade the localization cues, thus affecting the source localization. In response to this concern, the current study investigates how the binaural hearing aid algorithm affects source localization by adopting a cascaded structure of noise reduction technique (wiener filter) followed by filter bank summation (FBS) based spectral splitting and dynamic range compression for binaural dichotic presentation. Listening tests for seven different azimuth angles (-90º, -60º, -30º, 0º, 30º, 60º, and - 90º) were conducted on six listeners with normal hearing under different signal-to-noise ratio (SNR) conditions as well as on six subjects with mild bilateral sensorineural hearing impairment. Test stimuli included background glass-breaking sound and broadband noise for participants with normal hearing. In an experiment with hearing-impaired subjects, the glass-breaking sound served as one of the test stimuli. The result showed that these binaural hearing aid algorithms had no adverse effects on localization ability.*

***Keywords:** Binaural Hearing Aids, Interaural level difference, Interaural time Difference, Sensorineural Hearing Loss, Sound localization, Speech perception.*

I. INTRODUCTION

The sense of hearing is beneficial for directing the sense of vision and, consequently, bodily posture toward a direction that would merit paying greater attention. Even when a person is sleeping, or visual information is not accessible, the sense of hearing is always on, enabling it to construct a fundamental mental model of the physical world in connection to the human body.

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Binaural cues (interaural differences), or differences in the time of arrival or intensity of the sounds at the right and left ears, or monaural spectral cues (as an illustration, consider the frequency-dependent pattern of sound filtering brought on by the sound's angle of incidence with the external ear.), are the two main methods used to locate sounds [1]. Interaural differences serve primarily for left-right localization, whereas spectral cues are helpful for vertical and front-back localization. Headphone tests can determine the smallest interaural time or intensity difference a subject can detect reliably and evaluate lateralization ability. The study [2] proposes a method for binaural source localization based on ITDs and ILDs. In binaural recording, the two cues are obtained via a two-channel time-frequency representation and are combined to determine the azimuth of sources. Furthermore, this study recommended using an average parameter model and a parametric head model for heads whose HRTFs are unknown. Lastly, a comparison between the results of this methodology and a hierarchical binaural sound source localization system based on Bayes rules [3] demonstrates that the former yields more reliable and consistent results. Papers [4] suggest a combined configuration for an adaptive wiener filter and frequency division of speech signal setup for binaural dichotic presentation. The proposed approach lowers the masking and background noise effects. As a result, it increases speech intelligibility for deaf people. The Modified Rhyme Test (MRT) is performed on subjects with moderate SNHL to measure the intelligibility of processed speech. There are 300 sentences in the input speech. Every sentence includes a CVC word. Speech recognition scores improved by 0.639, 30.074, 30.401, 31.563, 32.28, and 32.935 at SNR values of ∞dB, 6dB, 3dB, 0dB, -3dB, and -6dB, respectively, as compared to unprocessed speech. Additionally, there was a significant 32.935% improvement in speech recognition accuracy at lower SNR values of -6dB. Hearing aids typically use a combination of Noise Reduction (NR) and Dynamic Range Compression (DRC). Paper [5] presents an integrated solution for NR and DRC. Each time, the segment's speech and noise levels are estimated to determine the solution. The NR is less active if the speech is dominant, and having as much DRC as possible is desirable. In contrast, in a noise-dominant segment, the NR is more active, and the idea is not to compromise this operation by applying DRC.

Experimental results confirmed that a series combination of NR and DRC degrades the SNR improvement and that the proposed solution offers a better SNR improvement than a serial concatenation. The article [6] presents a mobile sound localization setup to gauge an individual's localization proficiency. This configuration's horizontal (-70º to 70º azimuth) and vertical (-35º to 40º elevation) planes include the presentation of sounds inside a partial sphere. A head-mounted LED asks participants to point at the source of the audio they experience. Head movements are seen instantly and recorded. Depending on the study objective, the configuration may be modified for more complex or straightforward measurements, making it appropriate for various research problems. Subjects should naturally gesture toward the perceived sound while the loudspeakers remain hidden in this setup. Sound localization, an essential function for safety and communication, will be affected by multiple factors, including sound sources, listening environments, individual listening ability, and hearing devices if applied, as discussed in the review article [7] [18]. Compared to listeners with Normal Hearing (NH), listeners with Impaired Hearing (IH) and hearing devices have poorer localization performance. They will be affected more by adverse listening conditions due to reduced accessibility to localization cues caused by limitations of the auditory system and hearing devices. Hearing aids have consistently failed to improve localization performance and, in some cases, significantly impair sound localization. In general, bone conduction hearing aids do not boost sound localization capabilities, while binaural users do report some improvement. Although cochlear implants provide great hearing benefits to individuals with severe-to-profound sensorineural hearing loss, cochlear implant users have significant difficulty localizing sounds, even with two implants. Paper [8] aimed to compare speech perception in noise and horizontal localization with and without activating digital noise reduction (DNR) in hearing aids with and without ear-to-ear synchronization. Twenty-five listeners with mild-to-moderate bilateral sensorineural hearing loss, aged between 18 and 55, were the participants. Each participant's horizontal sound-source localization performance was measured using the root-mean-squared error. The SNR measured speech recognition in the presence of speech babble noise required for a 50% recognition score (SNR-50). Further, SNR-50 was measured with a noise source from four different directions and recorded in four aided conditions, with and without independent activation of the wireless link and DNR. Furthermore, SNR-50 was measured with a noise source from four distinct orientations and recorded under four assisted settings, with and without independent activation of the wireless connection and DNR. The activation of DNR and wireless synchronization in hearing aids showed a better performance in horizontal sound-source localization. The paper [9] aims to develop signal-processing algorithms for dynamic range compression and background noise reduction to improve hearing aid performance for listeners with sensorineural loss. Sliding-band compression (SLBC) is a technique developed to mitigate the shortcomings of single-band and multiband compressions. Quantile-based methods for noise estimation have improved single-input speech. These approaches include adaptive dynamic quantile tracking-based noise estimation (ADQTNE) and dynamic quantile tracking-based noise estimation (DQTNE). When considering the rise in PESQ scores for the various noises, ADQTNE and DQTNE provide an SNR advantage of 4-11 dB and 3-10 dB, respectively. Article [10] used both objective (PESQ) and subjective (MOS and the Modified Rhyme Test (MRT)) listening tests to assess the efficacy of the cascaded wiener filter design, spectrum splitting (using the FBS technique), and amplitude compression. According to the findings, speech intelligibility increased under all SNR conditions, with the highest gains in speech recognition score (27.29%) and response time reduction (1.581s) occurring at lower SNR levels (-6dB) as compared to unprocessed speech. The study [11] aimed to determine if binaural dichotic presentation using comb filters with complementary magnitude responses based on fixed bandwidth and auditory critical bandwidth could improve speech perception in subjects with moderate bilateral sensorineural hearing loss. It also sought to determine whether this could affect the ability to localize the source of sound. Six participants with normal hearing had simulated hearing loss, while eleven subjects with modest bilateral sensorineural loss underwent quiet listening assessments to measure consonant recognition and source direction identification. According to the experiments conducted on people with normal hearing, comb filters based on the auditory critical bandwidth produced more excellent identification scores and shorter response times. These comb filters significantly reduced response times and increased recognition scores by 14% to 31% (mean: 22%) in tests conducted on deaf participants. However, there was no noticeable difference in the ability to identify the direction of broadband sound sources. On the other hand, there was no appreciable variation in the capacity to locate broadband sound sources. The method described in Paper [12] uses a pair of binaural hearing aids to determine the direction of arrival (DOA) of sound sources using blind channel identification (BCI). It contrasts the adaptive principal component algorithm (APCA) with the adaptive Eigenvalue decomposition algorithm (AEDA) for calculating the impulse responses from the target to hearing aids, which are needed to calculate DOA. Both approaches undergo evaluation for efficacy across a range of reverberation durations, SNRs, and source placements. The findings indicate that noise and reverberation harm AEDA's DOA performance. Despite its relative immunity to noise, APCA may still handle significant reverberation. Paper [13] evaluates the particular and mixed effects of noise and reverberation on listeners' capacity to localize speech that has bilateral cochlear implants (BCIs) and normal hearing. Ten participants with NH and six individuals with BCIs took part. In simulated anechoic and reverberant settings (0.2, 0.6, and 0.9s RT60), all individuals underwent a virtual localization test in silence and at SNRs of 0, -4, and -8dB. BCI users also underwent testing at +8 and +4dB SNR. At nine simulated locations in the frontal horizontal plane (±90º), a three-word statement was uttered at 70 dB SPL with a noise source at 0º.

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Compared to listeners with NH, listeners with BCI experienced worse localization due to noise and reverberation at higher SNR (+4dB) and shorter RT60 (0.2 s) values.

II. PROPOSED WORK

Describing localization in three dimensions requires a coordinate system, as Figure 1 illustrates. The term "azimuth" (θ) describes the orientation of the sound source around the head and also refers to the horizontal plane that establishes the direction from left to right. Elevation (ϕ) refers to the vertical surface that indicates the direction of upward and downward motion. The x and z-axes in Fig.1 represent the horizontal surface, while the y and z-axes represent the vertical surface. Although sound enters both ears, the brain distinguishes between information obtained from binaural signals and monaural signals. The signals that reach the left and right ears provide information about the interaural time difference (ITD) and interaural level (intensity) difference (ILD, IID). As seen in Fig. 2, this suggests that the incident sound wave reaches our ears at different times and intensities. One ear receives the sound before the other as long as the source is not directly in front of the head. We obtain this information using both ear signals. The term "binaural cues" is used to describe them. Since ILD and ITD are the same for the left and right ears, sound sources that come directly in front of or behind a person do not provide them. Monaural signals are significant in this situation. Monaural cues, derived from the signal of one ear, indicate the spectral structure of the incident sound wave by the head, shoulders, torso, and, most crucially, the pinna.

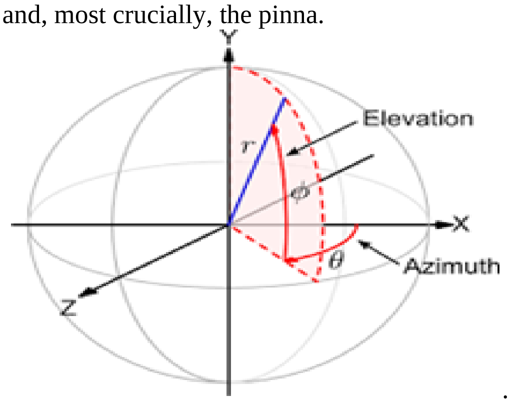

Fig. 1: Coordinate System Relative to the Head with Azimuth Θ & Elevation Φ

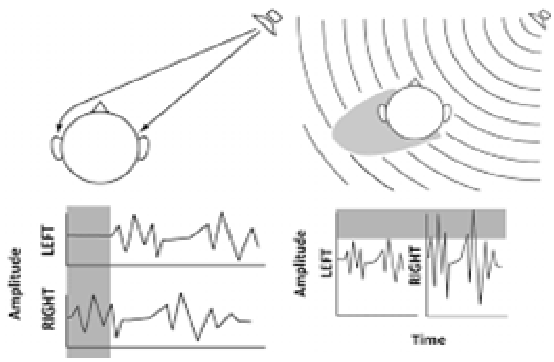

Fig. 2: Left & Right Discrimination with Time of Arrival & Intensity of Sound

Elevation, azimuth, and time all influence the head-related impulse response. HRIR is sampled in the data files both temporally and spatially. The discrete indices naz, nel, and nt specify azimuth, elevation, and time. A 3D array having dimensions of (25*50*200) is an HRIR h(naz, nel, nt). Table 1 shows the range of azimuth and elevation angle values for various places in the interaural polar coordinate system.

Table- I: Azimuth and Elevation Directions in 3D Space [14]

Azimuth	Elevation	Direction in 3D space
0^0	0^0	Ahead
0^0	90^0	Overhead
0^0	180^0	Behind
0^0	270^0	Below
90^0	0^0	To the right
-90^0	0^0	To the left

III. TESTS AND RESULTS

By using a cascaded structure of noise reduction approach, spectral splitting, and dynamic range compression, the current study aims to evaluate how effectively a binaural hearing aid algorithm enhances source localization. Two HRTFs were utilized in the experiment to generate spatial sounds. The public domain CIPIC HRTF database provides HRTFs for various azimuth and elevation configurations and explains the method used to quantify HRTFs and anthropometric features [15][16][17]. The HRTFs from this database for Subject 3, a KEMAR manikin participant, were used in the current experiment. They were for an elevation angle of 0º and the frontal azimuth angles ranging from -90º (left) to +90º (right). Six individuals with normal hearing in the face of broadband masking noise and six with mild sensorineural loss underwent hearing tests to study source localization ability. Individuals with normal hearing were exposed to broadband random noise as a mask when processing stimuli. The test was administered briefly (10 ms) at six SNR values: ∞dB, 6, 3, 0, -3, and -6dB. We did not use broad masking noise while evaluating subjects with hearing impairment. Participants may choose their comfort level with the binaurally transmitted sounds during each test. The present work compares direction identification outcomes in processed and unprocessed conditions. In the case of normal-hearing people, unprocessed speech is the input speech with different SNR, and processed speech is the output from the cascaded structure of the wiener filter followed by spectral splitting and dynamic range compression. In the case of deaf people, unprocessed speech is the clean input speech and processed speech is the output from the cascaded structure of the wiener filter followed by spectral splitting and dynamic range compression. Six participants exposed to broadband masking noise underwent hearing tests. Six SNR values were utilized to induce the noise: ∞ (no noise), 6, 3, 0, -3, and -6dB. The second round of testing, which didn't use masking noise, involved six participants with mild bilateral sensorineural loss. HRTFs were used in both trials to recreate surrounding sounds at 0° elevation and various azimuth angles (0°, ±30°, ±60°, and ±90°). The participants got a chart detailing these directions, as shown in Fig. 3. The stimuli processed for each angle are presented in a random order five times.

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The participant identified one of these seven angles as the source's direction. The responses are used as columns in a stimulus-response matrix to calculate the mean of each perceived angle. Test stimuli include background glass-breaking sound and broadband noise for subjects with normal hearing. The glass-breaking sound was the only test stimulus employed in the masking noise experiment. Therefore, each subject received 210 presentations with masking noise (7 angles x 5 repetitions x 6 SNR values) and 35 presentations without noise (7 angles x 5 repetitions). The stimulus-response matrix for the glass-breaking sound is depicted in Table 2, with responses from all six regular hearing participants combined. In an experiment with hearing-impaired subjects, the glass-breaking sound served as one of the test stimuli. Each person received 35 presentations (7 angles x 5 repetitions). Table 3 displays the stimulus-response matrix for the glass-breaking sound with the responses from all six deaf participants combined. With cascaded noise reduction strategy, spectral splitting, and dynamic range compression at various SNR levels and a compression ratio of 0.6, the average source direction identification by six participants with normal hearing is shown in Table 2 for unprocessed and processed speech. In comparison to unprocessed speech at SNR values of ∞ dB, +6 dB, +3 dB, 0 dB, -3 dB, and -6 dB, respectively, the mean of processed speech values improved to 1, 1.42, 1.58, 2, 2.15, and 3.43 in the sound source direction identification. Furthermore, advances in locating sound sources are considerable at lower SNR values. The results show that the subjects could determine the source direction using ITD and ILD signals from various bands. For six deaf participants, Table 4 compares their percentages of angle recognition scores under unprocessed and processed situations using the stimulus of glass-breaking sound. For six participants with hearing impairment, the unprocessed average stimulus-response matrix rates are 33.3%, 53.3%, 50%, 90%, 23.3%, 60%, and 66.7% at azimuth angles of -90°, -60°, -30°, 0°, 30°, 60°, and 90°, respectively. We can find that the localization performance is only up to 53.8% from the unprocessed stimulus-response matrix produced by performing listening tests. For azimuth angles of -90°, -60°, -30°, 0°, 30°, 60°, and 90°, the percentage of the average stimulus-response matrix that was processed is 43.3%, 60%, 23.3%, 96.7%, 33.3%, 60%, and 63.3%, respectively, among six deaf persons. According to the processed stimulus-response matrix acquired from the listening tests, the localization performance was up to 54.27%. The subjective assessment for seven separate azimuth angles on six listeners with normal hearing under different signal-to-noise ratio circumstances and six listeners with hearing impairment found no detrimental effects on source localization [19].

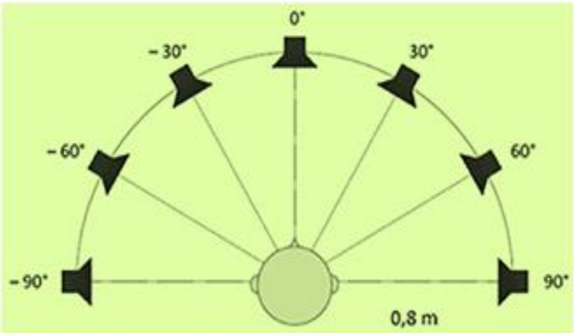

Fig. 3: Azimuth Perception Test Reference Chart

A. Graphical Analysis

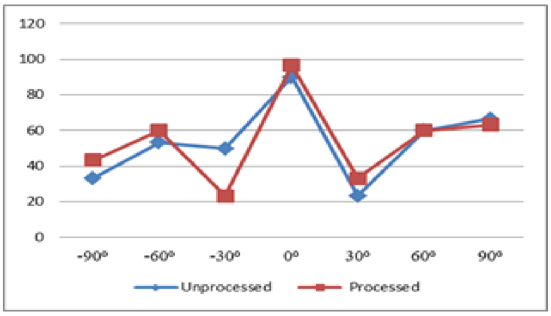

Fig. 4: Angle Determination Score (%) in Unprocessed & Processed Scenarios for 6 Hearing Impaired Participants

Fig.4 shows the angle determination score (%) for unprocessed speech and speech signals processed with the proposed scheme, employing a Wiener filter as a noise reduction technique. Six individuals with hearing impairments were in the trial. From this figure, we observed that subjective assessment for seven separate azimuth angles on six deaf listeners found no detrimental effects on source localization.

B. Spectrographic Analysis

Figs. 5 and 6 below show the deaf people's left ear, right ear, and wideband spectrum of unprocessed and processed glass breaking speech signal for a -90 degree angle. Below, Fig. 7 and 8 show the regular hearing of people's left ear, right ear, and wideband spectrum of unprocessed and processed glass breaking speech signal for -6 dB at -90 degree angle. Speech compression has no impact on the harmonic structure, according to processed speech spectrograms, which also demonstrate a significant reduction in background noise.

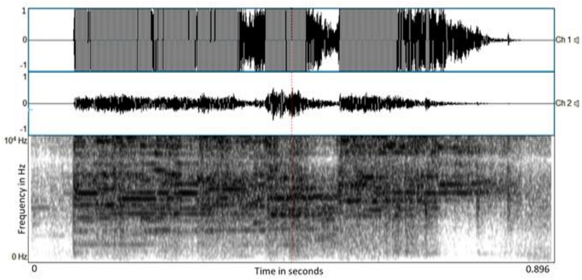

Fig. 5: Hearing Impaired People's Left Ear, Right Ear, and Wideband Spectrogram of Unprocessed Glass Breaking Speech Signal for -90 Degree Angle

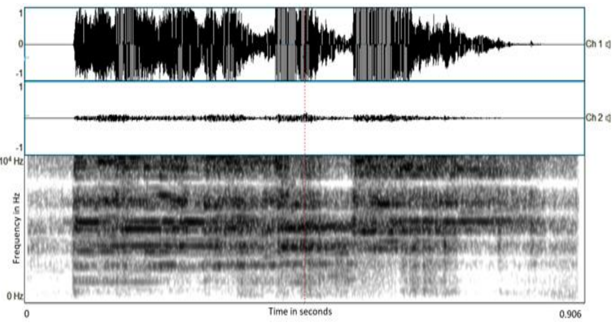

Fig. 6: Hearing Impaired People's Left Ear, Right Ear, and Wideband Spectrogram of Processed Glass Breaking Speech Signal for -90 Degree Angle

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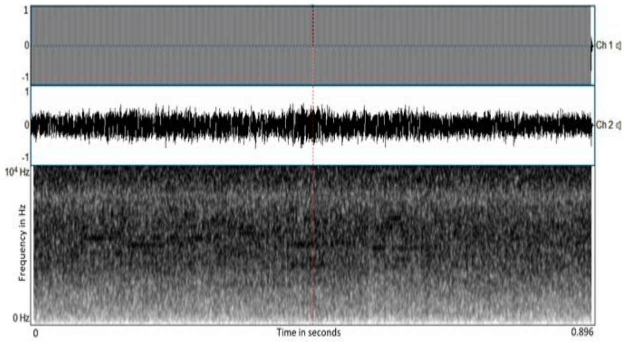

Fig. 7: Normal Hearing People's Left Ear, Right Ear, and Wideband Spectrogram of Unprocessed Glass Breaking Speech Signal for -6 Db At -90 Degree Angle

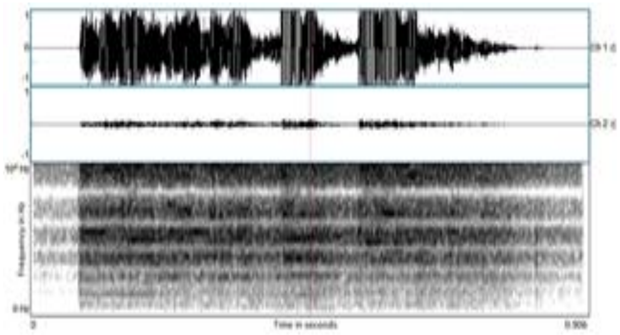

Fig. 8: Normal Hearing People's Left Ear, Right Ear, and Wideband Spectrogram of Processed Glass Breaking Speech Signal for -6 Db At -90 Degree Angle

Table-II: Average Source Localization Values for Processed and Unprocessed Speech for the Six Normal-Hearing Individuals at Various SNR Levels with the Sound of Breaking Glass

Angle (deg)	SNR (dB)											
	∞		6		3		0		-3		-6	
	Un processed	Processed	Un processed	Processed	Un processed	Processed	Un processed	Processed	Un processed	Processed	Un processed	Processed
-90°	17	16	19	20	18	20	15	18	18	22	19	24
-60°	17	17	16	18	16	16	16	17	15	17	19	21
-30°	16	18	18	19	17	19	17	20	18	20	15	19
0°	27	28	28	29	29	30	29	30	30	30	30	30
30°	13	15	19	20	20	21	19	20	17	19	19	22
60°	17	18	14	18	17	20	17	19	20	22	15	19
90°	19	21	21	21	21	23	21	24	20	23	16	22
Mean	18	19	19.29	20.71	19.71	21.29	19.14	21.14	19.71	21.86	19	22.43
Improvement	1		1.42		1.58		2		2.15		3.43	

Table- III: Source Localization Scores for Presentation Angle Versus Perceived Angle in Deaf Individuals. Each Angle has 30 Presentations (5 Presentations X 6 Subjects). Glass-Breaking Sound is the Test Material

Presented Azimuth angle(deg.)	Unprocessed Speech							Processed Speech						
	Perceived angle (deg.)							Perceived angle (deg.)						
	-90°	-60°	-30°	0°	30°	60°	90°	-90°	-60°	-30°	0°	30°	60°	90°
-90°	10	12	8					13	14	3				
-60°	5	16	9					5	18	7				
-30°	6	9	15					7	15	7	1			
0°			2	27	1						29	1		
30°					7	15	8				6	10	6	8
60°					3	18	9					4	18	8
90°						10	20					1	10	19

Table- IV: Angle Determination Score (%) in Unprocessed and Processed Scenarios for Six Deaf Participants

Presented Azimuth angle (deg.)	Unprocessed Speech							Processed Speech						
	Perceived angle (deg.)							Perceived angle (deg.)						
	-90°	-60°	-30°	0°	30°	60°	90°	-90°	-60°	-30°	0°	30°	60°	90°
-90°	33.3 %	40%	26.7%					43.3%	46.7%	10%				
-60°	16.7 %	53.3%	30%					16.7%	60%	23.3%				
-30°	20%	30%	50%					23.3%	50%	23.3%	3.33%			
0°			6.67%	90%	3.33%						96.7%	3.33%		
30°					23.3%	50%	26.7%				20%	33.3%	20%	26.7%
60°					10%	60%	30%					13.3%	60%	26.7%
90°						33.3%	66.7%					3.33%	33.3%	63.3%

IV. CONCLUSION

This paper examines the cumulative impact of a filter bank summation approach for performing spectral splitting of input signals for binaural dichotic presentation and dynamic range compression paired with a noise reduction strategy based on the Wiener filter on sound source localization. We can find that the localization performance is only up to 53.8% from the unprocessed stimulus-response matrix produced by performing listening tests. According to the processed stimulus-response matrix from the listening tests, the localization performance was up to 54.27%.

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At lower SNR values, sound source localization improves significantly. The results show that the subjects can recognize the source direction using ITD and ILD signals. According to the subjective evaluation of source localization for seven distinct azimuth angles on six listeners with normal hearing under various signal-to-noise ratio conditions and six listeners with hearing impairment, there were no negative impacts on source localization.

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DECLARATION STATEMENT

After aggregating input from all authors, I must verify the accuracy of the following information as the article's author.

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- **Ethical Approval and Consent to Participate:** The data provided in this article is exempt from the requirement for ethical approval or participant consent.
- **Data Access Statement and Material Availability:** The adequate resources of this article are publicly accessible.
- **Authors Contributions:** The authorship of this article is contributed equally to all participating individuals.

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