PERCEPTUAL EFFECT OF REVERBERATION ON MULTI-MICROPHONE NOISE REDUCTION FOR COCHLEAR IMPLANTS

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ABSTRACT

The combination of noise and reverberation make listening conditions difficult for cochlear implant (CI) users. The perceptual effect of reverberation was evaluated via speech intelligibility tests with CI users. A fixed directional microphone, an adaptive directional microphone and a beamformer post-filter were evaluated. Reverberation was varied by changing the target and noise distance and by simulating a highly reverberant room with concrete surfaces. CI performance expectedly degraded as the target distance was increased, but the benefit of noise reduction was unaffected by listening distance. In the highly reverberant condition, CI performance was severely degraded, but noise reduction benefit remarkably increased, especially for the beamformer post-filter algorithm. All directional processing algorithms were suitable for use in noisy reverberant conditions and the best outcome was provided by the post-filter condition.

Index Terms— noise reduction, speech enhancement, beamformer, cochlear implant, reverberation, post-filter

1. INTRODUCTION

While cochlear implant (CI) users can achieve good performance in quiet conditions, speech understanding becomes compromised in realistic environments that contain background noise and reverberation [1, 2]. Reverberation directly affects CI performance [3, 4] but also compromises the ability of noise reduction algorithms to enhance the signal, further compounding the effects of noise. Algorithms need to be robust to the effects reverberation encountered in common listening situations in order to provide practical noise reduction benefit for CI users.

Both single microphone and multi-microphone noise reduction techniques have been evaluated in CI listeners to improve performance. Single microphone noise reduction based on assumptions of statistical signal distributions can provide speech intelligibility benefit when background noise is unmodulated, but effectiveness is limited when the noise is competing speech [e.g. 5]. Alternatively, multi-microphone solutions can exploit the properties of a microphone array to spatially filter the signal. Typical approaches use directional microphones (beamformers) to form either fixed or adaptive spatial patterns, a common form is the generalised side-lobe canceller (GSC, [6]). They aim to preserve the target signal in front of the listener and attenuate noise from behind. Large benefits have been demonstrated using GSC-style algorithms for CI users when there is a single noise source in low reverberation [7-9]. However, performance is compromised at high levels of reverberation [10], which has been demonstrated in the application to CIs [7, 8, 11] and hearing aids [12].

To improve performance, a beamformer post-filter can be used to reduce direct and reverberation noise depending on the acoustic properties of the test environment. The design of such filters has been approached from a theoretical perspective under assumptions of known noise field coherence (e.g. [13, 14]), however, prior knowledge of the room acoustics is required, limiting practical application of these post-filters to CIs.

As an alternative to relying on assumptions of statistical signal distributions or coherence, Cao, et al. [15] used a post-filter based purely on spatial filtering. The principle of operation was based on a dual-beamformer stage that had a main beamformer aimed at the desired speech source and a reference beamformer designed to pick up all signals except the desired speech. The dual-beamformer outputs were used as speech and noise estimates in a modified Wiener post-filter. A similar approach was adapted for use in a CI system and was evaluated in low (T60=70 ms) and moderate levels of reverberation (T60=520 ms) [16, 17]. In a situation with 4 competing talkers in the rear hemi-field, the algorithm provided performance benefit in both levels of reverberation of more than 4 dB speech reception threshold (SRT) over the baseline GSC-style beamformer called Beam [18].

Yousefian and Loizou [19] proposed a dual-microphone coherence-based spatial filter for CI and demonstrated 5-10 dB SRT improvement over a fixed directional microphone in an anechoic room with one or two competing talkers. The principle of operation was based on spatial coherence functions for differential microphones [20]. The performance of coherence-based approaches are known to depend to a large extent on the acoustics of the environment such as the room reverberation, the orientation of the microphone array and the spatial distribution of sound sources, as well and the directivity of the microphones used to measure coherence [21]. This dependence may account for the degraded performance in reverberation demonstrated by Yousefian and Loizou [19] in normal hearing listeners. The benefit was substantially reduced to 0-2 dB SRT when evaluated in a moderately reverberant room (T60=465 ms) suggesting the coherence assumption was not as strong under this condition.

A spatial filtering approach based on the direct analysis of dual-microphone phase difference was proposed by Goldsworthy, et al. [22] and evaluated in CI listeners. The algorithm used phase difference to estimate direction of arrival of sounds and spatially filter the signal. The algorithm was evaluated in a room with reverberation (T60=350 ms) and competing time-reversed speech from three fixed locations (90, 180 and 270 degrees). The reported benefit over an omni-directional microphone was 5.8 to 10.7 dB SRT and over a fixed directional pattern was 2.2 to 7.0 dB SRT.

Given the detrimental effects of reverberation on the benefit that noise reduction algorithms can provide, this study was designed to explore the performance of CI listeners across a wide range of reverberant conditions. In particular, a spatial post-filter previously evaluated by Hersbach, et al. [17] was compared against fixed and adaptive beamformer baseline conditions. This was achieved by separately varying the listening distance of the target and noise signals, and using a room simulator to create an extreme reverberation condition. The reverberant room simulator was validated as part of the experiment.

2. METHODS

2.1. Processing conditions

Algorithms from the CP900 sound processor from Cochlear Ltd [23] were used as baseline conditions. They were Omni, an omni-directional microphone, Zoom, a fixed directional super-cardioid pattern with spatial null at 120° [23], and Beam, a GSC-style adaptive beamformer [18]. Additionally, a beamformer post-filter called SpatialNR, previously described by Hersbach, et al. [16] was evaluated with a gain threshold $\alpha = 3\,dB$ and smoothing time constants, attack $\beta_A = 5\,ms$ and release $\beta_R = 50\,ms$, chosen as the parameters that provided consistent benefit over a range of environments [17]. SpatialNR used the speech reference, $S_k[n]$, as the Zoom signal and noise reference, $N_k[n]$, from the Beam adaptive filter stage, where k is the frequency index and n is the time index of overlapping FFT windows. The signals were first smoothed in the dB domain (Eqn. 1, 2) before estimating the SNR, ${\xi^{dB}}_k[n]$ in Eqn. 3, and calculating the filter gains, $H_k[n]$ in Eqn. 4, that were finally applied to the Zoom signal in Eqn. 5.

$$\overline{S^{dB}}_k[n] = \beta_S {S^{dB}}_k[n] + (1-\beta_S)\overline{S^{dB}}_k[n-1], \quad \beta_S = \begin{cases} \beta_A, \overline{S^{dB}}_k[n] > \overline{S^{dB}}_k[n-1] \\ \beta_R, \text{otherwise} \end{cases} \tag{1}$$

$$\overline{N^{dB}}_k[n] = \beta_N {N^{dB}}_k[n] + (1-\beta_N)\overline{N^{dB}}_k[n-1], \quad \beta_N = \begin{cases} \beta_A, \overline{N^{dB}}_k[n] > \overline{N^{dB}}_k[n-1] \\ \beta_R, \text{otherwise} \end{cases} \tag{2}$$

$${\xi^{dB}}_k[n] = \overline{S^{dB}}_k[n] - \overline{N^{dB}}_k[n]. \tag{3}$$

$$H_k[n] = \frac{\xi_k[n]}{\alpha + \xi_k[n]}. \tag{4}$$

$$\hat{X}_k[n] = H_k[n] S_k[n]. \tag{5}$$

2.2. Reverberant room

The room used for testing was known as the "training room" located at Australian Hearing head office in Sydney, Australia, and was used in a previous evaluation of SpatialNR [17]. The dimensions of the room were 11.8m x 8.6m x 3.6m (LxWxH). Two adjoining walls were painted concrete brick surfaces with two wooden doors. Another wall was plastered over while the remaining wall was almost entirely glass. The floor was carpeted and the ceiling was treated with suspended absorbing panels. A loudspeaker circle was used to record impulse responses from 12 source angles (30 degree separation) and 2 source distances (1m, 3m).

2.3. Reverberation simulator

The reverberation simulator used was MCRoomSim [24]. It was a shoebox simulator that modelled the reflection and scattering of sound waves from six surfaces (floor, ceiling, 4 walls). The simulator was based on the work of Schimmel, et al. [25] with an important extension that allowed modelling of not only single microphone elements but microphone arrays, including accurate inter-sensor time delays [24]. The reverberation simulator was used to produce a set of impulse responses that modelled the real "training" room by setting the coefficients of absorption and scattering based on published material properties of the room's surfaces. In addition, a highly reverberant condition "Sim(Concrete)" was simulated by changing the properties of the surfaces in the model to simulate concrete walls, ceiling and floors.

Various reverberation properties were extracted from the impulse responses and calculated at both source distances (1m, 3m) that were simulated, with median values over all source angles provided in Table 1.

Table 1: Acoustic properties of the training room, T30-reverberation time, EDT-early decay time, C50-clarity 50ms, C7-clarity 7ms, DRR-direct-to-reverberant ratio.

Distance (m)	Impulse response	T30 (ms)	EDT (ms)	C50 (dB)	C7 (dB)	DRR (dB)
1	Recorded	460	20	17.2	12.0	11.1
	Simulated	622	15	18.4	14.9	14.7
	Sim(Concrete)	809	432	11.6	9.8	9.8
3	Recorded	581	412	9.2	3.3	-0.3
	Simulated	742	589	10.3	6.4	3.8
	Sim(Concrete)	887	1007	4.1	0.9	-0.4

2.4. Acoustic conditions

The reverberant properties of the speech and noise components were altered by changing the distance from the listening position and by changing the impulse response used to generate the stimuli (Table 1). Three configurations were created using the recorded impulse responses with speech/noise distances of 1m/1m, 1m/3m, and 3m/3m named LowR, MidR and HighR, indicating the level of reverberant energy in the signal. Impulse responses were generated using the simulator modelling the HighR condition (3m/3m), called HighRSim. Finally, a condition with very high reverberation was created by modelling the acoustic surfaces as concrete and simulating the 3m/3m distance, called ExtraHighRSim.

2.5. Test protocol

Speech intelligibility in 8 adult CI users was evaluated by obtaining SRTs of sentences in noise according to the AuSTIN [26]. Through an adaptive procedure the test finds the SNR required to understand 50% of the sentence material, and hence lower SRTs indicate better performance. The noise comprised four competing talkers that were spatially separated with one talker located in each quadrant of the circle. For each sentence that was presented during the test, the location of each talker was randomised amongst pairs of loudspeaker locations at 45°±15° within each quadrant. There were 20 test conditions in total (four sound processing conditions and five reverberation configurations.) One SRT was collected per condition for each subject. The order of testing was randomised for each subject, and spread across two visits to the laboratory. Statistical analysis was performed using repeated measures analysis of variance (RM-ANOVA) [27] with Student-Newman-Keuls post-hoc comparisons [28].

3. RESULTS

3.1. LowR/MidR/HighR

The comparison between LowR, MidR and HighR was made to analyse the effect of the physical distance between the target/noise sources and the listener. The mean SRTs for the LowR, MidR and HighR conditions are shown in Figure 1 for each processing condition.

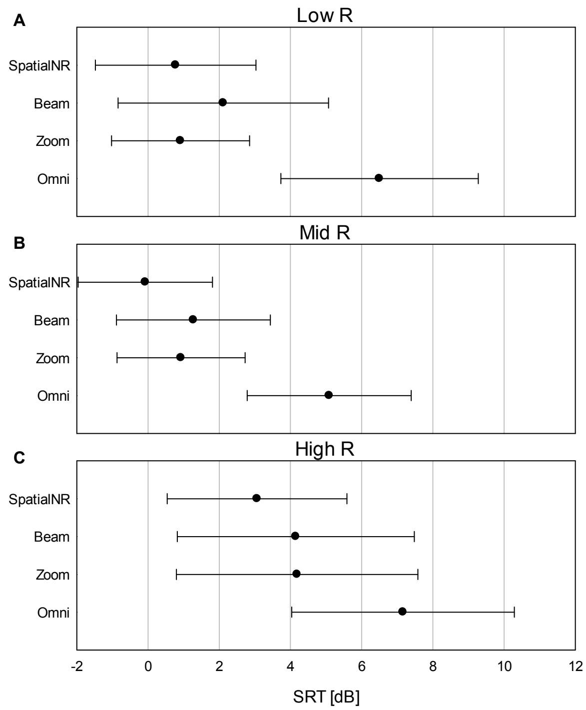

Figure 1: SRT results for each processing condition in (A) LowR, (B) MidR and (C) HighR reverberation conditions. Error bars show the 95% C.I.

The two-way RM-ANOVA with reverberation level and processing as factors revealed a significant main effect of reverberation level (F[2,14]=23.23, P<0.001) and a significant main effect of processing condition (F[3,21]=80.80, P<0.001).

Post-hoc comparisons were performed on the reverberation factor by averaging data across processing conditions. The 0.8 dB difference between the LowR and MidR conditions was not significant (P= 0.095). However, the HighR condition (with target speech at 3 m) resulted in significantly worse performance of 2.1 dB (P<0.001) compared to LowR and 2.8 dB (P<0.001) compared to MidR.

There was no significant interaction between the main factors (F[6,42]=1.43, P=0.227) indicating that the effect of processing was not dependent on the level of reverberation. To further analyse this, the SRT benefit over omni was calculated (Figure 2) and the two-way RM-ANOVA was re-run. The analysis revealed a significant main effect of processing (P=0.004). The main effect of reverberation was not significant (p=0.098), and the interaction term was not significant (P=0.659), further showing that while overall CI performance varied with reverberation, the benefit of directional processing was relatively consistent over the LowR, MidR and HighR levels of reverberation.

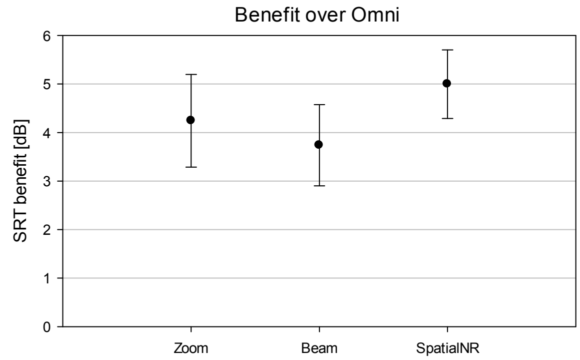

Figure 2: SRT Benefit over the Omni processing condition. Data have been averaged across three reverberation conditions LowR, MidR and HighR. Error bars show the 95% C.I.

Comparison of the processing condition factor showed that all pairs except Beam vs. Zoom resulted in significantly different comparisons. In particular, the benefit over omni (Figure 2) was significant for Zoom (4.2 dB, P<0.001), Beam (3.7 dB, P<0.001) and SpatialNR (5.0 dB, P<0.001). The benefit of SpatialNR over Zoom was 0.8 dB (P=0.043) and over Beam was 1.3 dB (P=0.005).

3.2. HighR/HighRSim

The comparison between HighR and HighRSim was made to validate the room simulator. The two-way RM-ANOVA with reverberation level and processing condition as main factors revealed no significant effect of reverberation level (F[1,7]=4.30, P=0.076). As expected, a significant main effect of processing condition (F[3,21]=43.23, P<0.001) was found. The interaction term, reverberation x processing was not significant (F[3,21]=0.791, P=0.513). Post-hoc comparisons revealed the level of reverberation did not produce a significant difference in performance in any of the processing conditions (omni P=0.937, Zoom P=0.112, Beam P=0.111 and SpatialNR P=0.937).

3.3. HighRSim/ExtraHighRSim

The comparison between reverberation levels HighRSim and ExtraHighRSim was made to analyse the effect of room acoustics on speech intelligibility. The mean SRTs are shown in Figure 3. The two-way RM-ANOVA with reverberation level and processing as main factors revealed a significant interaction term, reverberation x processing (F[3,21]=5.68, P=0.006). This indicates that the effect of changing the reverberation level had a different impact on the different types of processing.

In order to analyse the impact of reverberation on the benefit that each algorithm provided, the SRT benefit over Omni was calculated (Figure 4). A two-way RM-ANOVA on the SRT benefit over omni using processing condition and reverberation level as factors was performed. It revealed a significant interaction between the SRT benefit over Omni and the level of reverberation, consistent with the analysis of raw SRTs. Post-hoc analysis showed the SRT benefit over Omni in each of the processing conditions increased with ExtraHighRSim compared to HighRSim. The SRT benefit over Omni due to SpatialNR increased significantly by 7.1 dB (P=0.005). The increase in SRT benefit over omni due to Zoom (2.1 dB, P=0.369) and Beam (4.3 dB, P=0.053) failed to reach significance due to the high variability in results across the group in the ExtraHighRSim condition.

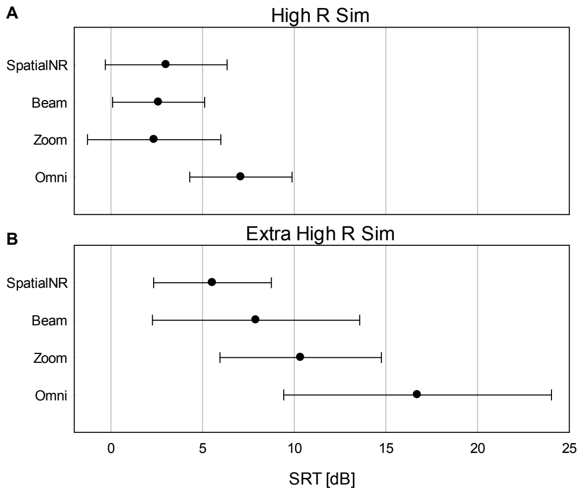

Figure 3: SRT results for HighRSim and ExtraHighRSim reverberation conditions. Error bars indication the 95% C.I.

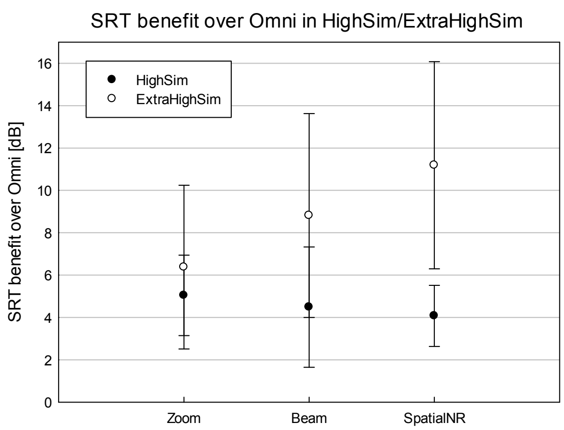

Figure 4: SRT benefit over the Omni condition in HighRSim and ExtraHighRSim levels of reverberation. Error bars show 95%

4. DISCUSSION

Changing the level of reverberant energy in the signal was initially achieved by altering the source to listener distance within the room. Comparing LowR and MidR conditions indicates the effect of changing the distance of the noise sources from 1 m to 3 m whilst keeping the target distance constant at 1 m, which did not affect intelligibility.

In the HighR condition, moving the target speech from 1 m to 3 m resulted in decreased performance. This suggests that the reverberant energy of the target speech played a significant role in speech intelligibility in the reverberant room when background babble noise was present, and perhaps had more impact than the competing babble noise itself. The effect was consistent across Omni, Zoom, Beam and SpatialNR such that the pattern of results obtained with the different algorithms was not significantly affected by the level of reverberation. These results suggest that the target to listener distance may play a key role in speech understanding in reverberant conditions where background noise is present.

Perceptually-based validation of the room simulator was performed by comparing the HighR and HighRSim results. There was no significant effect of changing the real impulse responses for the simulated impulse responses, and the performance of each algorithm was maintained with the change. This demonstrates that the simulator was useful for producing two-microphone impulse responses with relevant inter-microphone transfer functions that led to realistic processing of the microphone signals thereafter.

Comparing HighRSim and ExtraHighRSim conditions showed the effect of introducing concrete surfaces into the room. The comparison showed that performance in the Omni condition was significantly reduced with the introduction of concrete surfaces. This finding is supported by other studies that show CI performance in noise is reduced as the level of reverberation is increased [29, 30]. The comparison also showed the performance benefit due to each algorithm compared with Omni was maintained for Zoom and Beam, and increased for SpatialNR in the concrete room, demonstrating that all algorithms provided useful benefit in the highly reverberant condition. SpatialNR provided the most benefit, and the degree of benefit significantly increased as the reverberation was increased with the introduction of concrete surfaces into room. This finding is in contrast to other works, which contend that high levels of reverberation are detrimental to the performance of multi-microphone noise reduction algorithms [e.g. 7, 8, 10]. The current finding is not completely without precedence, however. Leeuw and Dreschler [31] found that while hearing aid performance decreased with listening distance in a reverberant room ($T60 \approx 900$ms), the benefit due to directionality was not significantly affected as the listening distance was increased. Ricketts and Hornsby [12] found the benefit of hearing aid directional microphones was not effected by listening distance in lower reverberation ($T60=300$ms) but was detrimental in higher reverberation ($T60=900$ms). Therefore, it is likely that the spatial configuration of the competing noise and physical location of target and noise sources within the reverberant room play an important role in the outcome and ultimate benefit that multi-microphone processing is able to provide.

5. CONCLUSION

The results of this experiment suggest that not only the acoustic properties of the room, but also the physical location of target and noise sources within the room, particularly the target to listener distance, play an important role in speech understanding for CI listeners. The benefit provided by all directional algorithms was maintained even with reasonable high levels of reverberation, suggesting that directional noise reduction algorithms, and in particular SpatialNR, provide robust benefits in terms of speech understanding in noisy reverberant environments.

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7. REFERENCES

[1] A. C. Neuman, M. Wroblewski, J. Hajicek, and A. Rubinstein, "Measuring speech recognition in children with cochlear implants in a virtual classroom," *J Speech Lang Hear Res,* vol. 55, pp. 532-40, Apr 2012.

[2] J. Muller-Deile, B. J. Schmidt, and H. Rudert, "Effects of noise on speech discrimination in cochlear implant patients," *Ann Otol Rhinol Laryngol Suppl,* vol. 166, pp. 303-6, Sep 1995.

[3] K. Kokkinakis, O. Hazrati, and P. C. Loizou, "A channel-selection criterion for suppressing reverberation in cochlear implants," *J Acoust Soc Am,* vol. 129, pp. 3221-3232, 2011.

[4] J. M. Desmond, L. M. Collins, and C. S. Throckmorton, "The effects of reverberant self- and overlap-masking on speech recognition in cochlear implant listeners," *J Acoust Soc Am,* vol. 135, pp. EL304-10, Jun 2014.

[5] A. A. Hersbach, K. Arora, S. J. Mauger, and P. W. Dawson, "Combining directional microphone and single-channel noise reduction algorithms: a clinical evaluation in difficult listening conditions with cochlear implant users," *Ear Hear,* vol. 33, pp. e13-23, Jul-Aug 2012.

[6] L. Griffiths and C. Jim, "An alternative approach to linearly constrained adaptive beamforming," *IEEE Trans Antennas Propag,* vol. 30, pp. 27-34, 1982.

[7] R. J. van Hoesel and G. M. Clark, "Evaluation of a portable two-microphone adaptive beamforming speech processor with cochlear implant patients," *J Acoust Soc Am,* vol. 97, pp. 2498-503, Apr 1995.

[8] V. Hamacher, W. H. Doering, G. Mauer, H. Fleischmann, and J. Hennecke, "Evaluation of noise reduction systems for cochlear implant users in different acoustic environment," *Am J Otol,* vol. 18, pp. S46-9, Nov 1997.

[9] J. Wouters and J. Vanden Berghe, "Speech recognition in noise for cochlear implantees with a two-microphone monaural adaptive noise reduction system," *Ear Hear,* vol. 22, pp. 420-30, Oct 2001.

[10] J. E. Greenberg and P. M. Zurek, "Microphone-array hearing aids," in *Microphone arrays, signal processing techniques and applications*, M. Brandstein and D. B. Ward, Eds., ed Berlin, Germany: Springer, 2001.

[11] M. Kompis, M. Bertram, P. Senn, J. Muller, M. Pelizzone, and R. Hausler, "A two-microphone noise reduction system for cochlear implant users with nearby microphones - Part II: Performance evaluation," *EURASIP Journal on Advances in Signal Processing,* vol. 2008, p. 451273, 2008.

[12] T. A. Ricketts and B. W. Hornsby, "Distance and reverberation effects on directional benefit," *Ear Hear,* vol. 24, pp. 472-84, Dec 2003.

[13] R. Zelinski, "A microphone array with adaptive post-filtering for noise reduction in reverberant rooms," presented at the IEEE International Conference on Acoustics, Speech and Signal Processing Las Vegas, Nevada, 1988.

[14] I. A. McCowan and H. Bourlard, "Microphone array post-filter based on noise field coherence," *IEEE Trans Speech Audio Process,* vol. 11, pp. 709-716, 2003.

[15] Y. C. Cao, S. Sridharan, and M. Moody, "Speech enhancement using microphone array with multi-stage processing," *IEICE Trans Fundamentals,* vol. E79A, p. 386, 1996.

[16] A. A. Hersbach, D. B. Grayden, J. B. Fallon, and H. J. McDermott, "A beamformer post-filter for cochlear implant noise reduction," *J Acoust Soc Am,* vol. 133, pp. 2412-20, Apr 2013.

[17] A. A. Hersbach, S. J. Mauger, D. B. Grayden, J. B. Fallon, and H. J. McDermott, "Algorithms to improve listening in noise for cochlear implant users," presented at the IEEE International Conference on Acoustics, Speech and Signal Processing, Vancouver, 2013.

[18] A. Spriet, L. Van Deun, K. Eftaxiadis, J. Laneau, M. Moonen, B. van Dijk, A. van Wieringen, and J. Wouters, "Speech understanding in background noise with the two-microphone adaptive beamformer BEAM in the Nucleus Freedom Cochlear Implant System," *Ear Hear,* vol. 28, pp. 62-72, Feb 2007.

[19] N. Yousefian and P. C. Loizou, "A Dual-Microphone Algorithm That Can Cope With Competing-Talker Scenarios," *IEEE Trans Audio Speech Lang Process,* vol. 21, pp. 145-155, Jan 2013.

[20] G. W. Elko, "Spatial coherence functions for differential microphones in isotropic noise fields," in *Microphone Arrays*, M. Brandstein and D. Ward, Eds., ed Berlin, Germany: Springer, 2001, pp. 61-85.

[21] R. Martin, "Small microphone arrays with postfilters for noise and acoustic echo reduction," in *Microphone Arrays*, M. Brandstein and D. B. Ward, Eds., ed Berlin, Germany: Springer, 2001, pp. 255-279.

[22] R. L. Goldsworthy, L. A. Delhorne, J. G. Desloge, and L. D. Braida, "Two-microphone spatial filtering provides speech reception benefits for cochlear implant users in difficult acoustic environments," *The Journal of the Acoustical Society of America,* vol. 136, pp. 867-876, 2014.

[23] S. J. Mauger, C. D. Warren, M. R. Knight, M. Goorevich, and E. Nel, "Clinical evaluation of the Nucleus® 6 cochlear implant system: Performance improvements with SmartSound iQ," *International Journal of Audiology,* vol. 53, pp. 564-576, 2014.

[24] A. Wabnitz, N. Epain, C. Jin, and A. van Schaik, "Room acoustics simulation for multichannel microphone arrays," in *Proceedings of the International Symposium on Room Acoustics*, 2010.

[25] S. M. Schimmel, M. F. Muller, and N. Dillier, "A fast and accurate “shoebox” room acoustics simulator," presented at the IEEE International Conference on Acoustics, Speech and Signal Processing, 2009.

[26] P. W. Dawson, A. A. Hersbach, and B. A. Swanson, "An Adaptive Australian Sentence Test in Noise (AuSTIN)," *Ear Hear,* p. 1, 2013.

[27] R. J. Shavelson, "Statistical reasoning for the behavioral sciences," 1988.

[28] M. Keuls, "The use of the „studentized range” in connection with an analysis of variance," *Euphytica,* vol. 1, pp. 112-122, 1952/07/01 1952.

[29] S. F. Poissant, N. A. Whitmal, 3rd, and R. L. Freyman, "Effects of reverberation and masking on speech intelligibility in cochlear implant simulations," *J Acoust Soc Am,* vol. 119, pp. 1606-15, Mar 2006.

[30] O. Hazrati and P. C. Loizou, "The combined effects of reverberation and noise on speech intelligibility by cochlear implant listeners," *Int J Audiol,* vol. 51, pp. 437-43, Jun 2012.

[31] A. R. Leeuw and W. A. Dreschler, "Advantages of directional hearing aid microphones related to room acoustics," *International Journal of Audiology,* vol. 30, pp. 330-344, 1991.