Perceptually Motivated, Intelligent Audio Mixing Approaches for Hearing Loss

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Statement of originality

I, Angeliki Mourgela, confirm that the research included within this thesis is my own work or that where it has been carried out in collaboration with, or supported by others, that this is duly acknowledged below and my contribution indicated. Previously published material is also acknowledged below.

I attest that I have exercised reasonable care to ensure that the work is original, and does not to the best of my knowledge break any UK law, infringe any third party's copyright or other Intellectual Property Right, or contain any confidential material.

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I confirm that this thesis has not been previously submitted for the award of a degree by this or any other university.

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Date: 6th of April 2023

COVID-19 Statement

During the last two years of my studies, my research has been greatly impacted by COVID-19. The pandemic has made it challenging for me to carry out laboratory experiments that require face-to-face interaction with participants, due to various government and university regulations that require social distancing.

My research demands a controlled environment with specific equipment and face-to-face interaction, which made it difficult to achieve optimal results in the face of these restrictions. As a result, participant recruitment and the execution of my studies have been significantly impacted.

In addition, since the majority of individuals with hearing loss were older people, there was a significant reluctance among them to take part in a study that necessitates close contact with the experimenter, due to the higher risk within this demographic. This situation added further challenges to the recruitment process, even after the lifting of restrictions.

Abstract

The growing population of listeners with hearing loss, along with the limitations of current audio enhancement solutions, have created the need for novel approaches that take into consideration the perceptual aspects of hearing loss, while taking advantage of the benefits produced by intelligent audio mixing.

The aim of this thesis is to explore perceptually motivated intelligent approaches to audio mixing for listeners with hearing loss, through the development of a hearing loss simulation and its use as a referencing tool in automatic audio mixing.

To achieve this aim, a real-time hearing loss simulation was designed and tested for its accuracy and effectiveness through the conduction of listening studies with participants with real and simulated hearing loss. The simulation was then used by audio engineering students and professionals during mixing, in order to provide information on the techniques and practices used by engineers to combat the effects of hearing loss while mixing content through the simulation.

The extracted practices were then used to inform the following automatic mixing approaches: a deep learning approach utilising a differentiable digital signal processing architecture, a knowledge-based approach to gain mixing utilising fuzzy logic, a genetic algorithm approach to equalisation and finally a combined system of the fuzzy mixer and genetic equaliser. The outputs of all four systems were analysed, and each approach's strengths and weaknesses were discussed in the thesis. The results of this work present the potential of integrating perceptual information into intelligent audio mixing production for hearing loss, paving the way for further exploration of this approach's capabilities.

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To my supervisor, Joshua Reiss thank you so much for trusting me and allowing me to start my research journey with you. Your valuable guidance and support throughout the years have been instrumental in shaping my academic and personal growth.

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List of abbreviations

AI	Artificial Intelligence
ASL	Adaptive Sentence List
BBC	British Broadcasting Corporation
BSA	British Society of Audiology
DAW	Digital Audio Workstation
dB	Decibel
dBHL	Decibel Hearing Level
DHLS	Differentiable Hearing Loss Simulation
DDSP	Differentiable Digital Signal Processing
DSP	Digital Signal Processing
DMC	Differentiable Mixing Console
EQ	Equaliser
FFT	Fast Fourier Transform
FIR	Finite Impulse Response
GA	Genetic Algorithm
GPU	Graphics Processing Unit
HA	Hearing Aid
HL	Hearing Loss
HLS	Hearing Loss Simulation
HRTF	Head-Related Transfer Function
IIR	Infinite Impulse Response
IoT	Internet of Things
MR-STFT	Multi-Resolution Short-Time Fourier Transform
MSS	Multi-scale Spectrogram
MFCC	Mel-frequency cepstral coefficients
NH	Normal Hearing
OOP	Object-Oriented Programming
RNID	Royal National Institute for the Deaf
SIMHL	Simulated Hearing Loss
SNR	Signal-to-Noise
SPL	Sound Pressure Level
STFT	Short-time Fourier Transform
VST	Virtual Studio Technology

Chapter 1

Introduction

1.1 Topic Area

Accessible broadcast audio for listeners with hearing loss is a topic area that has been gaining a lot of interest in the last few years, both from the research community, as well as the industry. Hearing loss is a widespread global phenomenon, as an estimated 20% of the total population is suffering from some form of hearing loss (1). Projections for a further increase in the future due to the growth of the ageing population, as well as the rise in cases of younger adults exhibiting exposure-related damage due to long-term and improper use of personal listening devices or recreational exposure, are also important factors that make a better understanding of hearing, as well as developing novel effective audio enhancement solutions even more critical (2).

One of the inevitable consequences of auditory ageing is the decline in hearing ability, which can introduce several perceptual difficulties for the affected individual, impacting their day-to-day activities and reducing their overall quality of life (3). Additionally, the population's increasing dependency on audio-visual means of entertainment including television, radio, computers, portable digital devices, etc., further highlights the impact of hearing loss, as well as the need for new ways to provide audiences with accessible audio.

One of the activities most impacted by hearing loss is TV watching and content streaming. More particularly, a very frequent complaint from TV viewers is that of unintelligible dialogue, especially when presented with competing maskers (e.g. multiple speakers, background ambient noise) or background music and sound effects (4). There are several factors affecting the way humans perceive and process speech, especially under challenging conditions and those include their ability to effectively analyse an auditory scene, as well as their familiarity with the speaker or the content (5). More specifically, audiences with hearing loss find it particularly difficult to understand dialogue in complex scenes with multiple sonic elements, such as when it is overlaid with sound effects, ambience or soundtrack (6), the crowd in sporting events, as well as when speech is presented in new, unfamiliar accents (7).

1.2 Project description & Objectives

The rapid evolution of audio technology has inspired many new approaches to audio enhancement for hearing loss, utilizing previously unavailable resources and introducing innovative approaches. This research project proposes the exploration of the perceptual differences between normal hearing (NH) listeners and listeners with mild to moderate hearing loss (HL) and more particularly the investigation of the loss's impact on audio, towards developing an effective automatic audio mixing model for broadcast audio enhancement. Research in this particular area within the wider topic of accessible audio is highly important, as it brings together existing studies on auditory perception and the psychoacoustics of hearing loss, with audio mixing principles and advances in the field of intelligent audio production.

The objectives of this research are:

- To simulate the effects of hearing loss on audio, by examining the auditory system from an audio signal processing perspective, as well as provide an audio input-output functionality, where the processed output signal would approximate the perception of audio by the affected ears.

- To use the simulation in audio production in order to form a deeper understanding of the effects of hearing loss on audio, extract mixing practices as well as utilize this knowledge towards the development of intelligent audio production methods and applications that could improve audio quality by providing effective enhancement.

- To investigate methods for effective enhancement, by utilising artificial intelligence principles in order to restore the HL simulation's output audio quality with minimum error.

- Finally, to utilise the above methods in order to develop and implement intelligent mixing approaches, that provide audio engineers, with the assistive tools that would help them produce two separate mixes (NH and HL enhanced mix), without requiring extra working hours or additional personnel. The diagram

in figure 1.1 presents the original idea proposed for this project.

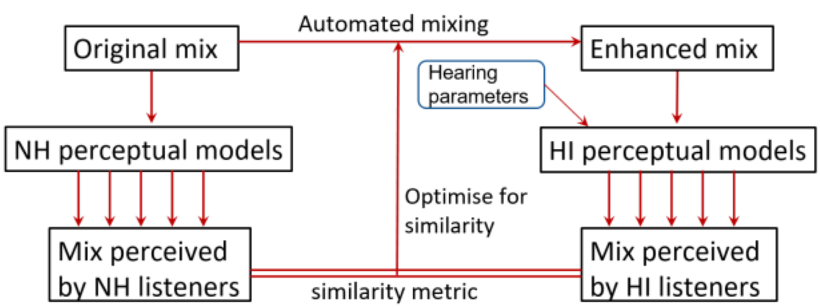

Figure 1.1: Diagram presenting the initial idea proposed by this PhD project.

1.3 Academic Impact

This thesis offers a significant contribution to the field of audio enhancement for individuals with hearing loss. It introduces a novel approach that integrates existing research into a practical format, the audio effects plugin. Additionally, it proposes innovative intelligent audio production approaches that utilise perceptual feedback from the audio effects plugin, in order to enhance audio for listeners with hearing loss.

The work presented in this thesis is important as it lays the groundwork towards bridging the gap between audio engineering and audiology, by providing a practical implementation of audiological and psychoacoustic research in a format accessible to sound engineers. This approach facilitates the application and evaluation of valuable research in real-world scenarios, contributing to the development of more effective audio enhancement techniques for individuals with hearing loss.

Furthermore, while significant progress has been made in understanding psychoacoustics through advancements in hearing aids, current research primarily confines itself to controlled laboratory environments and emphasizes speech comprehension and enhancement. The mixing approaches proposed in this thesis aim to find the balance between controlled laboratory settings and everyday listening, by allowing perceptually informed adjustments of multi-track mixes. Through the exploration of enhanced audio mixes for individuals with hearing loss, psychoacoustic principles can be expanded to encompass the perception of more intricate sonic content, such as broadcast audio and music.

1.4 Motivation

The motivation behind this approach comes from the increasing need for methods of audio enhancement that can effectively improve the overall audio mix quality, this way preserving elements that convey the meaning of the scene in the case of broadcast audio and are important for the listener's experience.

Hearing loss can make activities that rely on verbal communication or audio-reliant entertainment, such as watching television or listening to the radio, very challenging and unpleasant tasks. In TV listening, for instance, hearing loss can result in difficulties in understanding dialogue, particularly when competing sources are introduced, prompting affected individuals to increase their TV set's volume, which can lead to further speech intelligibility issues and distortion (8). Furthermore, communication impairments, including hearing loss, have been associated with decreased social participation and reduced engagement in activities that were once enjoyed, leading to social isolation(9).

A common subject of complaints to broadcasters is that the dialogue is masked by background noise/music or that speech intelligibility is poor. Therefore, there has been a growing need for methods of audio enhancement that can improve the overall audio mix quality. Previous approaches to accessible audio focus mainly on dialogue enhancement or background audio reduction (10), which have been shown to provide an effective solution towards improving intelligibility. However, simplifying the content or highlighting only particular elements in the mix, can negatively impact the overall listening experience thus disrupting the listener's immersion in the content.

Furthermore, simulating hearing loss is crucial, as it can allow researchers as well as sound engineers with normal hearing, to experience and better understand the perceptual effects of hearing loss. The ageing population as well as the evergrowing need for accessible audio is also highlighting the limitations of existing hearing loss models and simulations, as well as the gap between the audio engineering and auditory science communities.

Therefore, this project seeks to explore and utilise the advantages that mulitrack audio mixing can provide towards overall audio enhancement, with as few sacrifices to the listener's immersion and enjoyment as possible.

1.5 Scope of Research

This section outlines the scope of this research in more detail. As stated in the objectives of the research, this thesis focuses on utilising psychoacoustic knowledge of the perceptual manifestations of hearing loss, in order to develop intelligent mixing approaches for the production of enhanced audio. Therefore, this research project is divided into two main focus areas:

- Simulating hearing loss
- Exploring automated intelligent broadcast audio mixing approaches towards audio enhancement for hearing loss

The following sections describe the proposed approaches for each of these two areas in more detail.

1.5.1 Proposed Simulation

The simulation proposed in this research encompasses methods found in previous approaches in the literature, while reproducing four aspects of hearing loss, as well as providing an audio input/audio output capability. The development of the simulation consisted of three stages, the prototype offline simulation, the prototype real-time simulation and finally the virtual studio technology (VST) plugin version.

Similar existing developments utilizing the proposed comprehensive approach for simulation, include the 3D Tune-In Toolkit hearing loss and hearing aid simulation(11), HearLoss-Hearing Loss Demonstrator (12), as well as the Immersive simulation of hearing loss and auditory prostheses (13). Of these approaches, the only one available in a VST format is the 3D Tune-in Simulation (11), while the other two approaches are a commercially available wearable device (13) and a Windows-based standalone simulator application (12).

1.5.2 Proposed Intelligent Audio Mixing Approaches

Three different approaches were implemented and evaluated for the proposed intelligent audio mixing system. The first implementation was that of an equalisation system controlled by a genetic algorithm, the third implementation was that of a knowledge-informed Mamdani-type fuzzy logic system controlling the gain mixing of a multitrack and the final implementation combined the genetic equalisation and fuzzy gain mixing systems.

The genetic algorithm has been widely used in audio applications such as equalisation(14), synthesis (15), classification and audio feature extraction (16), providing a great optimisation tool.

Fuzzy logic architectures have been used in audio production for genre-based audio equalisation (17), speech enhancement (18), as well as proposed systems for intelligent mixing (19). Fuzzy logic is an attractive approach for intelligent audio mixing, as it provides a useful tool towards implementing knowledge-based practices.

1.6 Thesis Structure

Chapter 2 Describes the background and state of the art in the topic area. Previous approaches for hearing loss simulation are discussed and compared with the proposed simulation. Approaches in the field of intelligent audio mixing are also discussed along with their strengths and weaknesses.

Chapter 3 Discusses the design process of the hearing loss simulation from the prototype to the final format. A detailed analysis of the algorithmic implementation of the simulation is presented, along with important features modified throughout the design process.

Chapter 4 This chapter presents evaluation process of the simulation by analyzing two listening studies, their design, methodology and results as well as the conclusions drawn from the data analysis.

Chapter 5 Demonstrates the use of the simulation in the mixing process, as well as the extraction of processing parameters and standard procedures that will set the basis upon which the automated mixing approaches will be developed and tested in the following chapter. Parameters are determined through the recruitment of Tonmeister students from Surrey University as well as professional mixing engineers.

Chapter 6 Presents the proposed approaches explored for the automated mixing system. It discusses the design, implementation and evaluation of the four approaches explored in this thesis. The implementation of each approach is discussed in detail along with its strengths and weaknesses.

Chapter 7 Discusses the main findings and conclusions of the thesis, by revisiting the results presented in the previous chapters. It also describes the key biases and limitations of the methodology section in the thesis. Lastly, it highlights the main contributions of the thesis to the field as well as the future work that could further improve the research.

1.7 Associated Publications & Awards

Portions of the work detailed in this thesis, as well as additional projects, have been presented in national and international scholarly publications and conferences, as follows:

- Chapter 3: Section 3.1.1 on the design of the offline prototype was published as an e-brief paper and was presented at the 147th Audio Engineering Society's convention in New York on October 2019. (20)
- Chapter 3: Section 3.1.2 on the design of the real-time VST plugin was published as an e-brief paper and was presented at the 149th Audio Engineering Society's convention in New York on October 2020.(21).
- Chapter 3: Section 3.1.2 on the evaluation of the hearing loss simulation with participants with real and simulated hearing loss is currently being reviewed for publication.
- Chapter 6 The work found in section 6.2.1 of this chapter was presented at the Audio Diversity Network's first workshop at the University of Leeds in September 2021.[1]
- In collaboration with a neurologist in Greece, a case-control study on the investigation of frequency-specific loudness discomfort levels in listeners with migraine was published in Ear & Hearing journal. The project involved collaboration with University College Dublin in order to create an online testing platform that would measure the threshold of hearing and threshold of mild discomfort across 13 frequencies (22)

1.7.1 Awards and Highlights

- The hearing loss simulation VST plugin received the first place award at the Audio Engineering Society's MATLAB Student Plugin Competition in October 2020.

[1]https://auraldiversity.org/workshop1.html

- The hearing loss simulation VST plugin was a finalist for the Trailblazer Award for exceptional work in impact generated by Early Career Researchers and PhD students at Queen Mary University's Engagement and Impact Awards.

- The hearing loss simulation plugin was used in the production of a special episode of the British Broadcasting Corporation (BBC) series "Casualty", where it was used towards simulating a character's hearing loss and immersing the audience into the character's world. [2]

[2] https://www.bbc.co.uk/rd/blog/2020-07-casualty-jade-hearing-loss-aid-binaural

Chapter 2

Background

2.1 Simulating Hearing Loss

Various researchers in the field of psychoacoustics have explored and documented the perceptual effects of hearing loss, as well as the ways that they can impact audio quality and intelligibility in complex stimuli and rapidly changing sounds.

The first stage of the thesis is dedicated to the investigation of the perceptual aspects of hearing loss, as well as the design and implementation of a simulation reproducing these effects on audio in real-time. This was realised by combining existing psychoacoustic findings and previous simulation approaches towards developing a hearing loss simulation, in order to create an audio production-focused application that provides a real-time audio input-audio output functionality.

Modelling the perceptual characteristics of hearing loss has been made possible in the field of audiology and psychoacoustics both by obtaining physical measurements as well as conducting listening tests. The resulting perceptual models have been used to simulate hearing loss for normal hearing listeners, this way enabling researchers to identify the affected auditory functions as well as their impact on the audio quality, while facilitating the exploration of new approaches for effective audio enhancement.

Previous simulation methods include those based on the presentation of preprocessed stimuli, the use of additive noise to simulate threshold shifts, or those utilising a combination of spectral, dynamic, and temporal processing to simulate multiple aspects of hearing loss.

The four main aspects of hearing loss most commonly simulated in previous approaches in the literature are:

- ***Threshold Elevation***, a phenomenon commonly observed in individuals with presbycusis, a type of hearing loss that is associated with ageing. This phenomenon is characterized by an increase in the hearing threshold, which means that a higher sound level is required to detect sounds at a particular frequency (23). Threshold elevation usually affects higher frequencies, such as those in the range of 2-8 kHz. As a result, individuals with presbycusis may perceive sounds in these frequencies as attenuated. This can lead to difficulty in understanding speech, especially in noisy environments, and can also affect the perception of music and other complex sounds.
- ***Loss of Dynamic Range of Hearing*** a phenomenon attributed to a loss in the nonlinear function of the basilar membrane. The healthy basilar membrane presents a compressive nonlinearity, which enables it to respond more strongly to low-level sounds than to high-level sounds. This allows the membrane to amplify the softer sounds present in complex stimuli, such as speech and music, which are important for speech recognition and music perception. An ear affected by hearing loss can exhibit a rapid growth in the perceived loudness, starting at the elevated threshold of hearing and up to the threshold of total recruitment, which marks the level at which loudness is perceived equal to a normal hearing listener. This can reduce the dynamic range of hearing, negatively impacting everyday functions such as speech recognition in noisy environments (24).
- ***Loss of Spectral Resolution*** a phenomenon attributed to the widening of the auditory filters. Listeners with hearing loss present a reduced sharpness in their auditory filters' function, which introduces the effect of perceived spectral smearing, making audio content indistinguishable, especially in the presence of a competing masker. In healthy listeners, auditory filters

in the basilar membrane are sharply tuned, which enables the listener to distinguish complex sounds and correctly identify the frequencies of the stimuli presented to them. When spectral smearing occurs, spectral resolution is reduced which can affect the listener's ability to distinguish between different frequencies, thus making the audio appear "muddy" or distorted (25).

- ***Loss of Temporal Resolution***, a phenomenon attributed to neural asynchrony. Temporal resolution relates to our ability to track alterations in the time-based arrangements of sounds. A loss in temporal resolution can introduce difficulty in distinguishing sounds that are presented in rapid succession to the listener, while it can further impact their ability to distinguish between different sounds, as well as reduce speech intelligibility (26).

Various techniques for simulating hearing loss have been implemented and discussed in the literature. In the following section, a summary of the key approaches used for simulating hearing loss is provided:

Baer and Moore's (27) simulation of reduced frequency selectivity, employed an overlap-and-add method. More specifically, the short-term spectrum was computed for each analysis/synthesis frame of the audio, using an FFT and a Hamming window. A spectral smearing function was then applied, and the smeared spectrum was transformed back into the time domain using an inverse FFT. Finally, the waveforms from overlapping analysis frames were combined to generate the ultimate output. The results from the speech in noise evaluation study of this simulation technique present significant impact of smearing on intelligibility when noise is also present, however there was no significant impact when the speech was presented in quiet. One of the limitations of this approach is the fact that it uses offline processing to pre-process stimuli, so it needs adaptations to work in a real-time format, as well as the fact that it has only been tested with normal hearing listeners.

The 3D Tune-In hearing loss simulation (11) reproduces different types of hearing loss, based on individual audiograms. The simulation is available in a web-based format, a standalone application as well as a VST plugin and replicates the effect of non-linear audiogram attenuation with the option of using a Butterworth or gammatone filterbank, spectral smearing with the choice of utilising one of two approaches (27; 28), as well as temporal jitter. While this implementation is offered in a real-time, VST plugin format, its effectiveness and accuracy were not validated through evaluation with listeners. Furthermore, while its user interface offers a lot of options for customisation and different simulation models and methods, this could be counterintuitive in a non-research scenario such as in audio production, where it will most likely be used by sound engineers without prior knowledge of such models.

Braida and Lum's (29) real-time hearing loss simulator employs a filterbank approach to divide the input signal into frequency bands, which are then subjected to level-dependent attenuation. In order to calculate the necessary gain to be applied, the system measures the instantaneous level in each frequency band and compares it to the detection threshold of the impaired ear and the threshold of recruitment. The simulation was able to successfully apply level-dependent attenuation to the test stimuli in real-time based on the pre-defined static characteristics of the dynamic expansion mimicking the phenomenon of loudness recruitment. However, the study only hypothesizes the effects of the simulation on normal-hearing listeners, as listening experiments were not performed. Furthermore, this development does not address the effects of spectral smearing and temporal jitter that also occur with hearing loss, as well as their interaction with loudness recruitment.

Duchnowski and Zurek's (30) simulation of sensorineural hearing loss simulates two main characteristics of sensorineural hearing loss: elevated thresholds and loudness recruitment. The model works by dividing the input signal into 14 frequency bands, and the level of each band is compared to the impaired threshold at the centre frequency of that band. Signals below the threshold are attenuated to near zero, while signals within a certain range above the threshold are attenuated according to an expansive mapping of the input level. Signals above a certain threshold are passed without attenuation. While the technique was successful in simulating consonant reception in listeners with hearing loss during the pilot studies, the evaluation study showed that the implementation would be difficult to transform into a wearable format, thus its use was limited to laboratories and classrooms.

Gagné and Erber's (31) model of sensorineural hearing loss works by dividing the input signal into two independent frequency channels and using center-clipping devices to set the levels of the simulated thresholds of detection in each channel. By adjusting the level of the signal at the input and the amount of centre clipping, the system can simulate different audiometric configurations and degrees of hearing loss. The input-output intensity functions obtained at various hearing loss threshold settings resemble the loudness recruitment curves observed among listeners with sensorineural hearing loss. Through an evaluation study including audiometric tests, phonetically balanced word recognition tests, as well as vowel and consonant identification tests, the simulation yielded satisfactory results. However, it is noted that the selection of tests does not examine loudness perception through psychoacoustic measurements, and focuses on validating the audiometric thresholds as well as the perceptual effects on word recognition.

Graf's (28) simulation of the effects of sensorineural hearing loss simulates threshold elevation, loss of dynamic range, as well as spectral smearing. The simulation works by initially dividing the input signal into 14 frequency bands. The power in each band at a specific time is calculated, and a gain is determined for each band based on the power. The gain is a large attenuation for weak input sounds, and no attenuation is applied for levels above a certain range. The power spectrum is then smeared by convolving it with a Gaussian-like function. The evaluation study for this implementation included several tests to investigate each of the individual modules of the simulation, as well as consonant recognition. The results of the study report some discrepancies when compared to the literature results. One of the limitations of the study is that it only recruited normal-hearing participants and compared their results with literature results, which could have impacted the results due to the experiments taking place in different conditions.

Additionally, the simulation did not include a simulation of the temporal effects of hearing loss which could further impact the results.

Huckvale's (12) HearLoss hearing loss demonstrator is a Windows-only, standalone application that combines three audio files: speech, music and noise into its memory, and then processes the combined audio in the frequency domain to eliminate frequency components above a certain limit set by a frequency range slider, and to spread the energy across the frequency range according to the setting of the frequency selectivity slider. The technique used for spectral smearing is derived from the method explained by Baer and Moore (27). The simulation can be used for demonstration purposes, however, its use is limited as it cannot be used in real-time. Furthermore, the absence of an evaluation study for this implementation leaves us with limited insights into its performance and potential areas for improvement.

Moore and Glasberg's simulation(23) works by dividing the input signal into 13 separate frequency bands. In each band, the envelope of the signal is expanded in a way that would generate loudness sensations in a normal ear that were comparable to the sensations produced in an impaired ear with recruitment. Following this, the individual bands are recombined to create the final output signal. The main objective of this approach is to replicate the effect of hearing loss with recruitment in individuals with impaired hearing. By processing the envelope in each frequency band, the simulation aimed to mimic the characteristic features of loudness perception observed in such individuals. The simulation was evaluated through a listening study that included speech in quiet as well as speech in the presence of a single talker in various signal-to-noise ratios and hearing loss conditions. Some of the limitations of this development include the fact that it could only be used offline in order to pre-process stimuli for the tests and not in real-time, as well as the fact that its performance was only evaluated with normal hearing participants.

Zurek and Desloge's (13) immersive simulation of hearing loss and auditory prostheses creates an immersive experience for the user by modifying their detection thresholds for ambient sounds to a specified extent. To achieve this, the simulator employs a combination of passive attenuation, which is accomplished by using muff-type hearing protectors, and additive masking noise that is introduced through earphones placed within the muff. The acoustic signals that are captured by microphones located near each ear are then processed through a series of bandpass automatic gain control channels, which further modify the signals' characteristics. The processed signals are finally delivered to the earphones to complete the simulation of frequency-dependent hearing loss and loudness recruitment. The authors claim that auditory threshold and speech in noise data comparing the simulation against actual losses, along with subjective evaluations from listeners validate the simulation's performance in a variety of hearing losses. However, the implementation presents several limitations to its efficiency as well as its evaluation. One such limitation is its reliance in additional equipment(ear muffs) as well as additional disruptive signals (additive noise), which can further impact the perception of the listener this way affecting the accuracy of results. Furthermore, the implementation was only tested with a speech-in-noise test which cannot provide sufficient data on its effectiveness and accuracy, particularly in simulating threshold shifts and loudness recruitment.

Xu et al's (32) simulation of temporal fine structure distortion applied random phase shifts on audio signals while keeping the magnitude intact. More specifically, speech sentences were analyzed using the Short-Time Fourier Transform (STFT) to separate them into contiguous frequency bands. The resulting complex numbers were then converted into a combination of magnitude and phase information. To mimic cochlear implant coding strategies, the number of frequency bands was reduced from 64 to 6, using the n-of-m method. The 6 frequency bands were then used to reconstruct speech sentences with distorted phase values with random shifts. The simulation's effect was evaluated through a speech-in-quiet and noise test using both intact and temporally distorted sentences with normal-hearing listeners. The temporally distorted sentences were also used to measure the neural phase-locking activity in guinea pigs. Though the study was successful in demonstrating the importance of temporal fine structure for speech intelligibility, the simulation utilised was only used offline and in order to pre-process stimuli.

Additionally, the selection of tests did not examine performance in gaps-in noise and only tested the simulation with normal-hearing listeners.

Nagae et al's (33) simulation is based on the compressive gammachirp filter and utilizes level-dependent filter shapes and cochlear compression functions derived from notched-noise masking data. By reversing the level dependence of the compressive gammachirp filter, the researchers created an inverse compression used to resynthesize sounds that cancel the compression applied by the auditory system of normal-hearing listeners.One of the limitations of this development is that even with the frame-based processing it is still not real-time, which is limiting to its use-ability. Additionally, the study does not mention any evaluation procedure through listening tests, which could provide valuable insights into its effectiveness and accuracy.

A list of the previously mentioned simulation approaches is presented in Table 2.1. The table includes the aspect(s) of hearing loss simulated by each implementation along with the method used in each of the publications listed.

Reference	Method	Aspect Simulated
(33)	Compressive Gammachirp Filter	Loss of Compression, Threshold Elevation
(34), (13)(30)	Automatic Gain Control	Loss of Compression, Threshold Elevation
(29), (11), (35),(36),(28)	Multiband Dynamic Expansion	Loss of Compression
(36),(13)	Additive Threshold Noise	Threshold Elevation
(12),(11),(28),(37)	Gaussian Bell Convolution	Spectral Smearing
(23)	Envelope Expansion	Loss of Compression
(38)	Low-passed Noise Multiplication	Spectral Smearing
(37)	Sample Shift	Temporal Jitter
(32)	Random phase shifts	Loss of Temporal Resolution
(31)	Centre clipping	Loss of Compression, Threshold Elevation
(27)	Overlap-and-add	Spectral Smearing

Table 2.1: Previous simulation approaches for the four aspects of hearing loss.

The approaches presented in table 2.1 have replicated one or multiple aspects of hearing loss however, they present several limitations in their accuracy, efficiency and practicality. Such limitations include the inability to offer audio input/output functionality, a limitation typically observed in studies that simulate only one aspect of hearing loss, or model physical responses, thus receiving and outputting values that correspond to physical and neural attributes such as stapes velocity, basilar membrane displacement or nerve firing rates.

Though useful in clinical and psychoacoustic studies, this output format can not be easily integrated into audio production. Another limitation observed in some of the previous approaches is the inability to process and playback audio in real-time, usually observed in studies that are conducted using pre-processed stimuli or employ high-complexity algorithms (33; 34; 35; 12; 23; 28; 38; 37; 32). Additionally, many of the aforementioned approaches are developed and tested using simple stimuli such as noise signals and pure tones (39) and are evaluated under controlled listening and playback conditions, which can make them inapplicable to complex sounds and various listening environments and equipment and therefore real-life conditions.

Another significant limitation common to most of the hearing loss simulation approaches presented above, is their exclusive testing and evaluation with participants with normal hearing. This poses a challenge in accurately validating their effectiveness, particularly when relying solely on psychoacoustic measurement data of participants with real hearing losses from the literature, which have been obtained under different conditions.

Finally, the majority of the previous approaches with the exception of (11), lack the ability to be easily implemented in a digital audio workstation for use in audio production, which limits their access and use by audio mixing professionals.

2.2 Enhancing Audio for Hearing Loss

In order to counteract the effects of hearing loss defined in the previous section, various research and industry endeavours have been made towards exploring effective audio enhancement methods.

One of the most common ways of enhancing audio for hearing loss is by using hearing aids (HA). With appropriate fitting, hearing aids can provide an effective solution for people with hearing loss. In recent years HAs have evolved to provide better processing, and added features such as Bluetooth compatibility and related apps for user adjustments and easier integration with internet of things (IoT) devices. Furthermore, the exploration of incorporating artificial intelligence algorithms to facilitate the fitting process and introduce user preference learning and better adaptation to various listening environments, can further enrich their ability to improve the user's everyday life (40).

However, HAs still present limitations in their efficiency (41). The most basic HA models' limitations include their inability to restore intelligibility, as well as perform complex digital signal processing, which is affected by their battery requirements, and the necessity for minimal latency. Furthermore, they usually offer limited options for customisability upon their fitting, as well as limited connectivity with smart devices (41). Most high-end HA models can offer additional options for customisability and connectivity, as well as an improved frequency response and binaural processing, however, they can still present limitations in their ability to restore intelligibility, especially in sounds with rapidly changing acoustic properties (such as television sounds), or when used in noisy environments (8).

Another important limitation to HAs effectiveness is that even though it is suggested that in the United Kingdom 6.7 million people could benefit from using a hearing aid, only 2 million people use them, leaving approximately 4.7 million people without access to enhanced audio or relying on alternative enhancements (subtitles, assistive devices, etc.) (42). For the more high-end models, the cost is also an important deterring factor.

Besides HAs, recent studies have also explored alternative ways of providing accessible enhanced audio to all audiences, including HA and non-HA users. In an exploration of the preferred sound balances for hard-of-hearing individuals, Mathers (43) investigated the preferences of listeners when given a set of mixes with different ratios between different sonic elements such as speech, sound effects, and music. The study concluded that speech intelligibility was not as highly dependent on the background level for normal hearing listeners and that listeners with mild/moderate losses would benefit more from lip-reading.

However, the study mentions that listeners with severe and profound losses could benefit from further reductions in the background elements when assessing intelligibility improvements. A limitation of this study is that it focused mainly on the sound level balances between the sonic elements, therefore neglecting important factors such as masking due to spectral similarities between elements, or dynamic fluctuations over time, as well as the fact that it does not account for scenes that do not feature visible speakers or the possibility of audiences with visual impairments that cannot access lip-reading information.

Another study by Shirley et al. (44; 45) explored the potential of utilizing spatial separation technology and multiple speaker configurations, to assess listener preference by analysing dialogue clarity, enjoyment, and overall sound quality reports from participants with varying degrees of hearing loss as well as normal hearing participants. The participants were presented with a series of video clips with a Dolby Digital 5.1 encoded soundtrack, using the following configurations:

- centre channel for dialogue plus the left and right channels for background at standard relative levels
- centre channel plus left and right channel at -3dB
- centre channel plus left and right channel at -6dB
- centre channel only

Preference analysis among participants with hearing loss showed that reducing the level of the left and right background containing channels could improve intelligibility and enjoyability while using the centre speaker only showed the highest ratings for improved dialogue clarity. Limitations of this approach include the potential lack of compatibility with the user's reproduction system, as well as the necessity for additional modifications at the stage of production. Additionally, the attenuation or removal of content could result in a poorer experience and a lack of viewer immersion.

Focusing on providing the user with control of their mix's complexity at the user end stage, another study by Ward et al. (46) explored the potential of using an object–based approach to enhance broadcast audio. More specifically, the study suggests providing the user with control over the level balances between the audio elements in the mix, in order for them to be able to adjust the complexity of their auditory scene to a preferred and comfortable level. This was implemented by separating the mix into audio objects, each weighted with metadata corresponding to their narrative importance in the scene. By controlling a single button the user is able to then adjust the complexity of the overall audio. Early results from this study using the BBC taster platform, showed that 73% of the participants surveyed after the use of this development reported an improvement in enjoyment and the general understanding of the content.

Limitations of this study include the need for additional modifications in the audio content in order to be compatible with an object-based framework as well as the potential incompatibility of this technology with commercially available devices. Another consideration is that the proposed solution will provide a simplified version of the mix, affecting the overall balance of the mix and potentially impacting the viewer's experience.

2.3 Intelligent Audio Mixing

Audio mixing can be described as the process of balancing, treating, and combining multitrack material into a multichannel format, whether it was recorded, sampled, or synthesised (47). Mixing is a critical process in audio production since it is the stage where corrective and creative processing takes place, this way bringing the raw recorded material closer to its final form and preparing it for distribution. Several processing stages take place during mixing, which can include panning and levelling, corrective and creative equalisation, artificial reverberation as well as dynamic range compression and more.

Over the last decade, autonomous audio mixing models have become very popular both in research and the industry, with assistive mixing technology slowly becoming a new standard in the field of audio production (48). Through exploring the potential applications of artificial intelligence in audio production, various new approaches to mixing have emerged, including autonomous levelling, panning, equalisation, compression, reverb, as well as a combination of multiple processing methods. Some of the studies exploring autonomous mixing systems include and are not limited to:

- A 2008 study by Kolasinski (49) using genetic algorithm-based models for timbral classification and optimisation. The study utilises Euclidean distance information between spectral histograms, in order to calculate distances between a target and a mix while using genetic optimisation for coefficient calculation. The approach produced satisfactory results in multitrack mixing with a target. One of the limitations included a degradation in the algorithm's performance with increasing tracks.

- A 2015 study by Ma et al.(50), proposes a multiple linear regression approach to modelling experiment-derived ratio and threshold adjustments from audio engineers, toward implementing an intelligent dynamic compressor. Results from the evaluation study show that the proposed system's mixes were able to compete with and even outperform mixes made by semi-professionals.

- A 2013 study by De Man et al. (51), presents a knowledge-engineered autonomous mixing system, that uses semantic mixing rules derived from sound mixing textbooks. The rules are based on low-level signal features such as crest factor, loudness and hysteresis as well as instrument tags.

- A 2018 study by Ramirez et al. (52) exploring the use of deep learning for automatic, matched equalisation. More specifically the system proposed in this study uses an end-to-end learning approach in order to provide an approximation of a target equalisation, without the need to calculate the transfer function.

- A 2019 study by Moffat et al.(53) explores the use of a random forest approach to gain mixing for drums. More particularly the study proposes a reverse engineering approach for gain parameters, combined with feature vectors for each audio track.

- A 2020 dissertation by Fermin (19), explores the use of fuzzy logic, to create an automatic mixing system. The system is comprised of three modules, an equalisation module, a volume module and a panning module. The proposed approach was tested with a set of multitrack mixes with satisfactory results.

- A 2021 study by Steinmetz et al. (54) a multitrack mixing system that was based on a differentiable mixing console of neural audio effects. More specifically, the proposed approach provides solutions to issues such as limited training data and input source capability while producing human-readable mixing parameters and therefore the option of adjustability for the user.

Applications of machine learning-assisted audio production can also be found in the industry in the form of intelligent audio plugins (Izotope - Neutron[1], Sonible smart:eQ[2], Soundtheory - Gullfoss[3], etc.) or integrated directly into hardware (Midas Heritage D[4]), offering accessible and cost-efficient assistance to both musicians and non-expert users looking for improved sound quality, as well as professionals looking to save time by automating menial tasks while minimising manual intervention.

Though aspects of conventional audio mixing as an enhancement method for listeners with hearing loss have been proposed in the literature discussed in the previous section, the potential benefits of using artificial intelligence in order to automate the mixing process for this purpose still remain relatively unexplored. Applications of autonomous mixing for hearing loss enhancement could provide

[1] https://www.izotope.com/en/products/neutron.html
[2] https://www.sonible.com/smarteq3
[3] https://www.soundtheory.com/home
[4] https://www.midasconsoles.com/product.html?modelCode=P0BHN

time and cost-effective solutions since they require minimal manual intervention, this way facilitating the production of additional accessible content.

2.4 Chapter Summary

The chapter began by providing an overview of previous research on hearing loss simulation approaches. The main aspects of hearing loss were discussed as well as the impact they have on the listener's ability to perceive sound. Upon that, different simulation methods proposed in the literature were reviewed and their limitations were discussed.

Next, the chapter examined approaches to audio enhancement for listeners with hearing loss. Traditional and commercially available methods, such as hearing aids were discussed along with various novel research approaches to audio enhancement that utilise new technologies such as object-based audio.

Finally, the chapter explored recent advancements in intelligent audio production. Artificial intelligence algorithms were described and their use towards automating various aspects of the audio production process was presented. Several approaches that have been proposed in the literature were reviewed, such as deep learning, differentiable DSP, and random forest optimization.

The chapter also presented available tools that utilise intelligent audio production technology in the industry, including assistive software and hardware, this way highlighting the potential of AI-assisted audio production to provide time and cost-effective solutions for creating accessible content for listeners with hearing loss.

Chapter 3

Hearing Loss Simulation Design

3.1 Developing the Hearing Loss Simulation

3.1.1 Offline Prototype

The first stage of the PhD project consisted of the development of a hearing loss simulation prototype, which was implemented and evaluated using MATLAB. A diagram of the prototype simulation is given in Figure 3.1.

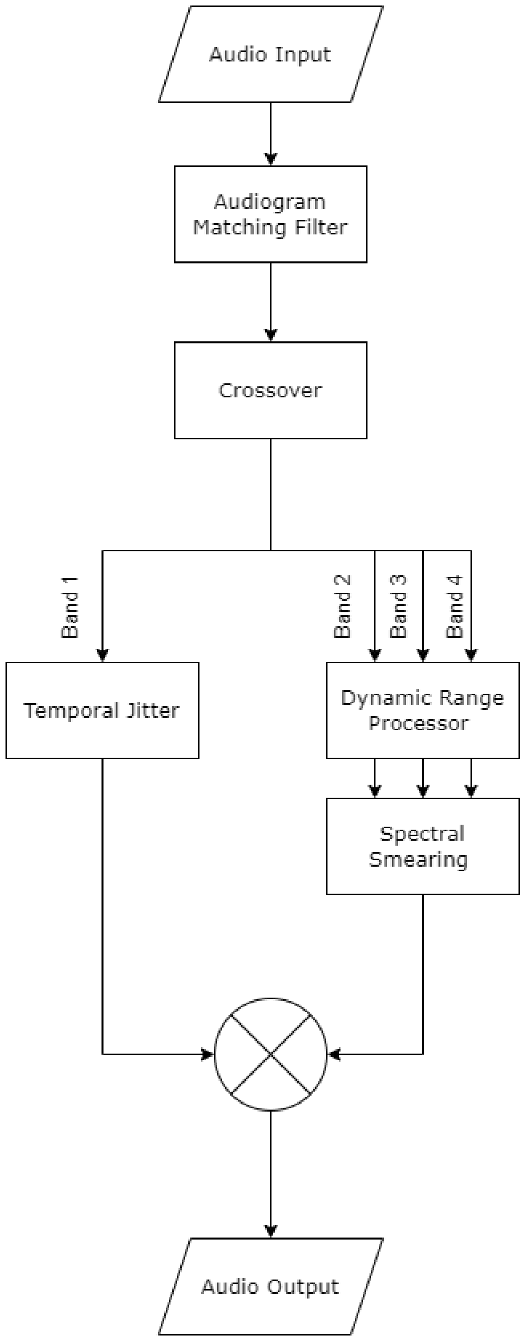

Figure 3.1: Block diagram of the prototype hearing loss simulation system.

The prototype simulation incorporated a combination of simulation approaches found in literature, in order to replicate the four perceptual aspects of hearing loss described in Chapter 2.

More specifically, the prototype development consisted of four different modules connected in series, each of which aimed to reproduce a different aspect of hearing loss. The first module of the prototype introduced an audiogram-matching filter, that used a frequency-sampling-based finite impulse response (FIR) filter (55), in order to mimic the effect of audiogram-specific attenuation corresponding to threshold elevations. Upon exiting the audiogram-matching filter, the signal was separated into four frequency bands through the use of an audio crossover filter with tuneable cut-off frequencies.

The three higher frequency-containing bands were then sent to an upwards expander. The purpose of this processor was to imitate the effects of loudness recruitment, which is a phenomenon that occurs when there is a loss of dynamic range (24). This phenomenon is commonly seen in individuals with damaged outer hair cells. The outcome of using this processor is a perception of a more linear and steep increase in loudness as the sound level rises above the hearing threshold. This is in contrast to the compressive nonlinear growth that is typically observed in individuals with normal hearing (24). The processor uses an upward expansion method based on the dynamic range expander presented in (56) to accomplish this. When the signal level exceeds the hearing threshold but is below the total recruitment threshold, the processor amplifies it. When the signal level reaches the total recruitment threshold, the processor returns to an in=out behaviour. Figure 3.2 presents a ramp function before and after the upwards expansion module.

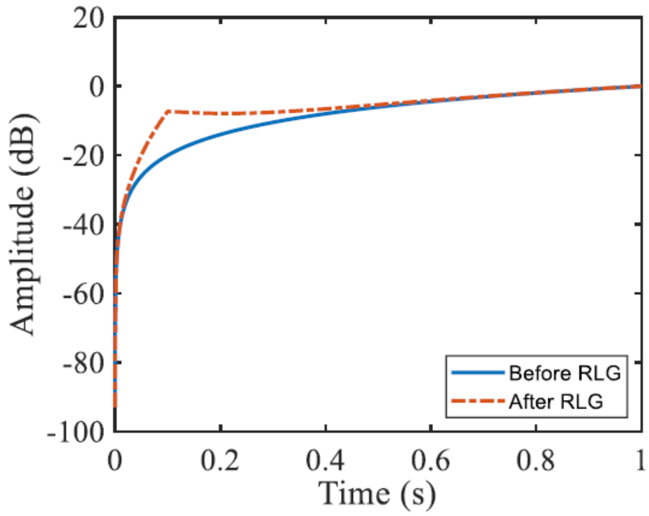

Figure 3.2: Amplitude over time plot in logarithmic scale, before (blue) and after (red) the upwards expansion applied rapid loudness growth (ramp signal).

After exiting the expansion module, the three high-frequency bands were then sent to a spectral smearing module which aimed at replicating the effect of loss of spectral resolution. The module generated 3 separate low-passed noise signals with different cut-off frequencies, which were then multiplied by the expanded signals containing the three high-frequency bands in the time domain. This resulted in a convolution of the low-passed noise signal with the high-frequency bands in the frequency domain, which produced a smeared representation of the frequency content. Figure 3.3 presents a 2 kHz pure tone before and after spectral smearing processing.

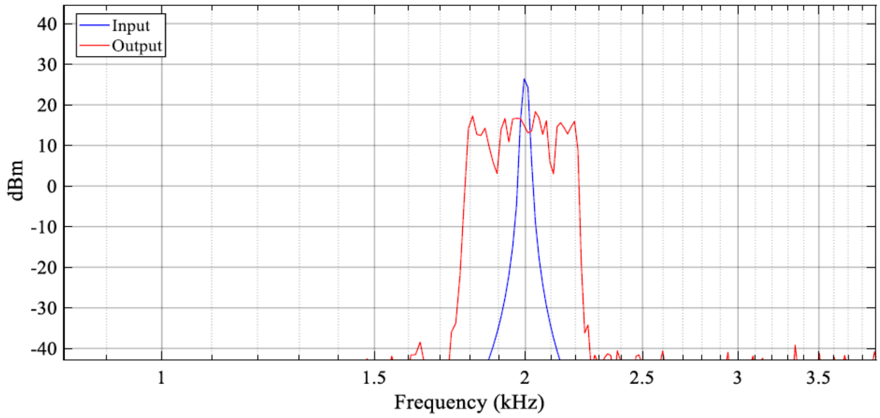

Figure 3.3: Input (before spectral smearing) and output (after spectral smearing) spectrum of 2kHz pure tone.

Each of the three bands was multiplied by its corresponding low passed noise signal and the cut-off frequencies were tuneable to provide the user with a choice of severity for the smearing (higher cut-off frequencies generated more evident smearing). At this stage the first band containing the low-frequency portion of the signal was sent to a temporal jitter unit, to simulate the loss of temporal resolution. To reproduce this aspect, a chorus-type processor is employed with a maximum delay value of 0.25 ms. This processing technique was chosen as a simple way to introduce small random delays in the signal, which has been shown to significantly reduce intelligibility (37). Figure 3.4 presents the amplitude over time for a triangular pulse signal before and after the processor.

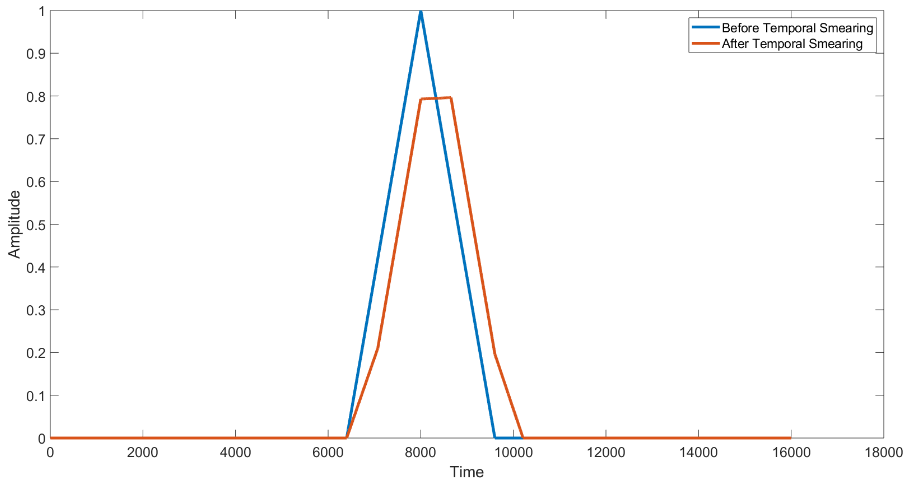

Figure 3.4: Input (before temporal jitter) and output (after temporal jitter) amplitude of a triangular pulse plotted over time.

Finally, all four bands were summed together to form the output of the simulation. Upon completion of the offline prototype version of the simulation, the code was adjusted in order to support real-time audio processing, thus making it easier to incorporate into practice. To achieve this in MATLAB, the code was modified to an Object-Oriented Programming (OOP) format. The streaming functionality of the model was then tested through MATLAB's inbuilt audio-testbench benchmarking application, to ensure there were no excessive delays or discontinuities in the audio streaming. This stage marked the basis for the formation of the real-time VST plugin.

3.1.2 Real-Time VST Plugin

The prototype simulation presented in the previous section was the first approach towards achieving real-time capability, however, it presented certain limitations in its response and efficiency, while it was also still limited within the MATLAB environment. To counteract these limitations the final version of the simulation was developed which moved the implementation away from MATLAB and configured in an audio effects plugin format for both Windows and macOS systems.

This version of the simulation is a stereo audio effects plugin, designed to provide an easy way for engineers to assess their mixes in real-time, using a VST-compatible digital audio workstation. The user interface of the simulation was designed using a compact and intuitive approach, offering bypass and mute options for each ear, as well as customisation options for intensity and feature selection for the attenuation and suprathreshold effects.

The user interface of the hearing loss simulation plugin can be seen in Figure 3.5.

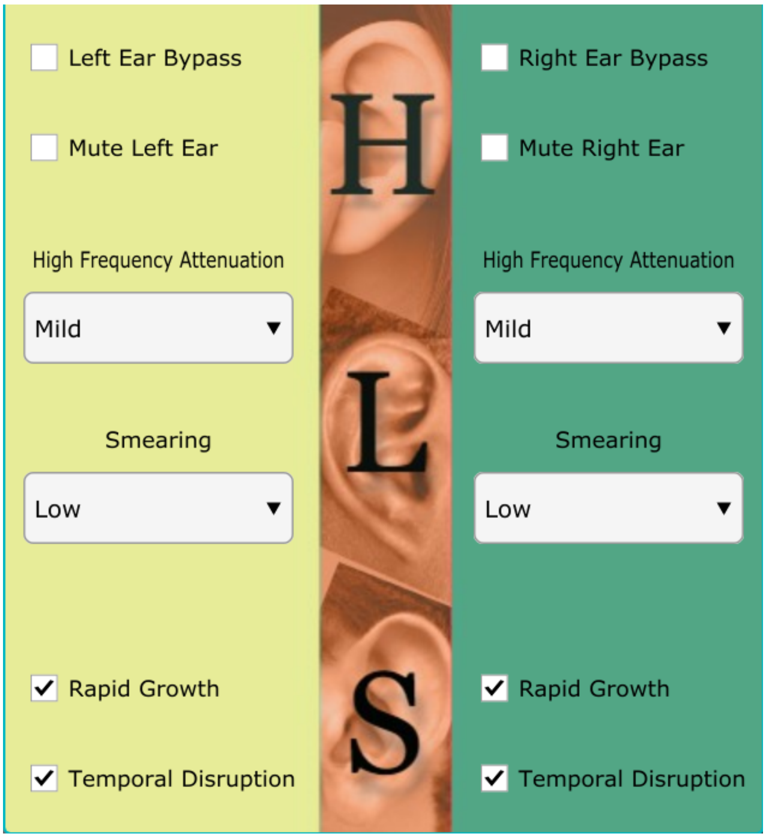

Figure 3.5: Hearing loss simulation user interface.

Figure 3.6 presents the flow diagram depicting the processing inside the real-time version of the simulation. The diagram presents the separation of the audio signal in its high and low-frequency components using the gammatone filterbank, as well as the specific processing applied to each frequency band.

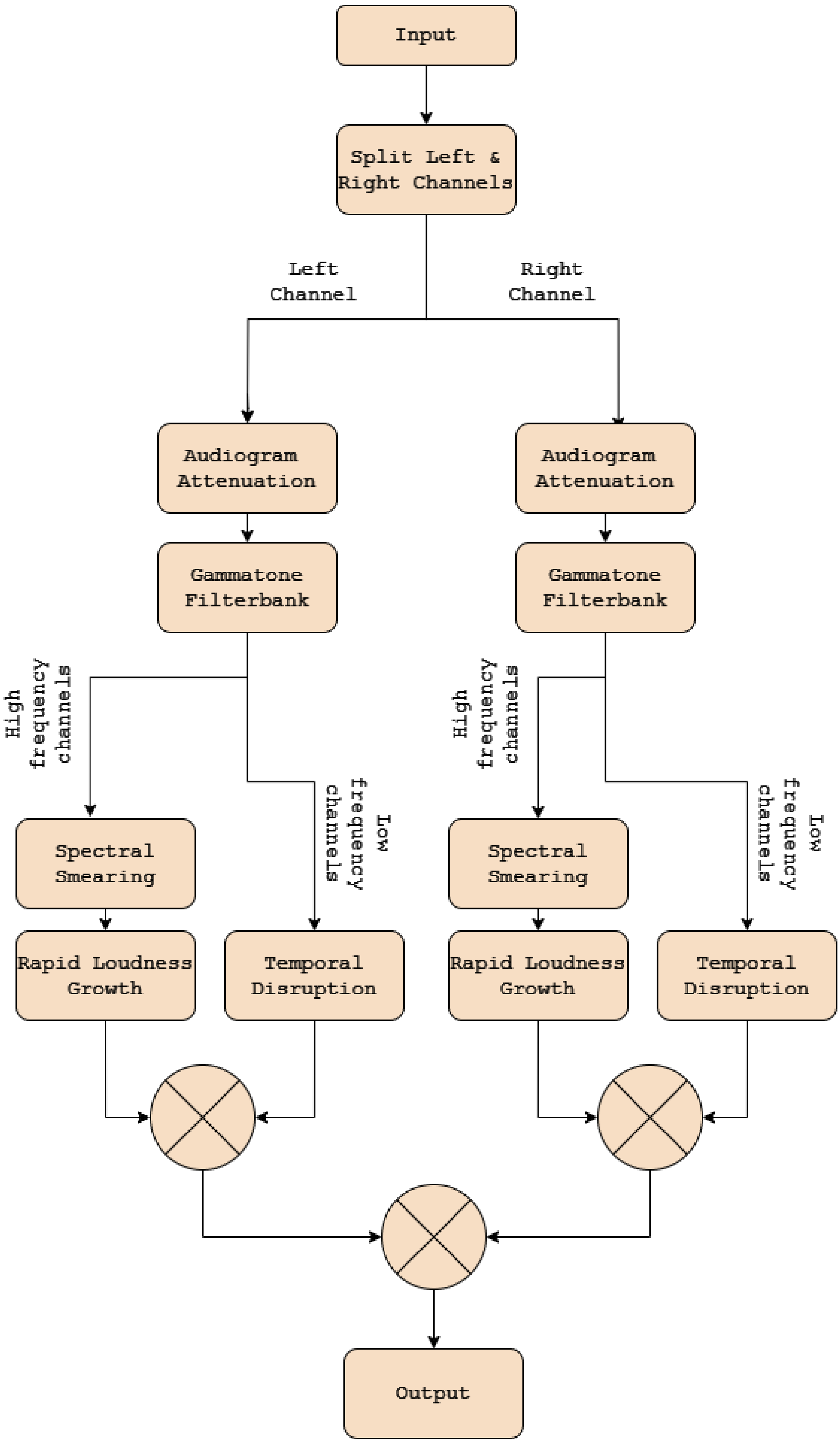

Figure 3.6: Hearing loss simulation plugin flow diagram.

The new version of the simulation also featured adjustments in processing, with the purpose of improving performance and practicality. The adjustments performed are divided into the following categories:

- **Format** - The new version of the simulation is implemented in a VST audio effects plugin format, offering both Windows and macOS compatibility.
- **Layout** - The new layout employed in the current version, features a simple and compact approach with increased customisability, offering a choice of intensity as well as feature selection.
- **User Interface** - The user interface of the simulation is simple and compact yet designed to offer a more user-friendly and pleasant appearance.
- **Processing** - The new version of the simulation features the following adjustments with regard to its processing. The first adjustment involves modifications to the method of simulating high-frequency attenuation. The approach presented in the prototype implementation, employed a frequency-sampling FIR filter, whereas the new version utilises a parametric equaliser approach. This simpler method was selected in order to improve the efficiency and speed of the simulation since it is a less computationally expensive approach that would reduce the simulation's overall latency.

  The second adjustment involves the addition of a 32-channel gammatone filterbank (57) in place of the simple crossover filter used in the prototype implementation, for each of the two channels of the stereo plugin. The gammatone filter bank was added to the implementation in order to simulate the band separation typically observed in the human cochlea, this way making the simulation more perceptually informed. The gammatone filterbank equation is given by:

  $$g(t) = at^{(n-1)} \cos(2\pi ft + \phi)e^{-2\pi bt} \tag{3.1}$$

  where:

- a is the amplitude
- n is the filter order
- t is time
- f is the centre frequency
- ϕ is the phase
- b is the bandwidth

Upon entering the gammatone filterbank, the signal is separated into 32 bands with a range of frequencies from 0 to half the sampling frequency. Bands with centre frequencies higher than 1000 Hz are then grouped to form the high-frequency portion of the signal and bands with centre frequencies lower than 1000 Hz are grouped to form the low-frequency portion signal. Another change in the real-time VST version of the simulation is the implementation of the rapid loudness growth. The upwards expander described in the prototype implementation was replaced by a modified dynamics expander MATLAB system object, in order to improve the stability of the system and reduce artefacts.

Lastly, the temporal jitter module was substituted with a temporal disruption processor, replacing the chorus-based delay approach with a random phase shift approach found in (32). The new temporal disruption processor works by decomposing the signal into phase and magnitude using the short-time Fourier transform (STFT) and then applying random phase shifts within the range of [-π/2, π/2] before recomposing the signal using the original magnitude with the modified phase. An equation describing the process is given in Equation 3.2.

$$x(t) = \mathcal{F}^{-1}\left[M(f) \cdot e^{j(\varphi(f)+\Delta(f))}\right] \tag{3.2}$$

where:

- $x(t)$ is the resulting time-domain signal after the processing

- $\mathcal{F}^{-1}$ is the inverse Fourier transform
- $M(f)$ and $\varphi(f)$ are the magnitude and phase components of the input signal $X(f)$, respectively, obtained through the STFT j is the imaginary unit
- $\Delta(f)$ is a random phase shift applied within the range of $[-\frac{\pi}{2}, \frac{\pi}{2}]$

This processor aims to disrupt the periodicity of the audio signal, replicating the effect of poor temporal resolution. The temporal disruption processing was only applied in the lower frequencies of the input signal. Figure 3.7 presents a 50 Hz pure tone with and without temporal disruption. The spectral-smearing processor remained the same as the prototype implementation.

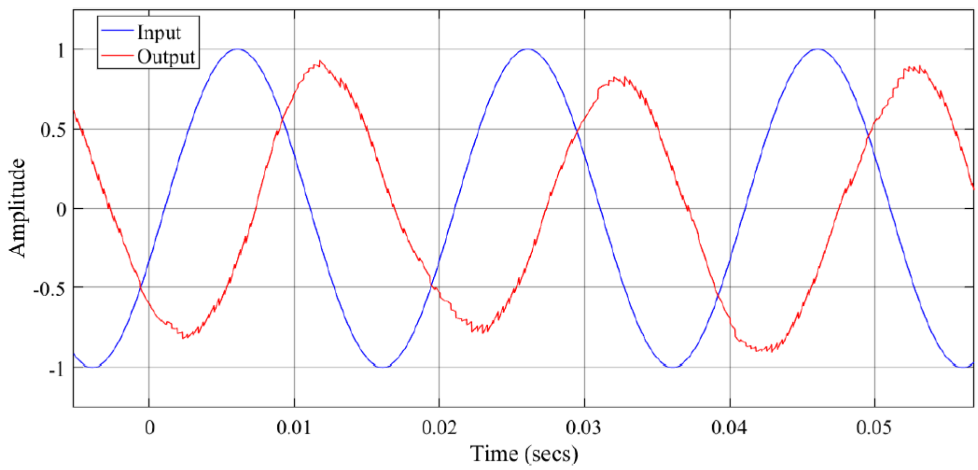

Figure 3.7: 50 Hz tone before (input) and after (output) temporal disruption.

- **Added features** -The new design of the simulation includes added features, which improve customisability and control for the user. Such features are bypass options and mute options for each ear, as well as bypass options and intensity level selection for the effects of high-frequency attenuation and spectral smearing. All remaining supra-threshold effects also offer an on/off functionality for each ear.

The new version of the simulation employs a more stable and better-targeted approach with additional features and capabilities compared to the prototype. The simulation is able to operate on any VST-compatible digital audio workstation with no issues, offering engineers a quick and easy way to assess and adjust their mixes before release.

However, as with all similar developments, this simulation also comes with certain limitations to its effectiveness and response. More specifically, due to the aim of providing a real-time capability, more complex and computationally expensive filters were replaced by simpler implementations, this way resulting in slightly poorer approximations. For future work beyond this thesis, IIR rather than FIR filters could be used, which could provide high quality without computational expense. Additionally, DSP-related limitations include the introduction of unwanted artefacts, due to the simultaneous processing of multiple aspects of the audio signal. Further limitations are those regarding the simulation's accuracy in replicating hearing loss, which arises from the difficulty in appropriately defining and accurately reproducing its perceptual aspects. Documentation of these effects relies mainly on physical measurements, performance assessment using audio tasks, as well as the individual's own perception and description of their loss's characteristics, all of which further complicate the development of a DSP approach attempting to simulate hearing loss.

Although this approach presents the aforementioned limitations in its efficiency and response, its intended use is not an attempt to accurately model hearing loss, but rather to approximate the degradation in audio quality documented in the literature, as closely as possible, using an audio signal processing approach. Additionally, an important goal of this implementation is to be functional and computationally efficient, in order to set the basis for designing and evaluating the intelligent mixing approaches.

Chapter 4

Hearing Loss Simulation Evaluation

4.1 Normal Hearing Participants and Participants with Simulated Loss

To evaluate the effectiveness of the hearing loss simulation, as well as document the effects it introduces in the performance of normal-hearing listeners on psychoacoustic tasks, a listening study was conducted, recruiting participants with normal hearing. Some of the challenges of evaluating a hearing loss simulation include:

- Lack of ground truth in measuring and documenting hearing loss: there is often no universally agreed-upon "gold standard" for measuring all of the perceptual aspects of hearing loss.
- Variability across individuals: Individual differences in hearing loss make it hard to predict how it can affect auditory functions. Due to this variability, it is difficult to assess how well hearing loss models function across different cases and to confirm the simulations' accuracy and applicability for specific types of losses.
- Complexity of auditory processing: Auditory processing is a complex and dynamic process that involves multiple stages of neural processing, and hearing loss can affect different stages in different ways. Simulating hearing

loss requires the modelling of these complex interactions with DSP, which can be challenging and may require simplifying assumptions due to the limitations of the algorithms used.

- Lack of objective measures: While there are subjective measures of hearing loss, such as self-reported difficulty hearing or speech understanding, there are few objective measures that can directly quantify the effects of hearing loss on auditory processing. This lack of objective measures makes it challenging to validate the accuracy of hearing loss simulations when testing them with normal hearing listeners.

4.1.1 Study Design

Participants

Participants were recruited through Queen Mary University using the following inclusion criteria:

- Be over 18 years old
- Have no prior history or diagnosis of any type of hearing loss
- Be proficient in English

A total of 12 participants completed the study, of whom 6 were in the normal hearing group and 6 were in the simulated hearing loss group. The demographic breakdown of the participants is presented in Table 4.1. The two groups of participants had similar demographic characteristics with regard to their gender and English proficiency.

Procedure

The study sessions took place at the Control Room studio at the Queen Mary University of London. The noise floor of the testing room was measured using a Velleman DVM805 sound pressure level meter with a reading of 30 dB SPL. Participants were asked to undergo standard audiometry at the beginning of their

	Simulated Hearing Loss Group	Normal Hearing Group
Total Participants	6	6
Female	2	2
Male	4	4
Proficient English Speakers	4	4
Native English Speakers	2	2

Table 4.1: Demographic breakdown of participants.

session in order to confirm that they qualify for the study, as well as a series of psychoacoustic tasks and finish with a speech-in-noise test. The equipment used for the study was an Interacoustics AS608e Screening Audiometer with assorted audiometric headphones, a pair of AKG K271 MkII headphones, and a Terratec DMX 6 Fire USB audio interface. The simulation was applied to the headphone feed using Reaper and internal routing was performed to connect the output feed from MATLAB to Reaper, using EarTrumpet[1] software. The simulation was set to "mild" high-frequency attenuation and "low" spectral smearing, with the temporal disruption and rapid loudness growth modules activated for both ears. These settings were chosen in order to emulate a mild high-frequency hearing loss typically observed in cases of presbyacusis (58).

Pure-Tone Audiometry The procedure followed for the audiometry was the one recommended by the British Society of Audiology (BSA) (59), as outlined in the manual for pure tone air and bone conduction. More specifically, audiometry was performed in the seven typical audiometric frequencies, ranging from 250 to 8000 Hz. Measurements began with the self-reported "better ear" if stated, starting at 1kHz first and at a clearly audible level and gradually increasing using 5 dB HL steps until a response was given. Following a response, the level was decreased in 10 dB HL steps and then increased at 5 dB HL steps until a response was given. A threshold was finalised once the participant gave a response at the same level, 2 out of 2 or 3 out of 4 times at the same level. The procedure was

[1]https://eartrumpet.app/

repeated for the rest of the frequencies and the other ear in the following order: 1kHz, 2kHz, 4kHz, 8kHz, 500Hz, and 250Hz. Measurements were repeated at 1kHz, for the ear that was tested first.

Psychoacoustic Tasks To perform the psychoacoustic tasks, the Psychoacoustics Matlab Toolbox (60) was used. More specifically the following tests were utilised:

- *Gap detection in white noise:* This test was performed using the staircase method of adaptive threshold estimation, in a 2-alternative forced choice configuration. The participant was presented with 2 noise intervals of 750ms duration and asked which of the two contains a short gap. The gap was then varied based on the participant's performance until the final threshold was derived.

- *Simultaneous masking:* The participant was presented with two consecutive band-passed noise signals one of which contained a 20-ms, 1-kHz sine tone located in its temporal centre. The participant was then asked to identify which noise sample contained the sine tone, while the level of the sine was varied based on their performance until the final threshold was derived. A staircase-type adaptive procedure with a 2-alternative force choice configuration was used for this test too.

Magnitude Estimation A magnitude estimation task (61) was selected in order to determine the growth of loudness of the participants at 1kHz. To perform the test, the online platform SAGE Edge ISLE[2] was utilised. More specifically, in magnitude estimation, participants assess and give numerical values to the perceived strength of a stimulus. The process for estimating magnitude is as follows. The participant is presented with a stimulus and is then asked to provide a number that corresponds to how loud the stimulus was perceived. For the purpose of this study, a variation of this procedure was utilised, where the participant was also presented with a modulus, also known as the standard

[2] https://isle.hanover.edu/Ch02Methods/Ch02MagnitudeEstimationTone_evt.html

stimulus. The modulus was presented to the participant first and was also given a specific value, by the experimenter.

For this study, participants were presented with two consecutive 1kHz tone samples and were told to use the first tone's loudness as their reference in order to estimate the second tone's loudness. A total of 11 different levels of gain were presented for the second tone in random order with 5 repetitions each. The minimum and maximum gain levels were set at 0.1 and 1 respectively and the modulus (first tone) was assigned the value of 50 which remained the same for all the examples. Participants were then provided with a scale of 0 to 100 and were asked to use the first tone as a reference in order to rate the second tone's perceived loudness.

Speech in Noise To perform the speech in noise task, Auditec's NU6 list was used [3]. The NU6 list is comprised of several 2-channel audio files containing spoken words on one channel and background multi-speaker "babble" on the other channel. To perform the speech-in-noise task, four signal-to-noise (SNR) conditions were tested, with 30 words in each condition. To create the stimuli, four NU6 lists were separated into left and right channels in order to obtain the background babble noise separately and combine them for the different SNR conditions. Once the two channels were separated, a speech-in-noise mixing function was used in MATLAB [4] to create the different SNR conditions. The SNR conditions used for this test were 15, 10, 5 and 0, and a total of 30 words were used for each condition. To perform the test, participants were asked to listen to the different lists through headphones and write down the words they heard. The total word count correctly identified in each condition was then calculated for each participant.

[3]https://auditec.com/tag/nu-6/

[4]https://uk.mathworks.com/matlabcentral/fileexchange/37842-speech-in-noise-mixing-signal-to-noise-ratio

4.1.2 Data Analysis

To compare performances between the normal hearing and simulated hearing loss group across the psychoacoustic tasks and magnitude estimation, a paired t-test was employed, since it is a statistical method specifically designed to compare the means of two related sets of data, such as the performance of the same group of individuals under two different conditions: one with normal hearing (Normal Hearing Group) and the other with simulated hearing loss (Simulated Hearing Loss Group). This test is appropriate when dealing with paired or matched data points, as it accounts for individual variations and aids in assessing whether any observed differences are statistically significant. For the speech in noise task, analysis of variance (ANOVA) with repeated measures was conducted in order to assess the impact of both SNR and Group (simulation/no simulation) factors, along with their interaction, on the participants' performance over time. This was chosen because it allows us to analyze within-subject factors (SNR and Group) with repeated measurements and assess their combined effects on the dependent variable, offering insights into how different conditions and groups influence the outcome.

4.1.3 Results

The analysis of the gaps in noise task data revealed significant differences between the Normal Hearing Group and the Simulated Hearing Loss Group. Participants in the Simulated Hearing Loss Group exhibited a significantly higher mean gap detection threshold (4.98) compared to the Normal Hearing Group (1.94), indicating that the simulated hearing loss condition impacted their ability to detect brief gaps in noise. Additionally, the t-statistic of -4.14, with 5 degrees of freedom, showed a significant difference between the groups. This suggests that the hearing loss simulation had a substantial impact on gap detection ability. The one-tail p-value of 0.004 (or two-tail p-value of 0.008) further supports the significance of this difference. The Pearson correlation coefficient of -0.44 suggests a negative correlation between group performance and hearing loss simulation, reinforcing the finding that simulated hearing loss led to decreased performance in the gaps in noise test.

Table 4.2 presents the measured thresholds in milliseconds for each of the 6 participants in the normal hearing and simulated hearing loss groups along with the average of the groups and the standard deviation, whereas Table 4.3 presents the results of the paired t-test for the gaps in noise data.

	Normal Hearing Group	Simulated Hearing Loss Group
	0.80	5.13
	3.76	4.19
	2.14	5.36
	1.40	6.36
	1.60	5.60
	1.93	3.24
Average	1.85	4.93
Standard Deviation	0.95	1.30

Table 4.2: Gap thresholds with mean and standard deviation for the normal hearing and simulated hearing loss groups.

Metric	**Normal HL Group**	**Simulated HL Group**
Mean	1.94	4.98
Variance	1.02	1.22
Observations	6	6
Pearson Correlation	-0.44	
Hypothesized Mean Diff	0	
df	5	
t Stat	-4.15	
P(T¡=t) one-tail	0.004	
t Critical one-tail	2.02	
P(T¡=t) two-tail	0.008	
t Critical two-tail	2.57	

Table 4.3: Results of Paired Two Sample t-Test for Means for the gaps in noise task.

In the simultaneous masking test, participants in the Simulated Hearing Loss Group exhibited a mean threshold of -25.87, while the Normal Hearing Group had a mean threshold of -33.25. This suggests that participants with simulated hearing loss required higher thresholds in order to detect a 1-kHz sine tone embedded in noise compared to those with normal hearing. The t-statistic of -3.10, with 5 degrees of freedom, indicates a significant difference in threshold between the two groups. The one-tail p-value of 0.01 (or two-tail p-value of 0.02) further confirms the significance of this difference. These findings suggest that the simulated hearing loss condition affected participants' ability to detect the sine tone. Table 4.4 presents the measured thresholds in dB for each of the 6 participants in the normal hearing and simulated hearing loss groups. Table 4.5 presents the results of the paired t-test for the simultaneous masking data.

	Normal Hearing Group	Simulated Hearing Loss Group
	-38.87	-21.62
	-30.87	-31.37
	-32.75	-27.37
	-32.87	-25.25
	-30.87	-25.37
	-33.25	-24.25
Average	-32.23	-25.93
Standard Deviation	3.71	2.8

Table 4.4: Simultaneous masking thresholds in dB with mean and standard deviation for the normal hearing and simulated hearing loss groups.

Metric	**Normal HL Group**	**Simulated HL Group**
Mean	-33.25	-25.88
Variance	8.66	10.77
Observations	6	6
Pearson Correlation	-0.75	
Hypothesized Mean Diff	0	
df	5	
t Stat	-3.10	
P(T¡=t) one-tail	0.01	
t Critical one-tail	2.02	
P(T¡=t) two-tail	0.03	
t Critical two-tail	2.57	

Table 4.5: Results of Paired Two Sample t-Test for Means for the simultaneous masking task.

The magnitude estimation test results reveal no significant differences between the normal hearing and simulated hearing loss groups. The statistical test results indicate that there is a Pearson correlation coefficient of 0.99 between the Simulated Hearing Loss Group and the Normal Hearing Group. This high positive correlation suggests that both groups generally had consistent responses when estimating the loudness of the 1kHz tones.

The t-statistic of 0.30, with 10 degrees of freedom, results in a p-value of 0.77 for the two-tailed test. This suggests that there is no significant difference in the magnitude estimation responses between the Simulated Hearing Loss Group and the Normal Hearing Group. In other words, the simulated hearing loss condition did not appear to have a significant impact on participants' ability to estimate the loudness of the tones when using the reference stimulus.Figure 4.1 presents the average magnitude estimations over actual gain for the two groups. Table 4.6 presents the results of the paired t-test for the magnitude estimation data.

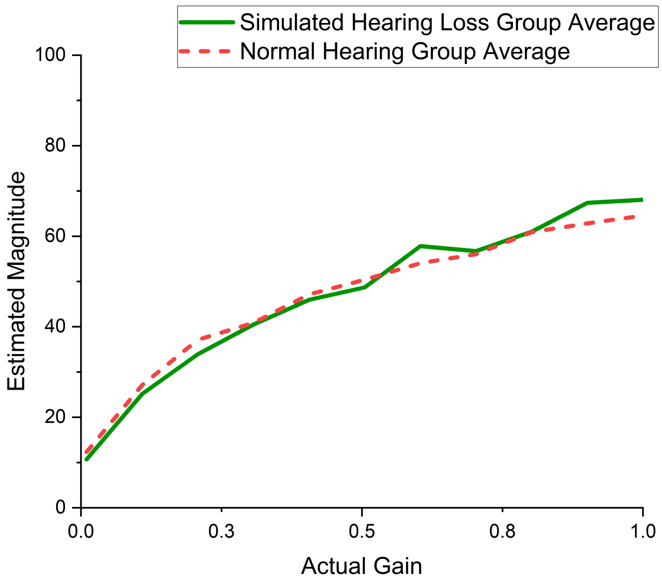

Figure 4.1: Average magnitude estimations plotted over actual gain values for the normal hearing and simulated hearing loss groups.

Speech-in-noise results present a significant reduction in the number of identified words for the simulated hearing loss group in all SNR conditions compared to the normal hearing group, with an observed decline with decreasing SNR.The

Metric	**Simulated HL Avg**	**Normal HL Avg**
Mean	46.9	46.66
Variance	327.66	262.55
Observations	11	11
Pearson Correlation	0.99	
Hypothesized Mean Diff	0	
df	10	
t Stat	0.30	
P(T¡=t) one-tail	0.39	
t Critical one-tail	1.81	
P(T¡=t) two-tail	0.77	
t Critical two-tail	2.23	

Table 4.6: Results of Paired Two Sample t-Test for Means for the magnitude estimation task.

repeated measures ANOVA assessed the impact of various factors and interactions on participants' performance over time in different conditions. Notably, there was a significant interaction effect between Group and Time, signifying that performance over time differed significantly between the Normal Hearing Group and the Simulated Hearing Loss Group Figure 4.2 presents a scatter plot of the total number of words identified by the participants in the two groups, while Figure 4.3 presents the average number of identified words in the various SNR conditions.

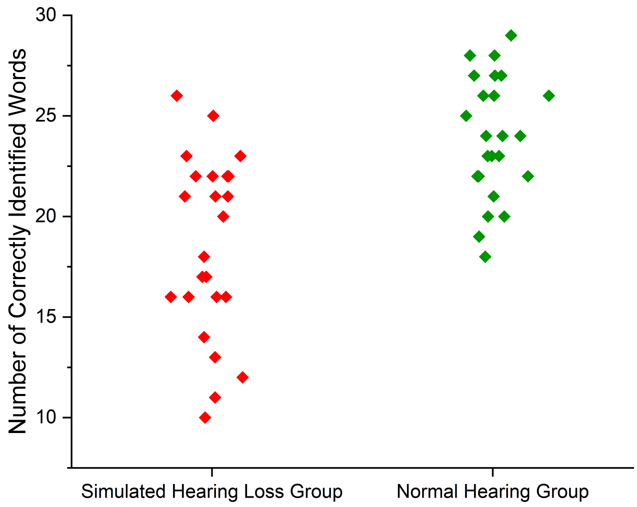

Figure 4.2: Total number of identified words in the normal hearing and simulated hearing loss groups.

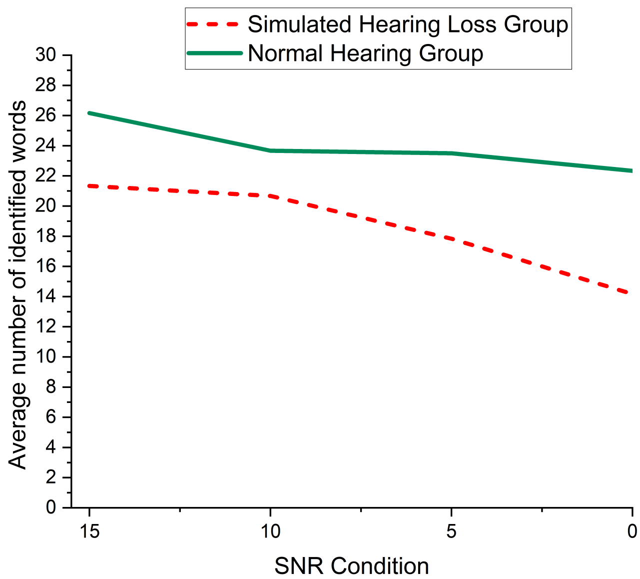

Figure 4.3: Average number of identified words in the different SNR conditions for the normal hearing and simulated hearing loss groups.

Metric	SumSq	DF	MeanSq	F	pValue	pValueGG	pValueHF	pValueLB
(Intercept):Time	24.19	5	4.83	1.38	0.27	0.30	0.27	0.30
SNR:Time	7.4	5	1.48	0.42	0.82	0.69	0.82	0.55
Group:Time	127.36	5	25.47	7.30	0.0004	0.01	0.0004	0.05
SNR:Group:Time	34.33	5	6.86	1.96	0.12	0.19	0.12	0.23
Error(Time)	69.76	20	3.48					

Table 4.7: Results of the Repeated Measures ANOVA for the speech-in-noise task.

4.1.4 Discussion

The results of this study present reduced performances in the majority of the tasks for participants in the simulated hearing loss group, compared to those in the normal hearing group. The average performance of the simulated hearing loss group in the gaps in noise test reveals an increase of approximately 61% in the measured duration of the identified gaps when compared against the normal hearing group. Simultaneous masking results also demonstrate a decrease in performance for the simulated hearing loss group, presenting an average increase of 22% in the measured thresholds required for successful identification of the pure tone in noise. Similarly, the identification of words in noise appears to be significantly impacted in the simulated hearing loss group, with an average decrease of 23% in the number of identified words across all SNR conditions. Both groups produced similar results on the magnitude estimation task which could be attributed to the rapid loudness growth module's processing not being very effective.

In conclusion, an expected decrease in the overall performance of the simulated hearing loss group was observed which is in agreement with results found in studies utilising the same simulation methods in the literature (32; 38). Based on the results, the hearing loss simulation's rapid loudness growth module parameters were modified in order to produce a more evident expansion. More specifically a threshold of recruitment was set at -10 dBFS with a ratio of 0.9 and attack and release times of 0.5 s and 1 s respectively. The settings for the expansion module were determined through testing, to achieve a perceptible rapid increase in the perceived loudness above the threshold of hearing without introducing any noticeable artefacts. A plot presenting the static characteristic of the revised upwards expansion module is presented in Figure 4.4.

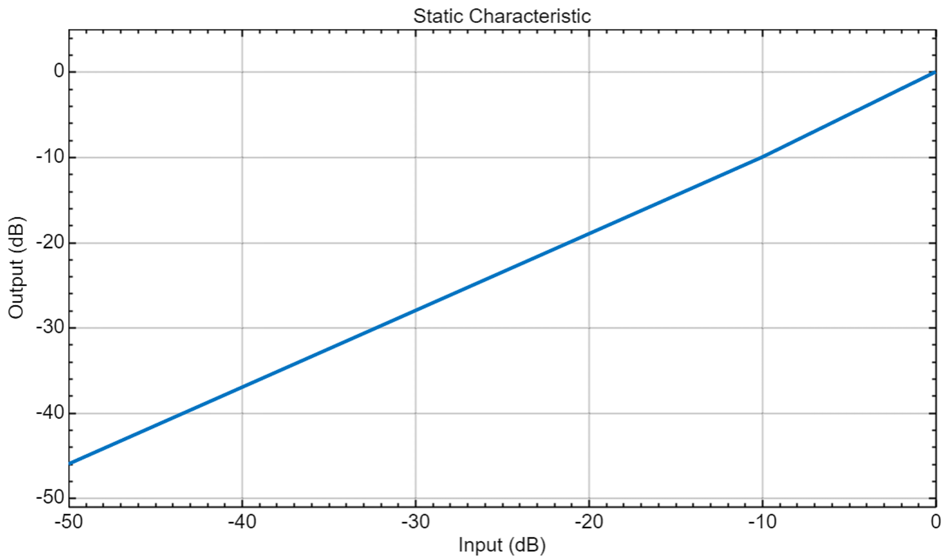

Figure 4.4: Static characteristic of the rapid loudness growth, upwards expansion unit.

Finally, the results prompt exploration of the simulation's accuracy through testing with listeners with real hearing loss, as the degradation in performance observed with the use of the simulation could be then compared against degradation caused by actual hearing loss. To investigate this, an additional study was conducted which is presented in the following section.

4.2 Participants with Real and Simulated Hearing Loss

To evaluate the accuracy of the hearing loss simulation against real hearing loss, an additional listening study was conducted in collaboration with Imperial College London, where the hearing loss simulation plugin was evaluated and compared with the 3DTI toolkit's hearing loss simulation plugin (11) designed by Imperial's audio design experience team in collaboration with the Diana Group From the University of Malaga. The primary aim of this study was not a formal evaluation, but rather an exploration of hearing loss profiles and an informal assessment of hearing loss simulation.The study sought to compare the experiences of participants with genuine hearing loss and those with simulated hearing loss across a range of psychoacoustic tasks and speech-in-noise assessment. The goal was to gauge how accurately these two implementations replicated the essential perceptual aspects of hearing loss. Moreover, the study aimed to identify the respective strengths and weaknesses of each approach. More particularly, participants in the hearing loss (HL) group were tested first in order to obtain their hearing profiles and use them as a hearing loss simulation template for the simulated hearing loss (SIMHL) group. Each HL group participants' hearing loss characteristics were used to test multiple SIMHL group participants.

4.2.1 Study Design

Participants

A total of 4 participants with hearing loss and 11 participants with normal hearing were recruited for this study. In order to qualify for the study all participants had to be 18 years old and over, as well as proficient in English. HL group participants also needed to meet the following additional criteria for inclusion:

- Have mild/moderate bilateral, hearing loss of sensorineural nature

- Have hearing loss located mainly in the higher frequencies

- If using hearing aids (HA), to be able to remove them for the duration of the study

Participants for the HL group were recruited by advertising the study to UK charities, such as RNID[5] and the Guildford Hard of Hearing Support Group [6] as well as by contacting relevant research groups. Participants for the SIMHL group were recruited through internal advertising at Imperial College London as well as Queen Mary University. The exclusion criteria for the study were:

- Incomplete participation
- Hearing loss characteristics that did not match the eligibility criteria
- Inability to perform the task due to lack of understanding or unable to reach audibility with extra amplification

Of the four participants in the HL group, two participants were excluded due to their inability to complete all the necessary tasks even with additional amplification. A total of 13 participants, 2 with hearing loss and 11 with normal-hearing and simulated hearing loss, successfully completed the study. Table 4.8 presents the assigned number of SIMHL listeners for each HL profile and in both simulations.

Implementation	**HL Profile 1**	**HL Profile 2**
QMUL Simulation	4	2
3DTI Simulation	3	2

Table 4.8: Number of participants with simulated hearing loss for each hearing loss profile and simulation.

Procedure

The duration of the study was approximately two hours and the sessions took place at the Turret Lab facilities of Imperial College London, the control room studio at Queen Mary University, as well as the performing arts technology

[5]https://rnid.org.uk/
[6]https://www.surreyinformationpoint.org.uk/Services/32/Guildford-Hard-Of-He

studios at Surrey University. These locations were chosen in order to facilitate participants by limiting their travel distance. All of the experiment sessions were run by the same experimenter.

The testing procedure was divided into 3 parts:

- Hearing screening which included audiometry and a questionnaire with general questions about the participants' hearing (see Appendix, A.1)
- Psychoacoustic tasks
- Speech in noise task using the adaptive sentence list method (ASL)

Audiometry To perform the audiometry the Natus-Aurical Aud audiometer was used along with the Oticon suite. The procedure followed was the same as outlined in the previous experiment described above. Audiograms obtained from participants with hearing loss were used to set the audiogram attenuation settings of the two hearing loss simulations.

Psychoacoustic Tasks For the psychoacoustic tasks, the PSYCHOACOUSTICS suite (62) was used. Stimuli for the psychoacoustic tasks and ASL experiment were presented through a set of Sennheiser TDH650 headphones. Two MOTU UltraLite Mk3 Hybrid audio interfaces were used for the reproduction of the test stimuli, as well as for applying the simulations to the testing signals. The application of the processing for the two simulations was performed in real-time on the headphone feed, by routing the testing software's output using Reaper[7] DAW.

The following psychoacoustic tests were selected, in order to provide information related to each of the modules of the simulations:

- *Gaps in noise task:* This task was selected in order to test the participants' temporal acuity, using a procedure of threshold determination based on the method described in the previous experiment. Participants were presented

[7]https://www.reaper.fm/

with two short noise intervals and were asked to identify which of the two contained a short gap, using a two-alternative forced choice method. The adaptive procedure for determining the threshold was the standard staircase procedure. The low and high cut frequencies for the main noise were set at 400Hz and 600Hz. An additional noise was added with a low and high cut at 50Hz and 100Hz. The main noise's spectrum level was set at 54 dB/Hz and the added noise's spectrum level at 25.2 dB/Hz. A total of eight turn-points were used and the final threshold was produced by calculating their geometric mean. The procedure was repeated for each of the two ears. To produce the temporal jitter and temporal disruption values for the two simulations, normal gaps were set at approximately 10 ms, based on (63) Values above 10 and up to 20ms were assigned a low temporal jitter setting at the 3DTI simulation, values between 20 and 30ms were assigned a medium setting and any value above 30ms was assigned a high temporal jitter setting.

- *Notched-Noise:* This task was selected in order to test the participants' frequency resolution. The procedure was based on the simplified method for auditory filter estimation as described in (64). Participants were presented with two short noise intervals and were asked to identify which of the two contained an extra tone through a two-alternative forced choice method and thresholds were derived using a standard staircase adaptive procedure. The target frequency tested was 1000 Hz with an initial presentation level of 65dB SPL and spectrum levels of the upper and lower noise bands were set at 42db/Hz. The test was performed using three symmetric and two asymmetric conditions for the Δf lower and Δf upper values. The terms Δf lower and Δf upper refer to the lower and upper frequency bounds, respectively, of the "notch" in the noise spectrum. The following values were used for this experiment:
  - symmetric: [0.0 & 0.0, 0.2 & 0.2, 0.4 & 0.4]
  - asymmetric:[0.2 & 0.4, 0.4 & 0.2]

These conditions were proposed as optimal for this test based on (64). Even though repeated measurements are suggested, each condition was only tested once due to time constraints as well as in order to minimise participants' fatigue. A total of eight turn-points were also used on this task and the final threshold was produced by calculating their geometric mean. Once all 5 conditions were calculated in both ears the Roex filter functionality of the PSYCHOACOUSTICS suite was used in order to produce the filter shape for each participant using the Roex filter equation. The roex filter coefficients and more specifically the $p$upper and $p$lower values that represent the asymmetry of the auditory filter shape were obtained from the calculation (65). The $p$upper and $p$lower values derived from the Roex suite was also used to determine the level of spectral smearing used in the hearing loss simulations. More specifically, average normal $p$upper and $p$lower values were calculated based on the measurements described in (66). Upper and lower widening factors were then determined for the 3DTI simulation by calculating the normal to-measured value ratio. To determine the smearing settings for our proposed simulation, ratio values up to 2 were assigned to the low smearing setting whereas ratio values larger than 2 were assigned to the high-frequency smearing setting.

- *Magnitude Estimation:* For the magnitude estimation task, the online platform SAGE Edge ISLE was utilised again following the same procedure as described in the previous study 4.2.1. The level of the first tone (modulus) was measured at 85 db SPL whereas the maximum output loudness of the test was measured at 90dB SPL.

- *Adaptive Sentence List:* For the speech in noise test the Spatial ASL test was selected. The test was performed using an iPad and participants were asked to listen to the spoken sentences and repeat back what they hear. The spatial speech-in-noise test is developed as an iOS application and is based on the ASL speech corpus [8]. Using an anechoic KEMAR head-related

[8] `https://www.sciencedirect.com/science/article/abs/pii/S0885230813000879`

transfer function (HRTF), the speech target and pink noise masker are both spatialized in a frontal position. The SNR for the up-down adaptive method is +12dB at the beginning. Each phrase contains three keywords, and when two or more are found, the solution is deemed accurate. After two reversals (change in the participant's response indicating a threshold or discrimination boundary, the SNR change's initial step, which is +-4dB, is lowered to +-2dB. The test ends after eight reversals and the 50% speech recognition threshold (SRT) is computed as the mean value of the last six reversals.

Calibration Calibration of the testing equipment was performed at the Turret Lab facility at Imperial College. The ambient noise floor of the lab was measured at 23 dB LAeq. To perform calibrations for all the experiments, a KEMAR head and torso simulator was used. For the Psychoacoustics suite, calibration was performed following the built-in instructions and using the calibrating stimulus provided in the software. The provided diffuse headphone mode was applied, to correspond to the headphones' frequency response. For the magnitude estimation task, calibration was performed by determining the modulus output level as well as the maximum output level which was measured at 85 and 90 dB SPL respectively. For the ASL test, calibration was performed by following the instructions and calibration noise provided with the software. The final levels were set at 80 dB SPL.

4.2.2 Data Analysis

In order to do an across-test comparison of the simulation effectiveness, a per cent error was computed for each test, so that the errors were always in the same unit. The percent error (ϵ) was computed as follows:

$$\epsilon = \frac{|T_{HL} - T_{SIMHL}|}{T_{HL}}$$

with T the measured threshold of a listener from the HL (T_{HL}) or a SIMHL (T_{SIMHL}) group. The SIMHL groups were compared only to the HL participant used to tune the simulation they listened through. Then, the percent errors were averaged across SIMHL listeners for a given simulation and HL participant. This allowed to compare which simulation was the best to simulate a deficit, which deficit was simulated the best and which HL listener was simulated the best.

Additionally, to compare the magnitude estimation curves numerically, the Stevens' power law was fitted for each participant, whose equation is:

$$\zeta = k * G^{\alpha}$$

with ζ the estimated magnitude, k the Stevens' power law constant, G the actual gains used during the test and α the Stevens' power law exponent. This function has been used to predict loudness curves (67; 68), and therefore seems appropriate to describe the current estimated magnitude. The two parameters of the equation, k and α, were varied to minimize the residual sum of the squares (RSS), computed as the sum of the squared differences between the participant's estimated magnitude and the power-law's estimated magnitude.

4.2.3 Results

The thresholds for the gap detection in noise task can be found in tables 4.9 and 4.10, for the two HL participants, as well as for the SIMHL ones using those hearing profiles. Note that SIMHL11 has been discarded from the analysis of this task, as they were deemed as an outlier after producing repeatedly irregular results. SIMHL11 was able to detect thresholds of less than 1 ms with the simulation turned on and 0 ms otherwise, which could be interpreted as having the ability to perceive the ramp that is present just before and after the gap.

The results present high variability for both simulations, particularly in the HL1 profile. Both simulations better approximate on average the thresholds obtained by HL1 as opposed to HL2 thresholds. Interestingly, the thresholds obtained by the SIMHL group through the HL2 profile are better compared to the other group, while the temporal distortion/disruption module was set at the worst setting level. The percent errors are displayed in the third row (or first row after the two header rows) of Table 4.11. The errors are indeed lower for HL1 but still at least as high as 23%.

Proposed Simulation			3DTI		
Participant	**Left Ear**	**Right Ear**	**Participant**	**Left Ear**	**Right Ear**
SIMHL1	5.16	26.01	SIMHL4	12.95	10.95
SIMHL2	13.68	14.08	SIMHL5	29.9	14.89
SIMHL3	11.26	18.1	SIMHL6	19.68	13.32
Average SIMHL	*10.03*	*19.39*		*20.84*	*13.05*
HL1	15.31	19.14		15.31	19.14

Table 4.9: Gaps-in-noise thresholds (in ms) for HL1 participant and the associated normal hearing participants with simulated hearing loss (SIMHL).

Proposed Simulation			3DTI		
Participant	**Left Ear**	**Right Ear**	**Participant**	**Left Ear**	**Right Ear**
SIMHL7	9.01	10.36	SIMHL9	8.76	10.95
SIMHL8	8.76	8.06	SIMHL10	7.41	15.74
Average	*8.88*	*9.21*		*8.08*	*13.34*
HL2	31.62	32.51		31.62	32.51

Table 4.10: Same as Table 4.9 but for HL2.

	HL1		HL2	
	Prop. Sim.	**3DTI**	**Prop. Sim.**	**3DTI**
Gap detection in noise	*34% / 23%*	*46% / 32%*	*72% / 72%*	*74% / 59%*
Tone detection in notched noise	*19% / 24%*	*12% / 18%*	*17% / 14%*	*14% / 7%*
Stevens' law exponent (α)	*42%*	*9%*	*138%*	*157%*
Stevens' law constant (k)	*5%*	*5%*	*18%*	*19%*

Table 4.11: Percent errors between the SIMHL groups and the HL participants for detection of gap in notch noise, detection of tone in noise, and magnitude estimation (Stevens' law exponent and constant) tasks. The cells containing two percentages show the left-ear and right-ear errors, respectively.

Figure 4.5 shows the outcomes of the tone detection in notch noise task for participants HL1 and HL2, respectively, plotted across all 5 notch conditions, and compared with the average results of the SIMHL listeners. The measured thresholds for HL1 (red lines top panel) demonstrate that their auditory filter at 1 kHz is very broad at each ear, as increasing the notch bandwidth in the noise did not decrease the measured masked threshold of the tone. The thresholds of the SIMHL group listening through HL1 simulation (blue and green lines top panel) decreases as the notch bandwidth increases, resulting in a increasing discrepancy between HL1 and the SIMHL group threshold as the the notch bandwidth widens. The masked thresholds measured with HL2 (red lines bottom panel) suggest that their auditory filters are narrower compared to HL1. The masked thresholds are about 40 to 45 dB for the larger notch bandwidths while the one of HL1 are above 55 dB whatever the notch bandwidth. The simulations better reproduce the loss of frequency selectivity of HL2 compared to HL1 as the blue and green lines are visually closer to the red lines in the bottom panel than in the top panel.

Table 4.11 displays the percent errors for both simulations on the detection of tone in noise task (fourth row, or second row after header rows). The percent error is computed as the mean across notch bandwidth conditions of the difference between HL participant and SIMHL group for a given simulation and ear. As observed on the graph the discrepancy between HL2 and the associated SIMHL groups is lower (as opposed to HL1) and the 3DTI simulation is more accurate for this task.

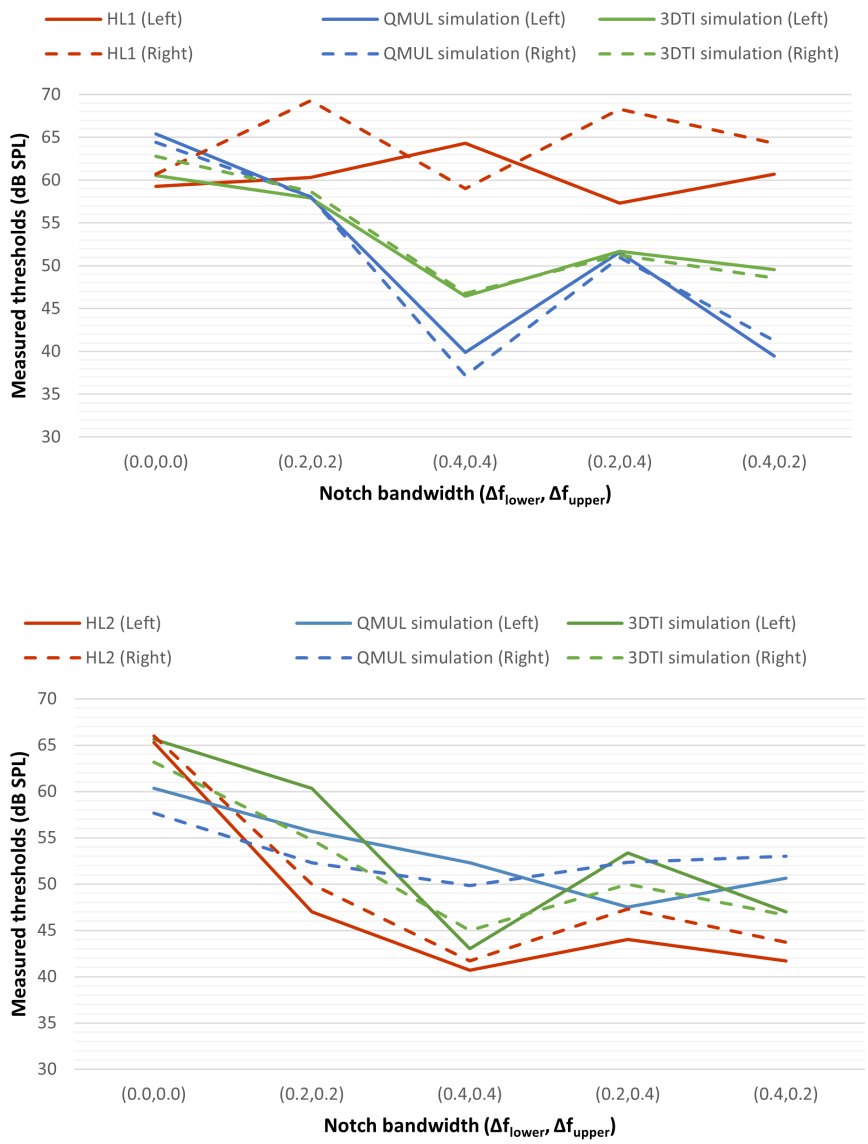

Figure 4.5: Measured thresholds for HL1 (upper panel) and HL2 (lower panel) compared to SIMHL groups through QMUL and 3DTI simulations.

Figures 4.6 and 4.7 show the estimated magnitudes for the two HL participants as well as the SIMHL participants listening through the two hearing loss profiles, for both simulations. Note that participant HL2 requested a reduction of -5 dBA in the reproduction level for this task, which was also applied to all SIMHL participants using this profile.

Magnitude estimation results show a good agreement between HL and SIMHL participants, with some differences observed in the low gain values, especially for participant HL2 who scores the highest value across all the participants.

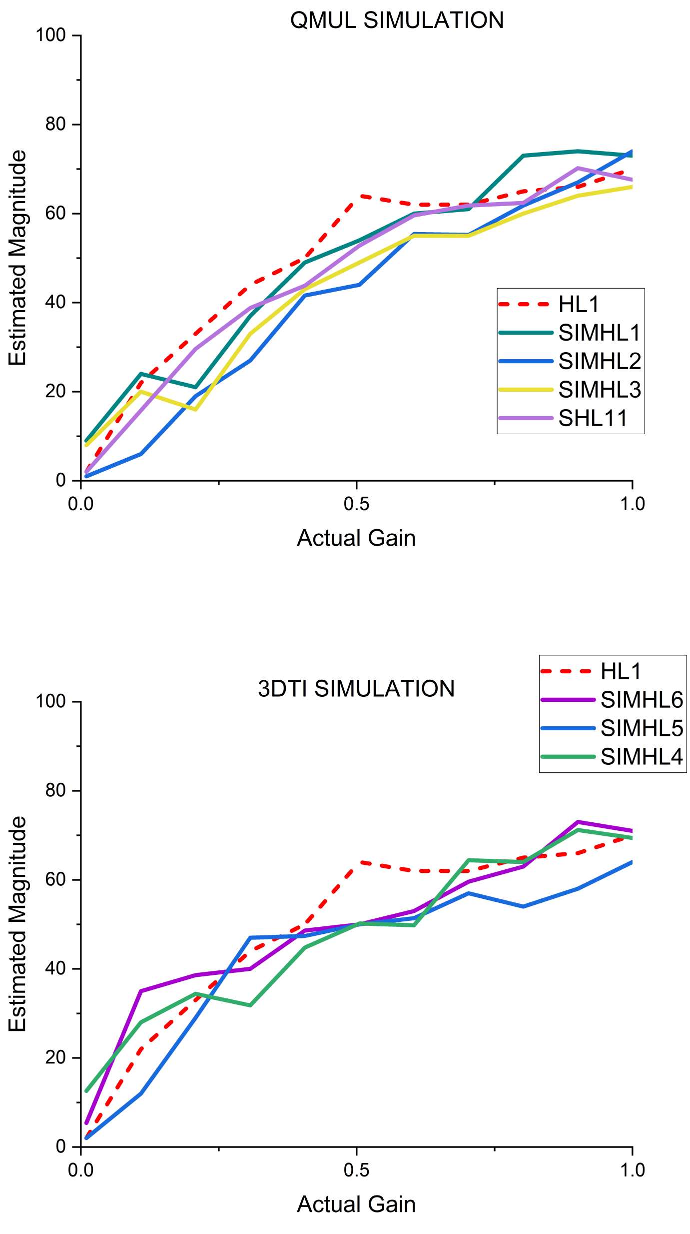

Figure 4.6: Estimated magnitude curve of listener HL1 (red dashed line) opposed to estimated magnitude curves of SIMHL listeners obtained using the QMUL (top panel) and 3DTI (bottom panel) simulation.

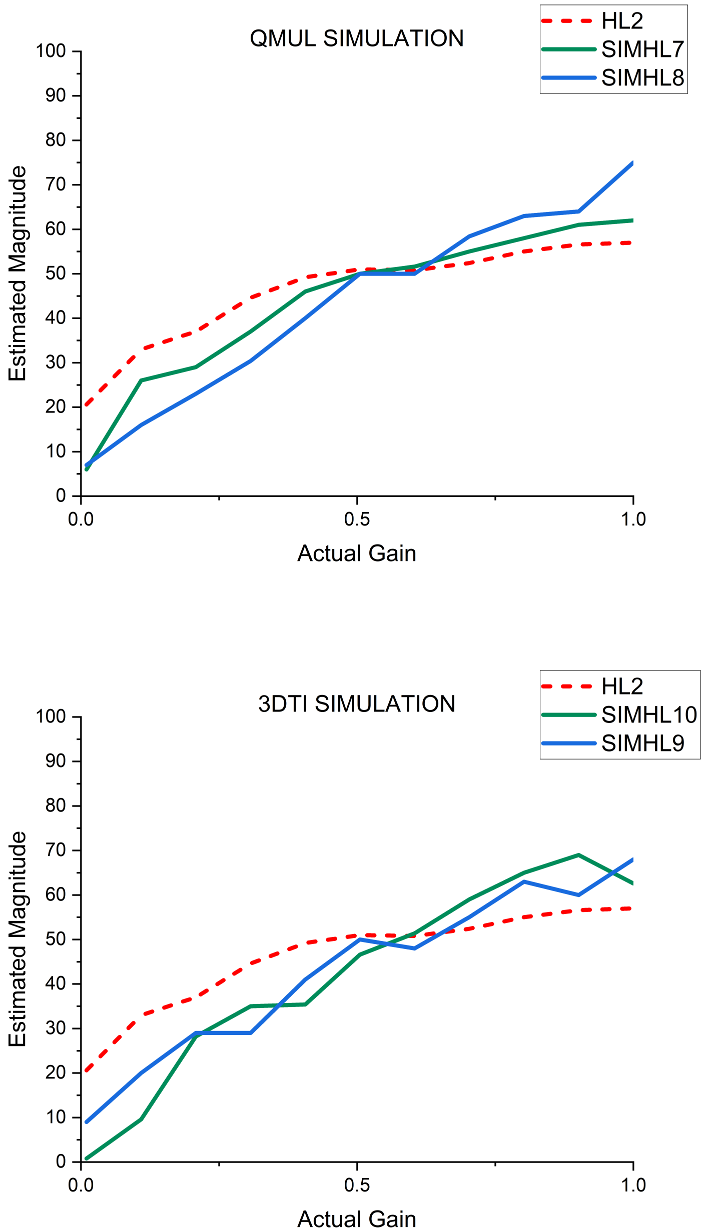

Figure 4.7: Estimated magnitude curve of listener HL2 (red dashed line) opposed to estimated magnitude curves of SIMHL listeners obtained using the QMUL (top panel) and 3DTI (bottom panel) simulation.

To compare the curves numerically, the Stevens' power law was fitted for each participant, whose equation is:

$$\zeta = k * G^{\alpha}$$

with ζ the estimated magnitude, k the Stevens' power law constant, G the actual gains used during the test and α the Stevens' power law exponent. This function has been used to predict loudness curves (69; 68), and therefore seems appropriate to describe the current estimated magnitude. The two parameters of the equation, k and α, were varied to minimize the residual sum of the squares (RSS), computed as the sum of the squared differences between the participant's estimated magnitude and power-law's estimated magnitude. The parameters values leading to the lowest RSS are displayed in Table 4.12.

Generally, the curves of HL2 and their associated SIMHL groups present a better fitting (opposed to HL1) as the RSS are on average lower. However, the Stevens's law parameters of the SIMHL groups associated to HL1 are closer to the ones of HL1. This is confirmed by the rather low percent errors for HL1 (shown by the last two rows of Table 4.11). The 3DTI simulation is providing quite an accurate simulation of HL1's rapid loudness growth, with only 9% and 5% of errors for α and k, respectively.

Participant	**Simulation**	k	α	**RSS**
HL1	-	**73.8**	**0.47**	**228**
SIMHL1	QMUL	77.8	0.59	193
SIMHL2	QMUL	75.1	0.82	118
SIMHL3	QMUL	68.7	0.60	180
SIMHL11	QMUL	73.2	0.57	110
SIMHL4	3DTI	70.6	0.48	180
SIMHL5	3DTI	70.5	0.42	147
SIMHL6	3DTI	65.2	0.49	317
HL2	-	**57.8**	**0.23**	**22**
SIMHL7	QMUL	64.1	0.44	40
SIMHL8	QMUL	78.8	0.68	66
SIMHL9	3DTI	66.9	0.55	183
SIMHL10	3DTI	70.8	0.66	99

Table 4.12: Parameters of the Stevens' power law for the fitted estimated magnitude curve of each participant as well as the RSS to provide the goodness of fitting.

To summarise the psychoacoustic task results, the simulation module providing the most reliable approximation of the HL participant's deficits is the frequency resolution deficit module for each simulator. The percent errors obtained for tone detection in noise task (being the task associated to this module) span from 14% to 24% for the QMUL simulation while the range is from 7% to 18% for the 3DTI simulation depending on the HL participant and their ear. On average across tasks and simulations, the thresholds of HL2 are not as well simulated as HL1's. Regarding the differences between simulations, it is hard to say whether there is a better one as none of them is consistently providing lower per cent errors compared to the other. Moreover, they both substantially fail in some ways, such as approximating the Steven's law exponent of HL2.

Results from the ASL intelligibility test present high variability. Tables 4.13 present the SRTs of both HL participants and the SIMHL participants for both simulations. Note that some SRTs are very high (e.g., SIMHL6 and SIMHL10), which means that participants were actually listening to the sentences in near-quiet conditions. They were then unable to understand speech degraded by the HL simulation, which made the SRT measurement impossible. It must be noted for the 11 SIMHL, only one performs better (i.e., lower SRT) than the associated HL participant.

Participant	Simulation	SRT
HL1	-	**-4.6**
SIMHL1	QMUL	-3.6
SIMHL2	QMUL	2.6
SIMHL3	QMUL	26.6
SIMHL11	QMUL	11.6
SIMHL4	3DTI	-8
SIMHL5	3DTI	17
SIMHL6	3DTI	207
HL2	-	**-10.6**
SIMHL7	QMUL	1.3
SIMHL8	QMUL	13
SIMHL9	3DTI	12.6
SIMHL10	3DTI	114

Table 4.13: SRTs for HL and SIMHL participants. Speech and noise were simulated at the same position, in front of the listener.

4.2.4 Discussion

The findings of this investigation demonstrate that both employed simulations exhibit a satisfactory level of accuracy in approximating spectral smearing for both hearing loss (HL) profiles and loudness growth, particularly for HL1 as illustrated in figures 4.5 and 4.6. However, the simulations do not effectively replicate temporal acuity for any of the HL profiles and loudness growth for HL2. Consequently, neither of the simulations impairs speech perception to the extent required to align the average Speech Reception Thresholds (SRTs) of SIMHL groups with those of individuals with HL.

Tone Detection in Notch Noise: In the context of tone detection in notch noise, participants in the SIMHL group, across both simulations, exhibit masked thresholds for the tone in notched noise that are comparable to or even superior (i.e., lower thresholds) to those observed in the HL participants. The measured values for the HL participants align closely with those reported in a previous study by (70). The disparity between the two groups becomes more pronounced with increasing noise bandwidth (Δ F). This implies that the simulations effectively approximate the filter characteristics around the target frequency, as evidenced by the absence of differences between the groups in the condition with no notch (i.e., $\Delta\ F = 0$). However, variations in band-pass filter slopes farther from the center frequency may elucidate the distinctions between the two groups. These variations have negligible influence on the thresholds recorded for $\Delta\ F$ equal to 0 since, in that circumstance, noise level in the band is primarily determined by the filter's configuration around the center frequency. The 3DTI simulation yields the smallest differences between HL and SIMHL participants, potentially attributable to its greater adaptability relative to the QMUL simulation or module-based implementation.

An alternative explanation, although speculative, for the widening gap between the two groups as the noise notch broadens may be an effect of training. Over successive conditions, SIMHL listeners exhibit progressive performance enhancement, suggesting an adjustment to the hearing loss simulation and an increasing familiarity with the task demands. It is imperative to note a significant limitation in this task, which was constrained to testing a single frequency (specifically, 1000 Hz) due to time constraints. Furthermore, it is worth acknowledging that while (64) advocates for conducting at least one repetition per condition, our approach in this study entailed examining each condition only once. This choice was deliberate, aimed at minimizing testing duration, and addressing potential issues related to participant fatigue and disengagement.

Gap Detection in Noise: Regarding the gap detection in noise task, the thresholds exhibited by participant HL2 are notably atypical and could almost be considered as an outlier. The protocol employed in our study mirrors that utilized by (71) and (72), both of which report average gap detection thresholds of 12.8 ms and 16.6 ms, focusing exclusively on binaural individuals with HL and the low-frequency conditions for equitable comparison. Participant HL2 displays thresholds approximately double the highest recorded threshold reported in these studies. This suggests that HL2 may possess a distinct temporal processing deficit, one that is not faithfully simulated by any of the simulations, which was intended to generate maximal temporal distortion. The SIMHL groups listening through the HL2 simulations exhibit the lowest thresholds within the study, approaching the range of thresholds observed in (71) and (72). This unexpected outcome indicates that the existing temporal distortion modules may not adequately incorporate relevant signal processing to emulate temporal resolution deficits, as evaluated through a gap detection in noise task. Furthermore, the substantial variability across SIMHL participants and ears suggests that these thresholds may be influenced more by participants' capacity to discern the auditory cues within the degraded stimuli than by the fidelity of the simulation in mimicking temporal processing deficits.

Magnitude Estimation: The findings from the magnitude estimation task were leveraged to deduce the parameters of Stevens' power law. The exponent values for HL1 and HL2 are determined to be 0.47 and 0.23, respectively. Both simulations encounter challenges in replicating the growth of the power function (i.e., the exponent) for HL2, where the value of 0.23 appears unusually low. It is notable that (69) posited a positive correlation between hearing threshold and the steepness of the estimated magnitude function, implying that higher exponents are associated with higher hearing thresholds. Surprisingly, participant HL2 possesses a lower hearing threshold compared to HL1 but exhibits a higher exponent.

Furthermore, the literature, as exemplified by (73) and (74), reports an exponent of 0.6 for the loudness curve of a 1-kHz tone in binaural listening and normal-hearing individuals. Given that both individuals with hearing loss in our study possess hearing thresholds surpassing those of normal-hearing individuals, one might anticipate exponent values exceeding 0.6. The discordance between the existing literature and our empirical findings may be attributed to the choice of reference stimuli. (69) demonstrated that the selection of reference stimuli can influence the growth of the magnitude function. In our study, the reference stimuli were set at 6 dBA (resulting in a gain coefficient of 0.5) below the maximum level. Participant HL2 necessitated a reduction in the maximum level due to loudness discomfort, implying that stimuli falling between the reference and maximum levels were likely situated near the upper limit of comfort. This might elucidate the plateaus observed in the estimated magnitude above the reference stimuli (corresponding to an actual gain of 0.5 on the x-axis) in Fig. 4.7. A similar pattern is observed for participant HL1 (Fig. 4.6), albeit with greater variation in estimated magnitude below the reference stimuli. In summary, the choice of reference stimuli may not have been ideal for HL2, and an approach incorporating sensation level (69) instead of absolute level might have yielded more favorable results. Therefore, results pertaining to participant HL2 should be interpreted with caution, as the test design may not align with their specific hearing capabilities.

For participant HL1 and the corresponding SIMHL groups, both simulations closely approximate loudness growth, with minimal percentage error, particularly for the 3DTI simulation, which boasts a more intricate loudness growth module than the QMUL simulation. Nevertheless, the QMUL simulation tends to overestimate the exponent, with the exponent derived from SIMHL2 data appearing anomalously high. The percentage error in the exponent would have been approximately 25% lower without SIMHL2 data, as opposed to the observed 42% reduction. This underscores the trade-off between complexity and efficiency, as the QMUL simulation performs nearly as well as the 3DTI simulation in this task, despite its lower intricacy.

Adaptive Sentence List: The measured SRTs demonstrate substantial variability, yielding inconsistent values across both hearing loss profiles and simulations. In most instances, HL participants outperform SIMHL participants, possibly attributable to the gradual adaptation and compensation for hearing deficits that individuals with real hearing loss develop over time. This adaptation is compounded by the fact that their hearing loss progresses gradually over the years, in contrast to the instantaneous hearing loss induced in normal-hearing listeners by the simulation. (75) have demonstrated the significance of hearing loss duration and age of onset as factors affecting speech intelligibility. At an equivalent degree of hearing loss, longer duration and earlier onset of hearing loss correlate with improved speech intelligibility. The experience of a normal-hearing individual listening through a hearing loss simulation can be likened, to some extent, to a sudden hearing loss. Consequently, the ability to adapt and compensate for a hearing deficit may exert a profound influence on the effectiveness of a simulation. This variability in measured SRTs and the superior performance of participants with actual hearing loss can be partially elucidated by this phenomenon.

The discrepancies between the two groups in the speech-in-noise task may stem from inaccuracies in the tuning of one or more components of the simulation. Most simulation modules are calibrated based on measurements taken at 1 kHz, with these parameters extrapolated across frequencies. This extrapolation, necessitated by practical constraints, may introduce additional disparities between HL and SIMHL participants when exposed to broadband stimuli. For instance, the widening of auditory filters with hearing loss, a phenomenon linked to the loss of outer hair cell functionality (76), is both frequency- and individual-dependent. Consequently, the approximations made during the calibration of modules related to frequency smearing may account for differences in results between HL and SIMHL groups in this specific task.

Another plausible explanation for these results pertains to the chosen procedure. The tasks undertaken by the SIMHL group, with the exception of the speech intelligibility test, entail comparisons between two intervals, necessitating the inclusion of a reference stimulus in addition to the target stimulus. In contrast, the speech intelligibility test provides no reference stimuli. Furthermore, as the sentences are degraded by the simulation, the auditory representation of words may diverge from the auditory experiences individuals encounter in daily life. These factors, albeit speculative, can elucidate the high variability in the measured SRTs obtained with the SIMHL group. A speech intelligibility test employing a closed-response set, where participants select the word they hear from a list, might have been a more suitable choice, as it would have provided reference stimuli.

An additional limitation arises from the fact that, among the participants in the SIMHL group, nine individuals had English as a non-native language. However, all SIMHL participants chosen for the study were proficient English speakers who had completed a university-level degree in the UK.

General Remarks on the Study: This study encountered significant challenges in recruiting a sufficient number of participants with the specific type of hearing loss under investigation. Despite diligent efforts to identify and enroll individuals with this specific profile, the sample size of this subgroup remained small, severely constraining the generalizability of our findings to the broader population of individuals with similar hearing loss characteristics. Consequently, the derivation of simulation parameters relied on data from a very limited dataset, raising valid concerns regarding the accuracy and representativeness of the simulated hearing loss profiles employed in the study.

Despite the recruitment challenges and the limited number of participants with the specific type of hearing loss, this study holds substantial value as one of the few investigations employing real-time simulations to replicate multiple aspects of hearing loss for comparing the performance of listeners with real and simulated hearing loss in psychoacoustic tasks. By adhering to rigorous methodologies, we managed to derive simulation parameters from the available, albeit limited, dataset of individuals with hearing loss. This enabled the exploration of simulated hearing loss in a larger cohort of participants. These findings enhance our comprehension of the validity of hearing loss simulations and can serve as a foundational framework for future research endeavours aimed at refining hearing loss simulations and enhancing the design of assistive technologies for individuals with hearing loss.

Chapter 5

Extracting Mixing Practices

5.1 Sound Engineer Recruitment

In order to gain insight into the mixing practices and adjustments that audio engineers would make when using the hearing loss simulation plugin in their work, two groups were recruited, one consisting of audio engineering students and one consisting of professional mixing engineers. The participants were tasked with mixing audio content while using the hearing loss simulation plugin, with the goal of documenting the mixing practices that they would use in their typical work settings. By observing and analyzing the mixing practices of these engineers, the study aimed to gain a deeper understanding of the impact of hearing loss on audio production and to identify potential strategies for mitigating the effects of hearing loss that could be later on used to design the proposed intelligent mixing approaches.

5.1.1 Tonmeister Student Mixes

This task was initially performed in collaboration with Surrey University, where it was also included in a Master's thesis. More specifically, final year Tonmeister students were recruited and asked to mix a set of 3 multi-tracks of varying content, once as they would normally mix and once while using a hearing loss simulation plugin in their master bus. The content for the mixes was a broadcast-type excerpt, a pop song as well as an orchestral song. A limitation that emerged during this phase of the study centered on the quality of the mixes created by the Tonmeister students. This issue was of significant importance, as the quality of these mixes would directly impact the validity and reliability of the entire research endeavor.

Upon close examination, it became clear that the mixing practices employed by the students did not align with established industry standards. In particular, their utilisation of equalisers and dynamic range compressors was not in accordance with the accepted norms and best practices in professional audio production, resulting in poorly processed audio.

The consequences of these suboptimal mixing practices were twofold. First and foremost, the resulting mixes were of noticeably poor quality. This compromised the integrity of the data collected during the experiment, as the output mixes were meant to serve as references for evaluating the impact of the hearing loss simulation on audio mixing.

Secondly, due to the low quality of the mixes, they could not be relied upon as a solid foundation for the development and refinement of the automatic mixing approaches.

As a result of these limitations, it became evident that further refinement of the experimental design was necessary. This included not only revisiting the criteria for participant selection but also providing clearer guidelines and training to ensure that the mixing practices adhered to industry standards.

5.1.2 Experienced Sound Engineer Mixes

To counteract the above-mentioned issue, another round of recruitment was put forth, this time recruiting industry-experienced mixing engineers. To ensure that the task would be focused and controlled, engineers were asked to use specific tools and limited capabilities and were provided with guidelines on how to perform the task (see Appendix A.2). More specifically, engineers were provided with a multi-track mixing session of a broadcast example using the digital audio workstation Reaper. The content comprised of 5 elements:

- narration

- child
- creature
- sound effects
- music

Engineers were instructed to use only two Reaper native audio effects plugins, an equaliser and a compressor plugin which were already placed on the tracks, as well as panning and levelling only, in order to complete the task. The engineers were instructed to first mix the content as they normally would in their practice and then activate the hearing loss simulation plugin already placed in the master bus and re-listen to their mix through the simulation. They were then instructed to perform the necessary adjustments in order to make the mix sound closer to the original mix when heard through the simulation. Participants were encouraged to bypass and listen to their mix without the hearing loss simulation plugin throughout the process, in order to ensure the mix retained its balance and were told to re-mix their content from the viewpoint of a sound engineer mixing for hearing loss and not as having the hearing loss themselves.

A total of three experienced mixing engineers performed the mixing task. The produced mixes were used to inform the intelligent mixing approaches.

Each of the engineers produced two separate mixes with two versions for each mix, one normal and one with the hearing loss simulation. The engineers were asked to provide their Reaper sessions along with the audio files for each session. For the purpose of developing and testing the genetic algorithm and knowledge-based approach, the plugin parameters of the broadcast mix example were extracted for both the HLS and normal mixes.

Results

In order to compare the techniques used by the engineers during the normal mix with those used in the simulated mix, the resulting mixes along with the effect parameters for each channel of the multitrack, were exported and analyzed for differences. The following section describes the key differences observed between the normal and HLS mixes produced by each of the three engineers.

Levels & Priority First, the fader levels were extracted for each track to determine the priorities assigned by each engineer to each of the sonic elements of the mix. Table 5.1 shows the fader values on each track for each of the three engineers.

Fader Values			
Track	**Engineer 1**	**Engineer 2**	**Engineer 3**
Narration	5	0	5.35
Child	4	-1.36	-6.93
Creature	4	-2.12	-7.51
Music	4	-2.7	-9.81
FX	2	-6.25	-8.26

Table 5.1: Fader values of each track for the three engineers.

The priority of each track as assigned by the three engineers based on the fader levels is depicted in Figure 5.1.

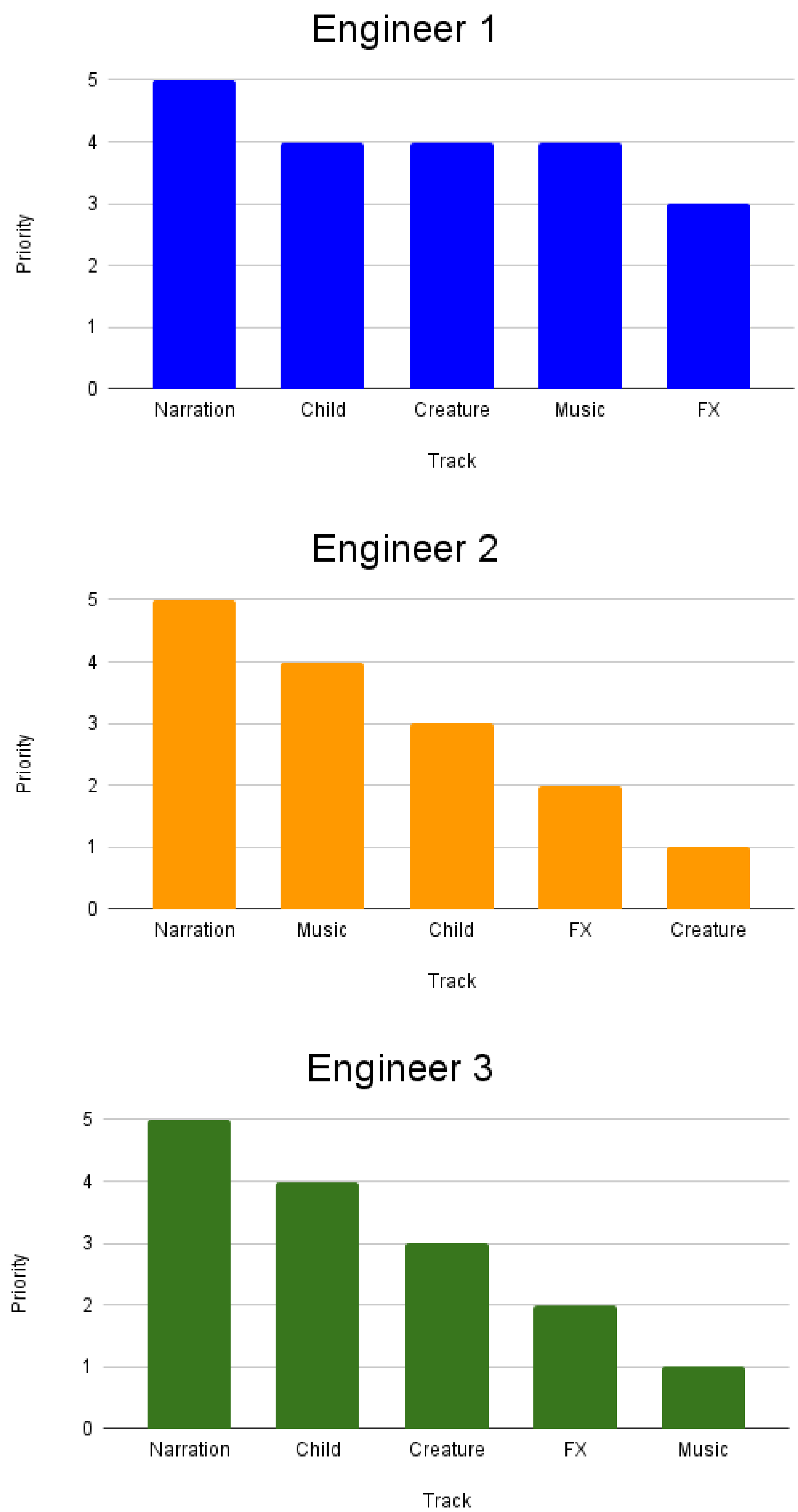

Figure 5.1: Track priority from 0 (low importance) to 5 (high importance) for each of the three engineers.

Equalisation First, the values of the equalisers used for the normal and HLS mixes were extracted for each of the 3 engineers. The equalisation data can be found in Table A.3 in Appendix.

For the "narration" track of the HLS mix, all three engineers enhanced the mid-high and high frequencies and employed low-cut filters. Engineer 1 and 3 used a more aggressive approach compared to Engineer 2 who used a more subtle enhancement.

Similar to the narration track, for the "child" track, all 3 engineers boosted the mid-high and high frequencies. Engineers 1 and 3 employed more dramatic boosts and low-cut filters whereas Engineer 2 only enhanced the high frequencies.

For the "creature" sounds, Engineer 1 and Engineer 2 follow approaches consistent with the narration, particularly focused on addressing the high-frequency hearing loss. Engineer 3 only applied a very subtle low-cut filter at 100 Hz.

For the "music" track, Engineer 1's approach was consistent with the prior content categories, primarily addressing the high-frequency loss, with an additional boost at 100 Hz in order to emphasize the low-frequency components of the music and potentially allow for some "spectral space" for the high priority elements of the mix. Engineer 2 employed a "bell-shaped" cut in the middle-high frequencies and a subtle high-frequency boost. Engineer 3 employed a more drastic high-frequency boost from 2kHz and above, along with a subtle bell-shaped mid-frequency cut and a low-cut.

In the case of the "FX" track, Engineer 1 and Engineer 2 followed strategies consistent with narration and creature sounds, emphasizing on high-frequency loss compensation with boosting the higher frequencies. Engineer 3 applied a more subtle low-cut and high-frequency boost with a small bell-shaped cut in the mid-high frequencies.

To examine the spectral differences between the normal mixes and the mixes made while using the simulation, the spectrograms of the two mixes were produced for the three engineers.

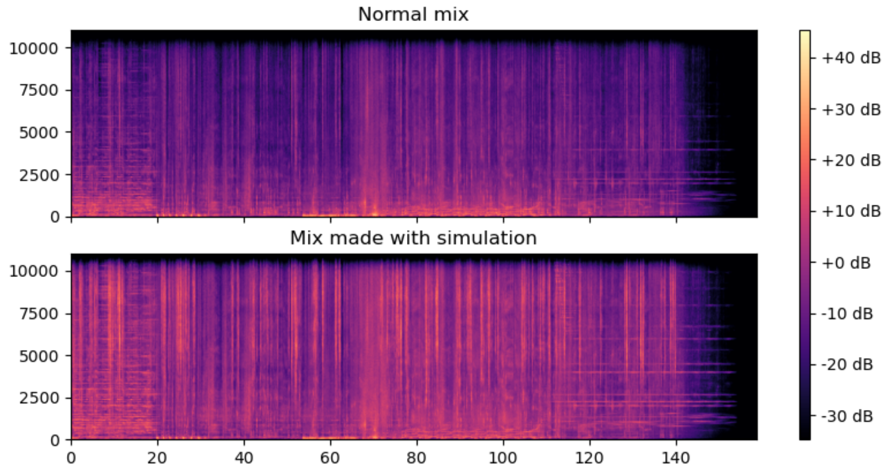

Figure 5.2: Spectrograms of the normal mix and the mix produced with the simulation for Engineer 1.

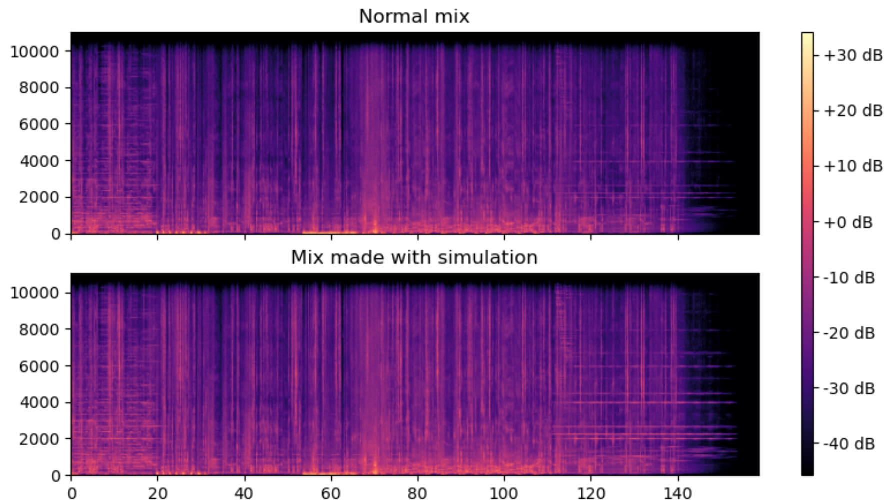

Figure 5.3: Spectrograms of the normal mix and the mix produced with the simulation for Engineer 2.

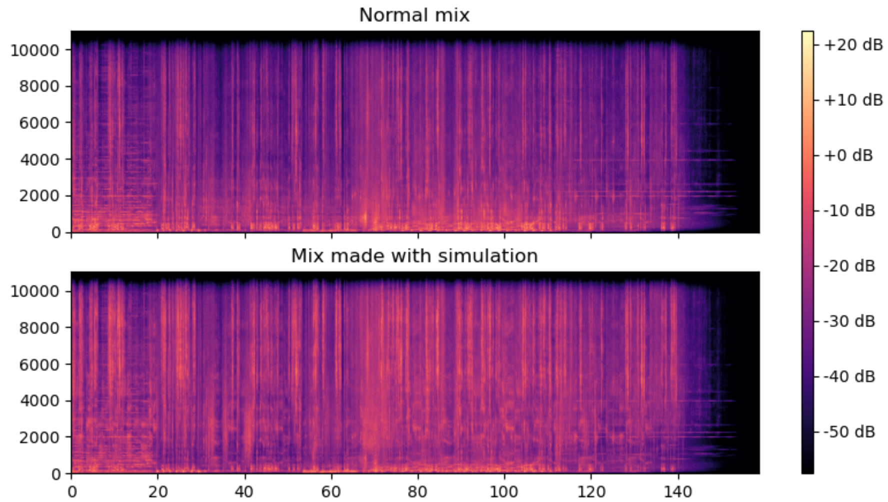

Figure 5.4: Spectrograms of the normal mix and the mix produced with the simulation for Engineer 3.

Compression A dynamic analysis of the resulting mixes was also performed. More specifically, to examine the differences between the normal and HLS mixes, the dynamic range of each mix was measured using the peak-to-loudness range. The resulting values for each mix are presented in table 5.2.

Engineer	Mix Version	Dynamic Range (dB)
Eng1	normal	39.2
Eng1	with simulation	33.3
Eng2	normal	40.1
Eng2	with simulation	39.1
Eng3	normal	35.3
Eng3	with simulation	34.1

Table 5.2: Dynamic ranges for the normal mixes and mixes performed through the simulation for the 3 engineers.

Discussion

In Table 5.2, a noteworthy observation emerges regarding the dynamic ranges of the audio mixes produced with and without the hearing loss simulation. It is evident that the dynamic ranges of the mixes created with the simulation are notably smaller in comparison to those produced without it. This divergence can be attributed to a key factor: the application of more pronounced compression techniques when working with the hearing loss simulation plugin. Compression data can be found in Table A.3 in Appendix.

This reduction in dynamic range can be seen as a strategic response by audio engineers to counteract the effects of the rapid loudness growth expansion unit inherent to the simulation. In an attempt to maintain a more consistent loudness profile throughout the mix, engineers may have found it necessary to employ compression more liberally, effectively narrowing the range between the loudest and softest elements within the audio tracks.

A deeper examination of the spectrograms, as depicted in Figures 5.2, 5.3, and 5.4, further illuminate the engineers' strategies in response to the hearing loss simulation. Notably, these spectrograms reveal that all three audio engineers took steps to enhance the high-frequency content of the mix when using the simulation compared to their normal mixing approach. This augmentation of high-frequency elements was likely a deliberate effort to counteract the perceptual effects of threshold shifts associated with hearing loss.

However, it's worth noting that there was variability in how the engineers approached this task. Two out of the three engineers demonstrated a degree of caution when boosting the high-frequency components in the mix. This cautious approach may have stemmed from their awareness of the potential risk associated with overemphasizing high frequencies. Such an overemphasis could result in an undesirable exaggeration of high-frequency distortion, particularly due to the presence of spectral smearing and temporal disruption units within the hearing loss simulation plugin.

While the mix data provided by the three sound engineers offer valuable insights for informing automated audio mixing techniques, it is imperative to acknowledge certain limitations inherent in drawing conclusions from this data. These limitations pertain primarily to the relatively limited number of participating engineers, which can hinder the extrapolation of findings to a broader context. Additionally, constraints on time resources imposed limitations on the number of mixes performed.

Moreover, an important consideration is the uniformity of tools employed by the engineers to conduct the mixing tasks. While this uniformity is beneficial for experimental control, it does not adequately account for potential variations arising from differences in the engineers' audio reproduction systems and individual hearing characteristics. These inherent diversities could introduce confounding factors that may influence the observed outcomes and should be taken into careful consideration when interpreting the results.

Chapter 6

Intelligent Mixing System Approaches

As described in the previous section, the next target of this research project is to investigate and implement audio enhancement methods, that can be used for designing an intelligent mixing model. The goal is to use the hearing loss simulation as an internal reference, in order to test various approaches of intelligent audio production that could be used to produce enhanced mixes for HL listeners, that would sound perceptually similar to how the original mix sounds to NH listeners while retaining the main elements and overall quality of the audio mix.

Two different approaches to aspects of intelligent audio production for hearing loss are presented in this chapter: an optimisation approach utilising a genetic algorithm approach to automatic equalisation, as well as a knowledge-based approach utilising fuzzy logic for automatic gain adjustment. The approaches were implemented and the strengths and weaknesses of each approach were presented along with the main results. Finally, a combined system utilising the two approaches together was designed.

6.1 Optimisation-Based Approach

6.1.1 Fundamentals of the Genetic Algorithm

The genetic algorithm (GA) is a metaheuristic optimisation algorithm that is based on the principles of natural selection and genetics and is commonly used to solve problems that are unable to be solved by traditional methods (77).

The genetic algorithm works by initiating a population of individuals representing a set of potential solutions, each assigned a fitness score based on its potential to be a good solution for the given problem. Individuals in the population that have the highest scores are selected for "reproduction" through "cross-breeding" in order to produce offspring sharing the best features from both "parents". This is repeated with the rest of the high-scoring individuals and a new generation is created with individuals possessing the best features of the previously selected "parents". The process is repeated for a selected number of generations until the system converges to the optimal solution (78).

Prior to running a GA, it is necessary to define the problem it is trying to solve in suitable coding. Potential solutions to the problem are represented as parameters or "genes" and a set of parameters is called a "chromosome". To solve a given problem, a fitness function needs to be defined which will evaluate the chromosomes. When a specific chromosome is evaluated by the fitness function, it produces a single numerical value that represents the individual's utility or ability. This value is intended to be directly proportional to the individual's overall fitness level (77).

Two important processes during the reproduction stage of a GA are crossover and mutation. Once two parents are selected, their chromosomes are recombined using techniques like crossover and mutation. Crossover involves cutting the chromosomes at a random point and swapping segments to create two new chromosomes, which are passed down to the offspring. Not all pairs of parents are subjected to crossover, and the likelihood of it happening is usually between 0.5 and 1. If crossover is not used, offspring are simply duplicated from their parents (79; 77).

Mutation is applied to each child after crossover, randomly altering genes with a low probability. It is commonly believed that crossover is a more crucial technique than mutation for efficiently exploring a search space. When a genetic algorithm has been properly executed, the population will progressively improve over successive generations, resulting in an increase in the fitness of both the best and the average individuals towards the global optimum (79; 77).

Convergence is the process by which the population becomes more uniform. A gene is said to have converged when all individuals in the population share the same value for that gene. The population is said to have converged when all of the genes have converged. The benefits of the GA approach include:

- Simple implementation that can be used to adjust plugins for easier integration into audio production
- Can handle constraints (e.g. equaliser gain values)
- Faster results compared to deep learning approaches
- Lack of need for large datasets as well as enhanced parameter tunability compared to black box approaches

Limitations of this approach include:

- Can be computationally expensive
- Is susceptible to premature convergence, this way producing suboptimal solutions
- Can be susceptible to local optima which causes the algorithm to fail to find the global optimum.

6.1.2 Genetic Equaliser Approach

This section will discuss the use of the GA towards effective audio enhancement for hearing loss. The problem that the GA is called to solve for this application was defined as the minimisation of the spectral differences between a raw input audio file going through the proposed hearing loss simulation and a target clean raw audio file. To achieve this, this approach was focused on reversing the spectral aspects of hearing loss imposed on the audio signal by the simulation.

More specifically, a class version of the hearing loss simulation plugin using the mild hearing loss profile was constructed in MATLAB and used to process the input file of a GA. The hearing loss simulation class features the same processing as the audio plugin, with the addition of a middle ear transfer function found in the "Matlab code for Calculation of the Loudness of Time-Varying Sounds Toolbox" (80) since the processing will only occur internally in the system. The proposed system implements a genetic algorithm to adjust the gains of a 1/3 octave graphic equaliser and utilises the Matlab system object standards-based graphic equaliser. Both the input and target files are derived from the mixing sessions performed by professional sound engineers, as described in Chapter 5. The system's input files are the raw unmixed audio files, which are then internally processed by the hearing loss simulation and the target audio files used in the optimization are processed stems obtained from the normal versions of the mixes performed by the engineers. The objective of the optimization is to adjust the gain values of the equaliser so that the processed signal from the hearing loss simulation matches the engineer-mixed stem made for normal hearing listeners.

Fitness Function To create the fitness function for the GA, the spectrogram loss method was utilised. More specifically, using the standards-based graphic equaliser MATLAB system object with the GA-derived parameters, the raw audio was processed and passed through the simulation and the similarity between the input and target audio files was compared by subtracting the two spectrograms and taking the Frobenius norm (81) of the resulting matrix.

The Frobenius norm of a matrix A is defined as:

$$\|A\|_F = \sqrt{\sum_{i=1}^{m}\sum_{j=1}^{n} |a_{ij}|^2} \tag{6.1}$$

where A is an $m \times n$ matrix, and $\|A\|_F$ denotes the Frobenius norm of A. This way the difference between the two sounds was summarized in a single numerical value and the goal of the GA is to minimize this value.

GA Parameters To implement this approach, a population of 100 individuals was initialised with a maximum number of generations set to 20 after several trials showed the system was not improving any further after the 20th generation. The mutation function selected was tournament selection and the mutation rate was set to 0.5. The crossover fraction was set to 0.6. These values were chosen to limit premature convergence observed in the first version of the system. Lower and upper bounds were set for each variable, with the upper bounds for the variables corresponding to frequency bands below 1000 Hz set to 5dB and the remaining variable bounds limits were set to 20dB, to avoid over-boosting the lower frequencies. The lower bounds were set to -20 dB for all variables. Figure 6.1 presents the spectrograms of the target stem audio and the GA equalised raw audio passing through the simulation.

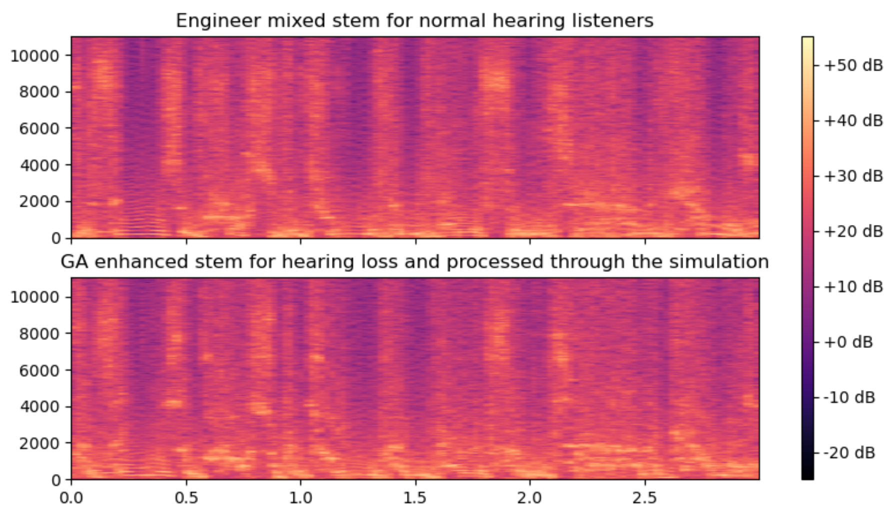

Figure 6.1: Spectrograms of the engineer mixed speech audio signal and the GA enhanced speech audio signal through the simulation.

The plot in Figure 6.2 shows the mean and best fitness values plotted over the generations of the genetic algorithm. The system appeared to achieve the closest approximation after twenty generations, with no significant improvement observed when increasing the number of generations.

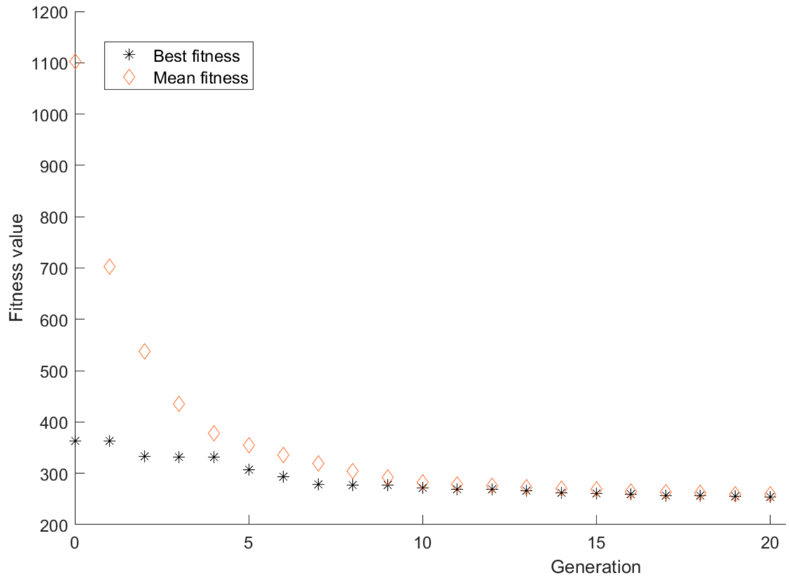

Figure 6.2: Mean and best fitness value of the genetic algorithm plotted over the number of generations.

Analytic tables showing the estimated gains across the center frequencies produced by the genetic equaliser can be found in section A.4 in Appendix.

Results from the proposed system, demonstrate the ability to successfully use the hearing loss simulation within the objective function of a GA to spectrally approximate a target signal. Additionally, the GA system can be easily implemented to control audio production tools such as the graphic equaliser used in the proposed system.

This result aligns with the findings in (82) as well as (14). In these studies, genetic algorithms (GAs) were suggested as effective tools for filter design, primarily focusing on coefficient estimation rather than parameter control. It's worth noting that employing GAs for parameter value estimation as in the proposed implementation might introduce additional constraints that could potentially limit their ability to approximate the optimal solution when compared to their use in filter design. However, this approach may facilitate a more seamless integration into existing audio processing tools. One of the limitations of this system is its longer processing times which is also observed in similar studies(82), which can be resolved by further increasing population size, allowing for a larger search space and a greater diversity of solutions or by utilising parallel computing in a multiple-core device. To improve this issue further work is required towards determining the best optimisation parameters or GA architecture for this specific application.

6.2 Knowledge-Based Approach

6.2.1 Fundamentals of Fuzzy Logic

Fuzzy logic is a concept presented by Lotfi Zadeh (83) and describes a type of mathematical logic that can be used when dealing with "fuzzy" or uncertain concepts where Boolean logic cannot be used. Fuzzy logic uses ambiguous or imprecise statements to simulate logical thinking, this way making it a powerful tool for simulating human behaviour. Particularly in the case of knowledge-based audio mixing systems, fuzzy logic can provide a way of translating mixing engineers' practices into a set of rules that can be then applied automatically to audio files, based on their specific characteristics as well as the mix requirements.

Fuzzy Architecture This section will discuss the components of the fuzzy system architecture as well as their role. The main components are listed below (84) :

- Rule Base, is the component that includes all the rules, membership functions, and conditionals the system relies on to make decisions
- Fuzzifier, is the system component, where crisp inputs are transformed into fuzzy sets, which are elements with a degree of membership.
- Inference Engine, where specific rules are applied to specific inputs. Rules are typically applied by following an "if-then" format, specifying the conditions under which the rule is applied as well as the resulting output.
- Membership Function, represents the degree of membership of a value in the set. There can be multiple types of membership functions including,

triangle, Gaussian, square and more, and the choice depends on the type of input. Examples of commonly used membership functions can be found in Figure 6.3

- Defuzzifier, where the fuzzy output sets are converted in the form of a crisp value. This marks the ultimate phase of a fuzzy logic system.

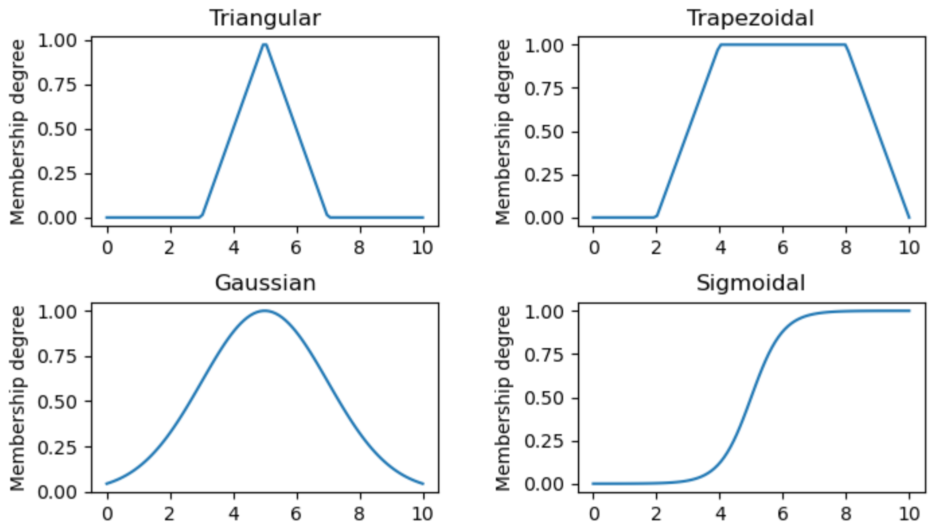

Figure 6.3: Examples of typical membership functions.

The general structure of a fuzzy logic system is presented in Figure 6.4.

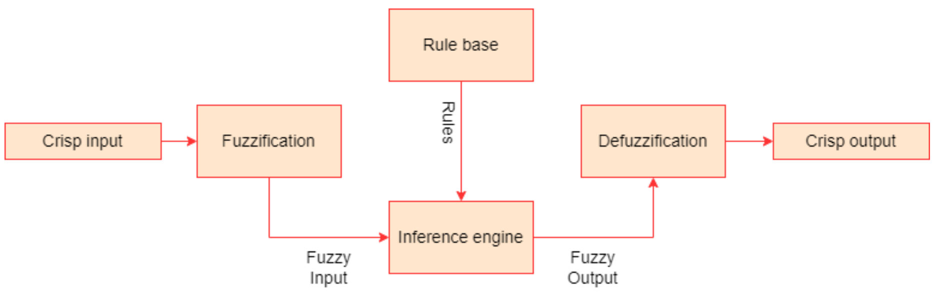

Figure 6.4: Fuzzy logic structure.

Some of the benefits of fuzzy logic used for a knowledge-based automatic mixing system include:

- The ability to handle uncertain data, which is a very useful concept in audio when working with audio descriptors that cannot be easily assigned a crisp value (e.g. bright, dull, boomy sound).
- Can be used to approximate human behaviour, which is very useful in audio production applications where decision-making is based on subjective or qualitative factors.
- Intuitive and easy to implement.

Limitations of fuzzy logic include:

- The performance of the system is highly dependent on the membership functions, which are difficult to choose.
- Can be computationally expensive
- In the case of audio production, they can require a large number of rules, which can make them complicated and difficult to debug.

6.2.2 Fuzzy Gain Approach

The following section describes the proposed fuzzy logic system in detail. The proposed fuzzy system is designed to automatically adjust the gain levels of five different audio tracks, obtained from the broadcast mix example described in Chapter 5, in this case: narration, child, creature, sound effects (sfx), and music. Each track is analyzed for its perceived loudness using the *integratedLoudness* function in Matlab, which provides a numerical value for the loudness of each track. The loudness values of each track are then averaged across the three different tracks of the three engineers who performed the mixes, in order to obtain an average loudness value for each track.

Next, the difference between the loudness of each track and the raw unprocessed audio loudness value is calculated. The reference loudness values are obtained from pre-mixed audio files, which are used as a benchmark for the desired loudness levels. These differences are then passed through a fuzzy logic system to determine the appropriate gain value for each track.

The fuzzy system takes into account the priority of each track, which is assigned by the user and is derived based on expert knowledge and standard broadcast mixing practices. The priority values are used as inputs along with the loudness differences to the fuzzy system. The fuzzy system then outputs a gain value for each track, which is then used to adjust the volume of the corresponding audio file. To create the fuzzy system a Mamdani-type, fuzzy inference system was implemented using the fuzzy logic designer application in MATLAB. Triangular membership functions were used for both the inputs as well as the output of the system. Figure 6.5 presents the surface plot of the implemented fuzzy gain system, a graphical representation of the system's output surface, which maps the input variables (priority, loudness) to the output variable (gain) based on the given set of fuzzy rules.

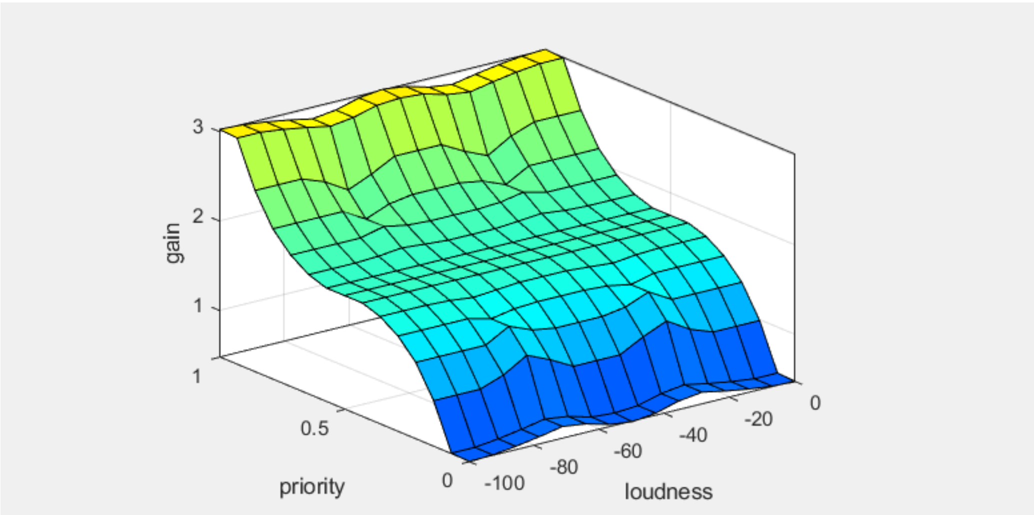

Figure 6.5: Surface plot of the fuzzy gain system.

Finally, the gain-adjusted audio files are mixed together to create a final audio mix. The final mixing is performed using a linear combination of the gain-adjusted audio files.

Table 6.1, presents a comparison of the average integrated loudness values of the five channels made by the engineers against the integrated loudness values of the channels produced by the fuzzy gain system. The error for each channel is calculated in per cent and given in the last column of the table.

Table 6.1: Integrated loudness values in dB FS for the engineer average and system-produced outputs, including absolute percentage error

Track	**Engineer Average**	**System**	**Abs Percentage Error (%)**
Narration	−22.87	−24.41	6.73
Child	−39	−40.17	3.00
Creature	−34.52	−24.35	29.58
FX	−38.06	−29.04	23.63
Music	−33.40	−34.80	4.20

The proposed system was able to approximate level balances between the five different tracks utilising the rule base constructed by the sound engineer data extracted in Chapter 5 as presented in (19). However, the approximations presented an average of approximately 13.8% error across the channels with *Creature* and *FX* channels being affected the most. These two channels were assigned medium priority by the engineers, which could indicate that the current system may present difficulty in assigning gains based on the priority membership function. The error could be improved by experimenting with different priority membership functions and priority assignment values.

6.3 Combined Approach

After testing the genetic equalization and fuzzy gain systems independently, a combined approach was adopted to create a more comprehensive system. A diagram of the final system is presented in Figure 6.6.

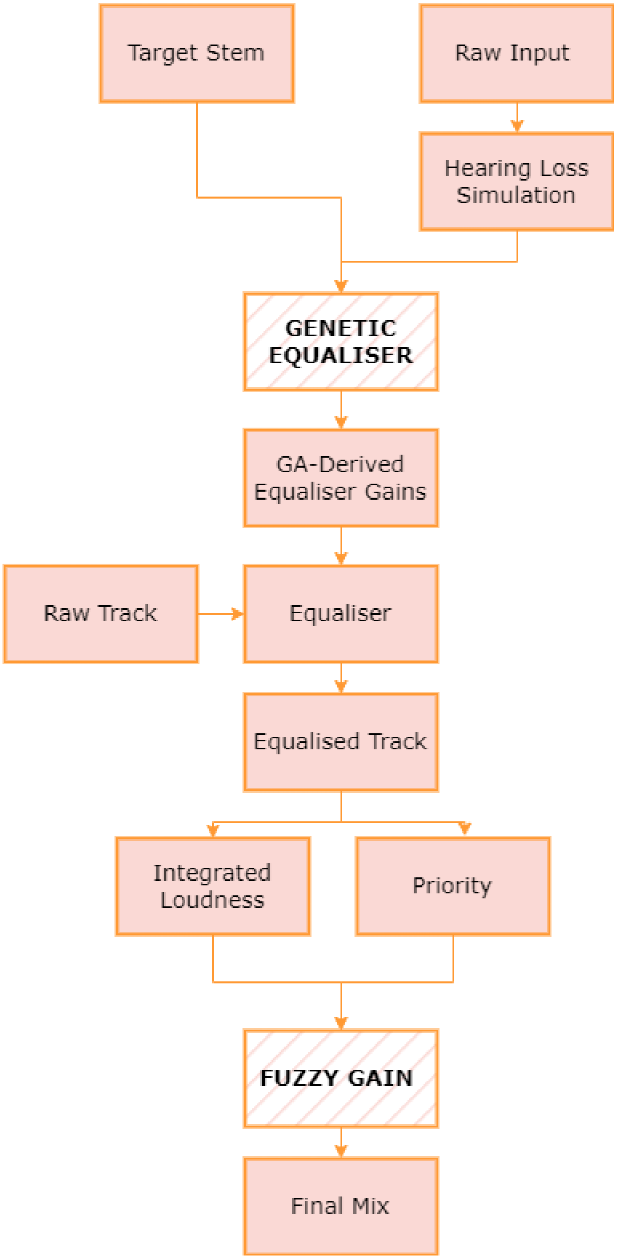

Figure 6.6: Combined system architecture.

First, the genetic algorithm is used to optimize the gain values of a 1/3 octave graphic equaliser for each stem of the target audio file. This is done by defining an objective function that measures the difference between the raw input and processed signal after applying the gain values. The optimization algorithm then searches for the gain values that minimize this difference. The optimized gain values are used to equalize each stem of the target audio file by applying the gain values to a 1/3 octave graphic equaliser using the *graphicEQ* function.

Next, the loudness levels of each stem are computed using the *integratedLoudness* function. The fuzzy logic system is then used to compute the gain values for each stem based on their priority and loudness levels. The priority levels are defined as a set of weights that determine the relative importance of each stem in the final mix. The loudness levels are used to adjust the gain values for each stem to achieve the desired loudness level for the final mix. These priority and loudness values are the inputs to the fuzzy inference system, which then outputs a gain value for each stem.

Finally, the gain values computed by the fuzzy logic system are used to mix the stems together to create the final mix. This is done by multiplying each stem by its corresponding gain value and adding the resulting signals together using the mix function.

When given the unprocessed audio tracks from the broadcast audio example, as well as the engineer-mixed stems as the target, the system was able to produce enhanced equalised stems which were then gain-mixed based on their loudness and user-defined priority. The final mix features high-frequency enhancement, as well as highlighting of the main sources. A spectrogram of the engineer-produced mix and the combined system-produced mix is presented in Figure 6.7

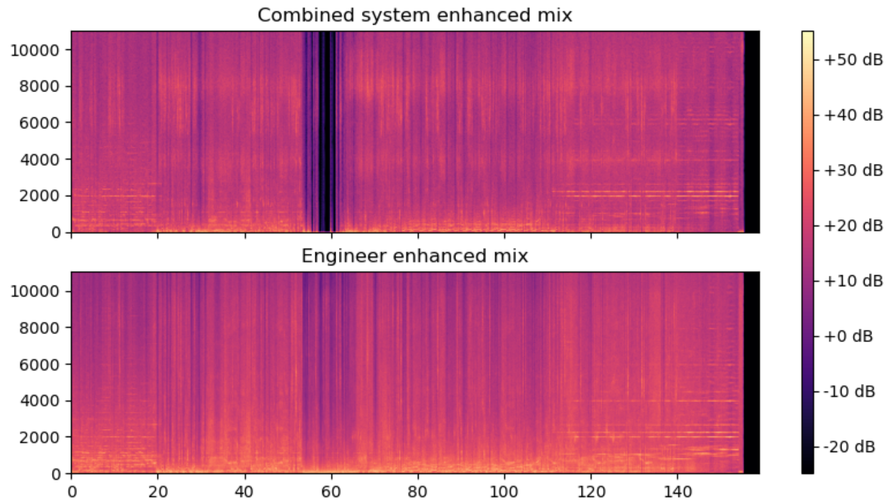

Figure 6.7: Spectrogram of the engineer-produced enhanced mix and the system-produced mix for hearing loss .

To measure the similarity between the engineer and system made mixes passed through the simulation with the mix produced for normal hearing listeners, the Mel-frequency cepstral coefficients (MFCC) of each track were produced and then compared for similarity using the cosine similarity metric. Table 6.2 shows the results of the cosine similarity mesaurements between the enhanced mix produced by Engineer 1 and the enhanced mix produced by the combined system passed through the simulation with the normal mix made by Engineer 1. The closer the number is to 1, the more similar the two audio files are in terms of their MFCC features. This comparison is important as it offers a perspective on how similar the enhanced mix would appear to listeners with hearing loss, in relation to how the normal mix would appear to normal hearing listeners, when evaluated using a perceptually-based metric.

Tracks Compared	**Cosine Similarity**
System enhanced mix through sim and Eng 1 Normal Mix	0.95
Eng 1 enhanced mix through sim and Eng 1 Normal Mix	0.97

Table 6.2: Cosine similarities between system and engineer enhanced mixes through the simulation compared to the normal mix made by Engineer 1.

Comparison of the combined system with the engineer produced stems when mixing through the hearing loss simulation, shows a more aggressive approach employed by the engineers compared to the system. More specifically, by examining the equalisation parameters found in Appendix sections A.3 and A.4, it can be observed that the engineers tend to employ higher boosts in the high frequencies compared to the genetic equaliser which appears to focus more on reducing the strength of the and mid-low frequencies. However it can be observed that the Engineer enhanced mix can approximate the normal mix more closely compared to the combined system enhanced mix, which however appears to be very close in performance as seen in Table 6.2.

6.4 Chapter Summary

This chapter proposed three different approaches to intelligent audio mixing for hearing loss. Each approach's fundamentals, design, methodology and evaluation were presented and the main benefits and drawbacks of each approach were discussed.

The first approach presented was that of GA-based equalisation. The GA system was able to incorporate the hearing loss simulation and approximate the engineer-produced target signal, however, limitations occurred in its accuracy and efficiency. Next, a fuzzy gain system was also presented, which was designed to automatically adjust the gain levels of multiple audio tracks based on expert knowledge derived from the professional engineers' mixes while using the hearing loss simulation, as well as perceived loudness levels, and then mix the gain-adjusted audio files together to create a final audio mix. The system was able to successfully incorporate expert knowledge and loudness information and derive the final audio mix. Finally, a combined system was implemented, integrating the GA equalisation with the fuzzy gain system. The combined system optimizes the gain values of a 1/3 octave graphic equaliser for each stem of the target audio file using a genetic algorithm. The fuzzy logic system then computes the gain values for each stem based on their priority and loudness levels, and the stems are mixed together using these gain values to create the final mix. The result is a final mix that is equalized and optimized for loudness and balance, based on expert knowledge and the characteristics of the target audio produced by professional sound engineers.

Chapter 7

Discussion & Conclusion

The field of audio enhancement for listeners with hearing loss has made significant strides in recent years, with several approaches described in chapter 2. Furthermore, the field of intelligent audio production is advancing rapidly, and its progress can be leveraged to improve audio enhancement for those with hearing loss. This can streamline the process and increase accessibility to wider audiences, without the need for additional personnel or working hours.

This thesis delved into a comprehensive study of the use of perceptually motivated intelligent audio mixing approaches for enhancing audio quality for individuals with hearing loss. The work presented encompasses a wide range of aspects, including the underlying background research, the motivation for pursuing this topic, the detailed design of the methodology, the implementation of the proposed approaches, as well as a thorough analysis of the results obtained.

The goal of the research presented in this thesis was to utilise intelligent audio production approaches towards improving the overall audio quality for individuals with hearing loss thus creating a more immersive and enjoyable listening experience.

7.1 Summary of Contributions

Highlights

- A fully working real-time hearing loss simulation plugin was developed, evaluated and used both in lab as well as real-life cases.
- The simulation was tested against a similar development as well as actual listeners with hearing loss, producing promising results on its effectiveness and accuracy.
- A series of automatic mixing approaches to audio enhancement for hearing loss were implemented and evaluated, and their key strengths and weaknesses were presented, paving the way for further exploration of the use of hearing loss simulations as components of audio enhancement implementations.

The scope of the research presented in this thesis was divided into two main areas of exploration: simulating hearing loss with digital signal processing and utilising the simulation to perceptually inform intelligent audio mixing approaches towards audio mixing for hearing loss.

The design, implementation and evaluation of a real-time audio effects plugin for hearing loss simulation were presented. More specifically, the various stages of its development were analysed from the prototype to the final version, and the key adjustments were presented along with the strengths and weaknesses of the final implementation. The simulation plugin in its final form was able to reproduce two degrees of hearing loss; mild and moderate, offering mute and bypass options for each ear, as well as customisable suprathreshold aspects including, spectral smearing, rapid loudness growth and temporal disruption, as well as the ability to process each of the two ears individually. Compared to similar developments in literature, the simulation was able to perform in real-time and in a format that could be easily implemented in most digital audio workstations, this way making it easier to use outside the lab and most importantly audio production.

To further enhance its usefulness, particularly in the field of machine learning, a differentiable version of the simulation was also designed and presented in A.5.

To evaluate the effectiveness and accuracy of the hearing loss simulation, two listening studies were conducted.

The first study recruited participants without hearing loss, separated them into two groups, and had them perform a series of psychoacoustic tasks and a speech-in-noise task, with one of the groups hearing through the hearing loss simulation plugin set to replicate mild symmetrical high-frequency hearing loss. The goal of this study was to measure the degradation in performance observed in the group using the hearing loss simulation.

Results from this study demonstrated a reduced performance in almost all of the tasks excluding the magnitude estimation, in which performance appeared to be similar for both groups. This result was attributed to inadequate performance of the rapid loudness growth module of the hearing loss simulation using envelope expansion, which was then modified to use upwards dynamic range expansion in order to achieve an improved approximation. Although the study was successful in demonstrating the decrease in performance using only normal hearing participants similar to existing studies in the field (23; 28; 32), the results cannot validate the simulation's accuracy against actual hearing losses.

The second study recruited two groups of participants: participants with normal hearing and participants with mild/moderate bilateral high-frequency hearing loss. The goal of this study was to compare the performance of the participants with hearing loss to that of participants without hearing loss who were using a hearing loss simulation. Participants with hearing loss were tested first. Based on their results, a profile of their hearing loss was derived which was then used to test participants from the normal hearing group. In this study, the proposed simulation was also compared with the 3DTI hearing loss simulation developed by the audio design experience team at Imperial College London, in collaboration with the Diana Group From the University of Malaga.

The listening study included a gaps-in-noise task to determine temporal resolution similar to the previous study, a notched noise task to determine frequency selectivity, a magnitude estimation task to determine the growth of loudness similar to the previous study and an adaptive sentence list task to determine the participants' ability to identify sentences in noise. Results showed that participants with simulated hearing loss using both implementations were able to achieve comparable results for the notched noise experiment as well as the magnitude estimation. However, the remaining tasks presented variable results.

There are several factors that could have influenced these results including poor approximation from both implementations due to DSP constraints, as well as the potential degree of training that occurs with long-term hearing loss as opposed to an instant temporary hearing loss imposed by the simulation. Results from the adaptive sentence list test were particularly varied, while it was observed that participants using either of the two simulations performed worse than participants with actual hearing loss. In addition to the degree of training mentioned above, it was also noted that for some participants in the simulated hearing loss group, English was not their native language, which could have also affected their performance in this task.

This study faced notable challenges in recruiting a satisfactory number of participants with the specific type of hearing loss under investigation. This limitation significantly restricts the extent to which our findings can be broadly applied to the larger population of individuals with similar hearing loss characteristics. Consequently, there are concerns regarding the accuracy and representativeness of the simulated hearing loss profiles used in the study, as they were derived from a notably limited dataset.

Moreover, the absence of guidelines and standardization in hearing loss simulation evaluation studies within the existing literature has not only influenced the study's design but has also posed challenges in comparing the obtained data with findings from similar studies.

Despite the recruitment and design difficulties, as well as the limited number of participants with the specific hearing loss type, this study retains substantial value as one of the rare investigations employing real-time simulations to recreate various aspects of hearing loss.

The hearing loss simulation designed and evaluated in this thesis was utilised to explore the practices and adjustments that experienced engineers perform to enhance their mixes for listeners with hearing loss. Tonmeister students from the University of Surrey were initially recruited to produce a normal and enhanced mix while listening through the hearing loss simulation, using a dataset of multitrack recordings. However, the quality of the resulting mixes was found to be inadequate. To address this limitation, professional sound engineers with mixing experience were recruited to produce a normal mix and a mix while listening through the simulation for two multitrack examples from the same dataset, a pop example and a broadcast example. The resulting mix sessions for the broadcast example were analyzed, and data were extracted to be used for the development of the fuzzy logic system. The processed stems from the normal mix session were utilized as targets for the genetic equalizer.

The extracted data from the professional sound engineers' sessions were analysed in order to identify and extract differences in production methods between the normal and enhanced mix. Results demonstrated that engineers were more likely to emphasise the high frequencies in the enhanced mixes compared to the normal ones, specifically in tracks considered to be of higher importance. Additionally, it was observed that the use of dynamic range compression was more prominent in the enhanced mixes, which could be attributed to the effect of rapid loudness growth applied by the simulation. Furthermore, the extracted data and practices from the engineers' mixes were used to facilitate the design and implementation of three intelligent audio mixing approaches.

Even though these results are of high importance towards understanding the effect of using a hearing loss simulation on the engineer's production approach, the small sample size limits the generalizability of the conclusions. It's important to acknowledge that the findings, while insightful, may not capture the full spectrum of methods and approaches that could be encountered in a larger, more diverse population of mixing engineers.

The first approach was that of optimisation and was tested through the use of a genetic algorithm for equalisation. The GA was used to solve the problem of minimising the spectral differences between a raw input audio file going through the proposed hearing loss simulation and a target clean raw audio file. A class version (a construct with its own properties and methods) of the hearing loss simulation plugin using the mild hearing loss profile was constructed in MATLAB and used to process the input file of a GA. The system successfully implemented a genetic algorithm to adjust the gains of a 1/3 octave graphic equalizer and utilised the Matlab system object standards-based graphic equalizer. Both the input and target files were derived from the broadcast audio mixing sessions performed by professional sound engineers while the fitness function for the GA was created using the spectrogram loss method.

Results from the proposed system demonstrated the ability to successfully use the hearing loss simulation within the objective function of a GA to spectrally approximate a target signal. One of the limitations of this system was longer processing times, which could be resolved by further increasing the population size or by utilising parallel computing in a multiple-core device. Additionally, further work is required towards determining the best optimisation parameters or GA architecture for this specific application as well as the implementation of alternative optimisation approaches including particle swarm optimisation that could offer a solution to the longer processing times. Finally, replacing the graphic equaliser with a parametric one in the system could yield a substantial improvement in the genetic algorithm's functionality. This adjustment would eliminate the limitations associated with the graphic equaliser, expanding the algorithm's scope for exploration and enabling targeted adjustments to specific frequencies with greater precision.

The next approach explored in this thesis was that of a knowledge-based system utilising fuzzy logic to implement a gain mixer. The proposed fuzzy logic gain mixer was designed to automatically adjust the gain levels of five different audio tracks: narration, child, creature, sound effects (sfx), and music, based on two inputs. The inputs were priority and integrated loudness and they were derived from analysing the stems produced by the professional sound engineers as described in Chapter 5. The loudness values of each track were measured using the *integratedLoudness* function in Matlab and the average loudness value for each track was obtained. The difference between the loudness of each track and the raw unprocessed audio loudness value was calculated and passed through a fuzzy logic system to determine the appropriate gain value for each track. The fuzzy system also took into account the priority of each track assigned by the user and would then output a gain value for each track. The gain-adjusted audio files were then mixed together to create a final audio mix. This approach presents an innovative use of the fuzzy logic algorithm in a knowledge-based system, with promising results towards automatic gain adjustment for audio mixing. The incorporation of user-assigned priorities allows for a level of control for the user and can be further automated by utilising importance metadata as seen in (46). This approach's performance could be further improved through the exploration of alternative loudness metrics, as well as more sophisticated membership functions for improved mapping.

Finally, the optimisation and knowledge-based approaches were combined to implement a genetic equaliser and fuzzy logic gain mixer system. The proposed solution combined both the optimization and knowledge-based approaches to create a hybrid system that included a genetic equalizer and fuzzy logic gain mixer. This system first equalized the audio stems according to a target input and then passed the processed files through the fuzzy logic gain mixer, which incorporated user-assigned priorities to generate a final mix. The system was able to effectively approximate the engineer-processed stems and produce a final mixture that took into account integrated loudness and priority information.This hybrid approach introduces both a fully automatic target-based equalisation, as well as user-centric integration through the use of the fuzzy logic gain mixer. Further advancements to this system include the introduction of automatic panning and compression. These enhancements will serve to complete the automation of traditional audio engineering production methods(47), while effectively addressing all of the effects introduced by the hearing loss simulation.

7.2 Further Work

While this thesis has yielded promising results in the area of utilising perceptually motivated intelligent audio-mixing for listeners with hearing loss, several important areas for future work and improvement in the proposed audio enhancement systems can be highlighted.

First, with regard to the evaluation of the hearing loss simulation, it is important to conduct more extensive listening tests with a larger and more diverse sample of individuals representing various types of hearing loss. This broader scope will enable a more accurate representation of the nuances associated with different hearing losses.

Additionally, addressing the spatial aspect of hearing loss should be a focal point for future development of the hearing loss simulation. This enhancement can contribute to a more comprehensive and realistic simulation of hearing impairments, taking into account the spatial implications of hearing loss.

With regard to extracting mixing practices, expanding the dataset by involving multiple sound engineers in the creation of mixes using the simulation is vital. This will provide a wealth of additional practices that can better inform the development of the automatic mixing models. The collaborative input of different professionals will enrich the database of practices and contribute to a more robust system.

The current automatic approaches, including the genetic equaliser and fuzzy logic gain mixer, should be subject to ongoing improvement. Optimizing their performance and efficiency is essential to provide more effective and precise audio enhancement. Furthermore, the exploration of machine learning and deep learning techniques is another important area for future work. These technologies, specifically employing the differentiable version of the simulation, should be considered for their potential to enhance the accuracy and automation of the audio processing.

Developing and designing appropriate loss functions that are informed by perceptual considerations is a significant research avenue. These loss functions can ensure that the machine learning models prioritize perceptual accuracy, which is essential for individuals with hearing loss.

In addition to objective metrics, such as spectral differences, the incorporation of subjective listening tests, including listener preference studies, is essemtial. These tests offer insights into how individuals with hearing loss perceive various audio enhancements, ultimately guiding the creation of more personalized and effective audio processing techniques.

A Appendix

A.1 Questionnaire for participants taking part in the accuracy assessment study described in Chapter ??

Imperial College London

Questionnaire about your hearing

Full Title of Project: Assessment of immersive audio for AR/VR interactions – Part 2: measurement and assessment

Name of Principal Investigator: Dr Lorenzo Picinali

This questionnaire purposes to complete the hearing screening. The questions are in line with the British Society of Audiology guidelines. If you prefer not answering to any question, leave the response box empty.

Question	Response
1. Have you been exposed to any loud sounds in the last 24 hours? *'Loud' can be determined by having to shout or use a raised voice to communicate at a distance of 1 metre or 3 feet.*	
2. Have you had surgery on your ears?	
3. Have you had a recent infection on your ears?	
4. Do you have tinnitus?	
5. Have you ever had a hearing test before?	
6. Do you have a better ear?	
7. Do you usually wear hearing aids? If so, in what kind of situations?	

______________________	______________	______________
Name of participant	Signature	Date
______________________	______________	______________
Name of person taking consent (if different from Principal Investigator)	Signature	Date
______________________	______________	______________
Principal Investigator	Signature	Date

1 copy for participant; 1 copy for Principal Investigator

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A.2 Mixing task guidelines provided to the professional mixing engineers.

Mixing Task Guidelines

Thank you for taking part in our mixing task! Below you will find a guideline for setting up and completing the task. If you have any questions or require any clarification about the task, please contact: **a.mourgela@qmul.ac.uk**

NOTE: PLEASE READ EVERYTHING ON THIS GUIDELINE BEFORE PROCEEDING TO THE TASK AS IT CONTAINS IMPORTANT INFORMATION ON SAFETY.

STEP 1 – SETTING UP

Before beginning the task you need to set up you workstation. If you don't already have it, please download and install Reaper. Reaper is free to use and fully functional on evaluation mode. You can download it from here:

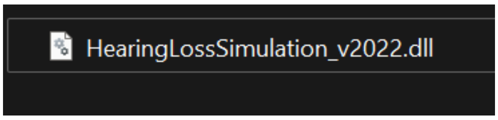
https://www.reaper.fm/download.php

Once Reaper is on your computer download the task materials zipped folder from your email that was sent to you.

Inside the compressed package you downloaded, you will find 2 folders named:

- "Plugin"
- "Mix Materials"

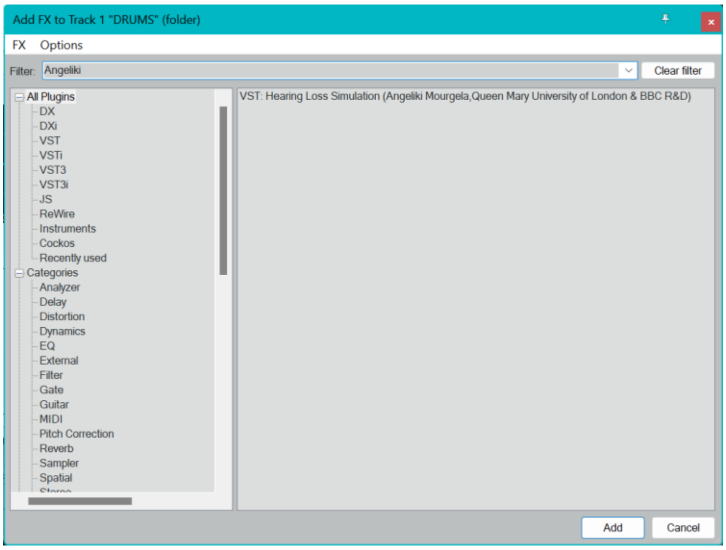
Open the folder named "Plugin" and you should see the following file:



This is an audio effects plugin that you will need to first add to your plugins folder before proceeding to the task. To do so, copy the file, go to your computer's hard drive, then Program Files, and find the vst plugins folder you are currently using. This could be under the name "VST Plugins", or under Steinberg>Vst Plugins, or if it doesn't exist create one and name it "VST Plugins"(if you do this you will need to add the folder to the vst plugin path in Reaper, the process is explained below).

Paste the file in your plugins directory. You may be asked if you want to perform this as administrator, if so click yes.

Now you should be able to see this plugin on Reaper's plugin list when you start the software, click on a track's FX button, and search for its name:

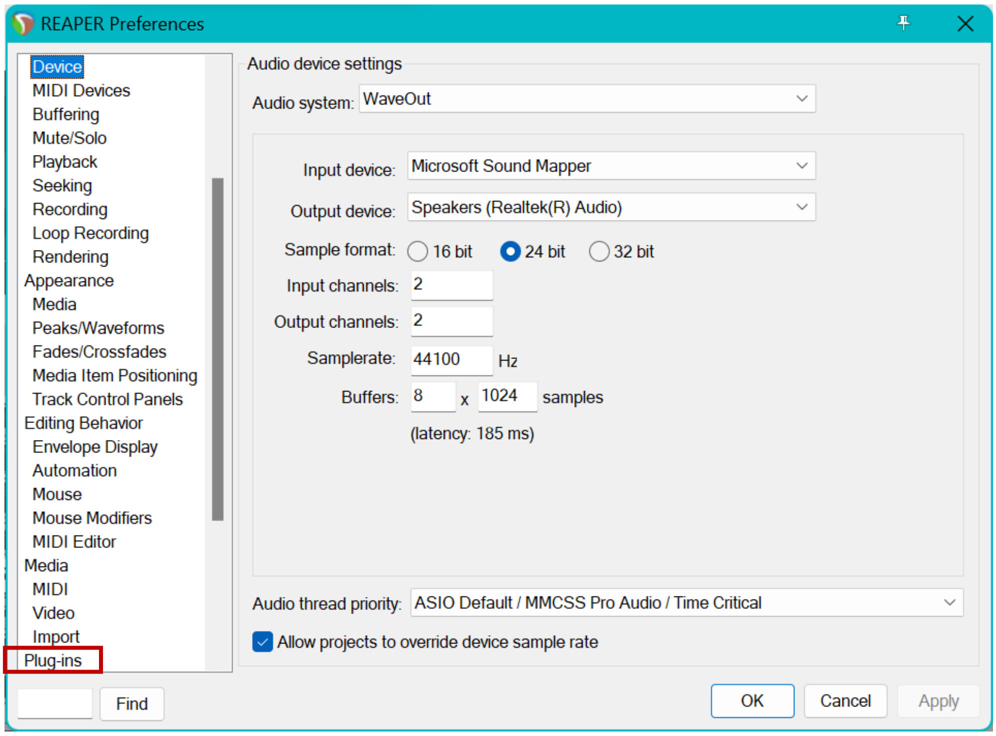


Troubleshooting:

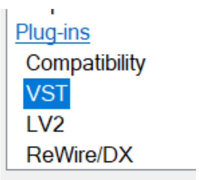
“I can’t see the plugin on the list”

Click “Control+P” and it will bring up the preferences menu

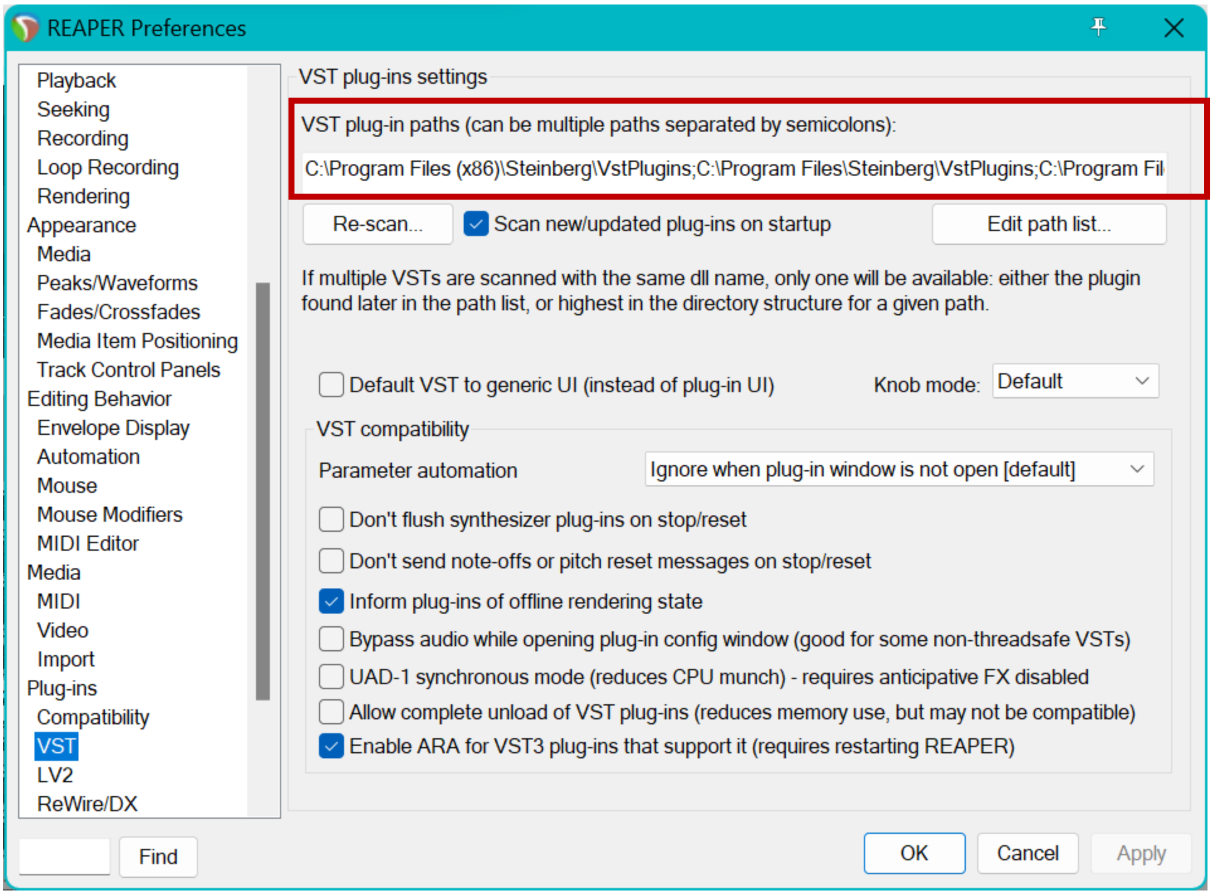
Go to the section Plug-ins



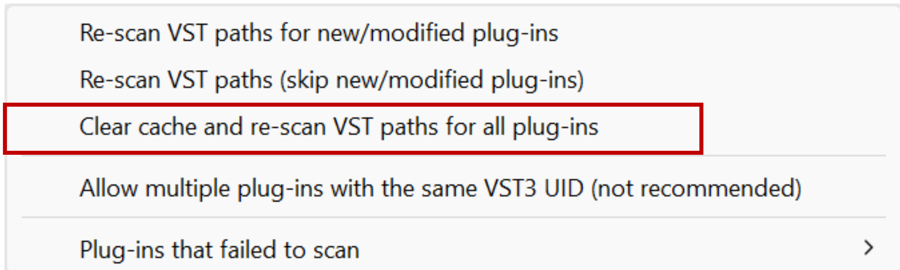
Then VST



This will bring up the following menu:



Make sure that the path contains the folder that you copied the vst plugin into, if not add the folder and click **Re-scan** > Clear cache and re-scan VST paths for all plug-ins



This should fix the issue, if not try to restart your computer and repeat the process.

If the issue persists, contact: a.mourgela@qmul.ac.uk

STEP 2: DOING THE TASK

Now that Reaper is setup and the plugin shows up in the FX list, you are ready to proceed to the mixing task. The total duration of the task depends on how much time you would like to spend on it, however each of the 2 songs should not take longer that 40 minutes to complete with both tasks considered.

To do this task you need to have access to a good and functional set of monitoring speakers and/or headphones and a quiet room.

To complete the task, you need to mix 2 different songs using the following guidelines:

THINGS YOU CAN DO:

- You can only use Reaper's native equaliser and compressor plugins that can be found on the tracks already.
- If you don't want to use either of these on certain tracks, you can bypass them.
- You can also use fader adjustments to modify the volume of each track as well as panning.

For the EQ:

- You can add as many bands and types of filters as you'd like
- You can use all of the parameters provided on the plugin to adjust your filters

For the compressor:

- You can adjust the ratio, threshold, attack and release values as well as the make-up gain
- You cannot adjust the knee size, pre-comp, low pass, high – pass and RMS size or change the detector input. No side chaining is allowed for this task as well.

THINGS YOU CAN NOT DO:

- You can not use automation.
- You cannot mute tracks.
- You cannot edit the items (e.g., chop, split, quantize, normalize items, change item's volume)
- You cannot adjust the master bus fader (always at 0)
- You cannot add any other plugins on the master bus (there is a hearing loss simulation plugin there already please leave it as is)

MAIN TASK PART 1

For this task you are asked to mix each of the to songs given in the mix materials folder using the sessions provided. Please copy the mix sessions in a separate folder on your computer and rename them as "Mix1_normal" and "Mix2_ Normal".

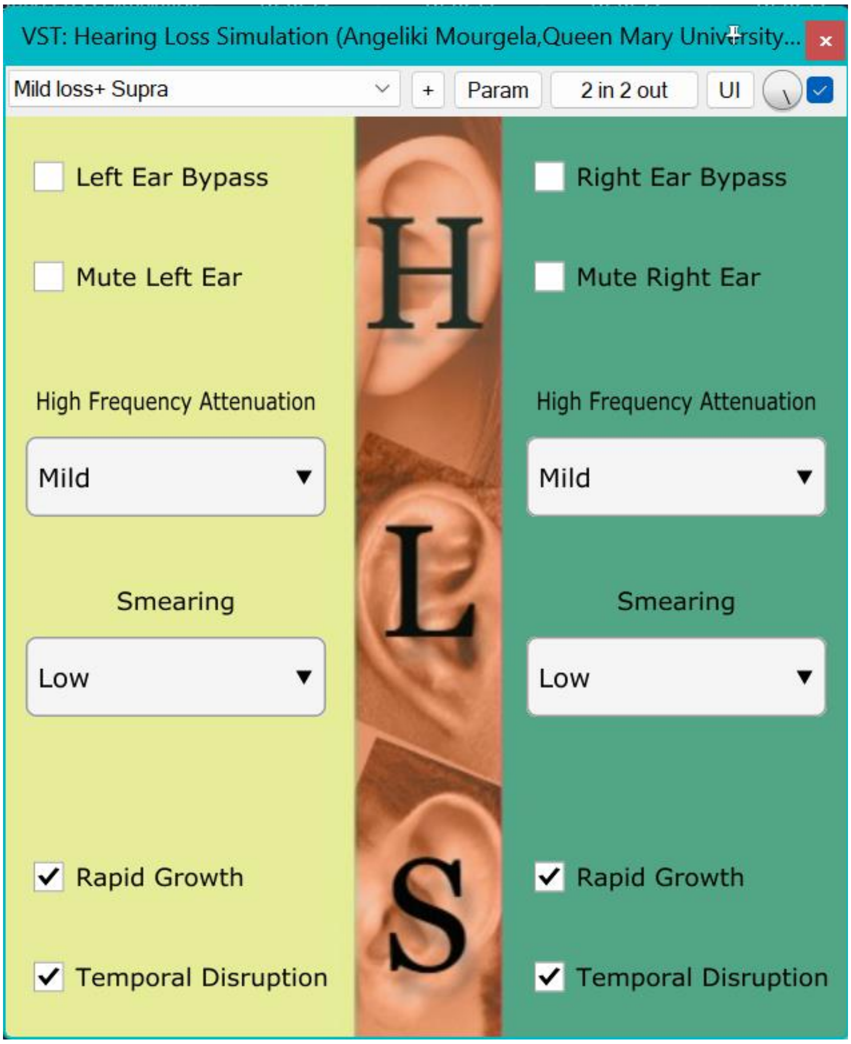
When opening the sessions, you will notice a bypassed plugin on the master bus, please do not activate the plugin for the first part of the task

Once you have finished your mixes for the two songs make sure to save them and then make another copy of the two sessions you just mixed (not the original unmixed ones) in a separate folder and rename them as "Mix1_hls" and "Mix2_hls".

MAIN TASK PART 2

Open the new sessions (Mix1_hls, Mix2_hls) and go to your master bus, where you will find the bypassed hearing loss simulation plugin. Activate the plugin and copy the following settings if not there already:

- High frequency attenuation (Mild on both ears)
- Smearing (Low on both ears)
- Rapid Loudness Growth (Checked on both ears)
- Temporal disruption (Checked on both ears)



Now listen to your mix again through the hearing loss simulation plugin. Your task is to identify the changes that are necessary for your mix to sound closer to your original mix, for a listener that has this type of hearing loss.

You can adjust the settings on all the plugins you used for your previous mix to reflect those changes. You can also activate any of the existing plugins that you may have bypassed for the previous mix.

PLEASE NOTE: You cannot add any more plugins.

Key things to remember:

You are not performing this task as a person that has hearing loss, but as a person mixing for listeners with hearing loss, therefore you can bypass and re-activate the hearing loss simulation plugin as much as you want while you make your adjustments, in order to ensure that your mix is balanced after the modifications. The goal for this plugin is to be used by you as a reference and make you aware of the difficulties that arise when a person with hearing loss hears your original mix.

CAUTION: The hearing loss simulation plugin attenuates the sound when activated, so make sure to adjust your speaker volume before bypassing it to avoid excessive levels. At no point should your mixing volume exceed safe levels. Please make sure to take frequent breaks to rest your ears as needed.

Once your two new mixes are done make sure to save them, compress them and upload them along with your previous two mixes on the following directory:

https://drive.google.com/drive/folders/1Z_Yq5vKoBlfq7axxhXGPaJf19IWjpX_7?usp=sharing

The final submission should include your four mix sessions as Reaper sessions along with their audio files in separate folders. You don't have to render your mixes.

Thank you again for taking part in this task!

A.3 Extracted equalisation and compression and data for the broadcast mix example.

Engineer	Mix	Track	Effect Type	Parameter Name	Parameter Value
1	Normal	Narration	Equaliser	Freq-Low Shelf	63.5
1	Normal	Narration	Equaliser	Gain-Low Shelf	-120
1	Normal	Narration	Equaliser	Q-Low Shelf	2
1	Normal	Narration	Equaliser	Freq-Band 2	116.4
1	Normal	Narration	Equaliser	Gain-Band 2	-5
1	Normal	Narration	Equaliser	Q-Band 2	0.41
1	Normal	Narration	Equaliser	Freq-Band 3	192.2
1	Normal	Narration	Equaliser	Gain-Band 3	3.7
1	Normal	Narration	Equaliser	Q-Band 3	2
1	Normal	Narration	Equaliser	Freq-Band 4	6115.2
1	Normal	Narration	Equaliser	Gain-Band 4	-3.6
1	Normal	Narration	Equaliser	Q-Band 4	0.36
1	Normal	Narration	Equaliser	Freq-Band 5	10411.4
1	Normal	Narration	Equaliser	Gain-Band 5	1.1
1	Normal	Narration	Equaliser	Q-Band 5	2
1	Normal	Narration	Equaliser	Global Gain	0
1	Normal	Narration	Equaliser	Bypass	normal
1	Normal	Narration	Equaliser	Wet	100
1	Normal	Narration	Equaliser	Delta	normal
1	HLS	Narration	Equaliser	Freq-Low Shelf	56.8
1	HLS	Narration	Equaliser	Gain-Low Shelf	-120
1	HLS	Narration	Equaliser	Q-Low Shelf	2
1	HLS	Narration	Equaliser	Freq-Band 2	129.2
1	HLS	Narration	Equaliser	Gain-Band 2	4.7
1	HLS	Narration	Equaliser	Q-Band 2	2
1	HLS	Narration	Equaliser	Freq-Band 3	1673
1	HLS	Narration	Equaliser	Gain-Band 3	0.1
1	HLS	Narration	Equaliser	Q-Band 3	2
1	HLS	Narration	Equaliser	Freq-Band 4	12071.5
1	HLS	Narration	Equaliser	Gain-Band 4	28.3
1	HLS	Narration	Equaliser	Q-Band 4	3.19
1	HLS	Narration	Equaliser	Freq-Band 5	182.4
1	HLS	Narration	Equaliser	Gain-Band 5	-2
1	HLS	Narration	Equaliser	Q-Band 5	2.4
1	HLS	Narration	Equaliser	Freq-Band 6	395.1
1	HLS	Narration	Equaliser	Gain-Band 6	-2.8
1	HLS	Narration	Equaliser	Q-Band 6	1.59
1	HLS	Narration	Equaliser	Global Gain	-0.1
1	HLS	Narration	Equaliser	Bypass	normal
1	HLS	Narration	Equaliser	Wet	100
1	HLS	Narration	Equaliser	Delta	normal
2	Normal	Narration	Equaliser	Freq-High Pass 1	96.5
2	Normal	Narration	Equaliser	Gain-High Pass 1	0
2	Normal	Narration	Equaliser	Q-High Pass 1	2
2	Normal	Narration	Equaliser	Freq-Band 2	255.2

2	Normal	Narration	Equaliser	Gain-Band 2		2.1
2	Normal	Narration	Equaliser	Q-Band 2		1.2
2	Normal	Narration	Equaliser	Freq-Band 3		1090.5
2	Normal	Narration	Equaliser	Gain-Band 3		-1.7
2	Normal	Narration	Equaliser	Q-Band 3		0.4
2	Normal	Narration	Equaliser	Freq-High Shelf 4		11657
2	Normal	Narration	Equaliser	Gain-High Shelf 4		7.1
2	Normal	Narration	Equaliser	Q-High Shelf 4		2
2	Normal	Narration	Equaliser	Freq-Band 5		2926.2
2	Normal	Narration	Equaliser	Gain-Band 5		3
2	Normal	Narration	Equaliser	Q-Band 5		0.8
2	Normal	Narration	Equaliser	Freq-Band 6		7137.3
2	Normal	Narration	Equaliser	Gain-Band 6		-3
2	Normal	Narration	Equaliser	Q-Band 6		0.8
2	Normal	Narration	Equaliser	Global Gain		0
2	Normal	Narration	Equaliser	Bypass	normal	
2	Normal	Narration	Equaliser	Wet		100
2	Normal	Narration	Equaliser	Delta	normal	
2	HLS	Narration	Equaliser	Freq-High Pass 1		96.5
2	HLS	Narration	Equaliser	Gain-High Pass 1		0
2	HLS	Narration	Equaliser	Q-High Pass 1		2
2	HLS	Narration	Equaliser	Freq-Band 2		255.2
2	HLS	Narration	Equaliser	Gain-Band 2		1.4
2	HLS	Narration	Equaliser	Q-Band 2		1.2
2	HLS	Narration	Equaliser	Freq-Band 3		1090.5
2	HLS	Narration	Equaliser	Gain-Band 3		-1.7
2	HLS	Narration	Equaliser	Q-Band 3		0.4
2	HLS	Narration	Equaliser	Freq-High Shelf 4		11463.6
2	HLS	Narration	Equaliser	Gain-High Shelf 4		9.1
2	HLS	Narration	Equaliser	Q-High Shelf 4		2
2	HLS	Narration	Equaliser	Freq-Band 5		2926.2
2	HLS	Narration	Equaliser	Gain-Band 5		3.7
2	HLS	Narration	Equaliser	Q-Band 5		0.8
2	HLS	Narration	Equaliser	Freq-Band 6		7380.6
2	HLS	Narration	Equaliser	Gain-Band 6		0.4
2	HLS	Narration	Equaliser	Q-Band 6		0.8
2	HLS	Narration	Equaliser	Global Gain		0
2	HLS	Narration	Equaliser	Bypass	normal	
2	HLS	Narration	Equaliser	Wet		100
2	HLS	Narration	Equaliser	Delta	normal	
3	Normal	Narration	Equaliser	Freq-High Pass 1		226.6
3	Normal	Narration	Equaliser	Gain-High Pass 1		0
3	Normal	Narration	Equaliser	Q-High Pass 1		2
3	Normal	Narration	Equaliser	Freq-Band 2		186.1
3	Normal	Narration	Equaliser	Gain-Band 2		4.9
3	Normal	Narration	Equaliser	Q-Band 2		2
3	Normal	Narration	Equaliser	Freq-Band 3		797

3	Normal	Narration	Equaliser	Gain-Band 3		-3.8
3	Normal	Narration	Equaliser	Q-Band 3		1.63
3	Normal	Narration	Equaliser	Freq-High Shelf 4		3600.5
3	Normal	Narration	Equaliser	Gain-High Shelf 4		0.2
3	Normal	Narration	Equaliser	Q-High Shelf 4		0.97
3	Normal	Narration	Equaliser	Freq-Band 5		3738.5
3	Normal	Narration	Equaliser	Gain-Band 5		-0.8
3	Normal	Narration	Equaliser	Q-Band 5		2
3	Normal	Narration	Equaliser	Global Gain		0
3	Normal	Narration	Equaliser	Bypass	normal	
3	Normal	Narration	Equaliser	Wet		100
3	Normal	Narration	Equaliser	Delta	normal	
3	HLS	Narration	Equaliser	Freq-High Pass 1		174.7
3	HLS	Narration	Equaliser	Gain-High Pass 1		0
3	HLS	Narration	Equaliser	Q-High Pass 1		2
3	HLS	Narration	Equaliser	Freq-Band 2		1040.8
3	HLS	Narration	Equaliser	Gain-Band 2		-11.9
3	HLS	Narration	Equaliser	Q-Band 2		2
3	HLS	Narration	Equaliser	Freq-Band 3		540.4
3	HLS	Narration	Equaliser	Gain-Band 3		-11
3	HLS	Narration	Equaliser	Q-Band 3		1.63
3	HLS	Narration	Equaliser	Freq-High Shelf 4		3986
3	HLS	Narration	Equaliser	Gain-High Shelf 4		7.9
3	HLS	Narration	Equaliser	Q-High Shelf 4		0.97
3	HLS	Narration	Equaliser	Freq-Band 5		2713.1
3	HLS	Narration	Equaliser	Gain-Band 5		1.1
3	HLS	Narration	Equaliser	Q-Band 5		2
3	HLS	Narration	Equaliser	Freq-Band 6		18217.6
3	HLS	Narration	Equaliser	Gain-Band 6		6.4
3	HLS	Narration	Equaliser	Q-Band 6		2
3	HLS	Narration	Equaliser	Global Gain		0
3	HLS	Narration	Equaliser	Bypass	normal	
3	HLS	Narration	Equaliser	Wet		100
3	HLS	Narration	Equaliser	Delta	normal	
1	Normal	Child	Equaliser	Freq-Low Shelf		70.8
1	Normal	Child	Equaliser	Gain-Low Shelf		-120
1	Normal	Child	Equaliser	Q-Low Shelf		2
1	Normal	Child	Equaliser	Freq-Band 2		300
1	Normal	Child	Equaliser	Gain-Band 2		0
1	Normal	Child	Equaliser	Q-Band 2		2
1	Normal	Child	Equaliser	Freq-Band 3		820.6
1	Normal	Child	Equaliser	Gain-Band 3		-1.1
1	Normal	Child	Equaliser	Q-Band 3		2
1	Normal	Child	Equaliser	Freq-High Shelf		2076.4
1	Normal	Child	Equaliser	Gain-High Shelf		2.2
1	Normal	Child	Equaliser	Q-High Shelf		2
1	Normal	Child	Equaliser	Global Gain		0

1	Normal	Child	Equaliser	Bypass	normal	
1	Normal	Child	Equaliser	Wet		100
1	Normal	Child	Equaliser	Delta	normal	
1	HLS	Child	Equaliser	Freq-Low Shelf		37.7
1	HLS	Child	Equaliser	Gain-Low Shelf		-120
1	HLS	Child	Equaliser	Q-Low Shelf		2
1	HLS	Child	Equaliser	Freq-Band 2		277.6
1	HLS	Child	Equaliser	Gain-Band 2		0.3
1	HLS	Child	Equaliser	Q-Band 2		2
1	HLS	Child	Equaliser	Freq-Band 3		1464.2
1	HLS	Child	Equaliser	Gain-Band 3		-1.1
1	HLS	Child	Equaliser	Q-Band 3		4
1	HLS	Child	Equaliser	Freq-High Shelf 4		5043.9
1	HLS	Child	Equaliser	Gain-High Shelf 4		30.9
1	HLS	Child	Equaliser	Q-High Shelf 4		2
1	HLS	Child	Equaliser	Freq-Band 5		77
1	HLS	Child	Equaliser	Gain-Band 5		3.8
1	HLS	Child	Equaliser	Q-Band 5		2
1	HLS	Child	Equaliser	Global Gain		-1
1	HLS	Child	Equaliser	Bypass	normal	
1	HLS	Child	Equaliser	Wet		100
1	HLS	Child	Equaliser	Delta	normal	
2	Normal	Child	Equaliser	Freq-Low Shelf		100
2	Normal	Child	Equaliser	Gain-Low Shelf		0
2	Normal	Child	Equaliser	Q-Low Shelf		2
2	Normal	Child	Equaliser	Freq-Band 2		300
2	Normal	Child	Equaliser	Gain-Band 2		0
2	Normal	Child	Equaliser	Q-Band 2		2
2	Normal	Child	Equaliser	Freq-Band 3		1000
2	Normal	Child	Equaliser	Gain-Band 3		0
2	Normal	Child	Equaliser	Q-Band 3		2
2	Normal	Child	Equaliser	Freq-High Shelf		5000
2	Normal	Child	Equaliser	Gain-High Shelf		0
2	Normal	Child	Equaliser	Q-High Shelf		2
2	Normal	Child	Equaliser	Global Gain		0
2	Normal	Child	Equaliser	Bypass	bypassed	
2	Normal	Child	Equaliser	Wet		100
2	Normal	Child	Equaliser	Delta	normal	
2	HLS	Child	Equaliser	Freq-Low Shelf		100
2	HLS	Child	Equaliser	Gain-Low Shelf		0
2	HLS	Child	Equaliser	Q-Low Shelf		2
2	HLS	Child	Equaliser	Freq-Band 2		300
2	HLS	Child	Equaliser	Gain-Band 2		0
2	HLS	Child	Equaliser	Q-Band 2		2
2	HLS	Child	Equaliser	Freq-Band 3		1000
2	HLS	Child	Equaliser	Gain-Band 3		0
2	HLS	Child	Equaliser	Q-Band 3		2

2	HLS	Child	Equaliser	Freq-High Shelf		3820.4
2	HLS	Child	Equaliser	Gain-High Shelf		10.6
2	HLS	Child	Equaliser	Q-High Shelf		2
2	HLS	Child	Equaliser	Global Gain		0
2	HLS	Child	Equaliser	Bypass	normal	
2	HLS	Child	Equaliser	Wet		100
2	HLS	Child	Equaliser	Delta	normal	
3	Normal	Child	Equaliser	Freq-High Pass 1		510.8
3	Normal	Child	Equaliser	Gain-High Pass 1		0
3	Normal	Child	Equaliser	Q-High Pass 1		2
3	Normal	Child	Equaliser	Freq-Band 2		300
3	Normal	Child	Equaliser	Gain-Band 2		0
3	Normal	Child	Equaliser	Q-Band 2		2
3	Normal	Child	Equaliser	Freq-Band 3		1732.5
3	Normal	Child	Equaliser	Gain-Band 3		-1.2
3	Normal	Child	Equaliser	Q-Band 3		2
3	Normal	Child	Equaliser	Freq-High Shelf		5000
3	Normal	Child	Equaliser	Gain-High Shelf		0
3	Normal	Child	Equaliser	Q-High Shelf		2
3	Normal	Child	Equaliser	Global Gain		0
3	Normal	Child	Equaliser	Bypass	normal	
3	Normal	Child	Equaliser	Wet		100
3	Normal	Child	Equaliser	Delta	normal	
3	HLS	Child	Equaliser	Freq-High Pass 1		510.8
3	HLS	Child	Equaliser	Gain-High Pass 1		0
3	HLS	Child	Equaliser	Q-High Pass 1		2
3	HLS	Child	Equaliser	Freq-Band 2		300
3	HLS	Child	Equaliser	Gain-Band 2		0
3	HLS	Child	Equaliser	Q-Band 2		2
3	HLS	Child	Equaliser	Freq-Band 3		810.7
3	HLS	Child	Equaliser	Gain-Band 3		-7.1
3	HLS	Child	Equaliser	Q-Band 3		2
3	HLS	Child	Equaliser	Freq-High Shelf		2043.4
3	HLS	Child	Equaliser	Gain-High Shelf		7.5
3	HLS	Child	Equaliser	Q-High Shelf		2
3	HLS	Child	Equaliser	Global Gain		0
3	HLS	Child	Equaliser	Bypass	normal	
3	HLS	Child	Equaliser	Wet		100
3	HLS	Child	Equaliser	Delta	normal	
1	Normal	Creature	Equaliser	Freq-Low Shelf		31.5
1	Normal	Creature	Equaliser	Gain-Low Shelf		-120
1	Normal	Creature	Equaliser	Q-Low Shelf		2
1	Normal	Creature	Equaliser	Freq-Band 2		72.7
1	Normal	Creature	Equaliser	Gain-Band 2		2.4
1	Normal	Creature	Equaliser	Q-Band 2		2
1	Normal	Creature	Equaliser	Freq-Band 3		684.3
1	Normal	Creature	Equaliser	Gain-Band 3		-0.9

1	Normal	Creature	Equaliser	Q-Band 3		2
1	Normal	Creature	Equaliser	Freq-High Shelf		4871.7
1	Normal	Creature	Equaliser	Gain-High Shelf		-0.8
1	Normal	Creature	Equaliser	Q-High Shelf		2
1	Normal	Creature	Equaliser	Global Gain		0
1	Normal	Creature	Equaliser	Bypass	normal	
1	Normal	Creature	Equaliser	Wet		100
1	Normal	Creature	Equaliser	Delta	normal	
1	HLS	Creature	Equaliser	Freq-Low Shelf		22
1	HLS	Creature	Equaliser	Gain-Low Shelf		-120
1	HLS	Creature	Equaliser	Q-Low Shelf		2
1	HLS	Creature	Equaliser	Freq-Band 2		67.8
1	HLS	Creature	Equaliser	Gain-Band 2		7.7
1	HLS	Creature	Equaliser	Q-Band 2		2
1	HLS	Creature	Equaliser	Freq-Band 3		206.7
1	HLS	Creature	Equaliser	Gain-Band 3		0.7
1	HLS	Creature	Equaliser	Q-Band 3		2
1	HLS	Creature	Equaliser	Freq-High Shelf 4		7993.6
1	HLS	Creature	Equaliser	Gain-High Shelf 4		26.6
1	HLS	Creature	Equaliser	Q-High Shelf 4		2
1	HLS	Creature	Equaliser	Freq-Band 5		1970
1	HLS	Creature	Equaliser	Gain-Band 5		4.7
1	HLS	Creature	Equaliser	Q-Band 5		3.9
1	HLS	Creature	Equaliser	Global Gain		-2.5
1	HLS	Creature	Equaliser	Bypass	normal	
1	HLS	Creature	Equaliser	Wet		100
1	HLS	Creature	Equaliser	Delta	normal	
2	Normal	Creature	Equaliser	Freq-Low Shelf		98
2	Normal	Creature	Equaliser	Gain-Low Shelf		-5.9
2	Normal	Creature	Equaliser	Q-Low Shelf		0.2
2	Normal	Creature	Equaliser	Freq-Band 2		300
2	Normal	Creature	Equaliser	Gain-Band 2		0
2	Normal	Creature	Equaliser	Q-Band 2		2
2	Normal	Creature	Equaliser	Freq-Band 3		1000
2	Normal	Creature	Equaliser	Gain-Band 3		0
2	Normal	Creature	Equaliser	Q-Band 3		2
2	Normal	Creature	Equaliser	Freq-High Shelf		5000
2	Normal	Creature	Equaliser	Gain-High Shelf		0
2	Normal	Creature	Equaliser	Q-High Shelf		2
2	Normal	Creature	Equaliser	Global Gain		0
2	Normal	Creature	Equaliser	Bypass	bypassed	
2	Normal	Creature	Equaliser	Wet		100
2	Normal	Creature	Equaliser	Delta	normal	
2	HLS	Creature	Equaliser	Freq-Low Shelf		98
2	HLS	Creature	Equaliser	Gain-Low Shelf		-5.9
2	HLS	Creature	Equaliser	Q-Low Shelf		0.2
2	HLS	Creature	Equaliser	Freq-Band 2		300

2	HLS	Creature	Equaliser	Gain-Band 2		0
2	HLS	Creature	Equaliser	Q-Band 2		2
2	HLS	Creature	Equaliser	Freq-Band 3		1000
2	HLS	Creature	Equaliser	Gain-Band 3		0
2	HLS	Creature	Equaliser	Q-Band 3		2
2	HLS	Creature	Equaliser	Freq-High Shelf		5000
2	HLS	Creature	Equaliser	Gain-High Shelf		0
2	HLS	Creature	Equaliser	Q-High Shelf		2
2	HLS	Creature	Equaliser	Global Gain		0
2	HLS	Creature	Equaliser	Bypass	normal	
2	HLS	Creature	Equaliser	Wet		100
2	HLS	Creature	Equaliser	Delta	normal	
3	Normal	Creature	Equaliser	Freq-High Pass 1		125.1
3	Normal	Creature	Equaliser	Gain-High Pass 1		0
3	Normal	Creature	Equaliser	Q-High Pass 1		2
3	Normal	Creature	Equaliser	Freq-Band 2		300
3	Normal	Creature	Equaliser	Gain-Band 2		0
3	Normal	Creature	Equaliser	Q-Band 2		2
3	Normal	Creature	Equaliser	Freq-Band 3		426.2
3	Normal	Creature	Equaliser	Gain-Band 3		-1.9
3	Normal	Creature	Equaliser	Q-Band 3		2
3	Normal	Creature	Equaliser	Freq-High Shelf		5000
3	Normal	Creature	Equaliser	Gain-High Shelf		0
3	Normal	Creature	Equaliser	Q-High Shelf		2
3	Normal	Creature	Equaliser	Global Gain		0
3	Normal	Creature	Equaliser	Bypass	normal	
3	Normal	Creature	Equaliser	Wet		100
3	Normal	Creature	Equaliser	Delta	normal	
3	HLS	Creature	Equaliser	Freq-High Pass 1		90.3
3	HLS	Creature	Equaliser	Gain-High Pass 1		0
3	HLS	Creature	Equaliser	Q-High Pass 1		2
3	HLS	Creature	Equaliser	Freq-Band 2		611
3	HLS	Creature	Equaliser	Gain-Band 2		-8.5
3	HLS	Creature	Equaliser	Q-Band 2		2
3	HLS	Creature	Equaliser	Freq-Band 3		291.1
3	HLS	Creature	Equaliser	Gain-Band 3		-1.1
3	HLS	Creature	Equaliser	Q-Band 3		2
3	HLS	Creature	Equaliser	Freq-High Shelf 4		3659
3	HLS	Creature	Equaliser	Gain-High Shelf 4		6.4
3	HLS	Creature	Equaliser	Q-High Shelf 4		2
3	HLS	Creature	Equaliser	Freq-Band 5		172.9
3	HLS	Creature	Equaliser	Gain-Band 5		2.9
3	HLS	Creature	Equaliser	Q-Band 5		2
3	HLS	Creature	Equaliser	Freq-Band 6		1656.8
3	HLS	Creature	Equaliser	Gain-Band 6		4.2
3	HLS	Creature	Equaliser	Q-Band 6		2
3	HLS	Creature	Equaliser	Freq-Band 7		3078.3

3	HLS	Creature	Equaliser	Gain-Band 7		5.6
3	HLS	Creature	Equaliser	Q-Band 7		2
3	HLS	Creature	Equaliser	Global Gain		0
3	HLS	Creature	Equaliser	Bypass	normal	
3	HLS	Creature	Equaliser	Wet		100
3	HLS	Creature	Equaliser	Delta	normal	
1	Normal	FX	Equaliser	Freq-Low Shelf		35
1	Normal	FX	Equaliser	Gain-Low Shelf		-120
1	Normal	FX	Equaliser	Q-Low Shelf		2
1	Normal	FX	Equaliser	Freq-Band 2		80.1
1	Normal	FX	Equaliser	Gain-Band 2		2.2
1	Normal	FX	Equaliser	Q-Band 2		2
1	Normal	FX	Equaliser	Freq-Band 3		835.7
1	Normal	FX	Equaliser	Gain-Band 3		-1.3
1	Normal	FX	Equaliser	Q-Band 3		2
1	Normal	FX	Equaliser	Freq-High Shelf		5000
1	Normal	FX	Equaliser	Gain-High Shelf		-0.7
1	Normal	FX	Equaliser	Q-High Shelf		2
1	Normal	FX	Equaliser	Bypass	normal	
1	HLS	FX	Equaliser	Freq-Low Shelf		24
1	HLS	FX	Equaliser	Gain-Low Shelf		-120
1	HLS	FX	Equaliser	Q-Low Shelf		2
1	HLS	FX	Equaliser	Freq-Band 2		68.1
1	HLS	FX	Equaliser	Gain-Band 2		4.3
1	HLS	FX	Equaliser	Q-Band 2		2
1	HLS	FX	Equaliser	Freq-Band 3		328.3
1	HLS	FX	Equaliser	Gain-Band 3		0.4
1	HLS	FX	Equaliser	Q-Band 3		2
1	HLS	FX	Equaliser	Freq-Band 4		1290.5
1	HLS	FX	Equaliser	Gain-Band 4		6.9
1	HLS	FX	Equaliser	Q-Band 4		2
1	HLS	FX	Equaliser	Freq-High Shelf		3753.8
1	HLS	FX	Equaliser	Gain-High Shelf		22.9
1	HLS	FX	Equaliser	Q-High Shelf		2
1	HLS	FX	Equaliser	Bypass	normal	
2	Normal	FX	Equaliser	Freq-High Pass 1		86.7
2	Normal	FX	Equaliser	Gain-High Pass 1		0
2	Normal	FX	Equaliser	Q-High Pass 1		2
2	Normal	FX	Equaliser	Freq-Band 2		300
2	Normal	FX	Equaliser	Gain-Band 2		0
2	Normal	FX	Equaliser	Q-Band 2		2
2	Normal	FX	Equaliser	Freq-Band 3		2838.1
2	Normal	FX	Equaliser	Gain-Band 3		-2.6
2	Normal	FX	Equaliser	Q-Band 3		2
2	Normal	FX	Equaliser	Freq-High Shelf 4		5767.2
2	Normal	FX	Equaliser	Gain-High Shelf 4		2.9
2	Normal	FX	Equaliser	Q-High Shelf 4		2

2	Normal	FX	Equaliser	Freq-Band 5		914.6
2	Normal	FX	Equaliser	Gain-Band 5		2.7
2	Normal	FX	Equaliser	Q-Band 5		2
2	Normal	FX	Equaliser	Bypass	normal	
2	HLS	FX	Equaliser	Freq-High Pass 1		86.7
2	HLS	FX	Equaliser	Gain-High Pass 1		0
2	HLS	FX	Equaliser	Q-High Pass 1		2
2	HLS	FX	Equaliser	Freq-Band 2		300
2	HLS	FX	Equaliser	Gain-Band 2		0
2	HLS	FX	Equaliser	Q-Band 2		2
2	HLS	FX	Equaliser	Freq-Band 3		2838.1
2	HLS	FX	Equaliser	Gain-Band 3		-2.6
2	HLS	FX	Equaliser	Q-Band 3		2
2	HLS	FX	Equaliser	Freq-High Shelf 4		5767.2
2	HLS	FX	Equaliser	Gain-High Shelf 4		2.9
2	HLS	FX	Equaliser	Q-High Shelf 4		2
2	HLS	FX	Equaliser	Freq-Band 5		914.6
2	HLS	FX	Equaliser	Gain-Band 5		2.7
2	HLS	FX	Equaliser	Q-Band 5		2
2	HLS	FX	Equaliser	Bypass	normal	
3	Normal	FX	Equaliser	Freq-High Pass 1		356.1
3	Normal	FX	Equaliser	Gain-High Pass 1		0
3	Normal	FX	Equaliser	Q-High Pass 1		2
3	Normal	FX	Equaliser	Freq-Band 2		300
3	Normal	FX	Equaliser	Gain-Band 2		0
3	Normal	FX	Equaliser	Q-Band 2		2
3	Normal	FX	Equaliser	Freq-Band 3		5519.2
3	Normal	FX	Equaliser	Gain-Band 3		-2.1
3	Normal	FX	Equaliser	Q-Band 3		2
3	Normal	FX	Equaliser	Freq-Low Pass 4		16480.1
3	Normal	FX	Equaliser	Gain-Low Pass 4		-0.6
3	Normal	FX	Equaliser	Q-Low Pass 4		2
3	Normal	FX	Equaliser	Bypass	normal	
3	HLS	FX	Equaliser	Freq-High Pass 1		356.1
3	HLS	FX	Equaliser	Gain-High Pass 1		0
3	HLS	FX	Equaliser	Q-High Pass 1		2
3	HLS	FX	Equaliser	Freq-Band 2		761.8
3	HLS	FX	Equaliser	Gain-Band 2		-2.3
3	HLS	FX	Equaliser	Q-Band 2		2
3	HLS	FX	Equaliser	Freq-Band 3		1511.1
3	HLS	FX	Equaliser	Gain-Band 3		-10.5
3	HLS	FX	Equaliser	Q-Band 3		1.71
3	HLS	FX	Equaliser	Freq-High Shelf		1773.2
3	HLS	FX	Equaliser	Gain-High Shelf		6.7
3	HLS	FX	Equaliser	Q-High Shelf		0.88
3	HLS	FX	Equaliser	Bypass	normal	
1	Normal	Music	Equaliser	Freq-Low Shelf		21.2

1	Normal	Music	Equaliser	Gain-Low Shelf		-120
1	Normal	Music	Equaliser	Q-Low Shelf		2.95
1	Normal	Music	Equaliser	Freq-Band 2		83.4
1	Normal	Music	Equaliser	Gain-Band 2		2.8
1	Normal	Music	Equaliser	Q-Band 2		2
1	Normal	Music	Equaliser	Freq-Band 3		1000
1	Normal	Music	Equaliser	Gain-Band 3		0
1	Normal	Music	Equaliser	Q-Band 3		2
1	Normal	Music	Equaliser	Freq-High Shelf		5000
1	Normal	Music	Equaliser	Gain-High Shelf		0
1	Normal	Music	Equaliser	Q-High Shelf		2
1	Normal	Music	Equaliser	Bypass	normal	
1	HLS	Music	Equaliser	Freq-Low Shelf		42.7
1	HLS	Music	Equaliser	Gain-Low Shelf		-120
1	HLS	Music	Equaliser	Q-Low Shelf		2
1	HLS	Music	Equaliser	Freq-Band 2		82.1
1	HLS	Music	Equaliser	Gain-Band 2		6.5
1	HLS	Music	Equaliser	Q-Band 2		2
1	HLS	Music	Equaliser	Freq-Band 3		281.3
1	HLS	Music	Equaliser	Gain-Band 3		-0.3
1	HLS	Music	Equaliser	Q-Band 3		2
1	HLS	Music	Equaliser	Freq-High Shelf 4		4926.8
1	HLS	Music	Equaliser	Gain-High Shelf 4		36.1
1	HLS	Music	Equaliser	Q-High Shelf 4		2
1	HLS	Music	Equaliser	Freq-Band 5		1279.3
1	HLS	Music	Equaliser	Gain-Band 5		0.8
1	HLS	Music	Equaliser	Q-Band 5		2
1	HLS	Music	Equaliser	Bypass	normal	
2	Normal	Music	Equaliser	Freq-High Pass 1		71.3
2	Normal	Music	Equaliser	Gain-High Pass 1		0
2	Normal	Music	Equaliser	Q-High Pass 1		2
2	Normal	Music	Equaliser	Freq-Band 2		511.6
2	Normal	Music	Equaliser	Gain-Band 2		-3.3
2	Normal	Music	Equaliser	Q-Band 2		1.6
2	Normal	Music	Equaliser	Freq-Band 3		1329
2	Normal	Music	Equaliser	Gain-Band 3		-3.6
2	Normal	Music	Equaliser	Q-Band 3		2
2	Normal	Music	Equaliser	Freq-High Shelf 4		12154.1
2	Normal	Music	Equaliser	Gain-High Shelf 4		3.3
2	Normal	Music	Equaliser	Q-High Shelf 4		0.2
2	Normal	Music	Equaliser	Freq-Band 5		6902.7
2	Normal	Music	Equaliser	Gain-Band 5		2.9
2	Normal	Music	Equaliser	Q-Band 5		1.2
2	Normal	Music	Equaliser	Freq-Band 6		3001.3
2	Normal	Music	Equaliser	Gain-Band 6		-2
2	Normal	Music	Equaliser	Q-Band 6		2
2	Normal	Music	Equaliser	Freq-Band 7		2758

2	Normal	Music	Equaliser	Gain-Band 7		2.1
2	Normal	Music	Equaliser	Q-Band 7		2
2	Normal	Music	Equaliser	Bypass	normal	
2	HLS	Music	Equaliser	Freq-High Pass 1		71.3
2	HLS	Music	Equaliser	Gain-High Pass 1		0
2	HLS	Music	Equaliser	Q-High Pass 1		2
2	HLS	Music	Equaliser	Freq-Band 2		511.6
2	HLS	Music	Equaliser	Gain-Band 2		-3.3
2	HLS	Music	Equaliser	Q-Band 2		1.6
2	HLS	Music	Equaliser	Freq-Band 3		1329
2	HLS	Music	Equaliser	Gain-Band 3		-3.6
2	HLS	Music	Equaliser	Q-Band 3		2
2	HLS	Music	Equaliser	Freq-High Shelf 4		9942.8
2	HLS	Music	Equaliser	Gain-High Shelf 4		8.5
2	HLS	Music	Equaliser	Q-High Shelf 4		0.2
2	HLS	Music	Equaliser	Freq-Band 5		6189.7
2	HLS	Music	Equaliser	Gain-Band 5		5.9
2	HLS	Music	Equaliser	Q-Band 5		1.2
2	HLS	Music	Equaliser	Freq-Band 6		3001.3
2	HLS	Music	Equaliser	Gain-Band 6		-2
2	HLS	Music	Equaliser	Q-Band 6		2
2	HLS	Music	Equaliser	Freq-Band 7		2388.1
2	HLS	Music	Equaliser	Gain-Band 7		10.7
2	HLS	Music	Equaliser	Q-Band 7		2
2	HLS	Music	Equaliser	Bypass	normal	
3	Normal	Music	Equaliser	Freq-Low Shelf		100
3	Normal	Music	Equaliser	Gain-Low Shelf		0
3	Normal	Music	Equaliser	Q-Low Shelf		2
3	Normal	Music	Equaliser	Freq-Band 2		300
3	Normal	Music	Equaliser	Gain-Band 2		0
3	Normal	Music	Equaliser	Q-Band 2		2
3	Normal	Music	Equaliser	Freq-Band 3		811.4
3	Normal	Music	Equaliser	Gain-Band 3		-2.8
3	Normal	Music	Equaliser	Q-Band 3		2
3	Normal	Music	Equaliser	Freq-High Shelf 4		5000
3	Normal	Music	Equaliser	Gain-High Shelf 4		0
3	Normal	Music	Equaliser	Q-High Shelf 4		2
3	Normal	Music	Equaliser	Freq-Band 5		2758
3	Normal	Music	Equaliser	Gain-Band 5		-1
3	Normal	Music	Equaliser	Q-Band 5		0.7
3	Normal	Music	Equaliser	Bypass	normal	
3	HLS	Music	Equaliser	Freq-Low Shelf		100
3	HLS	Music	Equaliser	Gain-Low Shelf		0
3	HLS	Music	Equaliser	Q-Low Shelf		2
3	HLS	Music	Equaliser	Freq-Band 2		1130.5
3	HLS	Music	Equaliser	Gain-Band 2		-4.7
3	HLS	Music	Equaliser	Q-Band 2		2

3	HLS	Music	Equaliser	Freq-Band 3		811.4
3	HLS	Music	Equaliser	Gain-Band 3		-2.8
3	HLS	Music	Equaliser	Q-Band 3		2
3	HLS	Music	Equaliser	Freq-High Shelf 4		2024.7
3	HLS	Music	Equaliser	Gain-High Shelf 4		2.7
3	HLS	Music	Equaliser	Q-High Shelf 4		1.44
3	HLS	Music	Equaliser	Freq-Band 5		2758
3	HLS	Music	Equaliser	Gain-Band 5		-1
3	HLS	Music	Equaliser	Q-Band 5		0.7
3	HLS	Music	Equaliser	Bypass	normal	

Engineer	Mix	Track	Effect Type	Parameter Name	Parameter Value
1	Normal	Narration	Compression	Threshold	-29
1	Normal	Narration	Compression	Ratio	5.2
1	Normal	Narration	Compression	Attack	9.9
1	Normal	Narration	Compression	Release	49
1	Normal	Narration	Compression	Pre-comp	0
1	Normal	Narration	Compression	resvd	0
1	Normal	Narration	Compression	Lowpass	8008
1	Normal	Narration	Compression	Hipass	145
1	Normal	Narration	Compression	SignIn	0
1	Normal	Narration	Compression	AudIn	0
1	Normal	Narration	Compression	Dry	-2.9
1	Normal	Narration	Compression	Wet	-2.9
1	Normal	Narration	Compression	Filter Preview	0
1	Normal	Narration	Compression	RMS size	0
1	Normal	Narration	Compression	Knee	1.5
1	Normal	Narration	Compression	Auto Make Up Gain	0
1	Normal	Narration	Compression	Auto Release	0
1	Normal	Narration	Compression	Legacy Attack/Knee (	0
1	Normal	Narration	Compression	Deprecated Broken A	0
1	Normal	Narration	Compression	Multichannel Mode	0
1	Normal	Narration	Compression	Metering Index	0
1	Normal	Narration	Compression	Bypass	normal
1	Normal	Narration	Compression	Wet	100
1	Normal	Narration	Compression	Delta	normal
1	HLS	Narration	Compression	Threshold	-19.6
1	HLS	Narration	Compression	Ratio	8.41
1	HLS	Narration	Compression	Attack	9.9
1	HLS	Narration	Compression	Release	49
1	HLS	Narration	Compression	Pre-comp	0
1	HLS	Narration	Compression	resvd	0
1	HLS	Narration	Compression	Lowpass	2652
1	HLS	Narration	Compression	Hipass	106
1	HLS	Narration	Compression	SignIn	0
1	HLS	Narration	Compression	AudIn	0
1	HLS	Narration	Compression	Dry	-3.3
1	HLS	Narration	Compression	Wet	-2.7
1	HLS	Narration	Compression	Filter Preview	0
1	HLS	Narration	Compression	RMS size	0
1	HLS	Narration	Compression	Knee	1.5
1	HLS	Narration	Compression	Auto Make Up Gain	0
1	HLS	Narration	Compression	Auto Release	0
1	HLS	Narration	Compression	Legacy Attack/Knee (	0
1	HLS	Narration	Compression	Deprecated Broken A	1
1	HLS	Narration	Compression	Multichannel Mode	0
1	HLS	Narration	Compression	Metering Index	0
1	HLS	Narration	Compression	Bypass	normal

1	HLS	Narration	Compression	Wet		100
1	HLS	Narration	Compression	Delta	normal	
2	Normal	Narration	Compression	Threshold		-35.1
2	Normal	Narration	Compression	Ratio		4.26
2	Normal	Narration	Compression	Attack		6.7
2	Normal	Narration	Compression	Release		100
2	Normal	Narration	Compression	Pre-comp		0
2	Normal	Narration	Compression	resvd		0
2	Normal	Narration	Compression	Lowpass		20000
2	Normal	Narration	Compression	Hipass		0
2	Normal	Narration	Compression	SignIn		0
2	Normal	Narration	Compression	AudIn		0
2	Normal	Narration	Compression	Dry		0
2	Normal	Narration	Compression	Wet		4.6
2	Normal	Narration	Compression	Filter Preview		0
2	Normal	Narration	Compression	RMS size		5
2	Normal	Narration	Compression	Knee		0
2	Normal	Narration	Compression	Auto Make Up Gain		0
2	Normal	Narration	Compression	Auto Release		0
2	Normal	Narration	Compression	Legacy Attack/Knee (		0.25
2	Normal	Narration	Compression	Deprecated Broken A		0
2	Normal	Narration	Compression	Multichannel Mode		0
2	Normal	Narration	Compression	Metering Index		0
2	Normal	Narration	Compression	Bypass	normal	
2	Normal	Narration	Compression	Wet		100
2	Normal	Narration	Compression	Delta	normal	
2	HLS	Narration	Compression	Threshold		-35.1
2	HLS	Narration	Compression	Ratio		4.26
2	HLS	Narration	Compression	Attack		6.7
2	HLS	Narration	Compression	Release		100
2	HLS	Narration	Compression	Pre-comp		0
2	HLS	Narration	Compression	resvd		0
2	HLS	Narration	Compression	Lowpass		20000
2	HLS	Narration	Compression	Hipass		0
2	HLS	Narration	Compression	SignIn		0
2	HLS	Narration	Compression	AudIn		0
2	HLS	Narration	Compression	Dry		0
2	HLS	Narration	Compression	Wet		4.6
2	HLS	Narration	Compression	Filter Preview		0
2	HLS	Narration	Compression	RMS size		5
2	HLS	Narration	Compression	Knee		0
2	HLS	Narration	Compression	Auto Make Up Gain		0
2	HLS	Narration	Compression	Auto Release		0
2	HLS	Narration	Compression	Legacy Attack/Knee (		0.25
2	HLS	Narration	Compression	Deprecated Broken A		0
2	HLS	Narration	Compression	Multichannel Mode		0
2	HLS	Narration	Compression	Metering Index		0

2	HLS	Narration	Compression	Bypass	normal	
2	HLS	Narration	Compression	Wet		100
2	HLS	Narration	Compression	Delta	normal	
3	Normal	Narration	Compression	Threshold		-41.3
3	Normal	Narration	Compression	Ratio		3.99
3	Normal	Narration	Compression	Attack		8.2
3	Normal	Narration	Compression	Release		16
3	Normal	Narration	Compression	Pre-comp		0
3	Normal	Narration	Compression	resvd		0
3	Normal	Narration	Compression	Lowpass		20000
3	Normal	Narration	Compression	Hipass		0
3	Normal	Narration	Compression	SignIn		0
3	Normal	Narration	Compression	AudIn		0
3	Normal	Narration	Compression	Dry		0
3	Normal	Narration	Compression	Wet		0
3	Normal	Narration	Compression	Filter Preview		0
3	Normal	Narration	Compression	RMS size		5
3	Normal	Narration	Compression	Knee		0
3	Normal	Narration	Compression	Auto Make Up Gain		0
3	Normal	Narration	Compression	Auto Release		0
3	Normal	Narration	Compression	Legacy Attack/Knee (		0.25
3	Normal	Narration	Compression	Deprecated Broken A		0
3	Normal	Narration	Compression	Multichannel Mode		0
3	Normal	Narration	Compression	Metering Index		0
3	Normal	Narration	Compression	Bypass	normal	
3	Normal	Narration	Compression	Wet		100
3	Normal	Narration	Compression	Delta	normal	
3	HLS	Narration	Compression	Threshold		-41.9
3	HLS	Narration	Compression	Ratio		3.99
3	HLS	Narration	Compression	Attack		8.2
3	HLS	Narration	Compression	Release		16
3	HLS	Narration	Compression	Pre-comp		0
3	HLS	Narration	Compression	resvd		0
3	HLS	Narration	Compression	Lowpass		20000
3	HLS	Narration	Compression	Hipass		0
3	HLS	Narration	Compression	SignIn		0
3	HLS	Narration	Compression	AudIn		0
3	HLS	Narration	Compression	Dry		0
3	HLS	Narration	Compression	Wet		0
3	HLS	Narration	Compression	Filter Preview		0
3	HLS	Narration	Compression	RMS size		5
3	HLS	Narration	Compression	Knee		0
3	HLS	Narration	Compression	Auto Make Up Gain		0
3	HLS	Narration	Compression	Auto Release		0
3	HLS	Narration	Compression	Legacy Attack/Knee (		0.25
3	HLS	Narration	Compression	Deprecated Broken A		0
3	HLS	Narration	Compression	Multichannel Mode		0

3	HLS	Narration	Compression	Metering Index		0
3	HLS	Narration	Compression	Bypass	normal	
3	HLS	Narration	Compression	Wet		100
3	HLS	Narration	Compression	Delta	normal	
1	Normal	Child	Compression	Threshold		-36.7
1	Normal	Child	Compression	Ratio		5.2
1	Normal	Child	Compression	Attack		9.9
1	Normal	Child	Compression	Release		49
1	Normal	Child	Compression	Pre-comp		0
1	Normal	Child	Compression	resvd		0
1	Normal	Child	Compression	Lowpass		8008
1	Normal	Child	Compression	Hipass		145
1	Normal	Child	Compression	SignIn		0
1	Normal	Child	Compression	AudIn		0
1	Normal	Child	Compression	Dry		-3
1	Normal	Child	Compression	Wet		-3.1
1	Normal	Child	Compression	Filter Preview		0
1	Normal	Child	Compression	RMS size		0
1	Normal	Child	Compression	Knee		1.5
1	Normal	Child	Compression	Auto Make Up Gain		0
1	Normal	Child	Compression	Auto Release		0
1	Normal	Child	Compression	Legacy Attack/Knee (		0
1	Normal	Child	Compression	Deprecated Broken A		0
1	Normal	Child	Compression	Multichannel Mode		0
1	Normal	Child	Compression	Metering Index		0
1	Normal	Child	Compression	Bypass	normal	
1	Normal	Child	Compression	Wet		100
1	Normal	Child	Compression	Delta	normal	
1	HLS	Child	Compression	Threshold		-34
1	HLS	Child	Compression	Ratio		3.77
1	HLS	Child	Compression	Attack		3
1	HLS	Child	Compression	Release		100
1	HLS	Child	Compression	Pre-comp		0
1	HLS	Child	Compression	resvd		0
1	HLS	Child	Compression	Lowpass		20000
1	HLS	Child	Compression	Hipass		0
1	HLS	Child	Compression	SignIn		0
1	HLS	Child	Compression	AudIn		0
1	HLS	Child	Compression	Dry		-3.1
1	HLS	Child	Compression	Wet		-3.1
1	HLS	Child	Compression	Filter Preview		0
1	HLS	Child	Compression	RMS size		5
1	HLS	Child	Compression	Knee		0
1	HLS	Child	Compression	Auto Make Up Gain		0
1	HLS	Child	Compression	Auto Release		0
1	HLS	Child	Compression	Legacy Attack/Knee (		0.25
1	HLS	Child	Compression	Deprecated Broken A		0

1	HLS	Child	Compression	Multichannel Mode		0
1	HLS	Child	Compression	Metering Index		0
1	HLS	Child	Compression	Bypass	normal	
1	HLS	Child	Compression	Wet		100
1	HLS	Child	Compression	Delta	normal	
2	Normal	Child	Compression	Threshold		0
2	Normal	Child	Compression	Ratio		1
2	Normal	Child	Compression	Attack		3
2	Normal	Child	Compression	Release		100
2	Normal	Child	Compression	Pre-comp		0
2	Normal	Child	Compression	resvd		0
2	Normal	Child	Compression	Lowpass		20000
2	Normal	Child	Compression	Hipass		0
2	Normal	Child	Compression	SignIn		0
2	Normal	Child	Compression	AudIn		0
2	Normal	Child	Compression	Dry		0
2	Normal	Child	Compression	Wet		0
2	Normal	Child	Compression	Filter Preview		0
2	Normal	Child	Compression	RMS size		5
2	Normal	Child	Compression	Knee		0
2	Normal	Child	Compression	Auto Make Up Gain		0
2	Normal	Child	Compression	Auto Release		0
2	Normal	Child	Compression	Legacy Attack/Knee (		0.25
2	Normal	Child	Compression	Deprecated Broken A		0
2	Normal	Child	Compression	Multichannel Mode		0
2	Normal	Child	Compression	Metering Index		0
2	Normal	Child	Compression	Bypass	bypassed	
2	Normal	Child	Compression	Wet		100
2	Normal	Child	Compression	Delta	normal	
2	HLS	Child	Compression	Threshold		0
2	HLS	Child	Compression	Ratio		1
2	HLS	Child	Compression	Attack		3
2	HLS	Child	Compression	Release		100
2	HLS	Child	Compression	Pre-comp		0
2	HLS	Child	Compression	resvd		0
2	HLS	Child	Compression	Lowpass		20000
2	HLS	Child	Compression	Hipass		0
2	HLS	Child	Compression	SignIn		0
2	HLS	Child	Compression	AudIn		0
2	HLS	Child	Compression	Dry		0
2	HLS	Child	Compression	Wet		0
2	HLS	Child	Compression	Filter Preview		0
2	HLS	Child	Compression	RMS size		5
2	HLS	Child	Compression	Knee		0
2	HLS	Child	Compression	Auto Make Up Gain		0
2	HLS	Child	Compression	Auto Release		0
2	HLS	Child	Compression	Legacy Attack/Knee (		0.25

2	HLS	Child	Compression	Deprecated Broken A		0
2	HLS	Child	Compression	Multichannel Mode		0
2	HLS	Child	Compression	Metering Index		0
2	HLS	Child	Compression	Bypass	bypassed	
2	HLS	Child	Compression	Wet		100
2	HLS	Child	Compression	Delta	normal	
3	Normal	Child	Compression	Threshold		0
3	Normal	Child	Compression	Ratio		1
3	Normal	Child	Compression	Attack		3
3	Normal	Child	Compression	Release		100
3	Normal	Child	Compression	Pre-comp		0
3	Normal	Child	Compression	resvd		0
3	Normal	Child	Compression	Lowpass		20000
3	Normal	Child	Compression	Hipass		0
3	Normal	Child	Compression	SignIn		0
3	Normal	Child	Compression	AudIn		0
3	Normal	Child	Compression	Dry		0
3	Normal	Child	Compression	Wet		0
3	Normal	Child	Compression	Filter Preview		0
3	Normal	Child	Compression	RMS size		5
3	Normal	Child	Compression	Knee		0
3	Normal	Child	Compression	Auto Make Up Gain		0
3	Normal	Child	Compression	Auto Release		0
3	Normal	Child	Compression	Legacy Attack/Knee (		0.25
3	Normal	Child	Compression	Deprecated Broken A		0
3	Normal	Child	Compression	Multichannel Mode		0
3	Normal	Child	Compression	Metering Index		0
3	Normal	Child	Compression	Bypass	normal	
3	Normal	Child	Compression	Wet		100
3	Normal	Child	Compression	Delta	normal	
3	HLS	Child	Compression	Threshold		0
3	HLS	Child	Compression	Ratio		1
3	HLS	Child	Compression	Attack		3
3	HLS	Child	Compression	Release		100
3	HLS	Child	Compression	Pre-comp		0
3	HLS	Child	Compression	resvd		0
3	HLS	Child	Compression	Lowpass		20000
3	HLS	Child	Compression	Hipass		0
3	HLS	Child	Compression	SignIn		0
3	HLS	Child	Compression	AudIn		0
3	HLS	Child	Compression	Dry		0
3	HLS	Child	Compression	Wet		0
3	HLS	Child	Compression	Filter Preview		0
3	HLS	Child	Compression	RMS size		5
3	HLS	Child	Compression	Knee		0
3	HLS	Child	Compression	Auto Make Up Gain		0
3	HLS	Child	Compression	Auto Release		0

3	HLS	Child	Compression	Legacy Attack/Knee (		0.25
3	HLS	Child	Compression	Deprecated Broken A		0
3	HLS	Child	Compression	Multichannel Mode		0
3	HLS	Child	Compression	Metering Index		0
3	HLS	Child	Compression	Bypass	normal	
3	HLS	Child	Compression	Wet		100
3	HLS	Child	Compression	Delta	normal	
1	Normal	Creature	Compression	Threshold		-32.6
1	Normal	Creature	Compression	Ratio		2.54
1	Normal	Creature	Compression	Attack		3
1	Normal	Creature	Compression	Release		100
1	Normal	Creature	Compression	Pre-comp		0
1	Normal	Creature	Compression	resvd		0
1	Normal	Creature	Compression	Lowpass		20000
1	Normal	Creature	Compression	Hipass		0
1	Normal	Creature	Compression	SignIn		0
1	Normal	Creature	Compression	AudIn		0
1	Normal	Creature	Compression	Dry		-3.2
1	Normal	Creature	Compression	Wet		-3.3
1	Normal	Creature	Compression	Filter Preview		0
1	Normal	Creature	Compression	RMS size		5
1	Normal	Creature	Compression	Knee		0
1	Normal	Creature	Compression	Auto Make Up Gain		0
1	Normal	Creature	Compression	Auto Release		0
1	Normal	Creature	Compression	Legacy Attack/Knee (		0.25
1	Normal	Creature	Compression	Deprecated Broken A		0
1	Normal	Creature	Compression	Multichannel Mode		0
1	Normal	Creature	Compression	Metering Index		0
1	Normal	Creature	Compression	Bypass	normal	
1	Normal	Creature	Compression	Wet		100
1	Normal	Creature	Compression	Delta	normal	
1	HLS	Creature	Compression	Threshold		-26
1	HLS	Creature	Compression	Ratio		2.63
1	HLS	Creature	Compression	Attack		3
1	HLS	Creature	Compression	Release		100
1	HLS	Creature	Compression	Pre-comp		0
1	HLS	Creature	Compression	resvd		0
1	HLS	Creature	Compression	Lowpass		20000
1	HLS	Creature	Compression	Hipass		0
1	HLS	Creature	Compression	SignIn		0
1	HLS	Creature	Compression	AudIn		0
1	HLS	Creature	Compression	Dry		-3.3
1	HLS	Creature	Compression	Wet		-2.9
1	HLS	Creature	Compression	Filter Preview		0
1	HLS	Creature	Compression	RMS size		5
1	HLS	Creature	Compression	Knee		0
1	HLS	Creature	Compression	Auto Make Up Gain		0

1	HLS	Creature	Compression	Auto Release	0
1	HLS	Creature	Compression	Legacy Attack/Knee (	0.25
1	HLS	Creature	Compression	Deprecated Broken A	0
1	HLS	Creature	Compression	Multichannel Mode	0
1	HLS	Creature	Compression	Metering Index	0
1	HLS	Creature	Compression	Bypass	normal
1	HLS	Creature	Compression	Wet	100
1	HLS	Creature	Compression	Delta	normal
2	Normal	Creature	Compression	Threshold	-19.9
2	Normal	Creature	Compression	Ratio	6
2	Normal	Creature	Compression	Attack	13.9
2	Normal	Creature	Compression	Release	100
2	Normal	Creature	Compression	Pre-comp	0
2	Normal	Creature	Compression	resvd	0
2	Normal	Creature	Compression	Lowpass	20000
2	Normal	Creature	Compression	Hipass	0
2	Normal	Creature	Compression	SignIn	0
2	Normal	Creature	Compression	AudIn	0
2	Normal	Creature	Compression	Dry	0
2	Normal	Creature	Compression	Wet	4.5
2	Normal	Creature	Compression	Filter Preview	0
2	Normal	Creature	Compression	RMS size	5
2	Normal	Creature	Compression	Knee	0
2	Normal	Creature	Compression	Auto Make Up Gain	0
2	Normal	Creature	Compression	Auto Release	0
2	Normal	Creature	Compression	Legacy Attack/Knee (	0.25
2	Normal	Creature	Compression	Deprecated Broken A	0
2	Normal	Creature	Compression	Multichannel Mode	0
2	Normal	Creature	Compression	Metering Index	0
2	Normal	Creature	Compression	Bypass	normal
2	Normal	Creature	Compression	Wet	100
2	Normal	Creature	Compression	Delta	normal
2	HLS	Creature	Compression	Threshold	-19.9
2	HLS	Creature	Compression	Ratio	6
2	HLS	Creature	Compression	Attack	13.9
2	HLS	Creature	Compression	Release	100
2	HLS	Creature	Compression	Pre-comp	0
2	HLS	Creature	Compression	resvd	0
2	HLS	Creature	Compression	Lowpass	20000
2	HLS	Creature	Compression	Hipass	0
2	HLS	Creature	Compression	SignIn	0
2	HLS	Creature	Compression	AudIn	0
2	HLS	Creature	Compression	Dry	0
2	HLS	Creature	Compression	Wet	4.5
2	HLS	Creature	Compression	Filter Preview	0
2	HLS	Creature	Compression	RMS size	5
2	HLS	Creature	Compression	Knee	0

2	HLS	Creature	Compression	Auto Make Up Gain		0
2	HLS	Creature	Compression	Auto Release		0
2	HLS	Creature	Compression	Legacy Attack/Knee (		0.25
2	HLS	Creature	Compression	Deprecated Broken A		0
2	HLS	Creature	Compression	Multichannel Mode		0
2	HLS	Creature	Compression	Metering Index		0
2	HLS	Creature	Compression	Bypass	normal	
2	HLS	Creature	Compression	Wet		100
2	HLS	Creature	Compression	Delta	normal	
3	Normal	Creature	Compression	Threshold		-37.2
3	Normal	Creature	Compression	Ratio		1.71
3	Normal	Creature	Compression	Attack		10.7
3	Normal	Creature	Compression	Release		146
3	Normal	Creature	Compression	Pre-comp		0
3	Normal	Creature	Compression	resvd		0
3	Normal	Creature	Compression	Lowpass		20000
3	Normal	Creature	Compression	Hipass		0
3	Normal	Creature	Compression	SignIn		0
3	Normal	Creature	Compression	AudIn		0
3	Normal	Creature	Compression	Dry		0
3	Normal	Creature	Compression	Wet		0
3	Normal	Creature	Compression	Filter Preview		0
3	Normal	Creature	Compression	RMS size		5
3	Normal	Creature	Compression	Knee		0
3	Normal	Creature	Compression	Auto Make Up Gain		0
3	Normal	Creature	Compression	Auto Release		0
3	Normal	Creature	Compression	Legacy Attack/Knee (		0.25
3	Normal	Creature	Compression	Deprecated Broken A		0
3	Normal	Creature	Compression	Multichannel Mode		0
3	Normal	Creature	Compression	Metering Index		0
3	Normal	Creature	Compression	Bypass	normal	
3	Normal	Creature	Compression	Wet		100
3	Normal	Creature	Compression	Delta	normal	
3	HLS	Creature	Compression	Threshold		-37.2
3	HLS	Creature	Compression	Ratio		1.71
3	HLS	Creature	Compression	Attack		10.7
3	HLS	Creature	Compression	Release		146
3	HLS	Creature	Compression	Pre-comp		0
3	HLS	Creature	Compression	resvd		0
3	HLS	Creature	Compression	Lowpass		20000
3	HLS	Creature	Compression	Hipass		0
3	HLS	Creature	Compression	SignIn		0
3	HLS	Creature	Compression	AudIn		0
3	HLS	Creature	Compression	Dry		0
3	HLS	Creature	Compression	Wet		0
3	HLS	Creature	Compression	Filter Preview		0
3	HLS	Creature	Compression	RMS size		5

3	HLS	Creature	Compression	Knee		0
3	HLS	Creature	Compression	Auto Make Up Gain		0
3	HLS	Creature	Compression	Auto Release		0
3	HLS	Creature	Compression	Legacy Attack/Knee (		0.25
3	HLS	Creature	Compression	Deprecated Broken A		0
3	HLS	Creature	Compression	Multichannel Mode		0
3	HLS	Creature	Compression	Metering Index		0
3	HLS	Creature	Compression	Bypass	normal	
3	HLS	Creature	Compression	Wet		100
3	HLS	Creature	Compression	Delta	normal	
1	Normal	FX	Compression	Threshold		-24.3
1	Normal	FX	Compression	Ratio		3.53
1	Normal	FX	Compression	Attack		3
1	Normal	FX	Compression	Release		100
1	Normal	FX	Compression	Pre-comp		0
1	Normal	FX	Compression	resvd		0
1	Normal	FX	Compression	Lowpass		20000
1	Normal	FX	Compression	Hipass		0
1	Normal	FX	Compression	SignIn		0
1	Normal	FX	Compression	AudIn		0
1	Normal	FX	Compression	Dry		-3
1	Normal	FX	Compression	Wet		-2.6
1	Normal	FX	Compression	Filter Preview		0
1	Normal	FX	Compression	RMS size		5
1	Normal	FX	Compression	Knee		0
1	Normal	FX	Compression	Auto Make Up Gain		0
1	Normal	FX	Compression	Auto Release		0
1	Normal	FX	Compression	Legacy Attack/Knee (		0.25
1	Normal	FX	Compression	Deprecated Broken A		0
1	Normal	FX	Compression	Multichannel Mode		0
1	Normal	FX	Compression	Metering Index		0
1	Normal	FX	Compression	Bypass	normal	
1	Normal	FX	Compression	Wet		100
1	Normal	FX	Compression	Delta	normal	
1	HLS	FX	Compression	Threshold		-23.9
1	HLS	FX	Compression	Ratio		3.2
1	HLS	FX	Compression	Attack		1.8
1	HLS	FX	Compression	Release		231
1	HLS	FX	Compression	Pre-comp		0
1	HLS	FX	Compression	resvd		0
1	HLS	FX	Compression	Lowpass		20000
1	HLS	FX	Compression	Hipass		0
1	HLS	FX	Compression	SignIn		0
1	HLS	FX	Compression	AudIn		0
1	HLS	FX	Compression	Dry		-3.2
1	HLS	FX	Compression	Wet		-3.3
1	HLS	FX	Compression	Filter Preview		0

1	HLS	FX	Compression	RMS size		5
1	HLS	FX	Compression	Knee		0
1	HLS	FX	Compression	Auto Make Up Gain		0
1	HLS	FX	Compression	Auto Release		0
1	HLS	FX	Compression	Legacy Attack/Knee (		0.25
1	HLS	FX	Compression	Deprecated Broken A		0
1	HLS	FX	Compression	Multichannel Mode		0
1	HLS	FX	Compression	Metering Index		0
1	HLS	FX	Compression	Bypass	normal	
1	HLS	FX	Compression	Wet		100
1	HLS	FX	Compression	Delta	normal	
2	Normal	FX	Compression	Threshold		0
2	Normal	FX	Compression	Ratio		1
2	Normal	FX	Compression	Attack		3
2	Normal	FX	Compression	Release		100
2	Normal	FX	Compression	Pre-comp		0
2	Normal	FX	Compression	resvd		0
2	Normal	FX	Compression	Lowpass		20000
2	Normal	FX	Compression	Hipass		0
2	Normal	FX	Compression	SignIn		0
2	Normal	FX	Compression	AudIn		0
2	Normal	FX	Compression	Dry		0
2	Normal	FX	Compression	Wet		0
2	Normal	FX	Compression	Filter Preview		0
2	Normal	FX	Compression	RMS size		5
2	Normal	FX	Compression	Knee		0
2	Normal	FX	Compression	Auto Make Up Gain		0
2	Normal	FX	Compression	Auto Release		0
2	Normal	FX	Compression	Legacy Attack/Knee (		0.25
2	Normal	FX	Compression	Deprecated Broken A		0
2	Normal	FX	Compression	Multichannel Mode		0
2	Normal	FX	Compression	Metering Index		0
2	Normal	FX	Compression	Bypass	normal	
2	Normal	FX	Compression	Wet		100
2	Normal	FX	Compression	Delta	normal	
2	HLS	FX	Compression	Threshold		0
2	HLS	FX	Compression	Ratio		1
2	HLS	FX	Compression	Attack		3
2	HLS	FX	Compression	Release		100
2	HLS	FX	Compression	Pre-comp		0
2	HLS	FX	Compression	resvd		0
2	HLS	FX	Compression	Lowpass		20000
2	HLS	FX	Compression	Hipass		0
2	HLS	FX	Compression	SignIn		0
2	HLS	FX	Compression	AudIn		0
2	HLS	FX	Compression	Dry		0
2	HLS	FX	Compression	Wet		0

2	HLS	FX	Compression	Filter Preview		0
2	HLS	FX	Compression	RMS size		5
2	HLS	FX	Compression	Knee		0
2	HLS	FX	Compression	Auto Make Up Gain		0
2	HLS	FX	Compression	Auto Release		0
2	HLS	FX	Compression	Legacy Attack/Knee (		0.25
2	HLS	FX	Compression	Deprecated Broken A		0
2	HLS	FX	Compression	Multichannel Mode		0
2	HLS	FX	Compression	Metering Index		0
2	HLS	FX	Compression	Bypass	normal	
2	HLS	FX	Compression	Wet		100
2	HLS	FX	Compression	Delta	normal	
3	Normal	FX	Compression	Threshold		0
3	Normal	FX	Compression	Ratio		1
3	Normal	FX	Compression	Attack		3
3	Normal	FX	Compression	Release		100
3	Normal	FX	Compression	Pre-comp		0
3	Normal	FX	Compression	resvd		0
3	Normal	FX	Compression	Lowpass		20000
3	Normal	FX	Compression	Hipass		0
3	Normal	FX	Compression	SignIn		0
3	Normal	FX	Compression	AudIn		0
3	Normal	FX	Compression	Dry		0
3	Normal	FX	Compression	Wet		0
3	Normal	FX	Compression	Filter Preview		0
3	Normal	FX	Compression	RMS size		5
3	Normal	FX	Compression	Knee		0
3	Normal	FX	Compression	Auto Make Up Gain		0
3	Normal	FX	Compression	Auto Release		0
3	Normal	FX	Compression	Legacy Attack/Knee (		0.25
3	Normal	FX	Compression	Deprecated Broken A		0
3	Normal	FX	Compression	Multichannel Mode		0
3	Normal	FX	Compression	Metering Index		0
3	Normal	FX	Compression	Bypass	normal	
3	Normal	FX	Compression	Wet		100
3	Normal	FX	Compression	Delta	normal	
3	HLS	FX	Compression	Threshold		0
3	HLS	FX	Compression	Ratio		1
3	HLS	FX	Compression	Attack		3
3	HLS	FX	Compression	Release		100
3	HLS	FX	Compression	Pre-comp		0
3	HLS	FX	Compression	resvd		0
3	HLS	FX	Compression	Lowpass		20000
3	HLS	FX	Compression	Hipass		0
3	HLS	FX	Compression	SignIn		0
3	HLS	FX	Compression	AudIn		0
3	HLS	FX	Compression	Dry		0

3	HLS	FX	Compression	Wet		0
3	HLS	FX	Compression	Filter Preview		0
3	HLS	FX	Compression	RMS size		5
3	HLS	FX	Compression	Knee		0
3	HLS	FX	Compression	Auto Make Up Gain		0
3	HLS	FX	Compression	Auto Release		0
3	HLS	FX	Compression	Legacy Attack/Knee (		0.25
3	HLS	FX	Compression	Deprecated Broken A		0
3	HLS	FX	Compression	Multichannel Mode		0
3	HLS	FX	Compression	Metering Index		0
3	HLS	FX	Compression	Bypass	normal	
3	HLS	FX	Compression	Wet		100
3	HLS	FX	Compression	Delta	normal	
1	Normal	Music	Compression	Threshold		-29.8
1	Normal	Music	Compression	Ratio		3.11
1	Normal	Music	Compression	Attack		3
1	Normal	Music	Compression	Release		100
1	Normal	Music	Compression	Pre-comp		0
1	Normal	Music	Compression	resvd		0
1	Normal	Music	Compression	Lowpass		20000
1	Normal	Music	Compression	Hipass		0
1	Normal	Music	Compression	SignIn		0
1	Normal	Music	Compression	AudIn		0
1	Normal	Music	Compression	Dry		-3.6
1	Normal	Music	Compression	Wet		-2.9
1	Normal	Music	Compression	Filter Preview		0
1	Normal	Music	Compression	RMS size		5
1	Normal	Music	Compression	Knee		0
1	Normal	Music	Compression	Auto Make Up Gain		0
1	Normal	Music	Compression	Auto Release		0
1	Normal	Music	Compression	Legacy Attack/Knee (		0.25
1	Normal	Music	Compression	Deprecated Broken A		0
1	Normal	Music	Compression	Multichannel Mode		0
1	Normal	Music	Compression	Metering Index		0
1	Normal	Music	Compression	Bypass	normal	
1	Normal	Music	Compression	Wet		100
1	Normal	Music	Compression	Delta	normal	
1	HLS	Music	Compression	Threshold		-27.5
1	HLS	Music	Compression	Ratio		2.9
1	HLS	Music	Compression	Attack		1.9
1	HLS	Music	Compression	Release		283
1	HLS	Music	Compression	Pre-comp		0
1	HLS	Music	Compression	resvd		0
1	HLS	Music	Compression	Lowpass		20000
1	HLS	Music	Compression	Hipass		0
1	HLS	Music	Compression	SignIn		0
1	HLS	Music	Compression	AudIn		0

1	HLS	Music	Compression	Dry		-2.8
1	HLS	Music	Compression	Wet		-2.8
1	HLS	Music	Compression	Filter Preview		0
1	HLS	Music	Compression	RMS size		5
1	HLS	Music	Compression	Knee		0
1	HLS	Music	Compression	Auto Make Up Gain		0
1	HLS	Music	Compression	Auto Release		0
1	HLS	Music	Compression	Legacy Attack/Knee (		0.25
1	HLS	Music	Compression	Deprecated Broken A		0
1	HLS	Music	Compression	Multichannel Mode		0
1	HLS	Music	Compression	Metering Index		0
1	HLS	Music	Compression	Bypass	normal	
1	HLS	Music	Compression	Wet		100
1	HLS	Music	Compression	Delta	normal	
2	Normal	Music	Compression	Threshold		0
2	Normal	Music	Compression	Ratio		1
2	Normal	Music	Compression	Attack		3
2	Normal	Music	Compression	Release		100
2	Normal	Music	Compression	Pre-comp		0
2	Normal	Music	Compression	resvd		0
2	Normal	Music	Compression	Lowpass		20000
2	Normal	Music	Compression	Hipass		0
2	Normal	Music	Compression	SignIn		0
2	Normal	Music	Compression	AudIn		0
2	Normal	Music	Compression	Dry		0
2	Normal	Music	Compression	Wet		0
2	Normal	Music	Compression	Filter Preview		0
2	Normal	Music	Compression	RMS size		5
2	Normal	Music	Compression	Knee		0
2	Normal	Music	Compression	Auto Make Up Gain		0
2	Normal	Music	Compression	Auto Release		0
2	Normal	Music	Compression	Legacy Attack/Knee (		0.25
2	Normal	Music	Compression	Deprecated Broken A		0
2	Normal	Music	Compression	Multichannel Mode		0
2	Normal	Music	Compression	Metering Index		0
2	Normal	Music	Compression	Bypass	bypassed	
2	Normal	Music	Compression	Wet		100
2	Normal	Music	Compression	Delta	normal	
2	HLS	Music	Compression	Threshold		-38.8
2	HLS	Music	Compression	Ratio		4.26
2	HLS	Music	Compression	Attack		3
2	HLS	Music	Compression	Release		100
2	HLS	Music	Compression	Pre-comp		0
2	HLS	Music	Compression	resvd		0
2	HLS	Music	Compression	Lowpass		20000
2	HLS	Music	Compression	Hipass		0
2	HLS	Music	Compression	SignIn		0

2	HLS	Music	Compression	AudIn		0
2	HLS	Music	Compression	Dry		0
2	HLS	Music	Compression	Wet		3.5
2	HLS	Music	Compression	Filter Preview		0
2	HLS	Music	Compression	RMS size		5
2	HLS	Music	Compression	Knee		0
2	HLS	Music	Compression	Auto Make Up Gain		0
2	HLS	Music	Compression	Auto Release		0
2	HLS	Music	Compression	Legacy Attack/Knee (		0.25
2	HLS	Music	Compression	Deprecated Broken A		0
2	HLS	Music	Compression	Multichannel Mode		0
2	HLS	Music	Compression	Metering Index		0
2	HLS	Music	Compression	Bypass	normal	
2	HLS	Music	Compression	Wet		100
2	HLS	Music	Compression	Delta	normal	
3	Normal	Music	Compression	Threshold		0
3	Normal	Music	Compression	Ratio		1
3	Normal	Music	Compression	Attack		3
3	Normal	Music	Compression	Release		100
3	Normal	Music	Compression	Pre-comp		0
3	Normal	Music	Compression	resvd		0
3	Normal	Music	Compression	Lowpass		20000
3	Normal	Music	Compression	Hipass		0
3	Normal	Music	Compression	SignIn		0
3	Normal	Music	Compression	AudIn		0
3	Normal	Music	Compression	Dry		0
3	Normal	Music	Compression	Wet		0
3	Normal	Music	Compression	Filter Preview		0
3	Normal	Music	Compression	RMS size		5
3	Normal	Music	Compression	Knee		0
3	Normal	Music	Compression	Auto Make Up Gain		0
3	Normal	Music	Compression	Auto Release		0
3	Normal	Music	Compression	Legacy Attack/Knee (		0.25
3	Normal	Music	Compression	Deprecated Broken A		0
3	Normal	Music	Compression	Multichannel Mode		0
3	Normal	Music	Compression	Metering Index		0
3	Normal	Music	Compression	Bypass	normal	
3	Normal	Music	Compression	Wet		100
3	Normal	Music	Compression	Delta	normal	
3	HLS	Music	Compression	Threshold		0
3	HLS	Music	Compression	Ratio		1
3	HLS	Music	Compression	Attack		3
3	HLS	Music	Compression	Release		100
3	HLS	Music	Compression	Pre-comp		0
3	HLS	Music	Compression	resvd		0
3	HLS	Music	Compression	Lowpass		20000
3	HLS	Music	Compression	Hipass		0

3	HLS	Music	Compression	SignIn		0
3	HLS	Music	Compression	AudIn		0
3	HLS	Music	Compression	Dry		0
3	HLS	Music	Compression	Wet		0
3	HLS	Music	Compression	Filter Preview		0
3	HLS	Music	Compression	RMS size		5
3	HLS	Music	Compression	Knee		0
3	HLS	Music	Compression	Auto Make Up Gain		0
3	HLS	Music	Compression	Auto Release		0
3	HLS	Music	Compression	Legacy Attack/Knee (		0.25
3	HLS	Music	Compression	Deprecated Broken A		0
3	HLS	Music	Compression	Multichannel Mode		0
3	HLS	Music	Compression	Metering Index		0
3	HLS	Music	Compression	Bypass	normal	
3	HLS	Music	Compression	Wet		100
3	HLS	Music	Compression	Delta	normal	

A.4 Extracted center frequency and gain level values for the GA equalisation system.

Table 1: Low Frequencies

Track	25 Hz	32 Hz	40 Hz	50 Hz	63 Hz	79 Hz	100 Hz	126 Hz	158 Hz	200 Hz
Narration	-16.2589	-3.24516	-0.46701	0.63826	-19.1196	-19.8876	-12.1399	-17.8673	-17.6276	-16.2045
Music	-3.47615	-7.04984	-15.955	-13.1465	-11.8527	-8.36933	-8.79533	-12.2842	-11.4766	-8.96481
FX	3.251021	-0.85848	-4.64922	-10.0816	-13.5245	-18.3756	1.952474	-2.95643	-13.6994	-7.31175
Creature	-11.8073	-17.4856	-12.6942	-17.5033	-14.4449	-7.77093	-4.9506	-10.2542	-2.73807	-17.674
Child	-2.70726	-12.3048	-2.97808	4.284527	1.479031	-10.2031	-14.5495	-8.2784	4.489012	-18.1473

Table 2: Mid Frequencies

Track	251 Hz	316 Hz	398 Hz	501 Hz	631 Hz	794 Hz	1000 Hz	1259 Hz	1585 Hz	1995 Hz
Narration	-19.0894	-18.4385	-19.5832	-17.7758	-12.1807	-19.9851	-12.4704	-18.5427	-10.9162	-18.1077
Music	-15.7262	-19.6816	-3.71328	-19.5978	-10.4557	-19.5625	-15.7913	-18.0299	-12.8698	-4.12739
FX	-7.04462	-17.2162	-7.16998	-19.595	-17.2311	-19.8429	-17.0722	-19.9107	-19.8833	-19.5113
Creature	-4.25659	-13.1539	-15.3825	-17.4701	-10.0417	-18.0532	-17.3673	-15.6658	-15.9478	-13.1311
Child	0.388975	-1.82121	-4.39177	-6.82143	-8.5657	-12.5086	-9.75658	-16.7215	-8.70788	-7.81747

Table 3: High Frequencies

Track	2512 Hz	3162 Hz	3981 Hz	5012 Hz	6310 Hz	7943 Hz	10000 Hz	12589 Hz	15849 Hz	19953 Hz
Narration	-14.7509	-17.9803	-1.73197	-11.9522	2.508265	3.879909	-17.7477	4.320808	-16.1325	5.063108
Music	-9.22202	-18.5643	-14.2849	-6.18039	-17.3768	-14.6886	-8.25154	-4.06849	-6.85881	19.25645
FX	-9.83723	-16.9923	6.402069	2.109286	-19.975	-1.34785	5.696855			
Creature	-17.2719	-14.1234	-18.0879	-16.0797	-14.6926	-13.51	-18.1572	-15.7962	-17.1322	-10.8092
Child	-4.3822	-1.11786	6.639855	10.8418	6.141834	13.4797	-3.11061	-8.63379	-14.894	16.12857

A.5 Differentiable Hearing Loss Simulation

The final improved version of the simulation suggested by (21) and presented in Chapter 3 served as the foundation for the design of a differentiable version of the hearing loss simulation in order to be used within a differentiable digital signal processing (DDSP) prototype system. Producing a differentiable version of the hearing loss simulation (DHLS) enables it to be used in machine learning-based systems this way encouraging further research beyond the field of audio production.

Four processing modules make up the DHLS, each of which simulates a different element of hearing loss. The frequency separation in the cochlea is first approximated using a gammatone filterbank. The design of these filters is based on the method suggested in (85), which generates the filterbank using a collection of FIR filters. Additionally, the DHLS applies spectral smearing, audiogram attenuation, and rapid loudness growth for frequencies beyond 1000 Hz. Finally, temporal jitter is applied to the low-frequency bands which are then summed with the high-frequency bands to form the output of the simulation.

The goal of the spectral smearing function in the system is to mimic the effect of decreased frequency resolution which results from the widening of the auditory filters. The proposed differentiable implementation applies spectral smearing by multiplying the signals in each band above 1 kHz by a white noise signal that has undergone low-pass filtering. A parametric equaliser with peaking filters focused on octave-spaced frequency bands is then used to achieve the audiogram attenuation, which corresponds to either mild or moderate hearing loss. By applying the frequency sampling method, feed-forward FIR equivalents are used to approximate the IIR biquad filters (86).

An upwards expansion module is then added to mimic the rapid increase in loudness observed in listeners with hearing loss. This is achieved by modifying the compressor architecture presented in (87). More specifically, this module uses a single-band expander with a shared attack and release time constant. Finally, the lower frequencies are processed by a temporal disruption unit that aims to mimic the effect of loss of temporal resolution. To achieve this, a low-pass filtered white noise signal is used to randomly disturb the index of audio samples in the low-frequency regions.

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