Aalto University
School of Electrical
Engineering

Parametric Spatial Audio: Uses in Hearing Assistive Devices

Janani Fernandez

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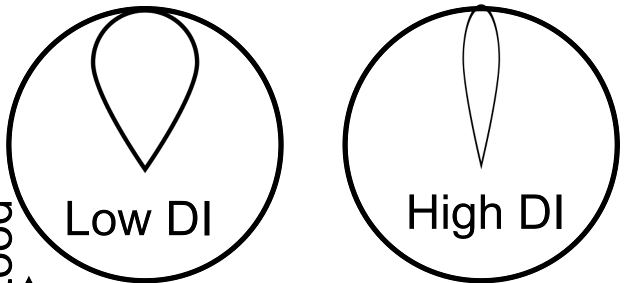
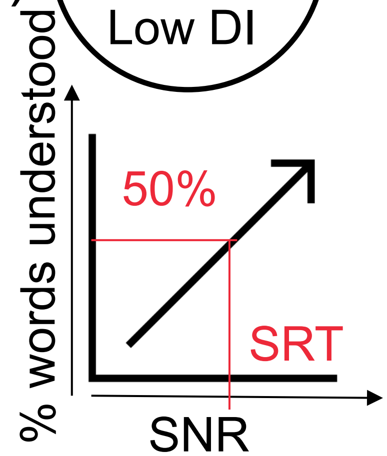
What do we mean by parametric spatial audio (for the purposes of this talk)?

- Parameters derived from the microphone array signals;
- They describe or include spatial information about the sound scene
  - Direction of Arrival
  - Direct to reverb ratio
  - Diffuseness
  - Number of sound sources

Hearing Assistive Device (HAD) metrics (a small selection)

- Signal to Noise Ratio (SNR)
- Directivity Index of Beamformers (DI)
- Speech Reception Threshold (SRT)
- Preservation of binaural cues
  - Interaural Time Difference (ITD)
  - Interaural Level Difference (ILD)
  - Interaural Coherence (IC)



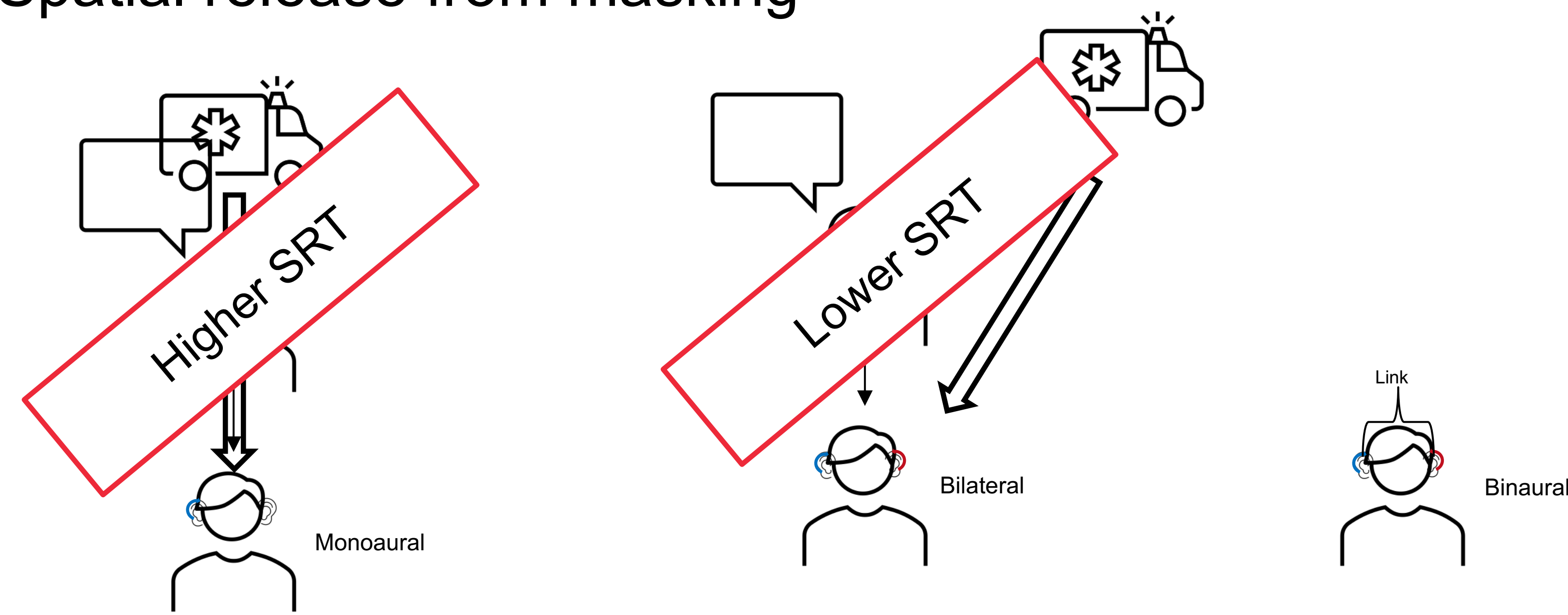


Why use parametric spatial methods in HADs?

- Signal to Noise Ratio (SNR)
  - Knowledge of the space helps choose parameters for processing
- Directivity Index of Beamformers (DI)
  - Make them adaptive instead of static
- Speech Reception Threshold (SRT)
  - Aside from the above, spatial release from masking
- Preservation of binaural cues
  - Interaural Time Difference (ITD)
  - Interaural Level Difference (ILD)
  - Interaural Coherence (IC)

Why use parametric spatial methods in HADs?

- Spatial release from masking



Hearing Aid Processing

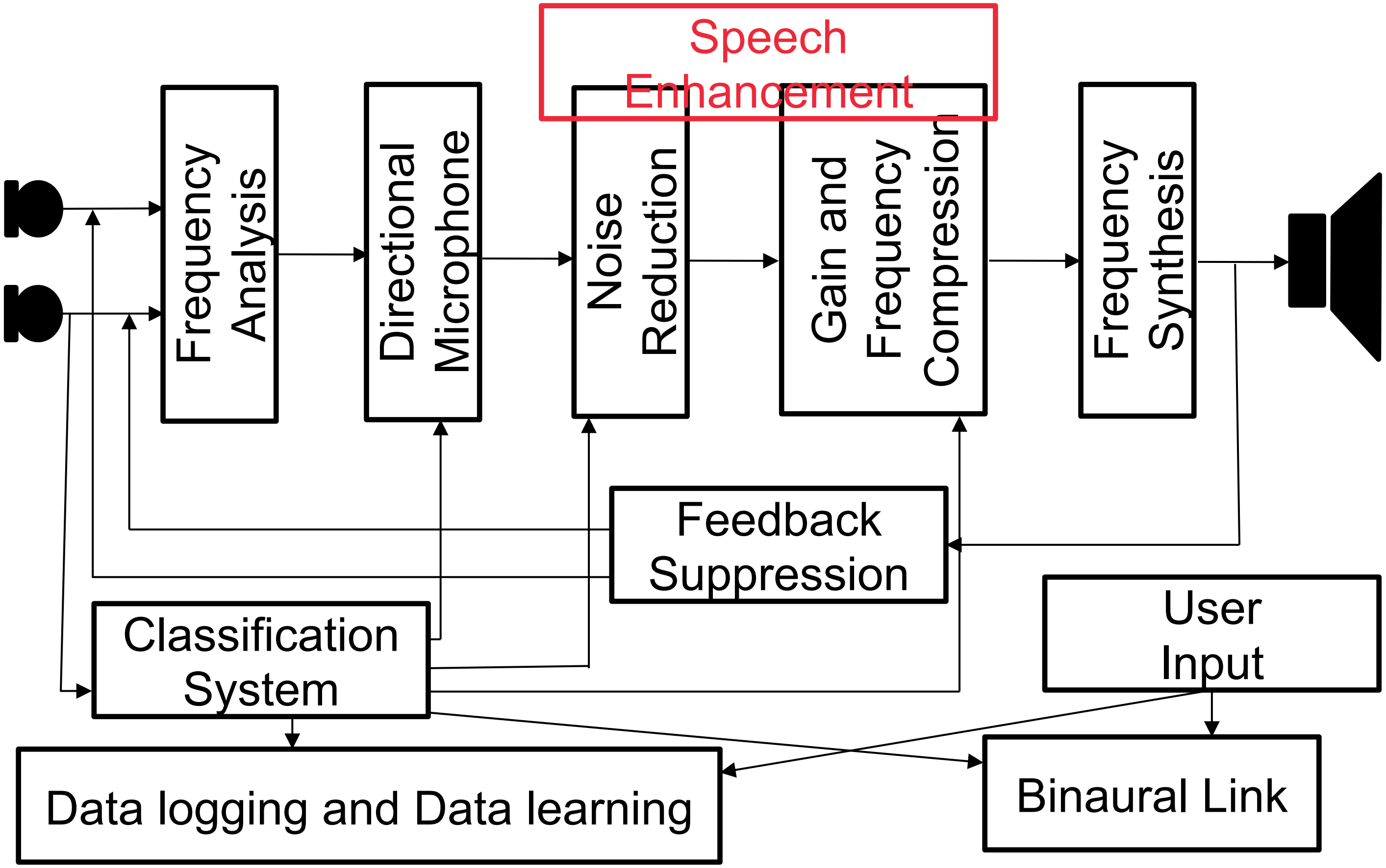
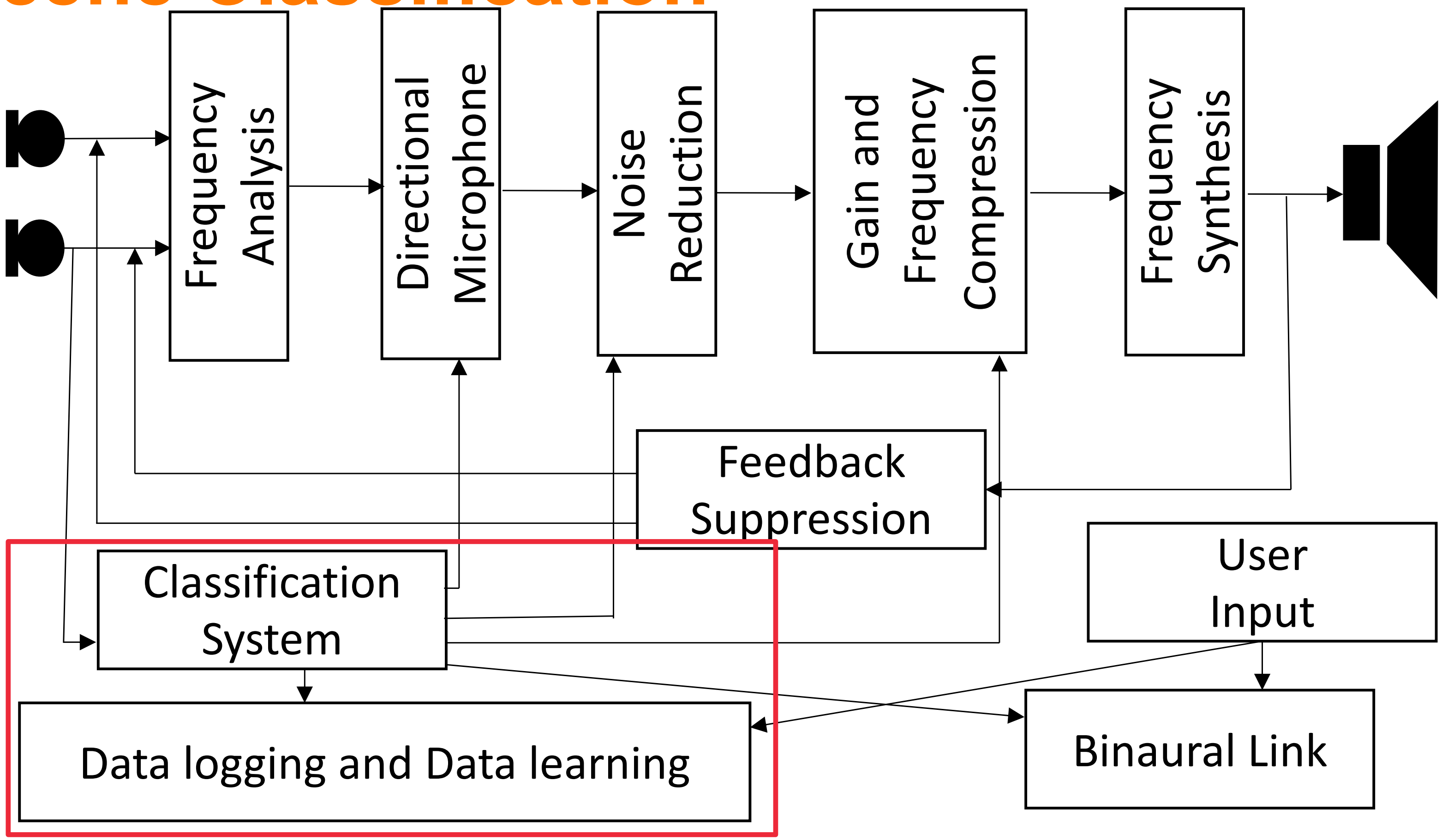


Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

On Device

On Device: Parametric Spatial Audio in Scene Classification



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On Device: Parametric Spatial Audio in Scene Classification

Speech Enhancement Algorithm Based on Sound Source Localization and Scene Matching for Binaural Digital Hearing Aids

Ruwei Li[1] · Dongmei Pan[1] · Shuang Zhang[1]

Received: 28 August 2017 / Accepted: 23 April 2018 / Published online: 27 April 2018

Abstract
At present, the speech enhancement algorithm in binaural digital hearing aids is mainly based on adaptive beamforming algorithm. This algorithm strongly depends on the environment. And the enhancement performance is not satisfactory, which makes it difficult for hearing loss people to get high intelligibility and comfort speech. To solve this problem, a binaural

- Li et al. (2018)
  - Uses DoA, distance and other parameters, for scene matching

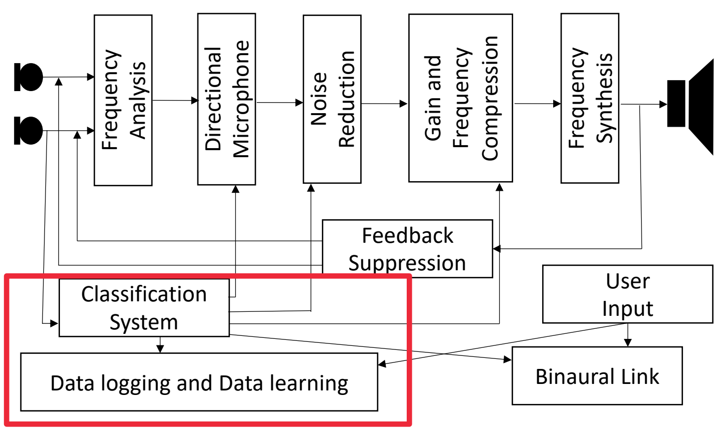


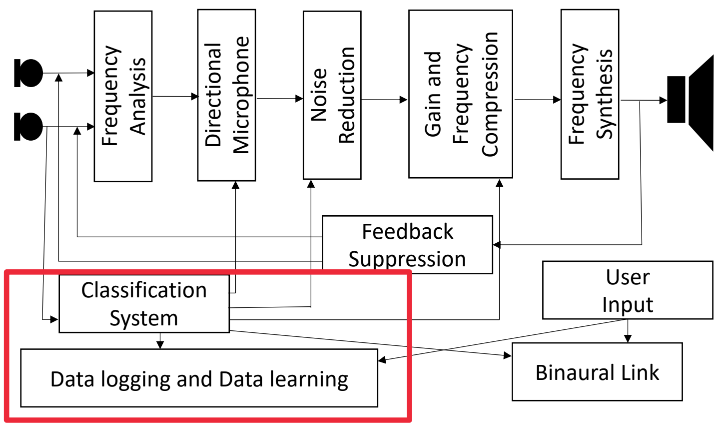
Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

Li, R., Pan, D., & Zhang, S. (2019). Speech enhancement algorithm based on sound source localization and scene matching for binaural digital hearing aids. *Journal of Medical and Biological Engineering*, *39*, 403-417.

On Device: Parametric Spatial Audio in Scene Classification

Improved Sound Classification by Means of Sound Localization in Hearing Devices

A dissertation submitted to
ETH Zurich
for the degree of
Doctor of Sciences

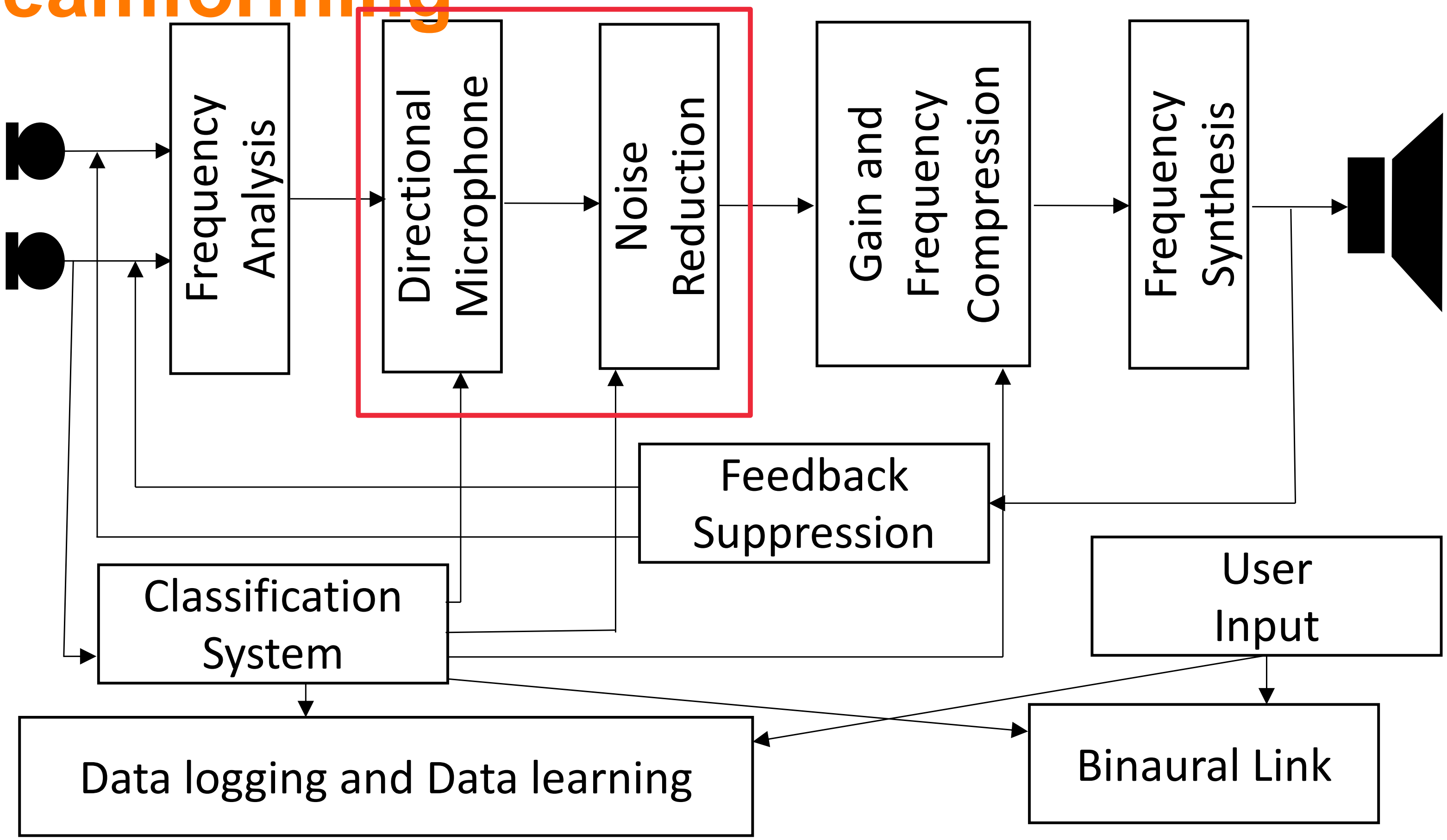


- Li et al. (2018)
  - Uses DoA, distance and other parameters, for scene matching
- Giurda (2020)
  - Doctoral thesis, focuses on spatial characteristics used for scene classification

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

Giurda, R. (2020). *Improved sound classification by means of sound localization in hearing devices* (Doctoral dissertation, ETH Zurich).

On Device: Parametric Spatial Audio for Beamforming



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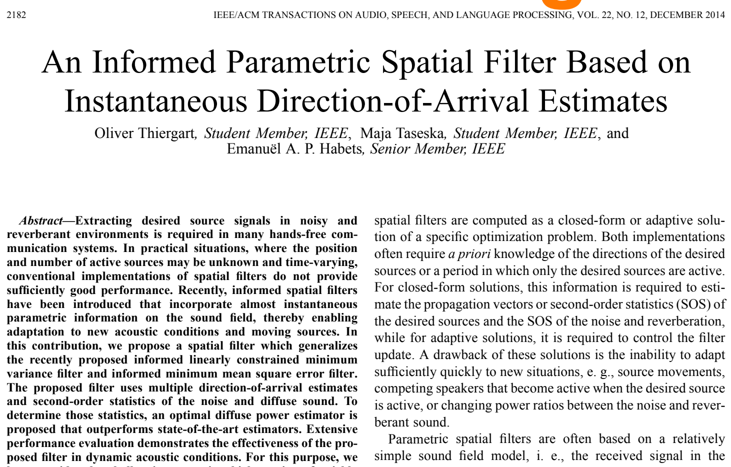
On Device: Parametric Spatial Audio for Beamforming



2182 IEEE/ACM TRANSACTIONS ON AUDIO, SPEECH, AND LANGUAGE PROCESSING, VOL. 22, NO. 12, DECEMBER 2014

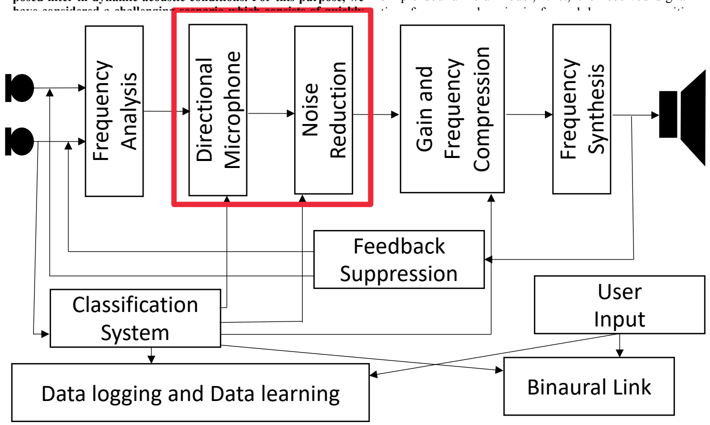
An Informed Parametric Spatial Filter Based on Instantaneous Direction-of-Arrival Estimates

Oliver Thiergart, *Student Member, IEEE*, Maja Taseska, *Student Member, IEEE*, and Emanuël A. P. Habets, *Senior Member, IEEE*

***Abstract*—Extracting desired source signals in noisy and reverberant environments is required in many hands-free communication systems. In practical situations, where the position and number of active sources may be unknown and time-varying, conventional implementations of spatial filters do not provide sufficiently good performance. Recently, informed spatial filters have been introduced that incorporate almost instantaneous parametric information on the sound field, thereby enabling adaptation to new acoustic conditions and moving sources. In this contribution, we propose a spatial filter which generalizes the recently proposed informed linearly constrained minimum variance filter and informed minimum mean square error filter. The proposed filter uses multiple direction-of-arrival estimates and second-order statistics of the noise and diffuse sound. To determine those statistics, an optimal diffuse power estimator is proposed that outperforms state-of-the-art estimators. Extensive performance evaluation demonstrates the effectiveness of the proposed filter in dynamic acoustic conditions. For this purpose, we**

spatial filters are computed as a closed-form or adaptive solution of a specific optimization problem. Both implementations often require *a priori* knowledge of the directions of the desired sources or a period in which only the desired sources are active. For closed-form solutions, this information is required to estimate the propagation vectors or second-order statistics (SOS) of the desired sources and the SOS of the noise and reverberation, while for adaptive solutions, it is required to control the filter update. A drawback of these solutions is the inability to adapt sufficiently quickly to new situations, e. g., source movements, competing speakers that become active when the desired source is active, or changing power ratios between the noise and reverberant sound.

Parametric spatial filters are often based on a relatively simple sound field model, i. e., the received signal in the



- Thiergart et al. (2014)
  - parametrically driven Wiener filter

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

Thiergart, O., Taseska, M., & Habets, E. A. (2014). An informed parametric spatial filter based on instantaneous direction-of-arrival estimates. *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, *22*(12), 2182-2196.

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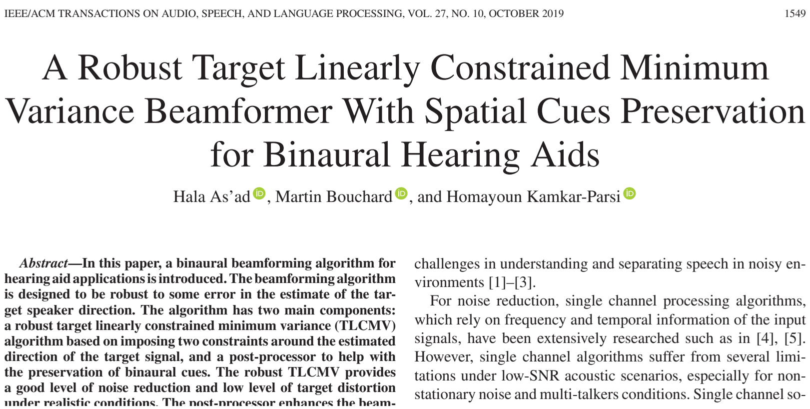
On Device: Parametric Spatial Audio for Beamforming


IEEE/ACM TRANSACTIONS ON AUDIO, SPEECH, AND LANGUAGE PROCESSING, VOL. 27, NO. 10, OCTOBER 2019 1549

A Robust Target Linearly Constrained Minimum Variance Beamformer With Spatial Cues Preservation for Binaural Hearing Aids

Hala As'ad, Martin Bouchard, and Homayoun Kamkar-Parsi

Abstract—In this paper, a binaural beamforming algorithm for hearing aid applications is introduced. The beamforming algorithm is designed to be robust to some error in the estimate of the target speaker direction. The algorithm has two main components: a robust target linearly constrained minimum variance (TLCMV) algorithm based on imposing two constraints around the estimated direction of the target signal, and a post-processor to help with the preservation of binaural cues. The robust TLCMV provides a good level of noise reduction and low level of target distortion under realistic conditions. The post-processor enhances the beam-

challenges in understanding and separating speech in noisy environments [1]–[3].

For noise reduction, single channel processing algorithms, which rely on frequency and temporal information of the input signals, have been extensively researched such as in [4], [5]. However, single channel algorithms suffer from several limitations under low-SNR acoustic scenarios, especially for non-stationary noise and multi-talkers conditions. Single channel so-

- Thiergart et al. (2014)
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- As'ad et al. (2019)
  - parametrically driven beamformer

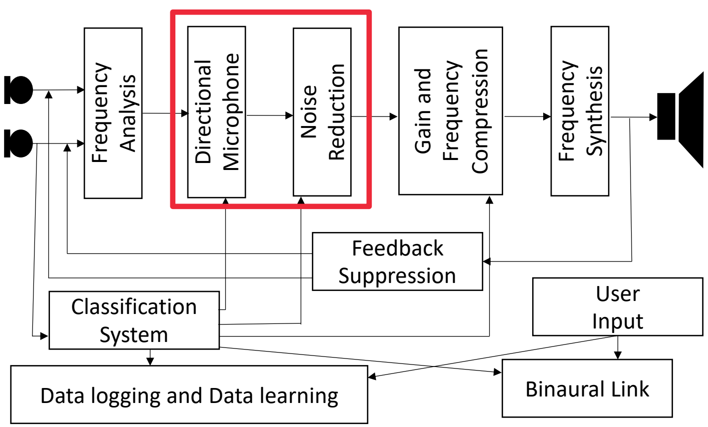

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

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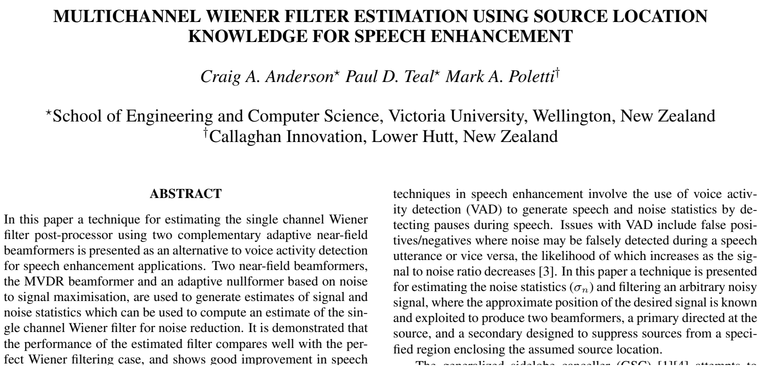
As' ad, H., Bouchard, M., & Kamkar-Parsi, H. (2019). A robust target linearly constrained minimum variance beamformer with spatial cues preservation for binaural hearing aids. *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, *27*(10), 1549-1563.

On Device: Parametric Spatial Audio for Beamforming



MULTICHANNEL WIENER FILTER ESTIMATION USING SOURCE LOCATION KNOWLEDGE FOR SPEECH ENHANCEMENT

Craig A. Anderson Paul D. Teal* Mark A. Poletti†*

*School of Engineering and Computer Science, Victoria University, Wellington, New Zealand
†Callaghan Innovation, Lower Hutt, New Zealand

ABSTRACT

In this paper a technique for estimating the single channel Wiener filter post-processor using two complementary adaptive near-field beamformers is presented as an alternative to voice activity detection for speech enhancement applications. Two near-field beamformers, the MVDR beamformer and an adaptive nullformer based on noise to signal maximisation, are used to generate estimates of signal and noise statistics which can be used to compute an estimate of the single channel Wiener filter for noise reduction. It is demonstrated that the performance of the estimated filter compares well with the perfect Wiener filtering case, and shows good improvement in speech

techniques in speech enhancement involve the use of voice activity detection (VAD) to generate speech and noise statistics by detecting pauses during speech. Issues with VAD include false positives/negatives where noise may be falsely detected during a speech utterance or vice versa, the likelihood of which increases as the signal to noise ratio decreases [3]. In this paper a technique is presented for estimating the noise statistics (σ_n) and filtering an arbitrary noisy signal, where the approximate position of the desired signal is known and exploited to produce two beamformers, a primary directed at the source, and a secondary designed to suppress sources from a specified region enclosing the assumed source location.

- Thiergart et al. (2014)
  - parametrically driven Wiener filter
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  - parametrically driven beamformer
- Anderson er al. (2014)
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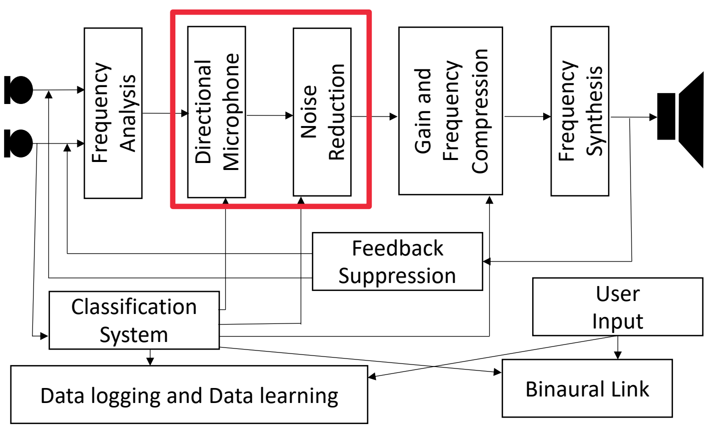

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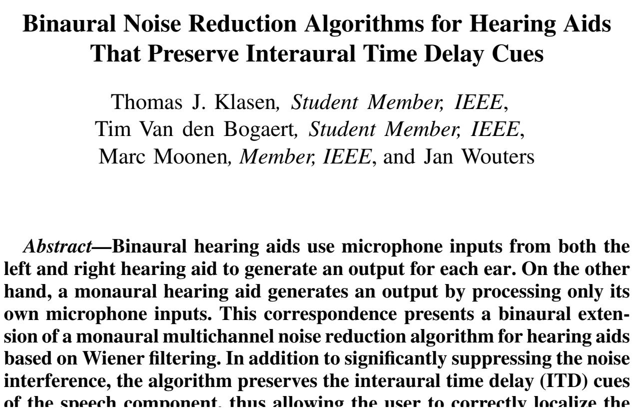
Anderson, C., Teal, P. D., & Poletti, M. A. (2014, June). Multichannel Wiener filter estimation using source location knowledge for speech enhancement. In *2014 IEEE Workshop on Statistical Signal Processing (SSP)* (pp. 57-60). IEEE

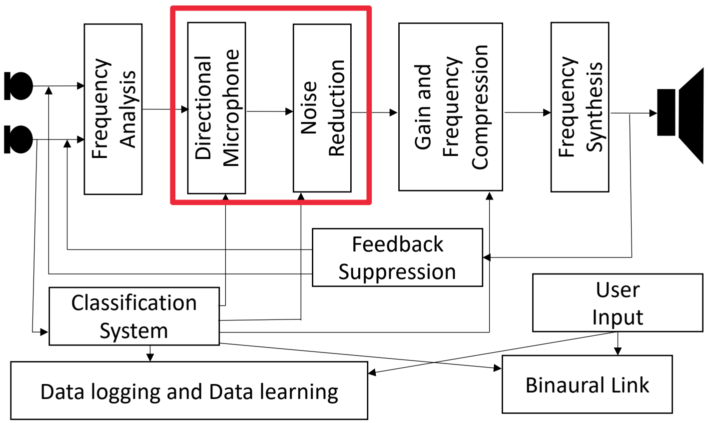
On Device: Parametric Spatial Audio for Beamforming



Binaural Noise Reduction Algorithms for Hearing Aids That Preserve Interaural Time Delay Cues

Thomas J. Klasen, *Student Member, IEEE*,
Tim Van den Bogaert, *Student Member, IEEE*,
Marc Moonen, *Member, IEEE*, and Jan Wouters

***Abstract*—Binaural hearing aids use microphone inputs from both the left and right hearing aid to generate an output for each ear. On the other hand, a monaural hearing aid generates an output by processing only its own microphone inputs. This correspondence presents a binaural extension of a monaural multichannel noise reduction algorithm for hearing aids based on Wiener filtering. In addition to significantly suppressing the noise interference, the algorithm preserves the interaural time delay (ITD) cues of the speech component, thus allowing the user to correctly localize the**



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  - Uses an adaptive “nullformer”.
- Klasen et al. (2007)
  - Binaural Wiener filter

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

Klasen, T. J., Van den Bogaert, T., Moonen, M., & Wouters, J. (2007). Binaural noise reduction algorithms for hearing aids that preserve interaural time delay cues. *IEEE Transactions on Signal Processing*, *55*(4), 1579-1585.

On Device: Parametric Spatial Audio for Beamforming

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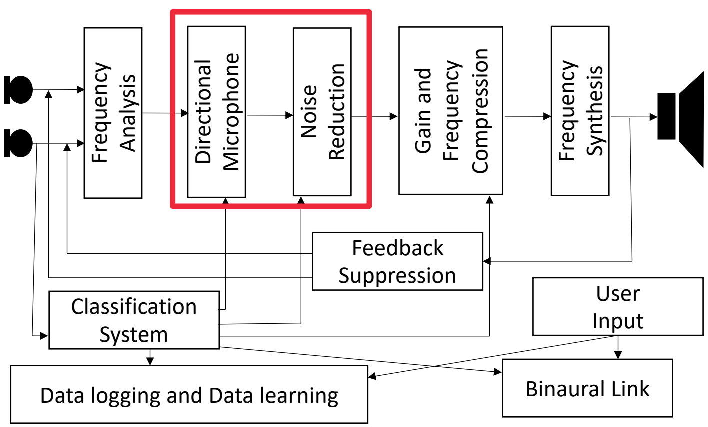

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  - parametrically driven beamformer
- Anderson et al. (2014)
  - Uses an adaptive "nullformer".
- Klasen et al. (2007)
  - Binaural Wiener filter
- …and many, many more

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

No Really. Many.

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AND MORE….

On Device: Parametric Spatial Audio in DRC

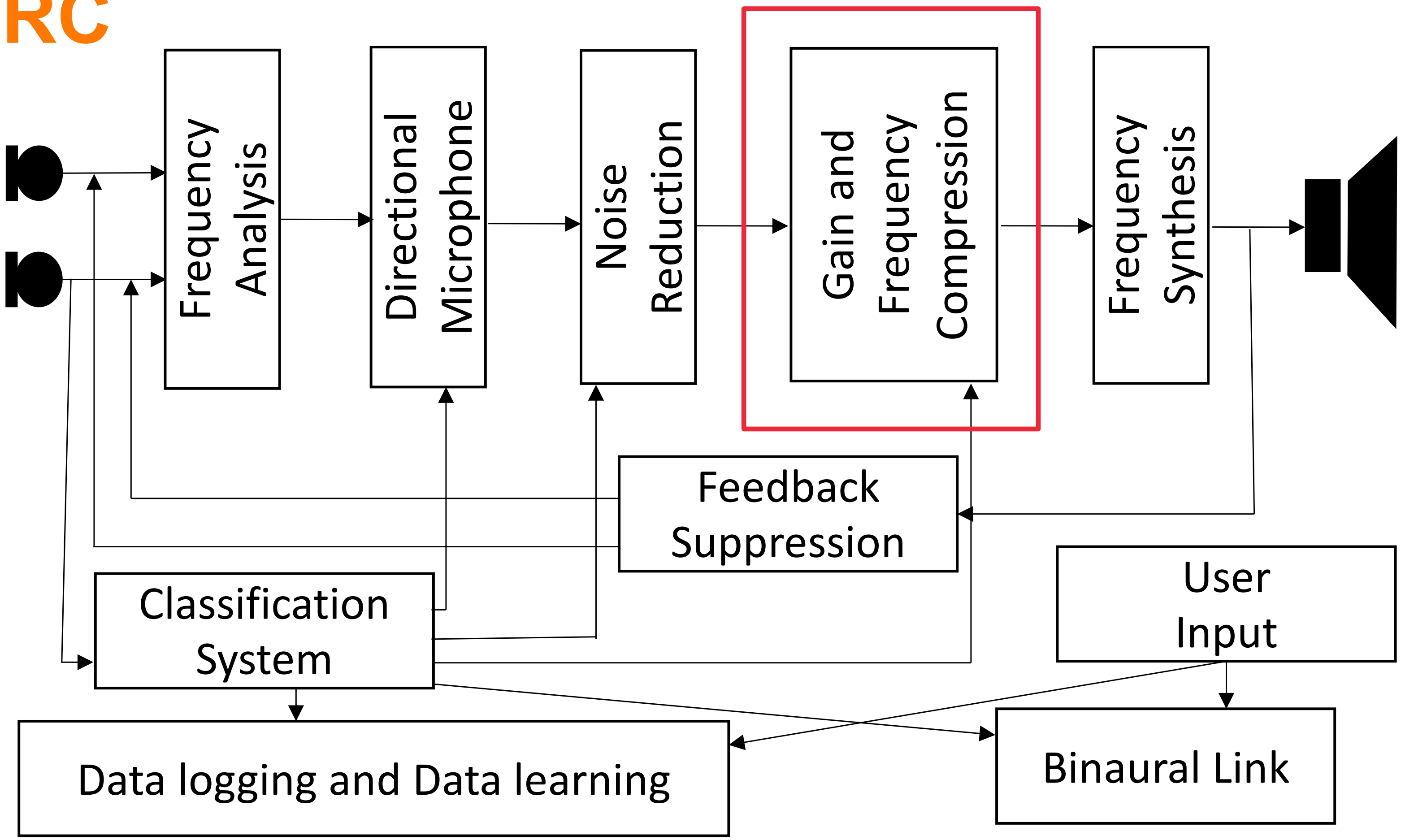

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

On Device: Parametric Spatial Audio in DRC

Original Article

Trends in Hearing
Volume 22: 1–12

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SAGE

Signal-to-Noise-Ratio-Aware Dynamic Range Compression in Hearing Aids

Tobias May[1], Borys Kowalewski[1], and Torsten Dau[1]

Abstract
Fast-acting dynamic range compression is a level-dependent amplification scheme which aims to restore audibility for hearing-impaired listeners. However, when being applied to noisy speech at positive signal-to-noise ratios (SNRs), the gain function typically changes rapidly over time as it is driven by the short-term fluctuations of the speech signal. This leads to an amplification of the noise components in the speech gaps, which reduces the output SNR and distorts the acoustic properties of the background noise. An adaptive compression scheme is proposed here which utilizes information about the SNR in different frequency channels to adaptively change the characteristics of the compressor. Specifically, fast-acting compression is applied to speech-dominated time-frequency (T-F) units where the SNR is high, while slow-acting compression is used to effectively linearize the processing for noise-dominated T-F units where the SNR is low. A systematic evaluation of this SNR-aware compression scheme showed that the effective compression of speech components embedded in noise was similar to that of a conventional fast-acting system, whereas natural fluctuations in the background noise were preserved in a similar way as when a slow-acting compressor was applied.

Keywords
wide dynamic range compression, signal-to-noise ratio, hearing-aid signal processing

- May et al. (2018)
  - used SNR ratios to choose the DRC parameters

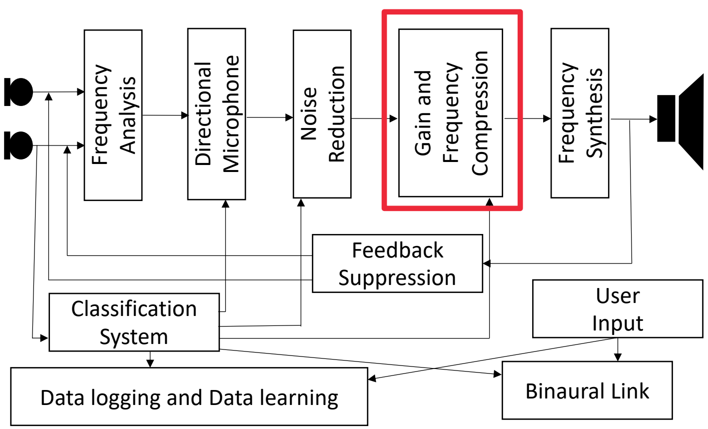

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

May, T., Kowalewski, B., & Dau, T. (2020). Scene-aware dynamic-range compression in hearing aids. *The Technology of Binaural Understanding*, 763-799.

On Device: Parametric Spatial Audio in DRC

Proceedings, APSIPA Annual Summit and Conference 2020 7-10 December 2020, Auckland, New Zealand

Localization Cues Preservation in Hearing Aids by Combining Noise Reduction and Dynamic Range Compression

Adrien Llave, Simon Leglaive and Renaud Séguier
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firstname.lastname@centralesupelec.fr

Abstract—**Dynamic range compression (DRC) and noise reduction algorithms are commonly used in hearing aids. They are known to have opposite objectives concerning the Signal-to-Noise Ratio (SNR) and to affect negatively the localization performance. Yet, the study of their interaction received few attention. In this work, we improve an existing combined approach of DRC and noise reduction to bridge the gap between the algorithms proposed independently in their respective communities. The proposed solution is then compared to state-of-the-art algorithms thanks to objective criteria assessing the spatial fidelity preser-**

follow the short-term speech level fluctuations. Therefore, the reverberation tail is considered as a soft speech period which has to be reinforced. The consequence is a lowering of the direct-to-reverberant energy ratio (DRR), leading to more internalization, image source diffusion for the listener as well as more front-back confusion and source localization splitting. The authors further showed by means of a perceptual evaluation that the Interaural Coherence (IC) [11] is an

- May et al. (2018)
  - used SNR ratios to choose the DRC parameters
- Llave et al. (2020)
  - Combined the above and parametric driven beamformers

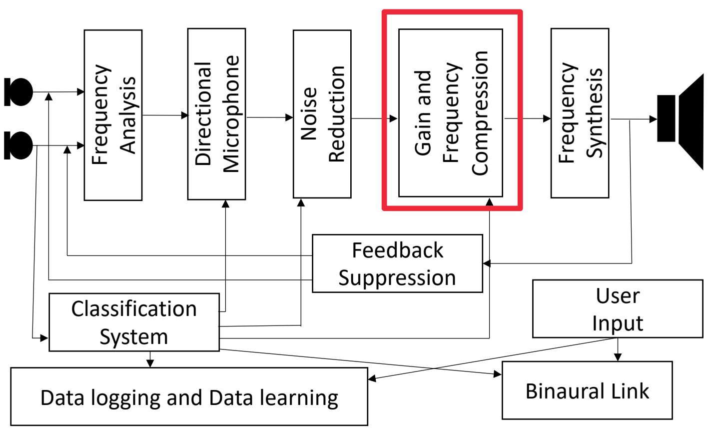

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

Llave, A., Leglaive, S., & Séguier, R. (2020, December). Localization cues preservation in hearing aids by combining noise reduction and dynamic range compression. In *2020 Asia-Pacific Signal and Information Processing Association Annual Summit and Conference (APSIPA ASC)* (pp. 686-693). IEEE.

On Device: Parametric Spatial Audio for improving SRT

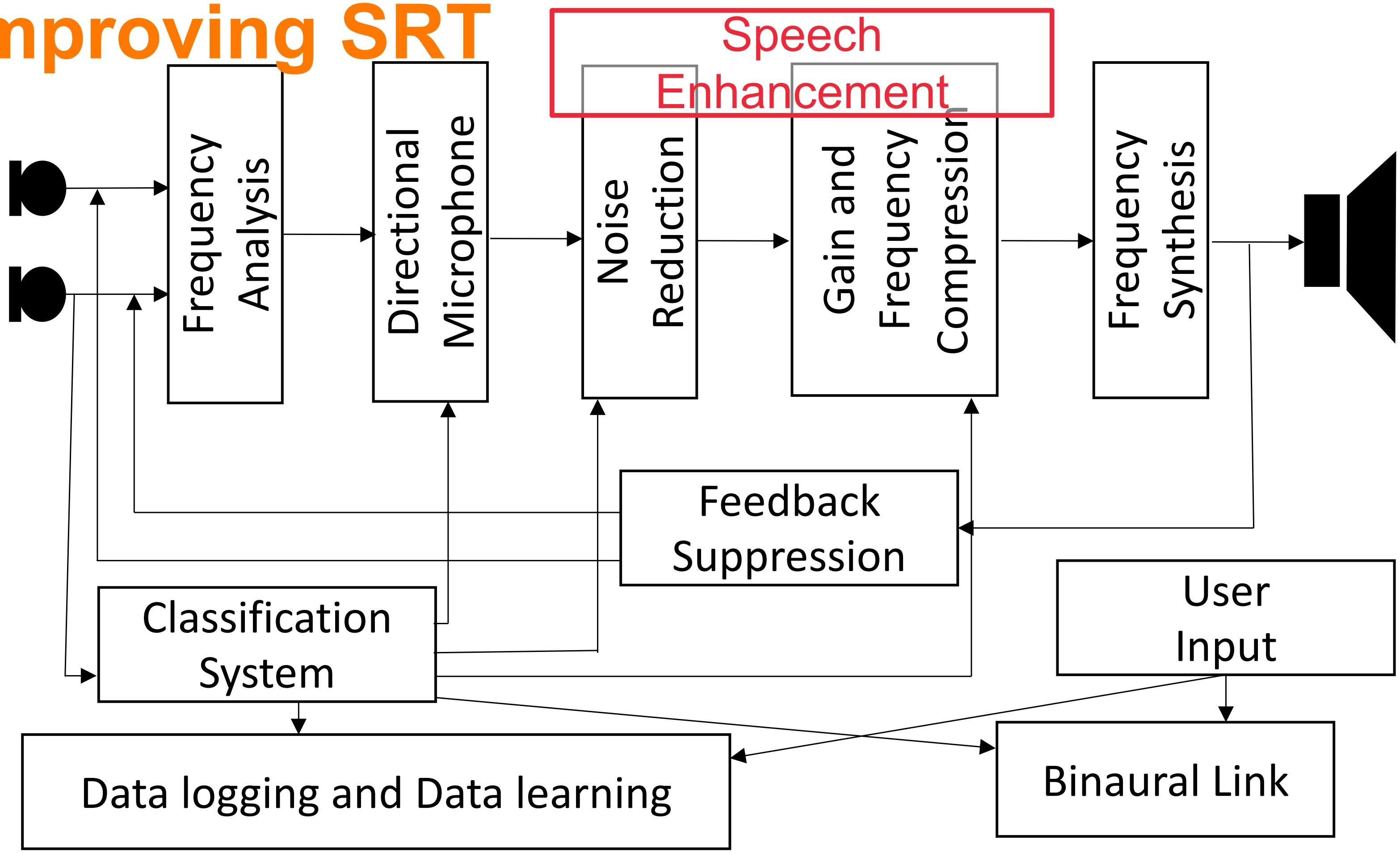

Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

On Device: Parametric Spatial Audio for Improving SRT


Available online at www.sciencedirect.com
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Speech Communication 76 (2016) 157–169
SPEECH COMMUNICATION
www.elsevier.com/locate/specom

Binaural rendering of microphone array captures based on source separation

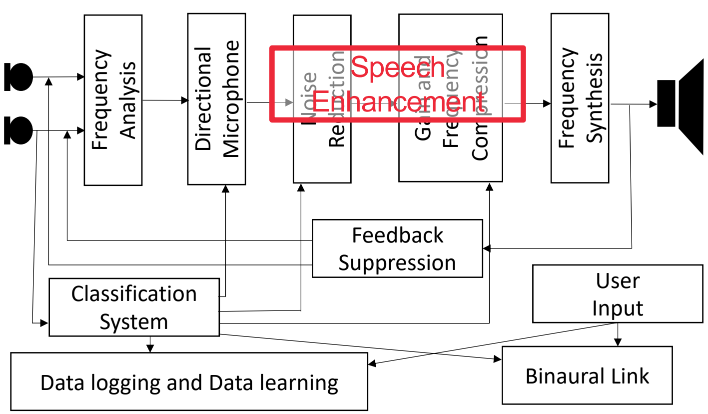
Joonas Nikunen [a,*,1], Aleksandr Diment [a,1], Tuomas Virtanen [a,1], Miikka Vilermo [b]

[a] *Tampere University of Technology, Department of Signal Processing, Tampere, Finland*
[b] *Nokia Research Center, Tampere, Finland*

Received 9 January 2015; received in revised form 24 July 2015; accepted 12 September 2015
Available online 21 September 2015

Nikunen et al. (2015)

- DoA informed machine learning based speech enhancement intended for multi-talker sound scenes.



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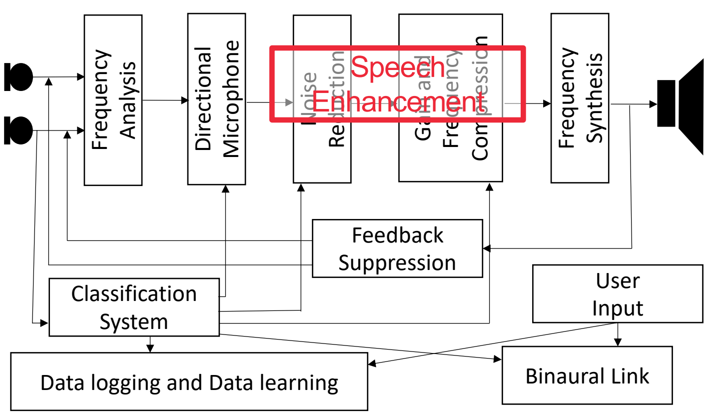
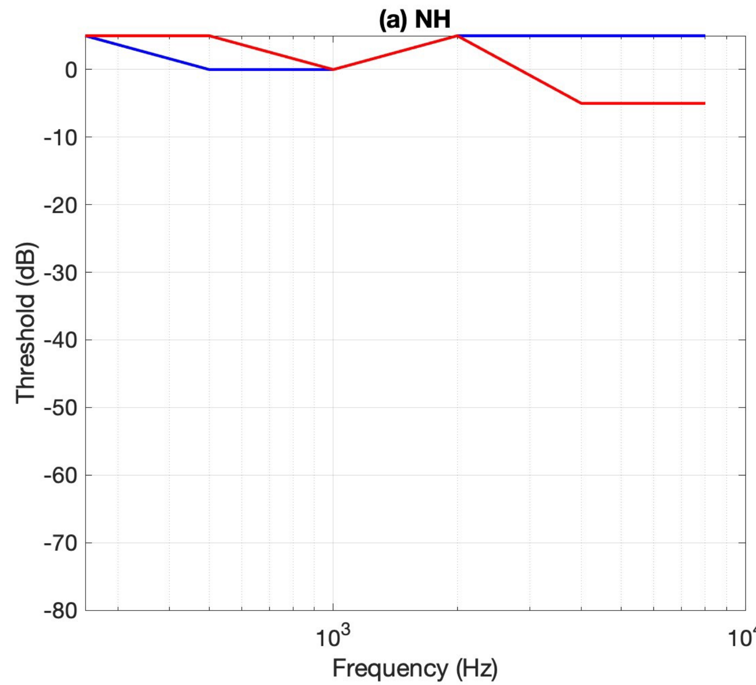
Nikunen, J., Diment, A., Virtanen, T., & Vilermo, M. (2016). Binaural rendering of microphone array captures based on source separation. *Speech Communication*, *76*, 157-169.

On Device: Parametric Spatial Audio in Pitch Shifting





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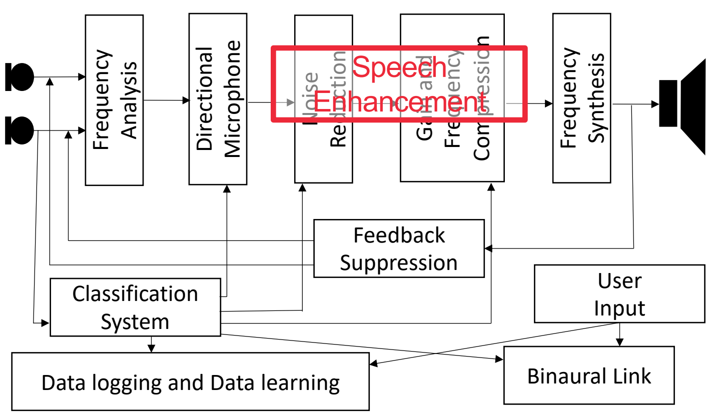


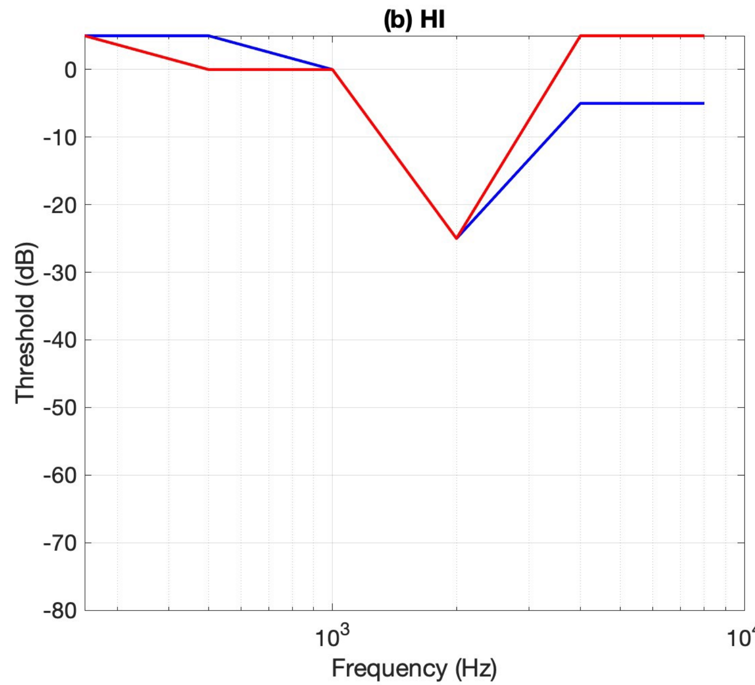
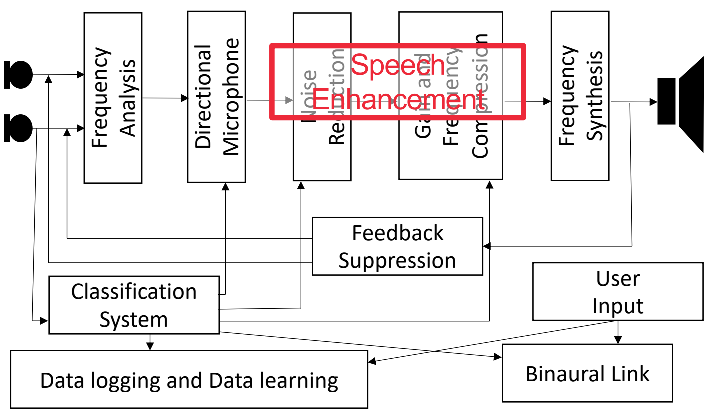


Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

On Device: Parametric Spatial Audio in Pitch Shifting



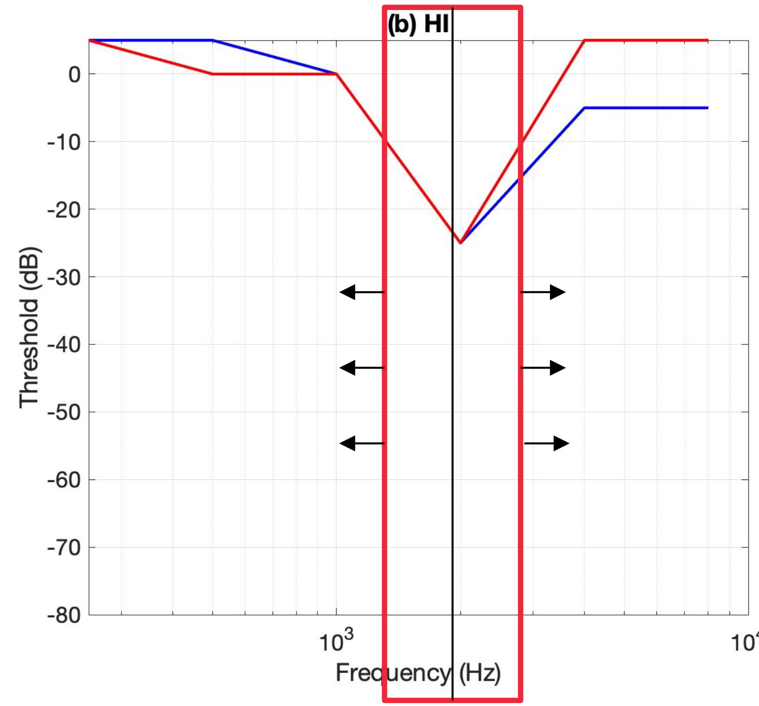


Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

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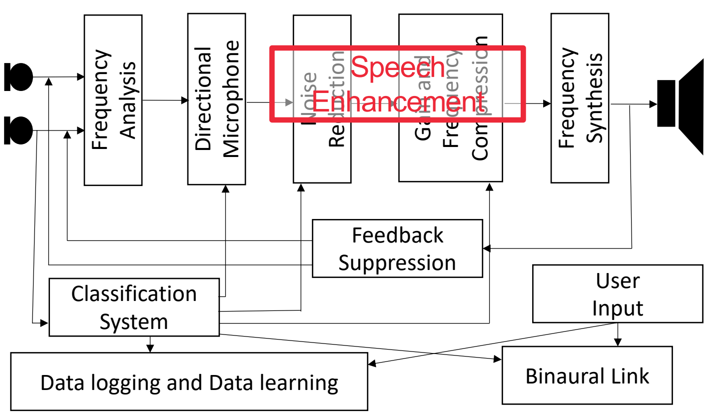
Superhuman spatial hearing technology for ultrasonic frequencies

Ville Pulkki, Leo McCormack & Raimundo Gonzalez

Ultrasonic sources are inaudible to humans, and while digital signal processing techniques are available to bring ultrasonic signals into the audible range, there are currently no systems which also



- Pulkki et al. (2021)
  - Pitch shifted bats to human hearing range

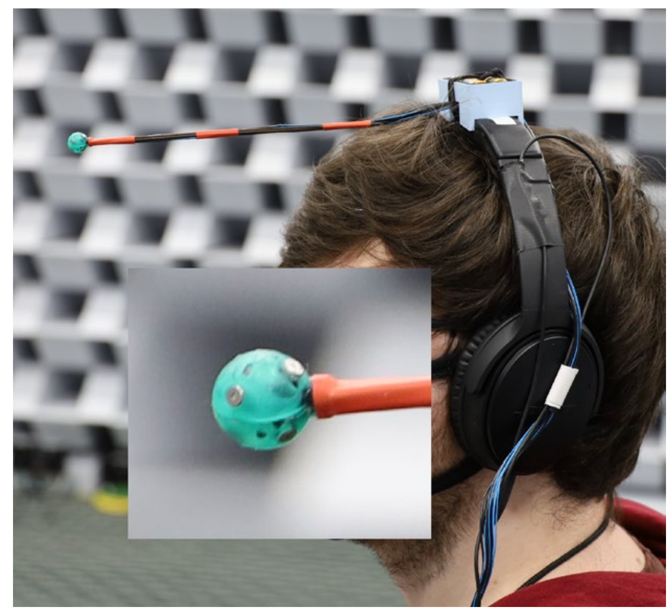
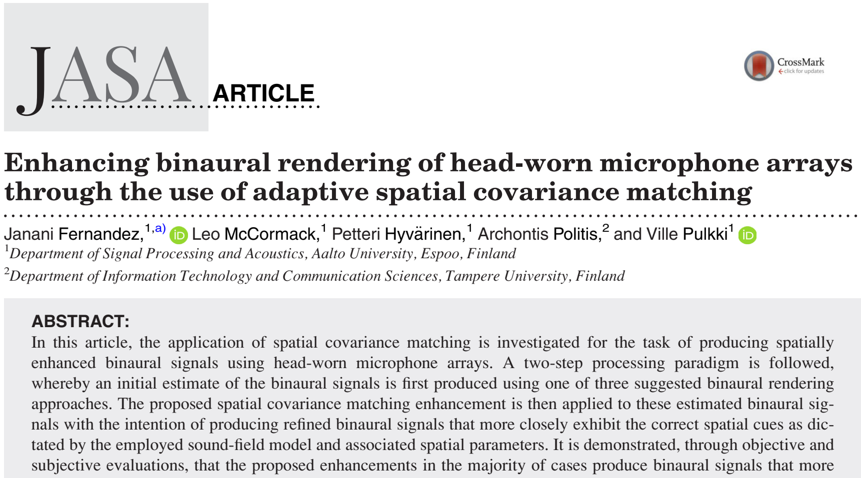


Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

Pulkki, V., McCormack, L., & Gonzalez, R. (2021). Superhuman spatial hearing technology for ultrasonic frequencies. *Scientific Reports*, *11*(1), 11608.

On Device: Parametric Spatial Audio



JASA ARTICLE

Enhancing binaural rendering of head-worn microphone arrays through the use of adaptive spatial covariance matching

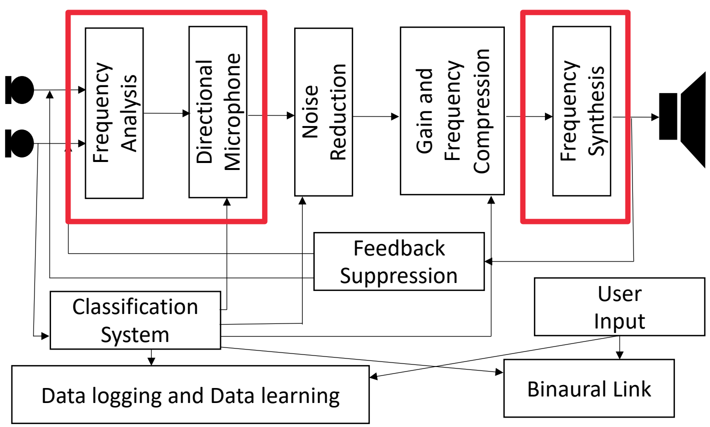
Janani Fernandez,[1,a)] Leo McCormack,[1] Petteri Hyvärinen,[1] Archontis Politis,[2] and Ville Pulkki[1]

[1]*Department of Signal Processing and Acoustics, Aalto University, Espoo, Finland*

[2]*Department of Information Technology and Communication Sciences, Tampere University, Finland*

ABSTRACT:
In this article, the application of spatial covariance matching is investigated for the task of producing spatially enhanced binaural signals using head-worn microphone arrays. A two-step processing paradigm is followed, whereby an initial estimate of the binaural signals is first produced using one of three suggested binaural rendering approaches. The proposed spatial covariance matching enhancement is then applied to these estimated binaural signals with the intention of producing refined binaural signals that more closely exhibit the correct spatial cues as dictated by the employed sound-field model and associated spatial parameters. It is demonstrated, through objective and subjective evaluations, that the proposed enhancements in the majority of cases produce binaural signals that more

- Fernandez et al. (2022)
  - Proposed a general framework for spatially enhancing binaural signals
  - Designed for head-worn devices
  - Can be used in AR/MR



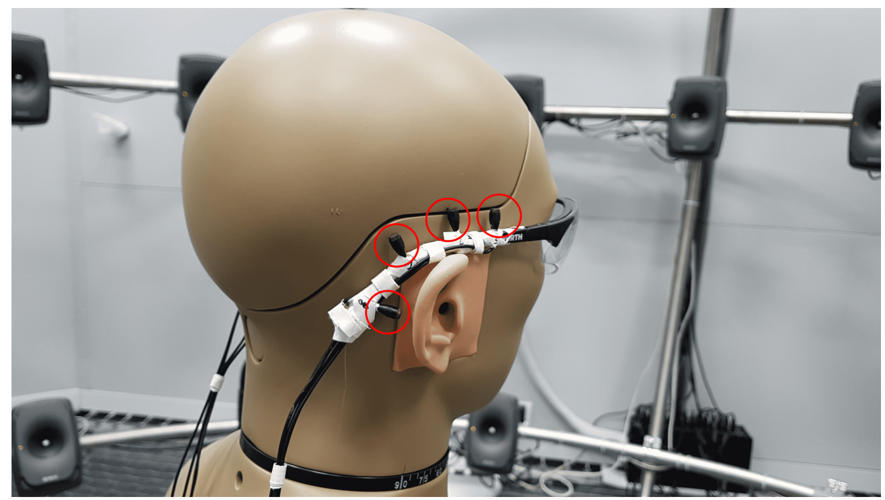
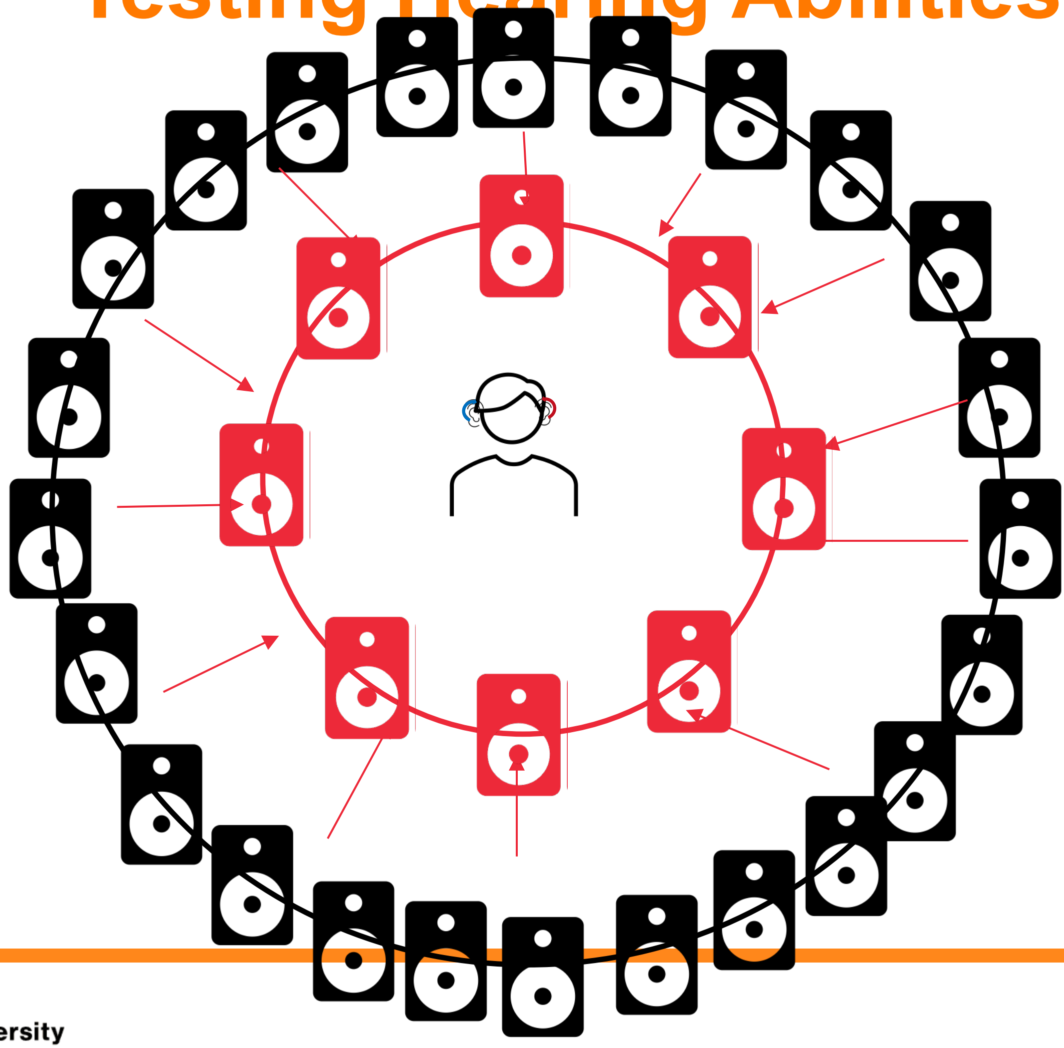


Figure adapted from Eddins, D. A. (2014). Sandlin's Textbook of Hearing Aid Amplification. 233.

Fernandez, J., McCormack, L., Hyvärinen, P., Politis, A., & Pulkki, V. (2022). Enhancing binaural rendering of head-worn microphone arrays through the use of adaptive spatial covariance matching. *The Journal of the Acoustical Society of America*, *151*(4), 2624-2635.

Off Device

Off Device: Spatialisation Algorithms in Testing Hearing Abilities



Aalto University
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Off Device: Spatialisation Algorithms in Testing Hearing Abilities

(S1) Original Article

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Volume 28: 1–17

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Toward Sound Localization Testing in Virtual Reality to Aid in the Screening of Auditory Processing Disorders

Melissa Ramírez[1,2], Johannes M. Arend[2], Petra von Gablenz[3], Heinrich R. Liesefeld[4] and Christoph Pörschmann[1]

Abstract

Sound localization testing is key for comprehensive hearing evaluations, particularly in cases of suspected auditory processing disorders. However, sound localization is not commonly assessed in clinical practice, likely due to the complexity and size of conventional measurement systems, which require semicircular loudspeaker arrays in large and acoustically treated rooms. To address this issue, we investigated the feasibility of testing sound localization in virtual reality (VR). Previous research has shown that virtualization can lead to an increase in localization blur. To measure these effects, we conducted a study with a group of normal-hearing adults, comparing sound localization performance in different augmented reality and VR scenarios. We started with a conventional loudspeaker-based measurement setup and gradually moved to a virtual audiovisual environment, testing sound localization in each scenario using a within-participant design. The loudspeaker-based experiment yielded results comparable to those reported in the literature, and the results of the virtual localization test provided new insights into localization performance in state-of-the-art VR environments. By comparing localization performance between the loudspeaker-based and virtual conditions, we were able to estimate the increase in localization blur induced by virtualization relative to a conventional test setup. Notably, our study provides the first proxy normative cutoff values for sound localization testing in VR. As an outlook, we discuss the potential of a VR-based sound localization test as a suitable, accessible, and portable alternative to conventional setups and how it could serve as a time- and resource-saving prescreening tool to avoid unnecessarily extensive and complex laboratory testing.

Keywords

spatial hearing, binaural interaction functions, sound localization abilities, auditory processing disorders, virtual reality

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Introduction

Hearing is a complex process. It entails the transduction of acoustic information arriving at the ears into neural impulses, their transmission through the auditory nerves, and their appropriate interpretation by the central nervous system (Werner et al., 2012). Sound localization and lateralization, auditory pattern recognition, temporal integration and discrimination, and speech understanding in challenging acoustic situations are just a few basic skills that rely on our auditory processing abilities (Bellis, 2003a; Chermak & Musiek, 1997). Musiek, 2013; de Wit et al., 2016; Geffner & Ross-Swain, 2019). Children and adults with APDs have impaired abilities to attend to, discriminate, organize, or comprehend auditory

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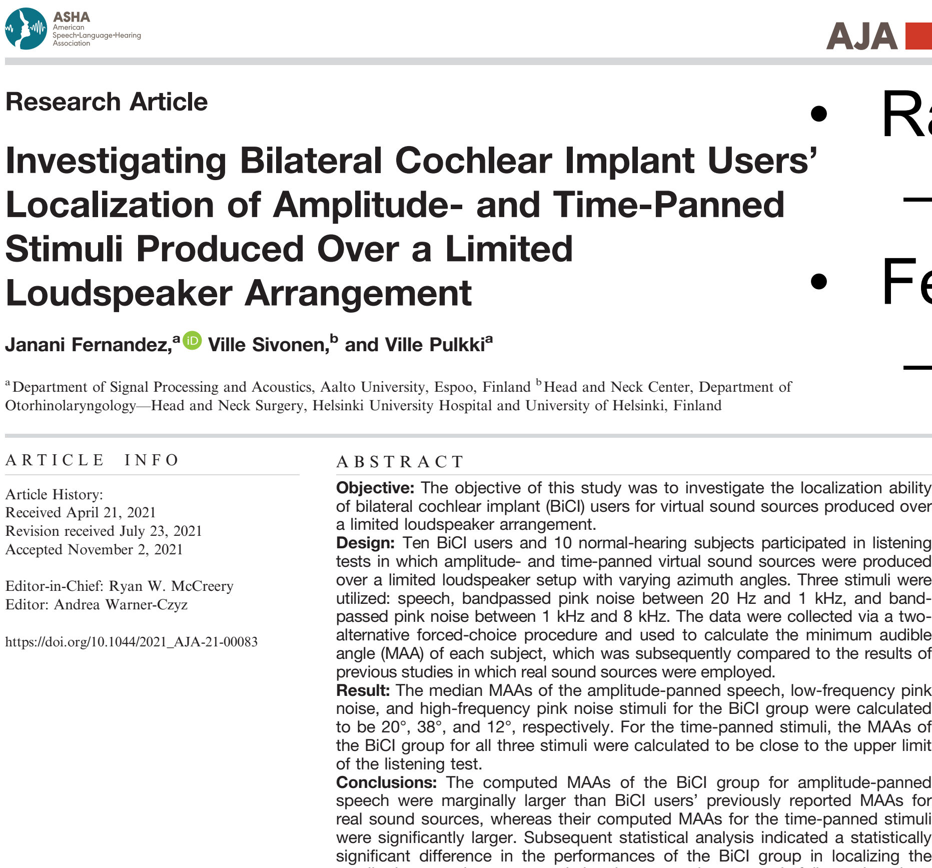
[4]Institute of Psychology, University of Bremen, Bremen, Germany

- Ramirez et al. (2024)
  - Used VBAP to test localisation

Ramírez, M., Arend, J. M., von Gablenz, P., Liesefeld, H. R., & Pörschmann, C. (2024). Toward Sound Localization Testing in Virtual Reality to Aid in the Screening of Auditory Processing Disorders. *Trends in Hearing*, *28*, 23312165241235463.

Off Device: Spatialisation Algorithms in Testing Hearing Abilities



ASHA American Speech-Language-Hearing Association

AJA

Research Article

Investigating Bilateral Cochlear Implant Users' Localization of Amplitude- and Time-Panned Stimuli Produced Over a Limited Loudspeaker Arrangement

Janani Fernandez,[a] Ville Sivonen,[b] and Ville Pulkki[a]

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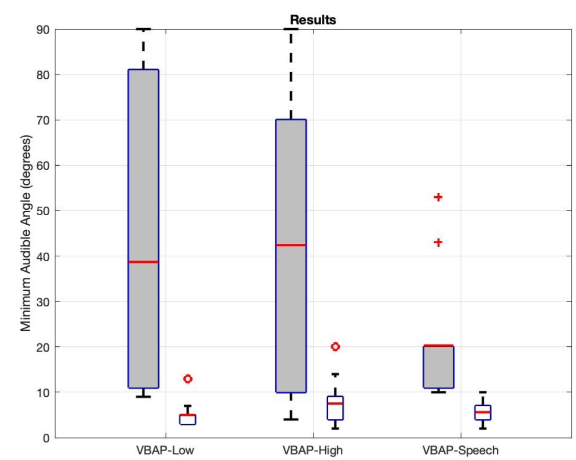
ABSTRACT

Objective: The objective of this study was to investigate the localization ability of bilateral cochlear implant (BiCI) users for virtual sound sources produced over a limited loudspeaker arrangement.
Design: Ten BiCI users and 10 normal-hearing subjects participated in listening tests in which amplitude- and time-panned virtual sound sources were produced over a limited loudspeaker setup with varying azimuth angles. Three stimuli were utilized: speech, bandpassed pink noise between 20 Hz and 1 kHz, and bandpassed pink noise between 1 kHz and 8 kHz. The data were collected via a two-alternative forced-choice procedure and used to calculate the minimum audible angle (MAA) of each subject, which was subsequently compared to the results of previous studies in which real sound sources were employed.
Result: The median MAAs of the amplitude-panned speech, low-frequency pink noise, and high-frequency pink noise stimuli for the BiCI group were calculated to be 20°, 38°, and 12°, respectively. For the time-panned stimuli, the MAAs of the BiCI group for all three stimuli were calculated to be close to the upper limit of the listening test.
Conclusions: The computed MAAs of the BiCI group for amplitude-panned speech were marginally larger than BiCI users' previously reported MAAs for real sound sources, whereas their computed MAAs for the time-panned stimuli were significantly larger. Subsequent statistical analysis indicated a statistically significant difference in the performances of the BiCI group in localizing the

- Ramirez et al. (2024)
  - Used VBAP to test localisation
- Fernandez et al. (2020)
  - VBAP to test localisation for CI users



Fernandez, J., Sivonen, V., & Pulkki, V. (2022). Investigating Bilateral Cochlear Implant Users' Localization of Amplitude-and Time-Panned Stimuli Produced Over a Limited Loudspeaker Arrangement. *American journal of audiology*, *31*(1), 143-154.

Off Device: Spatialisation Algorithms in Training and Improving Hearing Abilities

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Spatial rehabilitation using virtual auditory space training paradigm in individuals with sensorineural hearing impairment

Kavassery Venkateswaran Nisha*, Ajith Kumar Uppunda and Rakesh Trinesh Kumar

Department of Audiology, All India Institute of Speech and Hearing (AIISH), Mysore, India

Purpose: The present study aimed to quantify the effects of spatial training using virtual sources on a battery of spatial acuity measures in listeners with sensorineural hearing impairment (SNHI).

Methods: An intervention-based time-series comparison design involving 82 participants divided into three groups was adopted. Group I ($n = 27$, SNHI-spatially trained) and group II ($n = 25$, SNHI-untrained) consisted of SNHI listeners, while group III ($n = 30$) had listeners with normal hearing (NH). The study was conducted in three phases. In the pre-training phase, all the participants underwent a comprehensive assessment of their spatial processing abilities using a battery of tests including spatial acuity in free-field and closed-field scenarios, tests for binaural processing abilities (interaural time threshold [ITD] and level difference threshold [ILD]), and subjective ratings. While spatial acuity in the free field was assessed using a loudspeaker-based localization test, the closed-field source identification test was performed using virtual stimuli delivered through headphones. The ITD and ILD thresholds were obtained using a MATLAB psychoacoustic toolbox, while the participant ratings on the spatial subsection of speech, spatial, and qualities questionnaire in Kannada were used for the subjective ratings. Group I listeners underwent virtual auditory spatial training (VAST), following pre-evaluation assessments.

- Nisha et al. (2023)
  - Used amplitude panning to train better localisation

Nisha, K. V., Uppunda, A. K., & Kumar, R. T. (2023). Spatial rehabilitation using virtual auditory space training paradigm in individuals with sensorineural hearing impairment. *Frontiers in Neuroscience*,

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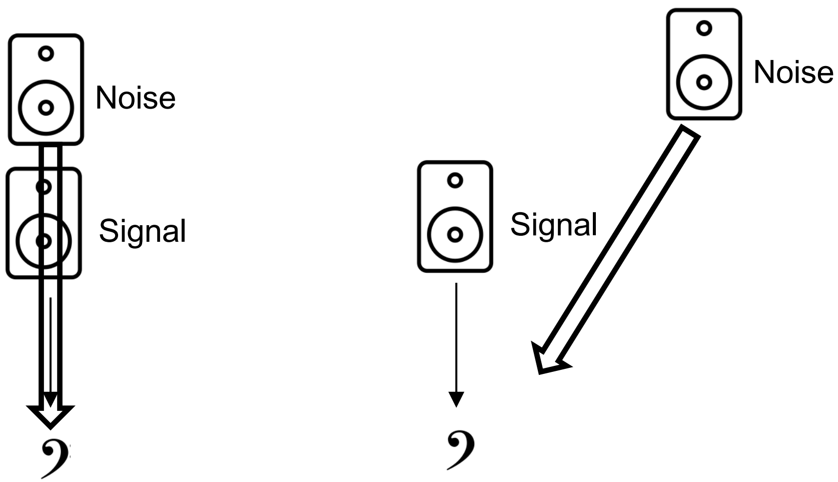
- Virtual Sound Environments (VSE)



Picture adopted from Minnaar, P., Favrot, S., & Buchholz, J. M. (2010). Improving hearing aids through listening tests in a virtual sound environment. *The Hearing Journal*, *63*(10), 40-42.

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- Virtual Sound Environments (VSE)
- Simon et al. (2017)
  - Investigated different spatialization methods paired with a HAD beamformer

University of Zurich[UZH]

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Comparison of Higher-Order Ambisonics, Vector- and Distance-Based Amplitude Panning using a hearing device beamformer

Simon, Laurent S R ; Wuethrich, Hannes ; Dillier, Norbert

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Simon, L. S., Wuethrich, H., & Dillier, N. (2017). Comparison of higher-order Ambisonics, vector-and distance-based amplitude panning using a hearing device beamformer.

Off Device: Spatialisation Algorithms in HAD Research and Developing / Testing

- Virtual Sound Environments (VSE)
- Laurent et al. (2017)
  - Investigated different spatialization methods paired with a HAD beamformer
- Oreinos et al. (2013)
  - Feasibility of using HOA for evaluating SNR and DI

2013 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics October 20-23, 2013, New Paltz, NY

EFFECT OF HIGHER-ORDER AMBISONICS ON EVALUATING BEAMFORMER BENEFIT IN REALISTIC ACOUSTIC ENVIRONMENTS

Chris Oreinos[*] *Jörg M. Buchholz*[*] *Jorge Mejia*[*]

National Acoustic Laboratories, Australian Hearing Hub, Sydney, Australia
The HEARing Cooperative Research Centre, Melbourne, Australia
Macquarie University, Faculty of Human Sciences, Department of Linguistics, Sydney, Australia
{chris.oreinos, jorg.buchholz, jorge.mejia}@nal.gov.au

ABSTRACT

Multi-channel loudspeaker systems have been proposed to assess the real-life benefit of devices such as hearing aids, cochlear implants, or mobile phones. This paper investigates to what extent sound fields recreated by Higher-Order Ambisonics (HOA) can be used to evaluate the performance of spatially selective multi-microphone processing schemes (beamformers) inside complex acoustic environments. Two example schemes are considered: an adaptive directional microphone (ADM) and a contralateral suppression bilateral beamformer (BBF), both implemented in the context of a hearing aid device. The acoustic scenarios consist of a single speech target (0^o) competing against three speech jammers ($\pm 90^o$ and 180^o) set either in an anechoic or in a reverberant simulated classroom ($T_{30} = 0.6s$). The HOA effect on the directional algorithm performance is quantified through: (a) the adaptive, frequency-dependent, algorithm gains, (b) the SNR improvement calculated in one-third octave bands, and (c) the processed target frequency response.

The HOA reconstruction errors influence the beamformers in mainly two ways; first, by altering the spatial characteristics of the sound field, which in turn modifies the adaptation of the algorithms, and second, by affecting the spectral content of the sources. The results suggest that although HOA (here 7^{th} order) does not degrade the broadband, long-term, intelligibility-weighted SNR improvement of the two beamformers, it imposes a low-pass effect on the processed target. This renders the HOA coding problematic above the system's cut-off frequency.

Index Terms— Ambisonics, beamforming, hearing aids, sound field synthesis

1. INTRODUCTION

creating VSEs. The physical and perceptual limitations of HOA are well known but its use to test multi-microphone signal enhancement schemes (i.e., beamformers) has not been studied.

This study evaluates the extent to which HOA-reconstructed sound fields can be used to successfully evaluate beamformers, herein implemented in the context of a hearing aid (HA) device. The whole sound path is considered, including scene generation, reproduction of virtual sources, capturing by HA microphones and beamformer processing. The simulation framework utilizes different distortion metrics to compare the beamformer benefit in a simulated reference and a HOA-decoded sound reproduction system.

2. METHODS

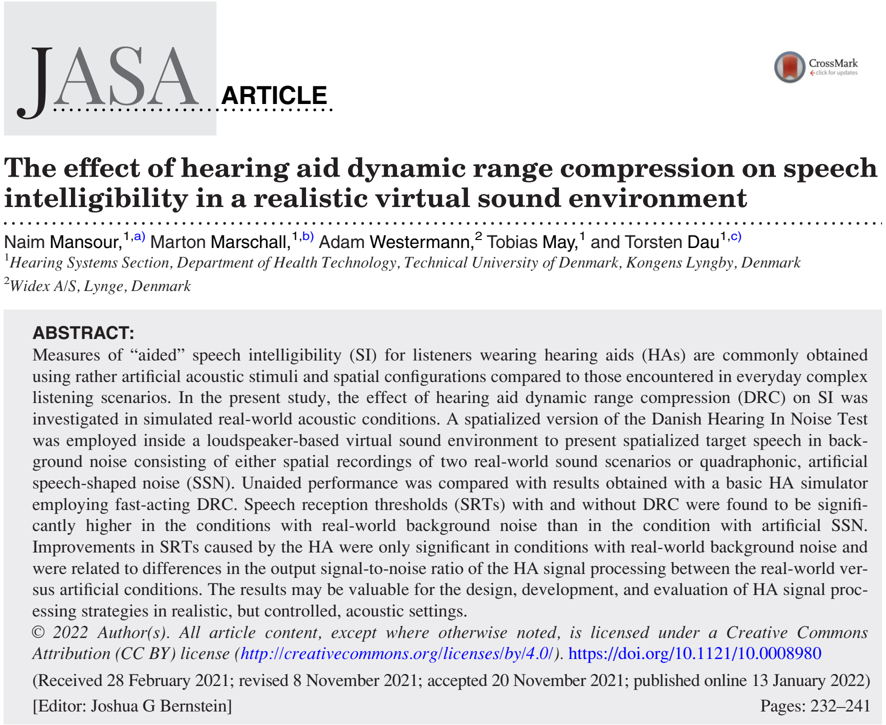
The first stage of the simulation framework (Fig. 1) models the acoustic scene inside which the beamformers operate. Here, it consisted of a target directly in front of a dummy listener and three jammers at $\pm 90^o$ and 180^o, all on the horizontal plane. In the anechoic scenario (path 1), the four talkers were modeled as free-space plane wave sources. In the reverberant scenario (path 2), a classroom ($T_{30} = 0.6s$, $6.7m$ x $9.5m$ x $3m$) was modeled using ODEON. The target was placed at $1m$ and the jammers at $2m$ distance from the listener (room's critical distance was $1.4m$). All jammers were normalized to the same RMS level. After the ODEON simulation, the LoRA (Loudspeaker-based Room Auralization) framework [6] was used to map the direct sound (DS) and specular early reflections (ER) of all talkers to a dense, 1784-node spherical array. The late reverberation was recreated using a virtual 196-node quasi-regular [7] virtual loudspeaker array according to the method described in [6].

Two methods were then used to reproduce the virtual scene

Oreinos, C., Buchholz, J. M., & Mejia, J. (2013, October). Effect of higher-order ambisonics on evaluating beamformer benefit in realistic acoustic environments. In *2013 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics* (pp. 1-4). IEEE.

Off Device: Spatialisation Algorithms in HAD Research and Developing / Testing

- Virtual Sound Environments (VSE)
- Laurent et al. (2017)
  - Investigated different spatialization methods paired with a HAD beamformer
- Oreinos et al. (2013)
  - Feasibility of using HOA for evaluating SNR and DI
- Mansoor et al. (2022)
  - Used HOA for their study


JASA ARTICLE

The effect of hearing aid dynamic range compression on speech intelligibility in a realistic virtual sound environment

Naim Mansour,[1,a)] Marton Marschall,[1,b)] Adam Westermann,[2] Tobias May,[1] and Torsten Dau[1,c)]
[1]*Hearing Systems Section, Department of Health Technology, Technical University of Denmark, Kongens Lyngby, Denmark*
[2]*Widex A/S, Lynge, Denmark*

ABSTRACT:
Measures of "aided" speech intelligibility (SI) for listeners wearing hearing aids (HAs) are commonly obtained using rather artificial acoustic stimuli and spatial configurations compared to those encountered in everyday complex listening scenarios. In the present study, the effect of hearing aid dynamic range compression (DRC) on SI was investigated in simulated real-world acoustic conditions. A spatialized version of the Danish Hearing In Noise Test was employed inside a loudspeaker-based virtual sound environment to present spatialized target speech in background noise consisting of either spatial recordings of two real-world sound scenarios or quadraphonic, artificial speech-shaped noise (SSN). Unaided performance was compared with results obtained with a basic HA simulator employing fast-acting DRC. Speech reception thresholds (SRTs) with and without DRC were found to be significantly higher in the conditions with real-world background noise than in the condition with artificial SSN. Improvements in SRTs caused by the HA were only significant in conditions with real-world background noise and were related to differences in the output signal-to-noise ratio of the HA signal processing between the real-world versus artificial conditions. The results may be valuable for the design, development, and evaluation of HA signal processing strategies in realistic, but controlled, acoustic settings.
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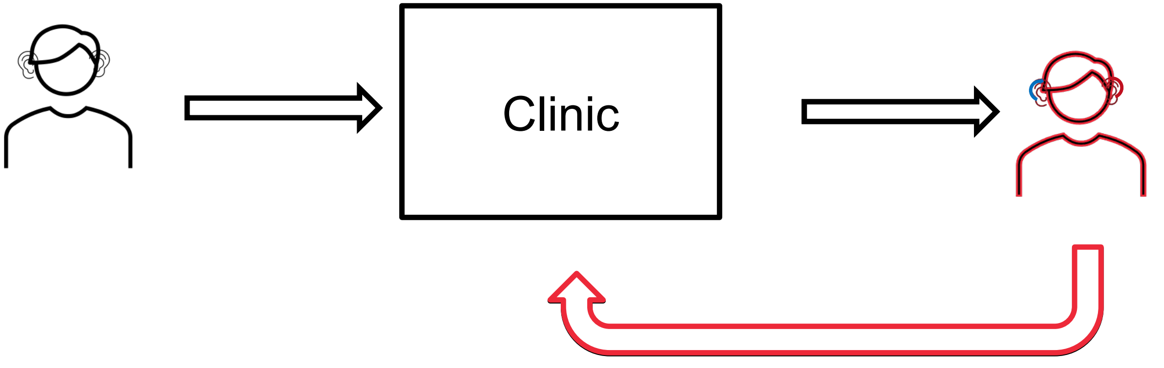

Off Device: Spatialisation Algorithms in HAD Fitting

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An examination of speech reception thresholds measured in a simulated reverberant cafeteria environment

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Abstract

Objective—There is increasing demand in the hearing research community for the creation of laboratory environments that better simulate challenging real-world listening environments. The hope is that the use of such environments for testing will lead to more meaningful assessments of listening ability, and better predictions about the performance of hearing devices. Here we present one approach for simulating a complex acoustic environment in the laboratory, and investigate the effect of transplanting a speech test into such an environment.

Design—Speech reception thresholds were measured in a simulated reverberant cafeteria, and in a more typical anechoic laboratory environment containing background speech babble.

Study Sample—The participants were 46 listeners varying in age and hearing levels, including 25 hearing-aid wearers who were tested with and without their hearing aids.

Results—Reliable SRTs were obtained in the complex environment, but led to different estimates of performance and hearing aid benefit from those measured in the standard environment.

Conclusions—The findings provide a starting point for future efforts to increase the real-world relevance of laboratory-based speech tests.

- Best et al. (2015)
  - Simulated complex sound scenes perhaps better measure of benefit

Best, V., Keidser, G., Buchholz, J. M., & Freeston, K. (2015). An examination of speech reception thresholds measured in a simulated reverberant cafeteria environment. *International Journal of Audiology*, *54*(10), 682-690.

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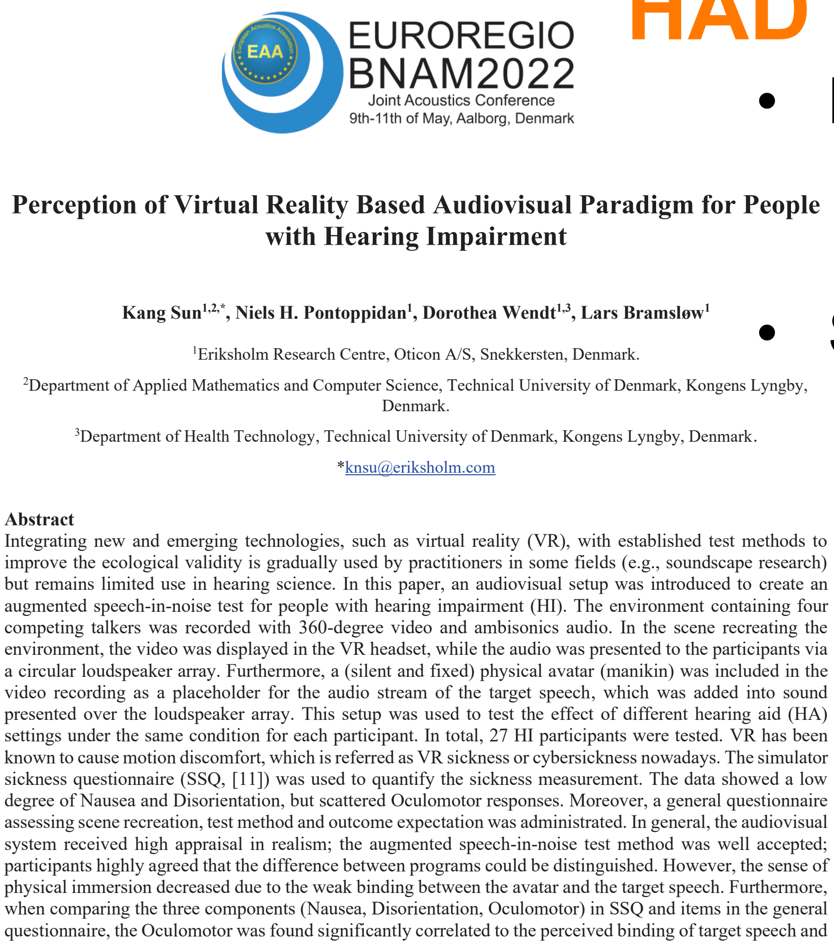

EUROREGIO BNAM2022
Joint Acoustics Conference
9th-11th of May, Aalborg, Denmark

Perception of Virtual Reality Based Audiovisual Paradigm for People with Hearing Impairment

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[1]Eriksholm Research Centre, Oticon A/S, Snekkersten, Denmark.

[2]Department of Applied Mathematics and Computer Science, Technical University of Denmark, Kongens Lyngby, Denmark.

[3]Department of Health Technology, Technical University of Denmark, Kongens Lyngby, Denmark.

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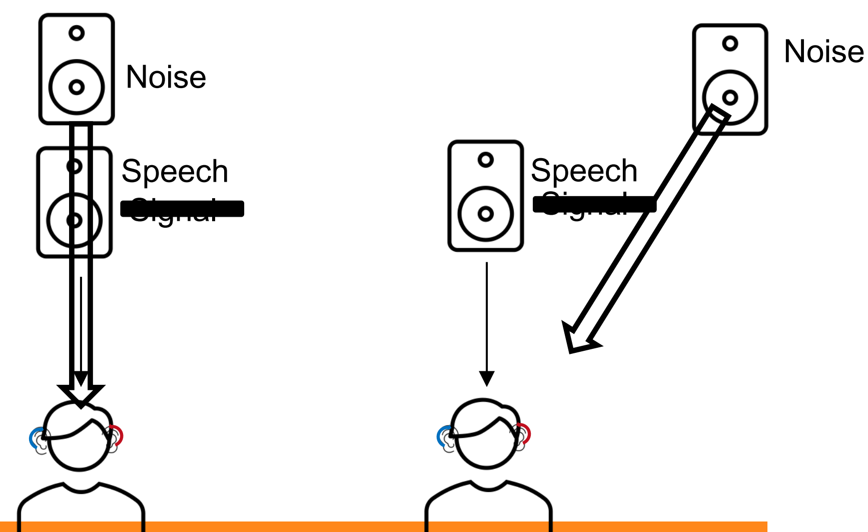
Abstract
Integrating new and emerging technologies, such as virtual reality (VR), with established test methods to improve the ecological validity is gradually used by practitioners in some fields (e.g., soundscape research) but remains limited use in hearing science. In this paper, an audiovisual setup was introduced to create an augmented speech-in-noise test for people with hearing impairment (HI). The environment containing four competing talkers was recorded with 360-degree video and ambisonics audio. In the scene recreating the environment, the video was displayed in the VR headset, while the audio was presented to the participants via a circular loudspeaker array. Furthermore, a (silent and fixed) physical avatar (manikin) was included in the video recording as a placeholder for the audio stream of the target speech, which was added into sound presented over the loudspeaker array. This setup was used to test the effect of different hearing aid (HA) settings under the same condition for each participant. In total, 27 HI participants were tested. VR has been known to cause motion discomfort, which is referred as VR sickness or cybersickness nowadays. The simulator sickness questionnaire (SSQ, [11]) was used to quantify the sickness measurement. The data showed a low degree of Nausea and Disorientation, but scattered Oculomotor responses. Moreover, a general questionnaire assessing scene recreation, test method and outcome expectation was administrated. In general, the audiovisual system received high appraisal in realism; the augmented speech-in-noise test method was well accepted; participants highly agreed that the difference between programs could be distinguished. However, the sense of physical immersion decreased due to the weak binding between the avatar and the target speech. Furthermore, when comparing the three components (Nausea, Disorientation, Oculomotor) in SSQ and items in the general questionnaire, the Oculomotor was found significantly correlated to the perceived binding of target speech and

- Best et al. (2015)
  - Simulated complex sound scenes perhaps better measure of benefit
- Sun et al. (2022)
  - Used a VR setup for SpIN testing



Sun, K., Pontoppidan, N. H., Wendt, D., & Bramsløw, L. (2022). Perception of virtual reality based audiovisual paradigm for people with hearing impairment. *BNAM*, *1*, 95-104.

Off Device: Spatialisation Algorithms in HAD Fitting

JASA ARTICLE

Investigating sound-field reproduction methods as perceived by bilateral hearing aid users and normal-hearing listeners

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ABSTRACT:
A perceptual study was conducted to investigate the perceived accuracy of two sound-field reproduction approaches when experienced by hearing-impaired (HI) and normal-hearing (NH) listeners. The methods under test were traditional signal-independent Ambisonics reproduction and a parametric signal-dependent alternative, which were both rendered at different Ambisonic orders. The experiment was repeated in two different rooms: (1) an anechoic chamber, where the audio was delivered over an array of 44 loudspeakers; (2) an acoustically-treated listening room with a comparable setup, which may be more easily constructed within clinical settings. Ten bilateral hearing aid users, with mild to moderate symmetric hearing loss, wearing their devices, and 15 NH listeners were asked to rate the methods based upon their perceived similarity to simulated reference conditions. In the majority of cases, the results indicate that the parametric reproduction method was rated as being more similar to the reference conditions than the signal-independent alternative. This trend is evident for both groups, although the variation in responses was notably wider for the HI group. Furthermore, generally similar trends were observed between the two listening environments for the parametric method. The signal-independent approach was instead rated as being more similar to the reference in the listening room.
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I. INTRODUCTION

Hearing assistive devices (HADs), such as hearing aids and cochlear implants, are typically custom fitted and calibrated for each individual user. These personalised fittings are usually performed at a clinic, where the surrounding sound sources and the acoustical characteristics of the environment may deviate from the situations the users may later encounter in their day-to-day lives. Indeed, it is common for users of HADs to report dissatisfaction with their devices when experiencing different real world scenarios.[1,2] This may be because established laboratory and clinical tests consider only simplistic sound scenes and static listening conditions,[3–7] despite several studies suggesting that such scenes

There are several existing reproduction methods in the literature, and while there is evidence of the perceptual accuracy of these methods, the accuracy generally relates to normal-hearing (NH) listeners experiencing the reproductions deployed in anechoic chambers.[13–18] There is, on the other hand, relatively little evidence of how these methods compare when deployed in rooms that are acoustically nonideal, especially in terms of how the methods perform when they are experienced by hearing-impaired (HI) listeners.[19–21] This article, therefore, focuses on the investigation of a subset of currently available sound-field reproduction methods that could be deployed within clinical settings, and the main objective is to characterize the perceptual differ-

- Fernandez et al. (2024)

  - Compares parametric methods to HOA

  - free-field and non free-field conditions

  - Both normal hearing participants and hearing aid users

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Top image from *Audio Visual Immersion Lab*. (n.d.). https://electro.dtu.dk/research/research-facilities/audio-visual-immersion-lab

Summary

- Many uses for parametric analysis on device
  - Scene Classification
  - Beamforming
  - DRC
  - Speech Enhancement
  - Pitch shifting
- Also many uses for parametric rendering off device
  - Testing hearing ability
  - Training Hearing Abilities
  - Testing devices
  - Fitting devices