MACQUARIE UNIVERSITY

DOCTORAL THESIS

Hearing aids and music

Rémi Marchand

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Audiology and Hearing Research Group
Department of Linguistics

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"Music is like a dream. One that I cannot hear."

Ludwig van Beethoven

Contents

Abstract **xi**

Declaration of Authorship **xiii**

Acknowledgements **xv**

1 Introduction **1**

1.1 Motivation . . . 1

1.2 Objectives . . . 2

1.3 Background . . . 3

1.3.1 Characteristic properties of speech . . . 3

1.3.2 Characteristic properties of music . . . 5

1.3.3 Auditory processing affected by hearing loss . . . 10

1.3.4 Digital hearing aid features . . . 11

1.3.5 Individual differences . . . 15

1.4 Outline . . . 15

2 Hearing aids and music: an analysis of the factors influencing hearing aid satisfaction **19**

2.1 Abstract . . . 19

2.2 Introduction . . . 20

2.3 Method . . . 24

2.3.1 Questionnaire . . . 24

2.3.2 Participants . . . 26

2.3.3 Statistical analysis . . . 27

2.4 Results . . . 28

2.4.1 Participants . . . 28

2.4.2 Hearing aids . . . 28
2.4.3 Listening conditions . . . 29
2.4.4 Experienced problems while listening to music with HAs . . . 31
2.4.5 Satisfaction with HAs while listening to music . . . 36
Overall satisfaction . . . 36
Effect of individual and group variables on overall satisfaction 36
Effect of encountered problems on satisfaction . . . 37
Effect of listening condition on satisfaction . . . 40
2.5 Discussion . . . 40
2.5.1 Listening habits . . . 41
2.5.2 Main problems experienced when listening to music with HAs 42
2.5.3 Factors influencing satisfaction while listening to music with HAs . . . 46
2.5.4 Limitations and outlook . . . 46
2.6 Conclusions . . . 47
2.7 Acknowledgements . . . 48
2.8 Supporting information . . . 48

3 Preferred frequency-specific gains and compression ratios in hearing aids for music listening 49

3.1 Abstract . . . 49
3.2 Introduction . . . 50
3.3 Method . . . 54
3.3.1 Participants . . . 54
3.3.2 Stimuli . . . 55
Music material . . . 55
Input stimuli . . . 58
Simulated hearing aid . . . 59
Instrumentation . . . 61
3.3.3 Procedure . . . 62
3.3.4 Statistical analyses . . . 63
3.4 Results . . . 65

3.4.1 Listening levels 65
3.4.2 Spectral tilt and CR preferences 66
3.4.3 Sound quality attributes, importance and satisfaction 68
3.5 Discussion 70
3.6 Conclusions 76
3.A Supporting information:
Simulation of the most common music listening scenario 78
3.A.1 Simulated stereo playback in a living room 79
3.A.2 Master HA processing 81
3.A.3 Additional information 84

4 Discussion, implications, limitations and future directions 85
4.1 Summary of major findings 85
4.2 Limitations and implications 89
4.3 Conclusion 91

A Overview of the full questionnaire 93

B Supplementary questions omitted in Chapter 2 99

C Additional information on the music material 103

D Instructions given to participants during the listening experiment 107

E Graphical user interface of the listening experiment 109

F Human Research Ethics Committee Letters 111

Bibliography 117

List of Figures

1.1 Factors influencing the perception of music with hearing aids 3

2.1 Age and degree of hearing loss of the participants 29
2.2 Music listening scenarios and conditions 30
2.3 Problems experienced while listening to music with hearing aids . . . 32
2.4 Hearing aid satisfaction with music 36
2.5 Hearing aid satisfaction in different music listening scenarios 41

3.1 Audiometric thresholds of the three groups of participants 55
3.2 Sound pressure levels of the musical excerpts 57
3.3 Examples of signal processing manipulations 61
3.4 Graphical user interface of the experiment, preference ratings 64
3.5 Differences in input levels between females and males 65
3.6 Surface fitted to preference ratings 67
3.7 Estimated preference of signal processing changes 69
3.8 Sound quality attributes, importance and satisfaction ratings 71
3.9 Signal flow diagram of the master hearing aid processing. 78
3.10 Signal flow diagram of the listening experiment. 79
3.11 Filters applied to aided and the acoustic paths 81
3.12 Signal flow diagram of the master hearing aid processing. 81
3.13 Example of sound pressure levels of a stimulus at the HA input and output 83

B.1 Importance of music 100
B.2 Genres of music participants listen to 100
B.3 Proportion of musicians among the participants 101
B.4 Problems experienced while playing music with hearing aids 101

E.1 Graphical user interface of the experiment, importance ratings of sound quality attributes 110
E.2 Graphical user interface of the experiment, satisfaction ratings of sound quality attributes 110

List of Tables

1.1 Review of potential factors influencing music perception with hearing aids . . . 17

2.1 Relationship between the problems experienced while listening to music with hearing aids and demographic data and hearing aid satisfaction 34
2.2 Exploratory principal component analysis for the frequency of experienced problems . . . 39
2.3 Multiple regression for the prediction of hearing aid satisfaction ratings 39

3.1 List of music stimuli . . . 58

4.1 List of retained factors to take in consideration when manipulating HA amplification specifically for music listening. . . . 92

A.1 Full list of questions and answer options . . . 93

C.1 Additional information on the music material downloaded for the listening experiment . . . 104
C.2 List of preferred pieces of music . . . 105

Abstract

Currently, hearing aids (HAs) are adjusted to compensate for individual hearing loss primarily to maximise the clarity and comfort of speech. Electroacoustic characteristics and settings of HAs may be ideal for speech recognition, but not for music enjoyment. The aims of this study were to better understand the musical listening habits of HA users, to identify the main issues they experience while listening to music and to develop signal processing recommendations specifically for music.

A survey on music listening with HAs was conducted and 151 respondents were recruited. The survey showed that HA users listen mainly to recorded music at home and use the HAs in their default program. 30% of the respondents were dissatisfied with the performance of their HAs and experience problems that need to be addressed. The most prevalent problems identified were related to the HA sound quality and suggest that frequency-specific gains and compression algorithms should be improved.

A follow-up study involved a controlled listening experiment to further understand the signal processing strategies preferred by participants when listening to instrumental music in relation to the problems highlighted by the survey. Manipulations of the signal processing comprise changes in frequency-specific insertion gains and compression ratios relative to those prescribed by the NAL-NL2 standard fitting formula. The preferred frequency-specific amplification provided by the HAs indicates that more gain at low frequency and less gain at high frequency than prescribed by NAL-NL2 standard fitting formulae is needed. The preferred amount of compression applied by the HAs depended on the music stimuli at the input of the HA with overall preference for more compression than prescribed by NAL-NL2.

Future research should derive amplification prescription schemes specifically targeted at optimizing the enjoyment of music with HAs. The optimal prescription will potentially depend on the acoustic characteristics of the specific music being listened to.

Declaration of Authorship

I, Rémi Marchand, state that this work has not previously been submitted for a degree or diploma in any university. To the best of my knowledge and belief, the thesis contains no material previously published or written by another person except where due reference is made in the thesis itself.

Chapter 2 describes an online survey that I have conceptualized, designed and implemented. During the questionnaire development, two focus group sessions were held during which I took the role of debate facilitator with the assistance of Katrina Freeston. Several discussions with Jörg Buchholz, Valerie Looi, Harvey Dillon and Maja Serman helped to perfect the design of the questionnaire. I carried out the data analysis and wrote the entire chapter. Valerie Looi, Jörg Buchholz and Harvey Dillon provided valuable feedback on the manuscript and the interpretation of the results.

Chapter 3 describes a listening experiment of which the design was based on discussions with Jörg Buchholz and Harvey Dillon. To conduct this experiment, I implemented a graphical user interface (GUI) that had to be incorporated in the National Acoustics Laboratories (NAL) master hearing aid developed by Cong-Van Nguyen. This allowed to vary in real-time the master hearing aid amplification parameters, which was a major and time consuming task of the experiment implementation. Jason Heeris provided me with highly helpful insights on the existing master hearing aid. Incorporated in the master hearing aid, I implemented the simulation of a semi-open fitting, and the simulation of a realistic playback of music stimuli with a stereo system inside a living room. This was made possible by the authors of the Ambisonic Recordings of Typical Environments (ARTE) Database who measured the room impulse responses and transfer functions used in this simulation. Jörg Buchholz wrote most of the program used for the calibration of the master hearing aid which I executed. I carried out the data collection and the data analysis. I also wrote the chapter with contributions from Jörg Buchholz and Harvey Dillon on the manuscript and the interpretation of the results.

Ethics review, guidance and approval have been obtained from:

- Australian Hearing Human Research Ethics Committee: 2016-22
- Macquarie University Human Research Ethics Committee: 5201700038
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Signed:

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Sydney, Australia

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List of Abbreviations

4FAHL	**4 F**requency **A**verage **H**earing **L**oss
ADC	**A**nalog-to-**D**igital **C**onverter
BTE	**B**ehing-**T**he-**E**ar (hearing aid)
CIC	**C**ompletely-**I**n-**Canal** (hearing aid)
CR	**C**ompression **R**atio
CT	**C**ompression **T**hreshold
GUI	**G**raphical **U**ser **I**nterface
HA	**H**earing **A**id
HATS	**H**ead **A**nd **T**orso **S**imulator
HL	**H**earing **L**oss
IR	**I**mpulse **R**esponse
ITC	**I**n-**T**he-**C**anal (hearing aid)
ITE	**I**n-**T**he-**E**ar (hearing aid)
LTAS	**L**ong-**T**erm **A**verage **S**pectrum
LTASS	**L**ong-**T**erm **A**verage **S**peech **S**pectrum
MUSHRA	**MU**ltiple **S**timuli with **H**idden **R**eference and **A**nchor
REIG	**R**eal-**E**ar **I**nsertion **G**ain
RIC	**R**eceiver-**I**n-**C**anal (hearing aid)
RIR	**R**oom **I**mpulse **R**esponse
RMS	**R**oot-**M**ean-**S**quare
WDRC	**W**ide **D**ynamic-**R**ange **C**ompression

Chapter 1

Introduction

1.1 Motivation

Currently, hearing aids are adjusted to compensate for individual hearing loss primarily to maximise the clarity and comfort of speech. Speech communication is an important aspect of our daily life. Nonetheless, other acoustic signals such as music may be very important for some people and have a significant social aspect. Hearing aids usually do not take into account the differences between speech and music. Thus, music listeners may be disappointed by the performance of their hearing aids when listening to music (e.g., Leek et al., 2008; Madsen and Moore, 2014b).

Electroacoustic characteristics and settings of hearing instruments may be ideal for speech recognition, but not for music enjoyment. Thus, hearing aids may process music inappropriately, since there are many acoustic differences between speech and music (Chasin and Russo, 2004; Hockley et al., 2010). There are large variations of acoustic properties between, and probably within, music genres. Moreover, it is still unclear how the impaired auditory processing of the individual listeners interfere with their perception of music. Finally, some aspects specific to each listener -such as their musical practice or their preferences- are of great influence. Modern hearing aids have sufficient processing power to not only recognise when music is present, but also to analyse its temporal and spectral characteristics and to adjust the hearing aid signal processing accordingly. Most of the manufacturers offer different processing programs for customers in need of a specific amplification for music. However, recent studies (Madsen and Moore, 2014b; Greasley et al., 2015) suggested that these so-called **music programs** may not improve significantly the experience of music listening.

1.2 Objectives

As described in the current Chapter, hearing aids are designed primarily to maximise the intelligibility of speech. In addition to the differences between speech and music, music presents much more variations than speech and there are many factors that may affect the enjoyment of music. The primary objectives of this thesis were:

1. To better understand the musical listening habits of hearing aid users, by identifying: (*where*) the listening environments in which they listen to music; (*how*) whether they use their hearing aids to do so and whether their hearing aids have a music program, or if they use another device; and identifying (*what*) the genre and type of music.

2. To identify the problems faced by hearing-impaired listeners when listening to music with their hearing aids and to determine which of these problems are detrimental for the enjoyment of music.

3. To better understand what are the factors that disturb the enjoyment of music amplified by hearing aids, and whether those factors are environmental, related to the hearing aid signal processing and fitting, induced by hearing loss or related to other individual aspects of the listeners.

4. To identify the hearing aid signal processing strategies preferred by hearing-impaired listeners when listening to music and determine whether these preferences vary from the hearing aid signal processing standards employed for speech communication.

5. To develop hearing aid signal processing recommendations specifically for music.

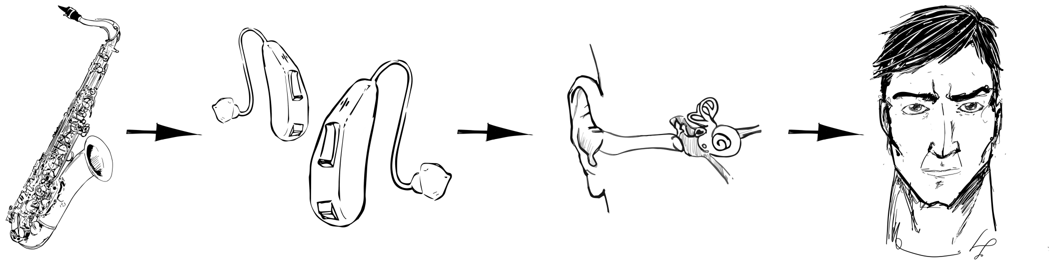

FIGURE 1.1: Factors influencing the perception of music with hearing aids can be divided into four categories: (1) the acoustic properties of the music, (2) the hearing aid signal processing and fitting, (3) the auditory processing affected by hearing loss and, (4) individual factors of the listener.

1.3 Background

Speech and music sound signals differ in many ways (e.g., Chasin and Russo, 2004; Hockley et al., 2010). The following sections will first outline some of these differences in more detail and then, provide an overview of potential factors that are likely to influence the perception of music while using hearing aids. An extensive list of references summarizes these factors in Table 1.1 into four categories (see Figure 1.1): (1) acoustics (sections 1.3.1 and 1.3.2), (2) hearing aid features (section 1.3.4), (3) auditory processing affected by hearing loss (section 1.3.3) and (4) individual factors (section 1.3.5).

1.3.1 Characteristic properties of speech

Speech is produced by air from the lungs passing the **vocal folds** in the larynx. The vibration of the vocal folds gives the speech its **fundamental frequency**, about 120 Hz for male speakers and 220 Hz for female speakers (Plack, 2014). This fundamental frequency is heard as a **pitch**. Using the muscles in the larynx, the frequency of the vocal folds vibration can be varied. Thus, it is possible to speak varying the fundamental frequency to give additional meaning to an utterance. Variation of fundamental frequency over time is referred to as **intonation** and, for instance, is an essential cue in many languages (e.g. English) to differentiate a statement from a question (e.g., a yes/no question exhibits a rising intonation while statements are more flat). Increase or decrease of fundamental frequency can also convey emotions. This is also done while singing to produce a musical melody.

The sound coming from the vocal folds is then modified by the **vocal tract** (i.e., the resonating structures in the throat, mouth and nose). These modifications are controlled by mobile articulators: pharynx, soft palate, lips, jaws and tongue. The motion of the articulators in different positions is responsible of the production of different **vowels**. Constrictions of the vocal tract are preventing or restricting the release of air and thus, produce **consonants**. Roughly, speech may be seen as sequences of vowels interrupted by consonants. The **phonemes** are the basic sounds of speech, such as individual vowels or consonants (which are generally the softer phonemes). The duration of a vowel typically ranges between 200 and 300 msec (House, 1961) while a burst may be as short as 5-10 msec (Schatz, 1954). A particular phoneme can also be described by its spectral shape containing intense regions called **formants** (i.e., broad peaks in the speech spectrum). Thus, vowels are complex tones with characteristic spectra depending on the position of the articulator. For example, the vowels *ee* and *oo* have a similar first formant (i.e., the lower frequency formant) and may be differentiated by their 2^{nd} and higher formants. We may note that the high-frequency components of speech are commonly weaker than the low-frequency components (Dillon, 2012) where the **loudness** (i.e., the psychological perception of the intensity of a sound) of speech most originates. Speech has a well-defined relation between loudness and its **intensity**, i.e., the physical quantity relating to the magnitude of a sound (Moore and Glasberg, 1996). Moreover, the acoustic characteristics and the waveform of an individual phoneme will be affected by the nature of the preceding phoneme and of the following phoneme, referred to as co-articulation.

It is now commonly assumed that the long-term average **frequency spectra** of speech signals are relatively uniform in shape. Pearsons et al. (1977) showed consistent spectral shapes across speakers. The sound source (i.e., the human vocal tract) is usually similar, even though differences are noticeable between voices of females, males and children. They highlighted that the main difference between females and males can be observed at frequencies below 200 Hz. For increased vocal effort, they highlighted a trend towards greater high frequency content at higher voice levels. For all three groups (females, males and children), the levels at low frequencies remain fairly constant. The spectrum of conversational speech has been characterized by Byrne et al. (1994) by **long-term average spectrum analysis** (**LTAS**) for 12 languages. They found similar LTAS for the languages they investigated. It is now standardized as the **Long-Term Average Speech Spectrum** (**LTASS**) (ANSI, 1997).

Despite the similarities mentioned above, the speech of individual talkers differ from one another. The individual characteristics of the vocal tract give to the speech its characteristic **timbre**, which allows us to differentiate the voices of two different speakers and to identify the voice of a familiar person. Moreover, one individual may speak very differently in various situations depending on the context. For instance, to guarantee adequate understanding by a listener, a speaker may vary the **sound level** of their voice. Pearsons et al. (1977) provided measures of speech levels in different noise environments at five vocal effort levels (described as "casual", "normal", "raised", "loud" and "shout"). The average speech levels at a distance of 1 m were found to vary from about 50 dB SPL (Sound Pressure Level) for soft speech to 85 dB SPL for shouted speech. Larger difference in voice levels between females, males and children and within individuals may be observed with increased vocal effort. For increased vocal effort, other signal variations can be observed such as in fundamental frequency, formant frequencies and formant bandwidth (Beechey et al., 2018). The tendency to increase one's vocal effort in the presence of competing noise is called the Lombard effect (Cooke et al., 2014). Variation in sound level may also provide expression. Increase in level is commonly used to emphasize one syllable, one word or one sentence. Finally, a speaker may vary the tempo of their speech to convey meaning, to adjust to the environment, or to adjust to a listener's hearing abilities. For example, if someone speaks quickly, it may convey urgency.

The characteristic properties of speech mentioned above, such as sound levels and spectrum of typical speech, have a direct effect on the frequency dependant gains that are applied in hearing aids (e.g., Buchholz, 2013).

1.3.2 Characteristic properties of music

Acoustical instruments are commonly subdivided into two groups: **percussive** and **non-percussive** instruments. Non-percussive instruments (also described as **melodic** instruments) are characterized by tones with a temporal part that is essentially cyclically repetitive. This part of the tones is usually referred to as its **steady state** and heard as a **pitch**. The temporal part of the tone preceding the steady state is called **attack transient** and the following part is called **decay transient**. Non-percussive instruments may also be classified into subgroups, e.g., wind instruments (wood and brass) or string instruments (bowed and plucked). The duration of the steady state and the attack and decay transients is greatly dependent of the subgroup of the instrument being played. For example, the duration of the transient time of the fundamental of a trumpet is about 10 ms independent of the frequency (Luce and Clark, Jr., 1967) while a pipe organ, with a fundamental of about 150 Hz, has a transient duration between 200 and 300 ms before settling into a steady state (Fletcher, 1976).

A specific musical instrument may also be described by its **spectral characteristics**. Sivian et al. (1931) measured the frequency spectrum of the individual instruments composing a symphony orchestra. Measurements of frequency spectra of many different musical instruments may be found in the literature: Luce and Clark, Jr. (1967) (trumpet, trombone and French horn); Suzuki (1986) (piano); Caldersmith and Jansson (1980) (acoustical guitar). Regarding its frequency response, a specific instrument can be characterized by its **timbre**. The term timbre involves spectral qualities of sound apart from pitch, while pitch refers to the height of a tone and is closely related to the fundamental frequency. Following this view, timbre is often described as the tone colour. The timbre being very different from one category of instrument to another, it may even be greatly different for two instruments within the same category as well. In an attempt to characterise different models of violin, Gabrielsson and Jansson (1976) used LTAS analysis to compare violins of different sound qualities.

The different categories of instruments differ not only in temporal and spectral characteristics but also in **sound levels**. Phillips and Mace (2008) measured individual acoustic instruments' sound levels in music practice rooms during individual rehearsals using a microphone clipped on the shoulder of the musician. Depending on the type of instrument being played, the average sound levels ranged from 87 dBA[1] (averaged across string instruments) to 95 dBA (averaged across brass instruments). Individual amplified instruments would present a larger range of potential

[1] A-weighted decibels, IEC (2003)

sound levels with the overall level depending on the degree of amplification provided. Lebo and Oliphant (1968) reported the sound level of a symphony orchestra concert measured from the audience location. During a fortissimo passage[2], the symphony orchestra showed an average sound level of 64 dBA. More recently, Killion (2009) reported measurements of sound levels of different live music conditions. This report showed sound levels up to 104 dBC[3] measured from the audience location during a fortissimo passage played by a symphony orchestra. The differences in sound levels reported in the two studies may be due to different measurement techniques, the specific acoustic properties of the different concert halls, and/or the location of the measurement microphone. With other genres of music, the musical instruments being played are very likely to be amplified. In Lebo et al. (1967), a rock-and-roll concert exhibited an average sound level of 78 dBA measured from the audience. Beach et al. (2013) reported average sound levels of 97 dBA at nightclubs and 92 dBA at live music venues, measured from dosimeters attached to audience members. The difference between the two reports may be explained by the improvements of amplifying systems and loudspeakers that has been made and allowed the sound to be presented at higher sound levels without distortion than during the earlier study. The intensity of music at given moment

Also, music may show a large **dynamic range** since it may alternate from soft to loud passages and vice versa in the same song. These dynamic changes can be estimated by describing the long-term dynamic range which refers to the difference between the softest levels and the peak levels of a signal. Another way to characterize the dynamic range of a signal is to measure its **crest factor**. It is defined as the ratio of the peak amplitude to the long-term **root-mean-square** (**RMS**) amplitude of the signal and describes its short-term dynamic range. More specifically, it characterizes the intensity of a peak of a signal regarding its overall level. We have seen that music may be played in many different conditions. Different instruments may be involved and amplified depending on the genre of music. Nonetheless, music listening is not limited to played instrument or live concert conditions and recorded music can be subject to reduction of its dynamic range. Indeed, live music is expressed within a

[2]i.e., a passage in the music dynamics played "loudly"
[3]C-weighted decibels, IEC (2003)

wide dynamic range whereas most CD recordings have a very small dynamic range. This is due to dynamic range **compression** which is used in the music industry to maximize loudness. While reducing the dynamic range of a signal, compression increases its RMS levels (for the same peak level) resulting in an increase in loudness (Arehart et al., 2011; Croghan et al., 2012). The amount of compression applied to commercial recordings has increased over time although it is argued that large amount of compression negatively affects the music signal quality. This trend for increased compression is now referred as the **loudness war** (Katz, 2007). The reduced dynamic range of recordings nay depend on the genre of music. During the mastering process, substantial amounts of compression limiting may be applied to pop or rock music whereas classical music and jazz tend to have minimal dynamic-range compression. Finally, recorded music can be played at very different sound levels; depending on many factors such as the preferences of the listener (adjustments of the volume control) and the devices being used (e.g., headphones, loudspeakers) as shown by Hodgetts et al. (2007) and Portnuff et al. (2011).

As discussed above, live music usually shows a larger dynamic range than speech since it may alternate between soft and loud passages within a rather short time frame. Within a piece of music, the intensity at which it is being played or its variations in intensity may be referred to as **dynamics**, from soft passages (e.g., piano) to loud passages (e.g., forte) and transitions (e.g., crescendo, decrescendo and sforzando). By contrast, speech rarely alternates from a whisper to a shout in the same context. Nonetheless, the amount of compression applied to some recordings within the audio industry may result in a smaller dynamic range than for speech signals. Moreover, music can be presented at much higher sound levels than speech in a live context or played at low levels in the background of a conversation.

We have also seen that there are a lot of differences between musical instruments. They exhibit different temporal and spectral characteristics and may be played at very different sound levels. Comparing the spectrum of speech (e.g., Pearsons et al., 1977 ; LTASS: Byrne et al., 1994) with the spectra of different musical instruments (e.g., Luce and Clark, Jr., 1967; Suzuki, 1986; Caldersmith and Jansson, 1980) highlights major differences. But the voice itself may be used as a musical instrument when we sing, and speech and sung signals exhibit spectral differences as well.

Sundberg (1970) highlighted different formant frequencies characteristic of speech and sung vowels. Jansson and Sundberg (1972) provided the LTAS of professional opera singers' voice and described the frequency region of a constant frequency peak so-called the **singer's formant**, i.e., a spectrum envelope peak near 3 kHz that is not present in speech. Investigating vocal intensity as a function of fundamental frequency of singers' voices, Titze and Sundberg (1992) showed higher sound levels for higher fundamental frequency. They also reported that professional tenor singers were able to produce 10 dB greater intensity than male non-singers.

The modulation spectrum of speech typically exhibits a maximum between 3 and 4 Hz (e.g., Houtgast and Steeneken, 1985), which reflects the syllabic rate of speech with an average syllable duration of around 250-300ms, but will depend on many parameters (see section 1.3.1). Music can also be characterised by its temporal characteristics that are also subject to variations. In addition to the attack and decay times of played notes that can vary widely (see above), **rhythm** describes patterns in time of music while **tempo** refers to the overall speed of a piece of music and involves the cyclical repetition of a **beat**. Thereby, the tempo can fluctuate within a piece of music, such as a slight speeding up or slowing down of the tempo to convey expression. The tempo from a piece of music to another may vary between 80 and 160 beats per minute (Moelants, 2002). Music exhibit other divisions of time such as the duration of a note as well as the duration between one note preceding another. In particular for music signals, these temporal aspects may be influenced by external factors such as the reverberation of a room or other acoustical features. In the case of speech, reverberation can be seen as a detrimental factor when late reflections of a speech signal interfere with the intelligibility of its subsequent words (e.g., Houtgast and Steeneken, 1985). In contrast, early reflections can aid speech intelligibility by increasing the effective speech level or signal-to-noise energy ratio (SNR), respectively (e.g., Bradley et al., 2003). In some cases, reverberation can be considered as an aspect of the music itself. What would an organ symphony sound like if it was performed in an environment deprived of reverberation?

As seen above, the acoustic properties of music signals vary far more than for speech signals and depends on many factors. An overview of these factors can be seen in the first column of Table 1.1.

1.3.3 Auditory processing affected by hearing loss

For most hearing losses, there are several deficits to be overcome. Some sounds are inaudible. Some sounds can be detected because part of their spectra is audible, but may not be clearly identified because other parts of their spectra (usually the high frequencies) remain inaudible. Typically, the range of levels between the weakest sound that can be detected and the most intense sound that can be tolerated (i.e., the listener dynamic range) is less for a person with a hearing impairment. Moreover, an increment of sound level produces a larger perceived loudness increase perceived for a hearing-impaired person than for a normal-hearing person, a phenomenon referred as **loudness recruitment** (Fowler, 1936; Steinberg and Gardner, 1937).

Another deficit induced by hearing loss is a reduction in **frequency resolution**. It is likely that an impairment of frequency selectivity would reduce the ability to perceive important frequency-related aspects of music. For example, several studies (e.g., Moore and Peters, 1992; Bernstein and Oxenham, 2006) suggest that hearing loss has a deleterious effect on the pitch perception of harmonic complexes. However, music cannot be reduced to a simple perception of pitch or melody variation, and is often presented in the form of polyphonic music, i.e., which contains several different melodic lines played or sung simultaneously. Reduced frequency selectivity is also associated with the increase of upward spread of masking, that is low-frequency sounds masking higher frequency sounds (e.g., Fastl and Zwicker, 2007, Chapter 4). Thus, an instrument with strong low-frequency energy may reduce the clarity or audibility of another instrument with higher frequency components.

The ability to detect changes in sounds over time is referred to as **temporal resolution**. Similarly to the reduction in frequency resolution, a hearing-impaired person has a decreased ability to hear a signal that rapidly follows or precedes a different signal. Glasberg and Moore (1992) investigated the relationship between loudness recruitment and gap-detection thresholds. They found that decreasing the envelope fluctuations of noise bands led to better detection of gaps. This suggests that the use of amplitude compression in hearing aids may be beneficial for gap detection (see section 1.3.4 below for further discussion on effects of compression in hearing aids).

Music has fluctuating envelopes and a reduced sensitivity to gaps may be detrimental for music perception. Thus, hearing impairment might make it harder to hear the onsets of musical notes and, in turn, individual notes being harder to hear (Rasch, 1978; Madsen and Moore, 2014a).

Hearing-impaired people are also less able to separate sounds on the basis of the direction from which they arrive. The ability to spatially separate sounds sources (i.e., **spatial resolution**) is a valuable cue for speech communication when a target speech is present among several sound sources. In the case of music, there is often no single target and a listener may direct their attention to several instruments at once. The primary aim of stereo and surround sound system is to recreate this sense of space associated with music.

Undoubtedly these deficits have an impact on the perception of music but it is still unclear how they affect it and to what extent. More generally, the perception of music and its enjoyment are affected greatly by the type of hearing loss and its duration (Madsen and Moore, 2014b), its severity and its spectral shape (Ricketts et al., 2008; Croghan et al., 2014). An overview of the aspects of the auditory processing that are affected by hearing loss and likely to influence the music perception can be seen in the third column of Table 1.1.

1.3.4 Digital hearing aid features

Hearing aids aim to overcome the deficits associated with a hearing loss, mainly when listening to speech in quiet and noise. We have seen in section 1.3.3 that there are several deficits induced by hearing loss that need to be overcome, even though the main effect of a hearing aid is to improve audibility.

Basically, a digital hearing aid is composed of at least two **microphones** that capture the sound of the environment, an electronic part and a **speaker/receiver** that transfers the amplified sounds to the ear so the wearer can hear them. The electronic part includes an **analog-to-digital converter** (**ADC**) that converts the analog signal captured by the microphone into a digital one, a processor that amplifies and processes the digital signal, and a battery providing the power supply. Digital processing enables manipulating the signals in many ways and its **algorithms** may use compression, expansion, feedback cancellation, noise-suppression, de-reverberation, etc.

Generally, the algorithms are designed to optimize speech understanding in quiet and noise. It is to be noted that there is currently no standard outcome measure of 'music enjoyment' with hearing aids, as opposed to speech intelligibility tests (e.g., Speech Intelligibility Index: ANSI, 1997).

To compensate for reduced dynamic range and loudness recruitment (see section 1.3.3), hearing aids have to amplify weak sounds more than they amplify intense ones. Such a reduction of a large dynamic range of levels in the environment to a smaller range of levels at the output of the hearing aids is processed by compression algorithms. In other word, compression acts on the amplification gain, turning it down as the sounds get stronger. In the case of speech understanding, a hearing aid must provide more amplification at frequencies where speech has the weakest components and where hearing losses are usually the greatest. A compressor can be fitted to a given hearing loss by standard gain prescription procedures (e.g., CAM2, Moore et al., 2010; NAL-NL2, Keidser et al., 2011). These procedures provide the insertion gain to apply depending on the audiogram of the wearer of the hearing aid, with the insertion gain being the amount of gain applied by the hearing aid in addition to the naturally occurring gain within the unaided concha and ear canal. The amount of compression is characterised by the compression ratio, which is defined as the ratio between the change in input level and the change in output level. It is also important to specify the time it takes for the compressor to react to an increase (attack time) or decrease (release time) in the input signal, leading to so-called fast- or slow-compression.

Characteristic properties of speech signals (see section 1.3.1) are influential in the prescription of compression applied by hearing aids. More specifically, compression ratios and thresholds (i.e., the input levels above which compression is applied) are tailored for the sound levels and dynamic range typical of speech signals. Also compressors often apply time constants that are designed for changes in the modulation spectrum of speech. Compression time constants either need to be slow enough not to affect the temporal envelope of speech, or to apply "syllabic compression" (Dillon, 2012, p. 181-182), i.e., to operate on the fluctuations of speech at a syllabic time scale. With respect to music signals, Madsen and Moore (2014a) showed that fast compression in hearing aids could help with the detection of a tone in the presence of another in asynchrony.

Nowadays, most if not all digital hearing aids use **wide dynamic-range compression** (**WDRC**). The term WDRC is used because this type of amplitude compressor adjusts the gain over a wide range of input sound levels. As we saw previously (see section 1.3.2), the dynamic range for music is often wider than for speech and in such cases WDRC can help hearing-impaired people to hear both soft and loud passages at a comfortable level. There are many ways to implement WDRC. Some compressors vary the gain for the entire signal at once (single-channel or wideband compression), while others process individual frequency-bands separately (multi-channel compression). More details on multi-channel compressors including their implementation can be found in Kates (2008), Zölzer (2008), Dillon (2012), and Buchholz (2013).

When using WDRC, there is a trade-off between distortion and audibility (Kates, 2008). WDRC can increase audibility for weak sounds but can also distort the temporal envelope of sounds. Madsen and Moore (2014b) investigated the effects of WDRC on the perceived clarity of individual musical instrument. They showed that the clarity with which an individual instrument/voice could be heard out from a mixture was reduced by the use of both fast- and slow-compression. They did not find a significant overall effect of compression speed, but a few subjects consistently rated clarity to be higher for slow than for fast compression. Croghan et al. (2014) investigated the effects of compression on music preferences and showed that different compression speed may be preferred for different genre of music.

Additionally, hearing aids can be helpful in compensating for decreased temporal resolution ability. Fast-acting compression, where the gain is rapidly increased during weak sounds and rapidly decreased during intense sounds, will make the weaker sounds more audible in the presence of preceding stronger sounds.So, in the case of speech, fast-acting compression will make the weaker sounds slightly more intelligible. Unfortunately, it will also make unwanted weak background noises more audible. To compensate for this, some hearing aids incorporate **noise suppression** algorithms or **expansion**. The basic principle is to amplify only sound levels above a certain threshold and progressively attenuate the levels below it. Nonetheless, the use of expansion may be detrimental for the perceived quality of the signal

(VanBuuren et al., 1999). Also, room reverberation can be amplified by fast-acting compression. In the previous studies, the time constants of WDRC (particularly the attack time) is likely to affect how distorted a sound is perceived to be because it determines the extent of overshoot and therefore influences how much the temporal envelope is changed by the compression. Also, modern hearing aids use multiple time constants in parallel to combine the advantages of slow and fast acting compressions. It is still unclear from the existing literature how dual-compression systems (i.e., compression system combining slow and fast acting compression) compare to single-compressors in terms of music preferences or sound quality.

There are many more hearing aid features that are likely to affect the sound quality of music. For example, Franks (1982) and Ricketts et al. (2008) investigated the impact of the frequency bandwidth of the applied amplification on musical preferences, and Parsa et al. (2013) investigated the effect of non-linear frequency compression (i.e., higher frequencies being reassigned to lower frequencies at which the auditory sensitivity is better) on the quality of music (and speech). The extent to which all these features are responsible for the perceived sound quality and distortion, and how to minimize it, is still to be determined. Distortion of music can also occur due to overload of input and/or output **transducers**. High sound levels, which are especially associated with live music, may also cause distortion when they exceed the upper limit of the dynamic range that the ADC can cope with. For speech, low frequency amplification is reduced to minimize upward spread of masking. This may be different for music. Other features such as **feedback cancellation** and **dereverberation** algorithms or **directional processing** may also be of interest. Finally, the coordination of the compressor and other hearing aid features across ears for bilateral amplification can have an impact on spatial perception (Wiggins and Seeber, 2011; Wiggins and Seeber, 2012; Hassager et al., 2017b; Hassager et al., 2017a), which may be particularly relevant for music listening. The second column of Table 1.1 gives an overview of the hearing aid features potentially influencing music perception.

1.3.5 Individual differences

The interest and experience in music of the listeners may vary a lot from one another and influence their appreciation of music as well (Leek et al., 2008; Madsen and Moore, 2014a). Some people listen to music in the background (Greasley and Lamont, 2016), others are music enthusiasts, and others are musicians who analytically listen to other musicians or even themselves playing (Killion, 2009; Parbery-Clark et al., 2011). Greasley and Lamont (2016) reviews the psychological factors of a listener that influence their preferences in music. In addition to individual preferences, cultural factors are to be taken into account (McDermott et al., 2016). Finally, the hearing aid experience of a listener has an impact on the satisfaction with the performance of their hearing aid while listening to music (Gatehouse, 1992; Leek et al., 2008; Dawes et al., 2014).

1.4 Outline

As part of this thesis, two studies were conducted and will be presented in the following chapters. Chapter 2 describes the design and results of an online survey whose conclusions guided the design of the listening experiment described in Chapter 3.

An extensive literature review is described above (section 1.3) along with an extensive list of references summarized in Table 1.1, broken down into four categories of factors potentially influencing the enjoyment of music with hearing aids: (1) acoustics, (2) hearing aid features, (3) auditory processing affected by hearing loss and (4) individual factors.

From literature review, it was concluded that the implementation of a survey on the use of hearing aids while listening to music was necessary. Chapter 2 describes the data obtained from 151 hearing aid users who took part in this online questionnaire. The survey showed that 30% of the respondents were dissatisfied with their hearing aid for music listening. Also, the survey results identified which of the problems reported by participants were contributing most to their dissatisfaction. These problems clustered in five sound quality attributes and suggest that in particular the prescribed frequency-specific gain and compression should be improved.

Based on the outcomes of the survey, a controlled listening experiment was designed to understand in what ways standard amplification prescription schemes could be varied to improve the performance of hearing aids with music, particularly in regards with the sound quality attributes identified previously. Twenty-six (26) hearing aid users participated in the study which is described in Chapter 3. Data shows that participants preferred amplification schemes different from those prescribed by the NAL-NL2 standard fitting formula. To conduct this experiment, a graphical user interface had to be integrated into a real-time hearing aid platform and the realistic playback of music stimuli with a stereo system inside a living room to an aided hearing-impaired listener had to be simulated. The details of these major and time consuming methods are described in the supplementary material of Chapter 3.

Finally, Chapter 4 provides a summary of the outcomes of the studies that shaped this thesis and discusses the implications of this research for hearing aid development, clinical practice and future research.

TABLE 1.1: Review of potential factors influencing music perception with hearing aids

Acoustics	**Hearing aid features**	**Auditory processing affected by hearing loss**	**Listener**
- Listening conditions (e.g., recorded or live music, self playing...) (Chasin, 2003; Chasin and Russo, 2004; Chasin, 2006; Hockley et al., 2010) - Environment (e.g., home, restaurant, church ...) - Presence of noise - Sound levels, dynamic range and loudness (Lebo and Oliphant, 1968; Lebo et al., 1967; Axelsson et al., 1995; Chasin, 2003; Chasin and Russo, 2004; Chasin, 2006; Killion, 2009; Hockley et al., 2010; Croghan et al., 2012; Schmidt, 2012; Croghan et al., 2014; Croghan et al., 2016; Wilson and Fazenda, 2016) - Signal envelope (VanBuuren et al., 1999; Croghan et al., 2012; Croghan et al., 2014) - Genre of music (Croghan et al., 2012; Higgins et al., 2012; Croghan et al., 2014; Madsen and Moore, 2014b)	- Type/model of HA (Dillon et al., 2003; Chasin, 2006) - ADC (Hockley et al., 2010; Schmidt, 2012) - Prescription formula (e.g., NAL-NL2, CAM2) (Byrne et al., 2001; Keidser et al., 2011; Moore et al., 2011) - Wide Dynamic Range Compression (attack and release time, compression ratio) (VanBuuren et al., 1999; Davies-Venn et al., 2007; Moore, 2008; Kates, 2010; Arehart et al., 2011; Moore et al., 2011; Croghan et al., 2012; Higgins et al., 2012; Croghan et al., 2014; Madsen et al., 2015) - Multichannel-compression (number of channels) (VanBuuren et al., 1999; Higgins et al., 2012; Croghan et al., 2014; Holube et al., 2016) - Expansion (VanBuuren et al., 1999)	- Spectral resolution, pitch perception and auditory filters (Pressnitzer and Patterson, 2001; Pressnitzer et al., 2001; Cousineau et al., 2014; Moore, 2012; Croghan et al., 2014) - Spectral shape of HL (Ricketts et al., 2008; Croghan et al., 2012; Croghan et al., 2014) - Severity of HL (Ricketts et al., 2008) - Type and duration of HL (Madsen and Moore, 2014a) - Loudness recruitment - Distortions - Temporal-fine structure processing - Auditory Scene Analysis (Pressnitzer et al., 2011; Madsen and Moore, 2014b; Madsen and Moore, 2014a; Madsen et al., 2015) - Spatial/binaural hearing (Minnaar, 2010)	- Age (Leek et al., 2008; Madsen and Moore, 2014a) - Personality/interest in music (Leek et al., 2008; Madsen and Moore, 2014a) - Expectations (Madsen and Moore, 2014a) - Attention, distractibility (e.g., background listening, concert...) - HA experience (Gatehouse, 1992; Leek et al., 2008; Dawes et al., 2014) - Musical expertise, training (Killion, 2009; Parbery-Clark et al., 2011; Greasley and Lamont, 2016) - Familiarity, preferences: individual (Wilson and Fazenda, 2016; Greasley and Lamont, 2016) and cultural (McDermott et al., 2016) - Situational appropriateness (e.g., genres and styles with environment) (Greasley and Lamont, 2016) - Gender - ...

Legend for references:
Investigations of music listening with hearing aids/music perception by hearing-impaired people
Investigations of music perception/measurement of music signals/music signals analysis
Investigations of speech perception/measurement of speech signals/speech signals analysis
Investigations of other use of hearing aids/perceptual impact of a hearing aid feature

Acoustics	Hearing aid features	Auditory processing affected by hearing loss	Listener
- Instruments (e.g., singer, piano, percussions...) (notions of timbre, attack...) (Pearsons et al., 1977; Uys et al., 2013; Madsen and Moore, 2014b; Luce and Clark, Jr., 1967; Suzuki, 1986; Caldersmith and Jansson, 1980; Gabrielsson and Jansson, 1976; Cousineau et al., 2014) - Dynamic changes of melody (Pressnitzer and Patterson, 2001; Madsen and Moore, 2014b; Cousineau et al., 2014; Madsen et al., 2015) - Type of music (solo instrument, band) (Madsen et al., 2015) - Dynamic changes of rhythm (Scheirer, 1998)	- Frequency bandwidth (Olson, 1947; Gatehouse, 1992; Franks, 1982; Leijon et al., 1984; Moore and Tan, 2003; Ricketts et al., 2008; Moore, 2012; Colucci, 2013; Moore, 2016; Volk et al., 2016) - Frequency compression/lowering (Uys et al., 2013; Mussoi and Bentler, 2015; Picou et al., 2015) - Music program - Directional processing - Noise suppression - De-reverberation - Time delay induced by processing (Groth and Søndergaard, 2004; Stone et al., 2008) - Receiver/transducer Fitting mode, vent effect, occlusion effect - Feedback suppression (Chasin, 2006; Hockley et al., 2010; Moore, 2016)	- ...	- ...

Legend for references:
Investigations of music listening with hearing aids/music perception by hearing-impaired people
Investigations of music perception/measurement of music signals/music signals analysis
Investigations of speech perception/measurement of speech signals/speech signals analysis
Investigations of other use of hearing aids/perceptual impact of a hearing aid feature

Chapter 2

Hearing aids and music: an analysis of the factors influencing hearing aid satisfaction

Rémi Marchand,[1,2] Jörg Buchholz,[1,2] Valerie Looi[3] and Harvey Dillon[1,2,4]

[1] *The HEARing Cooperative Research Centre*

[2] *Macquarie University*

[3] *Sydney Cochlear Implant Centre*

[4] *National Acoustic Laboratories*

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2.1 Abstract

Aims: To better understand the musical listening habits of hearing aid users, and to identify the main issues they experience while listening to music with their hearing aids. **Method:** After two successive focus group sessions were held, an online survey was developed, consisting of 42 multiple choice and open-ended questions. Study sample: 151 respondents were recruited from two large databases available at the National Acoustic Laboratories, providing detailed information about the respondents such as their age, type and degree of hearing loss, and information about their hearing aids. **Results:** The survey showed that hearing aid users are mainly listening to recorded music at home and use the hearing aids in their universal program. The most prevalent problems identified were difficulties in understanding lyrics, the soft passages of music being too soft, lack of clarity of the music, poor tonal quality, the music being too sharp/shrill and the music being too loud. Results suggest that compression algorithms should be improved. **Conclusions:** About 30% of the users are dissatisfied with the performance of their hearing aids with music and experience problems that need to be addressed. Future research should derive amplification prescription schemes specifically targeted at optimizing the enjoyment of music with hearing aids.

2.2 Introduction

Currently, hearing aids (HAs) are mainly adjusted to maximise the clarity and comfort of speech. Nonetheless, other acoustic signals such as music may be very meaningful for some people. Since the acoustic characteristics of music are different to speech, the electroacoustic characteristics and settings of hearing instruments that optimize speech perception may not be as suitable for music enjoyment. These differences between speech and music are often overlooked in hearing aid design and gain prescription (Chasin and Russo, 2004; Hockley et al., 2010) and, as a consequence, a significant number of HA users are disappointed by the performance of their hearing aids with music (e.g., Leek et al., 2008; Madsen and Moore, 2014b).

In a study involving phone call interviews with sixty-eight elderly HA users (Leek et al., 2008), almost 30% of the respondents reported that their hearing losses affected their enjoyment of music. By comparing their results with a mail questionnaire carried out 20 years before (Feldmann and Kumpf, 1988), Leek et al. (2008) concluded that developments in HA technology had reduced problems of music enjoyment but that some HA users still experienced problems when listening to music. Madsen and Moore (2014b) conducted a survey online in which 76% of the respondents reported that their HAs were helpful to listen to recorded music and 62% of the respondents found their HAs to be helpful to listen to live music. Also, their survey identified different problems when respondents were listening to music with their HAs such as distortion, acoustic feedback, insufficient or excessive gain, unbalanced frequency response, and reduced tone quality. They concluded that music enjoyment with HAs could be improved by reducing the overload on high input SPLs

(Sound Pressure Levels), extension of the low-frequency response, and improvement of feedback cancellation and automatic gain control systems. The population studied by Leek et al. (2008) consisted predominantly in retired male soldiers (7 females and 61 males) recruited from U.S. military health-care services, suggesting that hearing losses were most likely due to presbycusis and noise exposure. Some of the participants in Madsen and Moore (2014b) were recruited from the "Auditory" and "Earmail" mailing lists, two hearing science mailing lists, suggesting that some of the participants in the study were drawn from an expert population.

The very specific populations studied by Leek et al. (2008) and Madsen and Moore (2014b) question the applicability of their findings in the context of the overall population of HA users. Moreover, these studies identified existing problems when listening to music with HA but without investigating their impact on HA satisfaction. The impact of perceived problems on satisfaction could give indications on which are the main problems that need to be addressed to improve HA performance with music. Additionally, Madsen and Moore (2014b) assessed the type and degree of hearing loss (HL) of their participants with self-reported HL. One may argue that how strongly a HL is perceived may not correlate with its actual severity. Moreover, the accuracy of self-reported HL or information on the type of HA and fitting is often regarded as controversial in the literature.

There are a few quantitative studies investigating the effects of hearing aids on music perception, and most of these have focused on the signal processing involved in hearing aid design. To compensate for reduced dynamic range and loudness recruitment induced by hearing loss HAs have to amplify weak sounds more than they amplify intense ones. Such a reduction of a large dynamic range of levels in the environment to a smaller range of levels at the output of the hearing aids is achieved by wide dynamic-range compression algorithms (WDRC). Due to the differences in dynamic range of music and speech, research investigating the effect of hearing aid signal processing on music perception has focused on the impact of WDRC on music perception. The dynamic range for music is often wider than for speech and in such cases WDRC can help hearing-impaired people to hear both soft and loud passages at a comfortable level, although, there is a trade-off between distortion and audibility regarding the amount of compression that can be applied (Kates, 2008).

WDRC can increase audibility for weak sounds but can also distort the temporal envelope of sound signals (Croghan et al., 2012). Madsen et al. (2015) investigated the effects of WDRC on the perceived clarity of individual musical instrument with ten hearing-impaired subjects between 70 and 83 years of age. The five musical excerpts used in this study contained three to five musical instruments, including three classical music excerpts, one jazz excerpt and one pop music excerpt. They showed that the clarity with which an individual instrument/voice could be heard out from a mixture was reduced with the use of compression. Croghan et al. (2014) performed acoustic analyses of music excerpts and showed that increasing the amount of WDRC reduces the range of the amplitude distribution to a larger extent for fast than for slow WDRC. They also used a paired-comparison experiment to investigate effects of compression on music preferences. For both the classical and the jazz excerpts used as stimuli, compression with a fast release time was least preferred whereas there was a preference for linear over slow-compression only for the rock excerpt.

Additionally, hearing aids can be helpful in compensating for decreased temporal resolution ability. Fast-acting compression, where the gain is rapidly increased for weak sounds and rapidly decreased for intense sounds, will make the weaker sounds more audible in the presence of preceding stronger sounds. Unfortunately, it could also make unwanted weak background noises more audible. To compensate for this, some hearing aids incorporate noise suppression algorithms or expansion in addition to compression. The basic principle of expansion algorithms is to amplify only sound levels above a certain threshold and attenuate the levels below it. However, the use of expansion may be detrimental for the perceived sound quality (VanBuuren et al., 1999).

A compression system is generally fitted for a given hearing loss by standard gain prescription procedures, such as by CAM2 (Moore et al., 2010) or NAL-NL2 (Keidser et al., 2011). These procedures recommend the insertion gain and compression behaviour that should be applied by a HA depending on the individual audiogram. Moore and Sek (2013) showed an overall preference for the CAM2 fitting method over NAL-NL2 for music signals. Because the main difference between

CAM2 and NAL-NL2 lies in the greater upper cut-off frequency of the gain recommended by CAM2, they argue that extending the upper cut-off frequency would be beneficial for the sound quality of music. Moore et al. (2011) and Ricketts et al. (2008) also found that hearing-impaired listeners preferred music with additional gain in the extended high-frequency range, although listeners with steeply sloping audiograms tended to prefer a narrower bandwidth. However, Gatehouse (1992) supports the existence of a perceptual acclimatization of the hearing impaired with amplification of high-frequencies. This suggests that preferences over a given frequency-specific gain scheme for music listening may vary over time.

As an alternative to the standard fitting methods that are designed mainly to optimize the clarity and comfort of speech, most of the manufacturers offer different processing programs for music. However, recent studies (Madsen and Moore, 2014b; Greasley et al., 2015) have suggested that these so-called music programs may not significantly improve the experience of music listening. Vaisberg et al. (2017a), for instance, compared preferences for music listening between recordings of the output of different manufacturers' universal programs and music programs with a MUSHRA task (MUltiple Stimuli with Hidden Reference and Anchor, ITU-R, 2015). While for some manufacturers the music program did not significantly improve the perceived sound quality, some others were preferred. Thereby, they stated that the preference may have been related to the amount of low frequency gain that was provided, because the preferred HAs with a music program all provided more amplification at low frequency than the other HAs used in the study. However, there is still a limited understanding of what factors in HA signal processing strategies drive preferences.

Finally, some aspects specific to each listener such as their musical practice or their individual preferences are expected to influence music appreciation. Greasley and Lamont (2016) discusses the current understanding of the psychology of musical preferences. In addition, the large variation of acoustic properties between, and within, music genres implies that there may not be a 'one-fits-all' solution for the optimal HA processing and fitting for music listening, especially given the fact that music listening habits of HA users are yet to be described more in-depth. Finally, uncertainty remains on how the hearing loss of an individual listener affects their perception and appreciation of music, even though some general insights on the effect of hearing loss on music perception are provided in a review of literature by Moore (2016).

From the above review, it is clear that a significant number of HA users encounter problems, and are therefore dissatisfied with their HAs, when listening to music, but the main underlying factors are still not fully understood, and the music programs provided by current HAs do not provide adequate solutions. To address these limitations, the purpose of the present survey was to better understand the musical listening habits of HA users, to identify the main problems they experience while listening to music with HAs, and to identify the factors that influence their satisfaction while listening to music with HAs.

2.3 Method

The study was approved by the Australian Hearing and Macquarie University Human Research Ethics Committees.

2.3.1 Questionnaire

The hearing aids and music questionnaire was built upon the studies of Leek et al. (2008) and Madsen and Moore (2014b) and was designed with the aim to better understand the musical listening habits of HA users, to identify the main problems they experience while listening to music with their HAs, and to identify the factors influencing their satisfaction while listening to music with HAs.

The HA and music questionnaire consisted of 42 multiple-choice and open-ended questions, divided into seven sections, and was realized using an interactive, multi-layered design, in which questions were populated according to previous answers. The questionnaire was administered online on Survey Gizmo and took about 30 minutes to complete. When the participants were taking the survey, the name of the section to which the question belonged was displayed on top of the web browser page.

Section (1) "General information": participants were asked to provide their full name, date of birth, their hearing aid usage, and the date at which they received their most recent HA. The name, together with the date of birth, allowed to link their answers to the questionnaire with their demographic data from the databases described below. Answers were made anonymous prior to data analysis.

Section (2) "Interest in music, listening conditions and preferences": participants were asked about their musical engagement, such as the importance of music to them, the number of hours they spent listening to music per week in a number of different listening scenarios, and the genres of music they listen to. These questions were repeated for the period current to the questionnaire and prior to having a hearing loss (if applicable). The named listening scenarios included scenarios in which a listener can control the playback level of the music (e.g., "watching TV" or "in the car") as well as scenarios in which this is not the case (e.g., "live music" or "background music").

Section (3) "Musical experience and practice": participants were first asked if they were musicians or have ever practised a musical instrument, including singing. Participants who answered positively to this question were then asked eight additional questions about their musical practice. For brevity, the additional questions were not further considered in this study.

Section (4) "Use of hearing aids while listening to music": for all listening conditions that the participants indicated in section (2) that they would spent some time listening to music, they were asked if they listen in their normal HA setting, change the setting, adjust volume, use another assistive device, use different hearing aids, or remove the hearing aids. Additionally, the participants were asked if their HAs had a music program. If answered positively, they were then asked how often they use the music program, in what way it changed the sound quality of music (if applicable), how it could be improved, and how important a successful music program would be to them.

Section (5) "Problems experienced while listening to music with hearing aids": participants were first asked about the problems they experience while listening to music with their HAs and how often they experience these problems. For each indicated problem they were then asked how much this problem interferes with their enjoyment of music, in which listening scenarios it occurs (see section 2), and whether their hearing aids make the problem better or worse.

Section (6) "Problems experienced while playing music with hearing aids": participants who reported in section (3) that they were musicians were asked about the problems they experience with their HAs while playing music. For brevity, this section was not further considered in this study.

Section (7) "Music listening with hearing aids satisfaction": participants were asked about their overall satisfaction with the performance of their HAs when listening to music and their satisfaction when listening in their relevant listening scenarios indicated in section (2). The final question was related to the satisfaction with their HA when playing music, which, for brevity, was not further considered in this study.

A list of all the questions asked in the survey, together with the provided format of open and multiple-choice answers, is given in the Appendix A.

Two successive focus group sessions were held during the questionnaire development to ensure that all potential problems that hearing-impaired listeners experience while listening to music with HAs were included and that all questions were easily understandable. Each session was run by the first author, supported by an experienced audiologist, and included eight hearing-impaired participants with a broad range of hearing loss and age. The participants were recruited from the research participant database of the National Acoustic Laboratories and differed between sessions. After the first focus group session the questionnaire was revised and re-evaluated in the second session, from which the final questionnaire was then derived.

2.3.2 Participants

To participate in this study, participants had to be bilateral hearing aid users and more than 18 years of age. To recruit participants, a link to the online questionnaire was sent via email to 113 adult HA users selected from the research volunteer database of the National Acoustic Laboratories, and advertised through an electronic newsletter sent to 32,421 HA users from the client database of Australian Hearing, one of the main HA providers in Australia. The two databases provided detailed information on the participants, such as their age, gender, type and degree of hearing loss, and information about their hearing aids. Responses to the questionnaire were collected from the 24th of January to the 21st of May, 2017.

2.3.3 Statistical analysis

To investigate potential relationships between participant variables (e.g., age or degree of hearing loss) and the ratings in the questionnaire or answers from different questions, Kendall's τ correlation coefficients were calculated. All tests were two-sided, using the value of $p \leq .05$ for statistical significance. Cohen's standard was used to evaluate the strength of the relationships, where coefficients between .10 and .29 represent a weak association, coefficients between .30 and .49 represent a medium association and coefficients above .50 represent a strong association (Cohen, 1992). To assess group differences in satisfaction ratings from section (7), Kruskal-Wallis one-way non-parametric analysis of variance tests were conducted.

An exploratory principal component analysis was conducted to reduce redundancy among the frequency of occurrence of the different problems addressed in section (5). The frequency of problems was subjected to a principal component analysis and only the answers from respondents who provided an answer for all these problems were included. An initial analysis was run to obtain eigenvalues for each component in the data and to determine the number of components to retain in subsequent analyses. Applying Jolliffe's criterion, only components with eigenvalues larger than 0.7 were retained (Jolliffe, 1972). The subsequent analyses retained four components and made use of oblique rotation (oblimin, Field et al., 2012) to improve interpretability. Two problems ("the music seems lacking in bass" and "the music has too much bass") were excluded from the final principal component analysis because they were identified as impacting the reliability of the components extracted (Cronbach's α, Field et al., 2012). In order to investigate the contribution of the experienced problems to satisfaction ratings from section (7) a multiple regression analysis was conducted on factor scores (obtained from the principal component analysis described above) used as predictors of satisfaction ratings.

All the analyses were performed using R for Windows 3.4.1 with the integrated development environment RStudio version 1.0.153 (RStudio, Inc) and *pgrimess*, *clinfun*, *pastecs*, *car*, *corrplot*, *polycor*, *GPArotation*, *corpcor* and *psych* packages.

2.4 Results

2.4.1 Participants

A total of 187 responses to the HA and music questionnaire were collected. Out of these, 36 did not fulfil the selection criteria and were excluded: 3 respondents were cochlear implant users, 5 were not bilateral HA users, and 28 did not complete more than half of the questions. Thus, responses from 151 respondents were included for data analysis. As none of the questions were mandatory to proceed to a following question, the number of answers collected for some of the questions is less than 151, which is reported accordingly for each question.

Figure 2.1 shows the distribution of the degree of HL in the better ear of the participants, expressed as the 4 Frequency Average HL (4FAHL: average of audiometric pure tone thresholds at 0.5, 1, 2 and 4 kHz), as a function of age. With the seven youngest participants (who had severe losses) excluded, there was a weak correlation between age and hearing loss (Spearman's correlation coefficient $r_s = 0.20$; $p = .02$). With the younger participants included, the correlation decreased to $r_s = 0.06$ ($p = .49$). Eighty-four (84) participants had a symmetrical HL and 47 had asymmetrical HL. HL were categorised as asymmetrical if the difference in audiometric pure tone thresholds between the two ears was greater than 15 dB at either 0.5, 1, 2, 3 or 4 Hz. Most participants had a pure sensorineural HL (121 in the left ear and 127 in the right ear) while some had a mixed or conductive HL (16 in the left ear and 10 in the right ear). Categorisation of the type of HL was based on the information entered in the database by an audiologist (Katz et al., 2014). The group of participants comprised 61 females, 87 males, one intersex and two unstated.

2.4.2 Hearing aids

At least 126 respondents had the same model of HAs in both ears. More than half (55%) of the participants wore Behind-The-Ear (BTE) HA users, 27% of the participants were using Receiver-In-the-Canal (RIC) HAs and only a few participants were using Completely-In-the-Canal, In-The-Canal or In-The-Ear HAs (CIC/ITC/ITE, 5% in their best ear and 4.6% in their worse ear). Categorisation of the type of HAs was

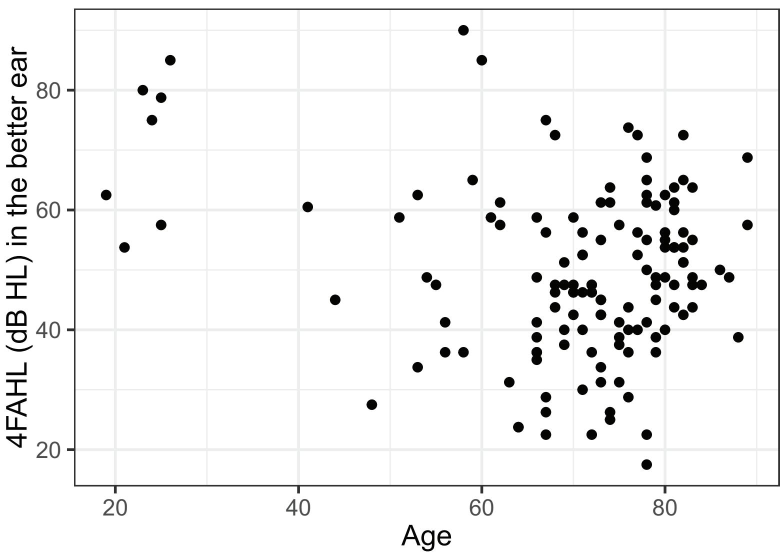

FIGURE 2.1: 4FAHL (4 Frequency Average Hearing Loss) in the better ear as a function of years of age.

based on the information entered in the database by an audiologist. For 12% no information was available on their HAs.

2.4.3 Listening conditions

In Question 5 (section 2), the participants were asked how many hours per week they were listening to music in seven different acoustic scenarios. A total of 145 answers were collected for this question. In the upper panel of Figure 2.2, each row shows the proportion of answers for each scenario and the different levels of frequency of listening are represented by the grey scale shown in the legend. The total number of answers collected for each scenario is shown on the left of the corresponding row. A weighted scoring system was applied to rank each scenario by frequency of listening. This was done by calculating the sum of weighted answers divided by the number of answers for a given scenario. Weighted answers were calculated with the following weights: 0='Never', 1='Less than 1 hour per week', 2='1 to 2 hours per week', 3='2 to 4 hours per week', 4='4 to 6 hours per week' and 5='More than 6 hours per week'. The weighted scores were then averaged across the number of responses for each scenario. The weighted average scores were then used to order the seven scenarios from most frequent (on the top row) to least frequent (on the bottom row). In the lower panel of Figure 2.2, a similar scoring system was applied to order the

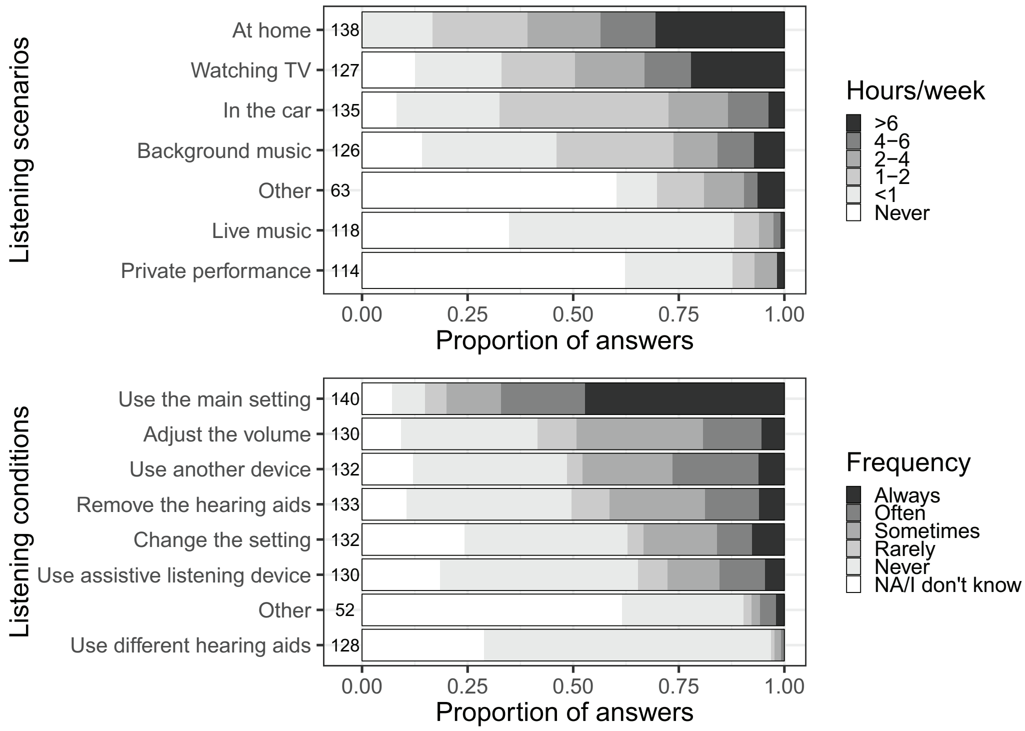

FIGURE 2.2: Proportion of answers for Questions 5 and 21 (section 2). ***Question 5:*** (upper panel) "How many hours a week do you listen to music in each scenario?" The scenarios are ordered from most frequent (top row) to least frequent (bottom row) with the total number of answers collected for each scenario shown on the left. ***Question 21:*** (lower panel) "How often do you listen to music in the following conditions?" The listening conditions are ordered from most frequent (top row) to least frequent (bottom row) with the total number of answers collected for each condition shown on the left.

listening conditions (Question 21, section 4) by frequency of listening.

In Question 21 (section 4), the participants were asked how often they were listening to music in eight different HA settings. A total of 151 answers were collected for this question. In the lower panel of Figure 2.2, each row shows the proportion of answers of each condition and the different levels of frequency of listening are represented by the grey scale shown in the legend. The conditions are shown on each row ordered by frequency of listening from most frequent (on the top row) to least frequent (on the bottom row). The first row shows that nearly half of the respondents (47%) always listen to music using their HAs with the universal program (i.e., with "the normal setting of my HAs"). The two following rows show that nearly half of the respondents reported ('Sometimes', 'Often' or 'Always') having to adjust the volume (49%), use another device than their HAs (48%) and 41% remove their HAs to listen to music . The 5th row shows that a third of the respondents (33%) reported that they 'Sometimes', 'Often' or 'Always' change the setting of their HAs and 42% 'Never' or 'Rarely' change it.

In Question 23 (section 4), the participants were asked if their HAs had a music program. It was described as "a setting specifically designed for music listening". Out of the 151 answers collected for this question, only 29 of the respondents (19%) reported that their HAs had a music program, 92 of the respondents (61%) reported than their HAs didn't have one and 30 respondents (20%) reported that they didn't know.

2.4.4 Experienced problems while listening to music with HAs

Questions 30 to 33 (section 5) were concerned with the problems experienced by the respondents when listening to music with their HAs. There were 16 problems investigated in these questions including an 'Other' option. In Question 30, the participants were asked how frequently these problems were experienced while listening to music with their HAs. A total of 143 answers were collected for this question. In the upper panel of Figure 2.3, each row shows the proportion of answers for each problem and the different levels of frequency are represented by the grey scale shown in the legend. A similar scoring system as for Figure 2.1 (upper panel) was applied to order the problems by their frequency of occurrence from most frequent (on the top row) to least frequent (on the bottom row). Only the problems that were reported as being experienced 'Rarely', 'Sometimes', 'Often' and 'Always' in Question 30 were further investigated in Questions 31 to 33.

In Question 31, the participants were asked to rate the impact of the problems reported in Question 30 on their music enjoyment. A total of 128 answers were collected for this question. In the middle panel of Figure 2.3, each row shows the proportion of answers of each problem and the different levels of interference of a problem on music enjoyment are represented by the grey scale shown in the legend. The problems are ordered by the frequency of occurrence as reported in Question 30. Kendall's τ correlation coefficients were calculated between the frequency of each problem (Question 30) and its interference on music enjoyment (Question 31). The corresponding Kendall's τ correlation coefficients are shown in the second column of Table 2.1, with the problems ordered from most frequent (on the top row) to least

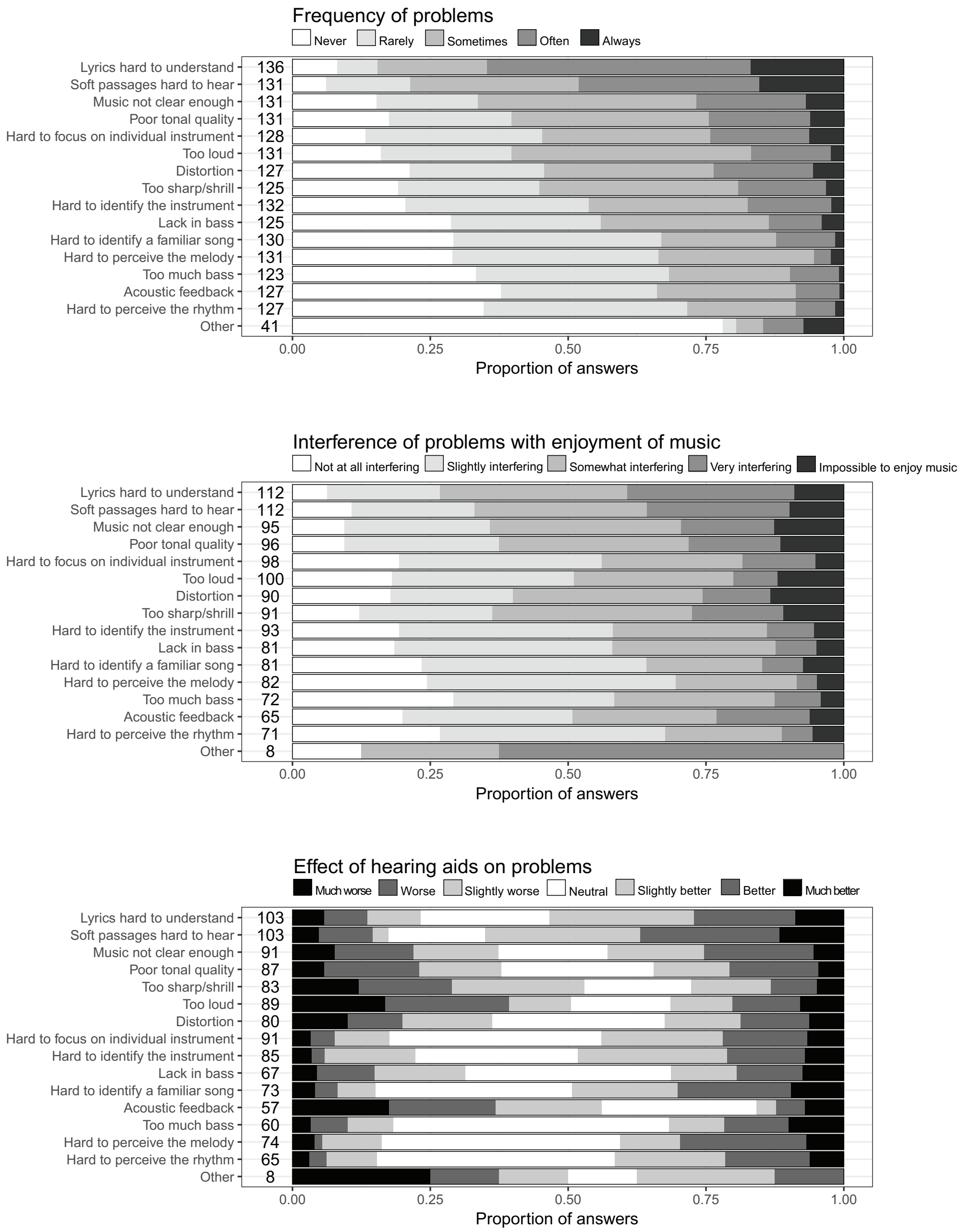

FIGURE 2.3: Proportion of answers for Questions 30, 31 and 33 (section 5). In each panel, the experienced problems are ordered from most frequent (top row) to least frequent (bottom row) with the total number of answers collected for each problem shown on the left. ***Question 30:*** (upper panel) "How often do you encounter each problem while listening music with your hearing aid(s)?" ***Question 31:*** (middle panel) "How far does each problem interfere with your enjoyment of music while listening with your hearing aid(s)?" ***Question 33:*** (lower panel) "For each problem, is it worse or better with your hearing aid(s)?"

frequent (on the bottom row). For most problems a strong positive correlation ($\tau > 0.5$, $p < 0.01$) was found, except for the problems "It is hard to identify the musical instrument(s) being played" ($\tau = 0.35$, $p < .01$) and "It is hard to focus on a specific instrument" ($\tau = 0.48$, $p < .01$), which were only moderately correlated, and "Other" ($\tau = 0.20$, $p = .55$), which showed no significant correlation. This suggests that for most of the problems, how negative a problem is perceived may be driven directly by how often it occurs.

In Question 33, the respondents were asked if their HAs were making the problems reported in Question 30 better or worse. A total of 122 answers were collected for this question. In the lower panel of Figure 2.3, each row shows the proportion of answers of each problem with the effect of the HAs on the problems presented by the grey scale shown in the legend. The white portion around the centre of each row shows the proportion of 'Neutral' answers for the corresponding problem. The grey scale on the right of the white portion in each row shows the different levels of improvement by the HAs and the grey scale on the left shows the different levels of the problem being worsened by the HAs. The problems are ordered according to their frequency of occurrence reported in Question 30.

TABLE 2.1: Correlation coefficients of the Kendall's τ correlations between the frequency of the different problems obtained in Question 30, section (5) and the interference of the problem on music enjoyment (Question 31, section (5); second column, the Satisfaction ratings (Question 40, section (7); third column) and the demographic data (columns 4 to 8). Significant relationships are indicated with ** for a significance level of $p < .01$ and * for a significance level of $p < .05$. The problems are ordered from most frequent (on the top row) to least frequent (on the bottom row) according to the weighted average of their frequency of occurrence.

Problem frequency	Problem interference	Satisfaction	4FAHL	Low-freq HL	High-freq HL	Slope of HL	Age
Lyrics hard to understand	**0.58****	**-0.24****	**0.32****	**0.27****	**0.25****	-0.01	-0.02
Soft passages hard to hear	**0.55****	**-0.36****	**0.23****	**0.22****	**0.16***	-0.01	-0.12
Music not clear enough	**0.64****	**-0.47****	**0.18***	**0.24****	0.08	-0.09	-0.08
Poor tonal quality	**0.64****	**-0.43****	0.05	0.12	-0.04	-0.12	-0.02
Hard to focus on individual instrument	**0.48****	**-0.26****	**0.16***	0.14	0.12	0.00	-0.03
Too loud	**0.51****	-0.04	-0.02	-0.01	-0.06	-0.08	-0.03
Distortion	**0.56****	**-0.43****	0.13	**0.24****	0.01	**-0.19****	-0.11
Too sharp/shrill	**0.53****	**-0.27****	0.09	**0.20****	-0.05	**-0.24****	0.00
Hard to identify the instrument(s)	**0.35****	**-0.25****	**0.16***	**0.18***	0.08	-0.03	-0.08
Lack in bass	**0.64****	**-0.23****	-0.10	0.01	**-0.19****	-0.13	-0.08
Hard to identify a familiar song	**0.54****	**-0.21****	**0.23****	**0.26****	**0.16***	-0.03	-0.11
Hard to perceive the melody	**0.57****	**-0.22****	**0.24****	**0.29****	**0.16***	-0.08	0.05
Too much bass	**0.58****	**-0.14****	0.14	**0.18***	0.08	-0.08	**-0.14***
Acoustic feedback	**0.52****	**-0.17****	0.05	0.11	0.00	-0.12	**-0.19****
Hard to perceive the rhythm	**0.56****	**-0.34****	**0.18***	**0.20***	0.09	-0.07	-0.01
Other	0.20	-0.14	0.12	0.23	-0.05	-0.26	0.16

Abbreviations: 4FAHL (4 Frequency Average Hearing Loss), HL (Hearing Loss).

The reported effect of HAs on music listening varies strongly across problems and listeners, with many problems being improved in some listeners but made worse in other listeners. For instance, the majority of the participants answered that their HAs improve understanding the lyrics (53%) and hearing the soft passages (65%), while 23% and 17%, respectively, responded that their HAs made these problems worse. Some problems such as lack of clarity, poor tonal quality and distortion were rated as often as improved (43%, 34% and 33%) as worsened (37%, 38% and 36%) by their HAs. Some of the most frequent problems, such as the music being too loud and the music being too sharp/shrill, were found to be more negatively affected by the HAs (51% and 53%) than positively (31% and 27%). For two frequent problems, "The soft passages are hard to hear" and "The music is too loud", the HAs were rated as having effects in opposite directions: the HAs were generally rated as being helpful to hear the soft passages while making the music too loud.

To understand the main individual variables that may have affected the frequency of the problem that the respondents encountered when listing to music with their HAs, a correlation analysis was performed between the frequency of the problems (Question 30) and the demographic variables of the respondents. The considered variables were 4FAHL, the low and high frequency HL (i.e., the average over the audiometric thresholds at 250 and 500 and 1, 2 and 4 kHz, respectively), the slope of the HL, and the age of the respondents. The slope of the HL was calculated as the difference between the low-frequency HL and the average of the audiometric thresholds at 2 kHz, 3 kHz and 4 kHz. The resulting Kendall's τ correlation coefficients for the different problems are summarized in Table 2.1. Unsurprisingly, problems such as understanding the lyrics, hearing the soft passages or lack of clarity were positively correlated with 4FAHL ($\tau = 0.32$, 0.23 and 0.18; $p < .01$ respectively). Is is worth noting the negative correlation found between the slope of the HL and the music being too sharp/shrill ($\tau = -0.24$, $p < .01$) which suggest that those with the most sloping audiograms were most likely to complain about the music being too sharp/shrill. However, most of the correlation coefficients shown in Table 2.1 indicate a small relationship between the frequency of a given problem and the demographic variables.

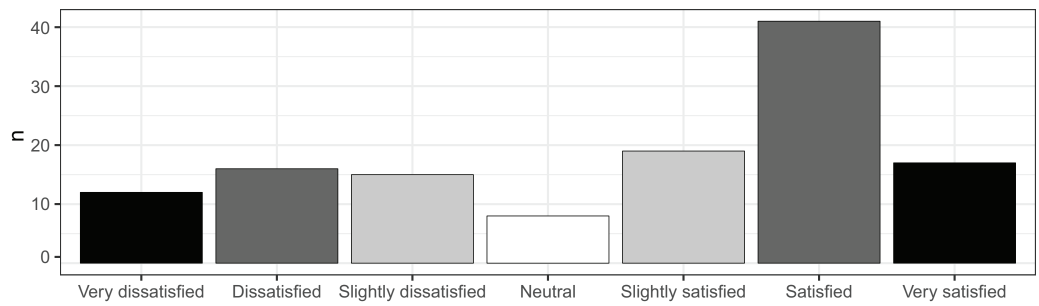

FIGURE 2.4: Proportion of answers for Question 40 (section 7). "How satisfied are you with the performance of your hearing aid(s) while listening to music?"

2.4.5 Satisfaction with HAs while listening to music

Overall satisfaction

In Question 40 (section 7), participants were asked to rate their overall satisfaction with the performance of their HAs while listening to music on a 7-point Likert scale from 'Very dissatisfied' to 'Very satisfied'. One hundred and twenty eight (128) responses were obtained for this question and are shown in Figure 2.4. More than half (60%) of the respondents reported being 'Slightly satisfied', 'Satisfied' or 'Very satisfied' and a third (34%) of the respondents reported being 'Slightly dissatisfied', 'Dissatisfied' or 'Very dissatisfied'.

Effect of individual and group variables on overall satisfaction

To understand the main individual variables that may have affected the respondents' satisfaction with their HAs when listening to music (Question 40), a correlation analysis was performed between the overall satisfaction ratings and the demographic variables described above (Table 2.1). None of the Kendall's τ correlations between the satisfaction ratings and the different demographic variables were significant ($p > 0.05$). Similarly, Kruskal-Wallis tests showed no significant group differences in satisfaction ratings with gender ($\chi^2(2) = 0.58$, $p = .75$, with $n = 49$ females, $n = 76$ males and $n = 1$ intersex), type of HL in the better ear ($\chi^2(2) = 0.80$, $p = .67$, with $n = 107$ sensorineural HL, $n = 7$ mixed HL and $n = 1$ conductive HL), type of HAs in the better ear ($\chi^2(2) = 0.44$, $p = .80$, with $n = 71$ BTE, $n = 35$ RIC and $n = 6$ ITE/CIC/ITC) or between participants having or not a music program (Question

23: $\chi^2(2) = 2.03$, $p = .36$, with $n = 25$ with a music program, $n = 81$ without and $n = 22$ who do not know whether their HAs have one). However, Kruskal-Wallis tests showed significant group differences in satisfaction ratings with symmetry of HL ($\chi^2(1) = 4.17$, $p = .04$). Focused comparisons of the mean ranks between the two groups showed that participants with asymmetrical HL (mean satisfaction score +0.21, $SD = 1.89$, $n = 38$) were less satisfied than those with symmetrical HL (mean satisfaction score +0.93, $SD = 1.85$, $n = 72$).

Effect of encountered problems on satisfaction

To understand which of the problems that the respondents encountered with their HAs when listening to music affected their satisfaction ratings the strongest, a correlation analysis was performed between the satisfaction ratings and the frequency the problems were encountered (Question 30, section 5). The resulting correlations were significant for almost every problem, except for "The music is too loud" and "Other". The corresponding Kendall's τ correlation coefficients are shown in the third column of Table 2.1. Note that a negative correlation coefficient indicates that the more often respondents experience a problem the more likely they are to be dissatisfied with the performance of their HAs while listening to music.

To further understand the main factors that drive the frequency of occurrence of the different problems, an exploratory principal component analysis was conducted. Two problems (Acoustic feedback and "Other") were excluded from this analysis due to the small number of answers collected for these problems and their weak correlations with the other problems, and the problems "The music seems lacking in bass" and "The music has too much bass" were identified as impacting the reliability of the components and were excluded from the final analysis. The frequency of the remaining problems was subjected to a principal component analysis conducted on the matrix of their Kendall's τ correlation coefficients. Only the answers from respondents who provided answers for all of these twelve problems were included, resulting in a sample size for this analysis of $n = 101$. The four components in combination explained 74% of the variance. Table 2.2 shows the factor loadings after rotation for the frequency of occurrence for each of the 12 problems retained in the final analysis. Factor loadings greater than 0.30 are shown in bold font to highlight the frequency of which problem contributes the most for each component. The frequency of the problems that cluster on the same components suggest that component 1 represents 'Sound quality', component 2 represents 'Identification', component 3 represents 'Definition/Clarity' and component 4 represents 'Loudness'. The three last rows of Table 2.2 show the eigenvalues, the proportion of explained variance, and the Cronbach's α reliability measure for each of the four components. Components 1, 2 and 3 had all high reliability with Cronbach's α =.9, .87 and .8. Cronbach's α could not be calculated for component 4 as this component is composed of only one item ("The music is too loud").

To obtain the factor scores for all the respondents included in the principal component analyses ($n = 101$), the regression method was used. The inverse of the matrix of Kendall's τ correlation coefficients between the frequency of the 12 experienced problems was multiplied with the matrix of factor loadings. The resulting factor scores were used as predictors in a multiple regression analysis with the outcome being the satisfaction ratings of the respondents when listening to music with their HAs (Question 40). Since the frequency of the "music being too loud" didn't show any significant correlation with the satisfaction ratings (Table 2.1, $p = -.04$) and was the only problem driving the 'Loudness' component 4, this component was excluded from the regression model and only components 1, 2 and 3 ('Sound quality', 'Identification', and 'Definition/Clarity') were used as predictors of the satisfaction ratings. The multiple regression was conducted with only the answers from respondents who were included in the principal component analysis and also responded to Question 40, resulting in a sample size for this analysis of $n = 88$. The results from the multiple regression analysis are shown in Table 2.3. The 2 first columns indicate the model statistics multiple R^2 and adjusted R^2. The following columns show the estimate of the b-values, their associated standard error, the standardized β estimates and the associated significance level of the contribution of a given predictor. All three components were shown to make a significant contribution to satisfaction ratings ($p < .05$). The model statistics R^2 indicates that 36% of the variability in the

TABLE 2.2: Summary of exploratory principal component analysis results for the frequency of experienced problems ($n = 101$)

Problem	*Oblimin rotated standardised factor loadings* Component 1 *Sound quality*	Component 2 *Identification*	Component 3 *Definition/Clarity*	Component 4 *Loudness*
Music too sharp/shrill	**0.85**	-0.18	0.08	-0.03
Poor tonal quality	**0.84**	0.12	-0.05	-0.01
Distortion	**0.83**	0.19	-0.14	0.03
Music not clear enough	**0.69**	-0.02	**0.32**	0.10
Hard to perceive the melody	0.01	**0.89**	0.04	-0.05
Hard to identify a familiar song	0.06	**0.88**	-0.07	0.04
Hard to perceive the rhythm	-0.01	**0.73**	0.12	0.05
Hard to identify the instrument(s)	0.03	**0.56**	**0.35**	-0.01
Hard to understand the words in lyrics	-0.14	0.10	**0.85**	0.01
The soft passages are difficult to hear	0.23	0.03	**0.69**	-0.01
Hard to focus on individual instrument	0.18	0.09	**0.61**	0.11
Too loud	-0.01	-0.01	-0.01	**1.00**
Eigenvalues	2.85	2.75	2.12	1.07
% of variance	0.24	0.23	0.18	0.09
α	0.9	0.87	0.8	-

Note: Factor loadings over .30 appear in bold.

TABLE 2.3: Summary of multiple regression with factor scores as predictors of the satisfaction ratings from Question 40 ($n = 88$)

	Multiple R^2	Adjusted R^2	b	SE b	Standardised β estimates	Significance $p-$value
Regression model	0.36	0.34				<.001
Intercept			2.47	0.37		<.001
Component 1 - Sound quality			-0.93	0.14	-0.64*	<.001
Component 2 - Identification			-0.45	0.18	-0.26*	.012
Component 3 - Definition/clarity			-0.45	0.14	-0.31*	.002

satisfaction ratings is accounted for by the three predictors 'Sound quality', 'Identification' and 'Definition/Clarity'. The adjusted $R^2 = 34\%$ is very close to the observed R^2, indicating that the cross-validity of the model is good. The column with the standardised β provides an estimate e of the importance of the predictors in the model. The standardized β value for 'Sound quality' ($\beta = -0.64$) indicates a relatively strong influence of this component on the satisfaction ratings. The standardized β values for "Identification" ($\beta = -0.26$) and "Definition/Clarity" ($\beta = -0.31$) indicate that these components are approximately equally important for the model but less important than "Sound quality".

Effect of listening condition on satisfaction

In Question 41 (section 7), the participants were asked to rate their satisfaction (with the same 7-point Likert scale as used for Question 40, section 7) with the performance of their HAs while listening to music in the seven different listening scenarios from Question 5 (section 2). A total of 128 answers were collected for this question. In Figure 2.5, each row shows the proportion of answers for each scenario and the different levels of satisfaction are represented by the grey scale shown in the legend. The listening conditions are ordered according to the frequency the respondents listen to music with their HAs (Question 5). The scenarios "At home", "Watching TV" and "In the car" obtained the highest satisfaction ratings with 64%, 61% and 55% of satisfied respondents (i.e., who answered 'Slightly satisfied', 'Satisfied' or 'Very satisfied'). The scenarios "Private performance", "Live music" and "Background music" obtained a smaller proportion of satisfied respondents with 40%, 47% and 41%. Interestingly, by comparing the upper panel of Figure 2.2 and Figure 2.5 it can be seen that the listening conditions in which the respondents listen to music most of their time are also the conditions in which they were most satisfied with their HAs.

2.5 Discussion

The present survey was aimed at understanding the musical listening habits of HA users, to identify the main problems they experience while listening to music with

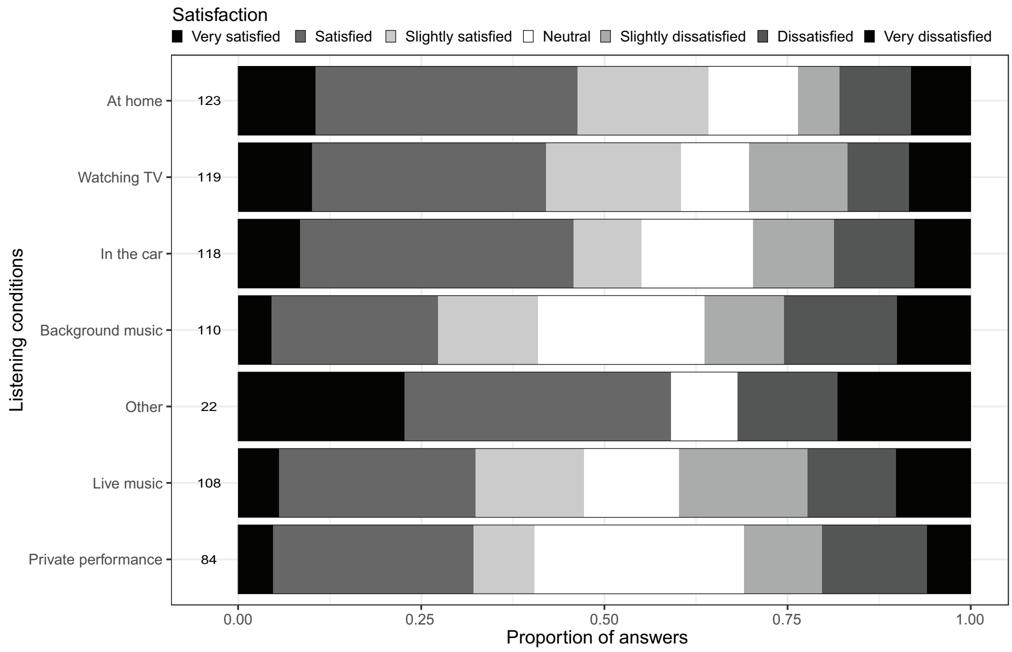

FIGURE 2.5: Proportion of answers for Question 41 (section 7). "How satisfied you are with the performance of your hearing aid(s) while listening to music in different scenarios?" The scenarios are ordered from most frequent (top row) to least frequent (bottom row) with the total number of answers collected for each scenario shown on the left

their HAs, and to identify the factors influencing their satisfaction while listening to music with HAs.

2.5.1 Listening habits

The three most frequent scenarios in which listeners were listening to music were "at home", "while watching TV", and "in the car". In these scenarios the music is recorded and reproduced with a sound system, which allows the listener to have control over the music being played, for instance by adjusting the volume. This is in contrast with the two least frequently visited scenarios of "Live music" and "Private performance", where music instruments are played live and often amplified, and are largely out of control of the listener (besides moving into a quieter corner or leaving the location). Despite the lesser frequency of "Live music" listening, 53% of the respondents answered listening to "Live music" up to an hour a week and 12% more than an hour a week. One should emphasise the frequency of listening in a given scenario may not reflect the relevance of the scenario in term of overall music appreciation. To the knowledge of the authors, there are no other studies that investigated the proportion of live music attended to in comparison with the overall time spent by HA users listening to music in any scenario. Independent of the listening scenario, the present study found that a majority of HA user mainly listened to music using the default program of their HAs (47% responded 'Always'). 33% reported that they 'Sometimes', 'Often' or 'Always' change the setting of their HAs. However, what changes in their HA setting they are operating is still unclear since only 19% of the respondents reported that their HAs had a music program while 61% of the respondents reported than their HAs didn't have one. In Madsen and Moore (2014b), 40% of the participants reported having a music program but did not report on whether or not those music programs were used.

2.5.2 Main problems experienced when listening to music with HAs

"Understanding lyrics" was reported as the most frequently experienced problem, but at the same time, 53% of the respondents indicated that their HAs were helpful for this problem. Some respondents also indicated in an open-ended question that the music itself or the way a singer would sing was a potential cause for difficulties in understanding the words in lyrics and not necessarily their HL nor their HAs. This is in general agreement with Feldmann and Kumpf (1988) and Leek et al. (2008), who also reported lyrics understanding as a common complaint of people with HL. However, this problem may be more reflective of the difficulty of understanding speech in background noise than of a problem specific to music listening. This is supported here by the correlation found between the degree of HL (4FAHL) and the frequency of this problem ($\tau = 0.32$, $p < .01$), which may simply reflect the general observation that speech intelligibility is correlated with HL, although the correlation coefficients here indicate only a small to medium relationship. Given that HAs are primarily designed to improve speech understanding, this might also explain the positive outcome of HAs with understanding lyrics.

While the "soft passages being hard to hear" and "the music being too loud" were also identified as frequent problems, HAs seemed to have opposite effects on each of these problems. Most participants reported that HAs were helpful for hearing the soft passages (65%) while making the music also too loud (51%). Madsen and Moore (2014b) found similar results in their survey in which 25% of the respondents reported that their HAs made the louder parts of music too loud and only 28% reported that they could hear soft passages without the louder parts being too loud. This suggests that most HAs provide enough amplification at low levels to allow soft sounds to be audible but may not provide adequate gain compression, thus making the higher sound levels uncomfortably loud.

Even though providing more compression at higher levels would allow to amplify low levels without over-amplifying high levels, it may create other problems. In a subjective listening experiment Madsen et al. (2015), for instance, concluded that the use of multi-channel WDRC was reducing the clarity of individual musical instruments when compared to linear amplification, although VanBuuren et al. (1999) reported that a small amount of compression could be applied without degrading the sound quality. In the present survey, 44% of the respondents reported that focusing on individual instrument was easier with HAs while 17% reported the opposite effect.

Kates (2008) generalised that applying compression provides a trade-off between audibility and distortion due to the changes in the temporal envelope of the signal. Thereby, not only the amount of compression applied, but also the number of channels and the time constants of the compression algorithms effect signal distortions. Croghan et al. (2014) performed an acoustical analysis of a number of music samples and showed that the reduction of the dynamic range was larger for fast than for slow WDRC. Moore et al. (2011) investigated the impact of different parameters of multi-channel compression on sound quality (clarity and pleasantness judgements) with speech and music signals. Clarity judgements were higher for slow than for fast-acting compression for input levels of 80 and 65 dB SPL but not for the lowest input level tested (i.e., 50 dB SPL). Compression speed had no effect on pleasantness for input levels of 50 and 65 dB SPL, but pleasantness judgements were slightly higher for slow than for fast-acting compression for the highest input level at 80 dB SPL. However, many participants didn't show consistent preferences for compression speed. In a paired-comparison listening experiment, Croghan et al. (2014) showed that preferences were influenced by the changes in dynamic range caused by different compression speeds and depending on the genre of the music sample. For both the classical and the jazz samples, compression with a fast release time was least preferred, whereas there was a preference for linear over slow-compression only for the rock music excerpt.

Signal distortions can be caused by other factors than compression such as output limiting, noise reduction algorithms, microphone directionality, the use of frequency lowering (Mussoi and Bentler, 2015; Uys et al., 2013) or could occur at the input of the HA, e.g., microphone distortion or introduced by an analog-to-digital converter (Hockley et al., 2010). Moore (2012) provides a review of some of these HA features and how they may influence music perception. However, it is hard to know exactly how much each of these features is responsible for the distortion reported in the present survey. From the results of the principal component analysis (subsection 2.4.5), problems of distortion and lack of clarity clustered on the same component along with poor tonal quality and the music being too sharp or shrill, and 53% of the respondents reported that the HAs were affecting negatively the music being too sharp or shrill. Additionally, Moore et al. (2011) and Ricketts et al. (2008) found that hearing-impaired listeners preferred music with extended high-frequency range (i.e., 7.5 kHz upper cut-off frequency preferred over 5 kHz and 10 kHz preferred over 4 kHz respectively), although listeners with steeply sloping audiograms tended to prefer a narrower bandwidth. The latter is supported in the present survey by the negative correlation found between the slope of the HL and the music being too sharp/shrill ($\tau = -0.24$, $p < .01$). That is, those with the most sloping audiograms were most likely to complain about the music being too sharp or shrill.

Half of the respondents in the present survey reported that the lack in bass in the music was 'Never' or 'Rarely' a problem for them. However, Vaisberg et al., 2017b identified that a lower cut-off frequency was preferred when comparing the sound quality of different music programs. The amount of bass amplification that a HA receiver can provide is not only dependent on the prescribed gain but is greatly influenced by the type of fitting. While semi-open fittings are the most common fitting with current HAs the amount of gain that can be provided at low frequencies is very limited and low-frequency stimulation is generally achieved acoustically from the vent path without passing through the HA amplification.

Also, open fittings present the disadvantage of limiting the amount of high-frequency amplification that can be achieved without feedback oscillation (Dillon, 2012). Unfortunately, no information could be obtained regarding the type of fitting of the respondents that participated in the present survey, but a majority of them is expected to be fitted with semi-open domes because they are the most common fitting domes currently used, at least for the less severe HLs. In the survey, a small proportion of respondents reported experiencing acoustic feedback (25% ’Sometimes’ and only 9% ’Often’ or ’Always’) and in Madsen and Moore (2014b), only a third of the respondents "had experienced feedback when listening to music". This suggests that feedback cancellation algorithms in semi-open fittings deal reasonably well with this problem. However, some noise or feedback cancellation algorithms may still incorrectly identify some of the music signals as noise or feedback (e.g., flutes, Chasin and Russo, 2004). Therefore, the general approach with music programs in current HAs is to deactivate these features and to vary the characteristics of compression algorithms away from the prescription formulae, either with more linear amplification or with slower acting compressor systems.

Madsen and Moore (2014b) and Greasley et al. (2015) observed that current music programs may not improve significantly the experience of music listening. In Madsen and Moore (2014b), 40% of the respondents reported having a music program in their HAs whereas in the present study only 19% of the respondents reported having a music program and 20% didn’t know. Nearly half of the respondents (47%) reported to always listen to music using their HAs with the universal program, i.e., without changing the program manually. Nowadays, some HAs have the capabilities to change programs automatically and a music program may be activated without the user being necessarily aware of it. Also, some people who reported having a music program may not use it on a regular basis. Even so, the potential lack of an effect of having a music program on satisfaction suggests that current music programs do not significantly improve the experience of listening to music and need to be further optimized.

2.5.3 Factors influencing satisfaction while listening to music with HAs

The answers collected in this survey showed that more than half of the respondents (60%) were satisfied with the performance of their HAs while listening to music. Most of them listen mainly to recorded music, at home and without changing the setting of their HAs. A large proportion of respondents stated that their HAs were useful in addressing some of the music listening problems they had such as making soft passages easier to hear (65%), and being able to better understand the lyrics (53%). However 34% of respondents were dissatisfied with their HAs for music listening. This survey identified several problems that need to be addressed. A principal component analysis followed by a regression model showed that 36% of the variability in the satisfaction ratings was accounted for by three predictors each clustering problems related to either 'Sound quality', 'Identification' or 'Definition/Clarity'. Out of these three contributors of satisfaction, 'Sound quality' had the strongest influence on satisfaction ratings and was clustering together the frequency of the music being too sharp/shrill, the poor tonal quality, occurrence of distortions and the lack of clarity. This is supported by Madsen and Moore (2014b) who reported that 53% of their participants reported that their HA made music sound distorted, 29% found that their HA worsened the sound quality of music and 21% found their HA to make music too bright or shrill. Addressing these four problems should be a priority in HA signal processing design for music listening. However, it is not clear whether this can be done without impacting the problems associated with 'Identification' or 'Definition/Clarity'. As discussed above (subsection 2.5.2) problems associated with distortion and clarity could be addressed with improved compression systems while the music being too sharp/shrill and tonal quality are musical attributes associated with frequency specific gains. As shown in Table 2.1, the music being too sharp/shrill, the music lacking clarity and distortions are associated with the degree and slope of HL. Thus, special attention should be given to the effects of HL on the sound quality of the music processed by HAs.

2.5.4 Limitations and outlook

The questions addressed in this article are only a portion of the questionnaire and some other questions were not addressed here due to space limitations. These include aspects such as open ended questions, musical experience and practice or problems experienced while listening to music with HAs. Although, the results presented here provide some new insights into the use of HAs for music listening, the large number of questions was a limitation in the study as it increased considerably the duration to take the survey. To avoid participants giving up the survey, none of the questions was made mandatory to proceed to a following question. As a result, the number of answers collected vary across questions and 25 respondents gave up the survey before completing half of it. However, 131 respondents completed the survey and 20 answered at least half of the questions. Additionally, the questionnaire did not assess the potential effect of the type of HA fitting. While semi-open fittings are the most common fitting with current HAs the amount of gain that can be provided at low frequencies is very limited which could have a detrimental effect on the sound quality of music. Thus, the impact of HA fitting type on satisfaction remain unclear. However, the survey allowed us to identify problems that were strongly contributing to HA satisfaction and which are therefore a priority for addressing. Conducting a follow-up study with a control group of normal-hearing listeners answering some of the questionnaire items (e.g., listening conditions) would provide a useful comparison on the extent of the difficulties associated with hearing loss. Finally, to better address the limitations of the present study more controlled studies inside the laboratory would be required.

2.6 Conclusions

The present survey showed that HA users are mainly listening to recorded music at home and generally use the HA in their universal program. They are most satisfied with the performance of their HAs in scenarios with recorded music, where they have control over the music presentation, such as its loudness. However, about 30% of the respondents are dissatisfied with the performance of their HAs with music and experience problems that need to be addressed, confirming findings from other studies. The most frequent problems identified were difficulties in understanding lyrics, the soft passages of music being too soft, lack of clarity of the music, poor tonal quality, the music being too sharp/shrill and the music being too loud. A principal component analysis revealed that satisfaction was mainly related to the overall 'Sound quality' provided by the HAs and, to a lesser extent, to the 'Identification' of music instruments and lyrics as well as to the 'Clarity' (or 'Definition') of the music. Thus, future research should derive amplification prescription schemes specifically targeted at optimizing the enjoyment of music with HAs. Particular attention should be given to compression system algorithms and frequency-specific gain.

2.7 Acknowledgements

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2.8 Supporting information

Appendix A. Overview of the full questionnaire with the provided format of open and multiple-choice answers.
Appendix B. Answers collected for supplementary questions omitted in Chapter 2.

Chapter 3

Preferred frequency-specific gains and compression ratios in hearing aids for music listening

Rémi Marchand,[1,2] Jörg Buchholz[1,2] and Harvey Dillon[1,2,3]

[1] *The HEARing Cooperative Research Centre*

[2] *Macquarie University*

[3] *National Acoustic Laboratories*

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3.1 Abstract

Aims: To further understand the hearing aid compression characteristics preferred by participants when listening to music in relation to the problems highlighted by the survey previously conducted (see Chapter 2). **Method:** This study involved a controlled listening experiment with instrumental music stimuli processed by a 15-band dual compression system implemented in a hearing aid simulation. Manipulations of the signal processing comprised changes in compression ratio and frequency-specific amplification relative to those prescribed by the NAL-NL2 standard fitting formula. **Results:** Differences in chosen listening levels were observed with gender, females choosing lower input levels than males. The preferred frequency-specific amplification provided by the hearing aid (HA) indicates that more gain at low-frequency and less gain at high-frequency than prescribed by NAL-NL2 standard fitting formulae is needed. The preferred amount of compression applied by the HA depended on the music stimulus level at the input of the HA with overall preference for more compression than prescribed by NAL-NL2. **Conclusions:** The current study showed that it is possible to manipulate the signal processing of HAs in order to obtain good satisfaction outcomes for instrumental music in relation to the problems highlighted by the previously conducted survey. Future research intending to derive amplification prescription schemes specifically targeted at optimizing the enjoyment of music with HAs should give particular attention to gender differences, the effect of HA fitting and the sound levels of the music signals at the input of the HAs.

3.2 Introduction

Standard prescription formulae (e.g., NAL-NL2, Keidser et al., 2011; CAM2, Moore et al., 2010; and DSL, Scollie et al., 2005) give insertion gain recommendations for specific frequency bands and are primarily aiming at improving speech intelligibility while adjusting loudness to that perceived by a normal-hearing listener. One could argue that different frequency-specific amplification may be desirable for signals other than speech, such as music, which may include amplification of frequency regions that are less important for speech. For instance, Keidser et al. (2007) varied the (linear) real ear insertion gain (REIG) at 250 Hz to evaluate the preferred low-frequency gain and thus, the effective HA bandwidth, for speech that was amplified according to the NAL-NL1 formula (Byrne et al., 2001). They found that providing no extra gain (0 dB) at 250 Hz was the most frequently preferred condition, followed by the conditions providing 6 dB and then 12 dB of extra gain. However, they did not observe any degradation of speech recognition with increase of low-frequency gain. Franks (1982) investigated the effects of HA bandwidth on preference judgements with a music signal, and showed that the highest preference ratings were given to the conditions with the lowest cut-off frequency. Similarly, Moore and Tan (2003) investigated the effects of HA bandwidth variations on the perceived naturalness of music and speech signals. Their results indicate that increasing the lower cut-off frequency resulted in a degradation of naturalness for both type of signals. However, the critical lower cut-off frequency for degradation of perceived naturalness was higher for speech than for music signals. This could be interpreted as low-frequency being more important for music sound quality than it is for speech signals.

Moore et al. (2011) investigated how clarity judgements of speech signals (a female talker and a male talker) were affected by variations of the high-frequency bandwidth of a simulated HA fitted with CAMEQ2-HF (Moore et al., 2010). They found that clarity judgements of speech were higher for the 7.5 and 10 kHz bandwidths than for the 5 kHz bandwidth. However, when investigating the effects of high-frequency bandwidth on pleasantness judgements with three musical excerpts, no significant difference in pleasantness was associated with reduction of high-frequency bandwidth. Ricketts et al. (2008) conducted paired-comparisons of preferred sound quality for a HA simulation fitted with NAL-NL1 with two different upper cut-off frequencies. For both the music and the movie soundtrack excerpts, results identified that bandwidth preferences were correlated with the slope of hearing loss at high frequencies (from 4 to 10 kHz), with steep sloping hearing loss associated with preference for narrower bandwidth and vice versa. However, the observed preference for narrower bandwidth associated with steep hearing loss may be, at least, partly explained by an insufficient acclimatization period provided by Ricketts et al. (2008). As shown by Gatehouse (1992), hearing-impaired subjects need to perceptually acclimatize to amplification of high-frequencies.

In addition to linear insertion gains, digital HAs incorporate compression algorithms to compensate for the reduced dynamic-range induced by hearing loss. Compressor systems are characterised by their non-linear amplification depending on the input levels. For input levels below the compression threshold (CT) the system provides a constant linear gain. For input levels above the CT, it provides dynamic-range compression: for each dB increase in input level the output level increases by 1/CR dB, where CR designates the compression ratio. Compressor systems are also characterised by their time constants, i.e., the speed at which the compressor gains are adjusted. The attack time constant is the duration it takes for the compression system to decrease the gain of the system when input levels increase above the CT. The release time constant is the duration it takes for the compression system to increase the gain of the system when input levels decrease below the CT. The compression systems used in HAs are referred to as wide dynamic-range compression (WDRC) as they are designed to adjust gains over a wide range of input levels. Nowadays, most if not all of the HA compression systems are implemented as multichannel compression, meaning that they combine a filter bank with more or less independent compression in each frequency channel. VanBuuren et al. (1999) investigated the effect of CR applied on one, four and 16 channels on pleasantness judgements for four musical excerpts. They found that increasing CR lead to a decrease in pleasantness. The observed deleterious effect of compression on pleasantness was stronger with increasing number of compression channels. However, CRs were the same for all the participants and may have not been suitable for individual hearing losses.

Moore et al. (2011) investigated how pleasantness judgements were affected by compression time constants with three musical excerpts processed by a five-channels HA simulation. They found little effect of compression speed on pleasantness at input levels of 50 and 65 dB sound pressure level (SPL), but pleasantness decreased with increasing compression speed for higher input levels (i.e., 80 dB SPL).

Croghan et al. (2014) used a paired-comparison experiment to investigate the effects of compression on preference judgements with WDRC implemented in a three- or 18-channel HA simulation. For both the classical and the jazz excerpts, compression with a fast release time was least preferred whereas there was only a preference for linear over slow-compression for the rock excerpt. Also they performed acoustic analyses of music excerpts and showed that increasing the amount of compression reduces the range of the distribution of the short-term levels to a larger extent for fast than for slow WDRC.

Madsen et al. (2015) investigated the effect of WDRC on clarity judgements of individual musical instrument with ten hearing-impaired subjects between 70 and 83 years of age. The five musical excerpts used in this study contained three to five musical instruments, including three classical music excerpts, one jazz excerpt and one pop music excerpt. They showed that the clarity with which an individual instrument/voice could be heard out from a mixture was reduced with the use of compression. In Chapter 2, an online survey was conducted to better understand the musical listening habits of HA users and to identify the main issues they experience while listening to music. While the soft passages being hard to hear and the music being too loud were identified as frequent problems, HAs seemed to have opposite effects on each of these problems. Most commonly, HAs were reported to be helpful to hear the soft passages (65% of the responses) while making the music too loud (51%).

Another survey (Madsen and Moore, 2014b) found similar results to those presented in Chapter 2: 25% of the respondents reported that their HAs made the louder parts of music too loud and only 28% reported that they could hear soft passages without the louder parts being too loud. This suggests that the HAs may be providing enough amplification at low levels to allow soft sounds to be audible but may not provide adequate gain compression, thus making the higher sound levels uncomfortably loud. In summary, improved compression systems may be needed for better appreciating music with HAs, but too aggressive compression algorithms could be detrimental for sound quality. Kates (2008) generalised that applying compression is a trade-off between audibility and distortion.

The aim of the current study was to further understand the signal processing strategies preferred by participants when listening to music in relation to the problems presented above and, in particular, the problems highlighted by the survey previously conducted (see Chapter 2). Despite understanding lyrics being the most frequent problem reported in that survey, most participants also reported HAs as being helpful for understanding lyrics. This was not the case for a number of other problems. In particular, HAs were reported to have a detrimental effect on sound quality when listening to music (i.e., the music being too sharp/shrill, poor tonal quality, distortions and music lacking clarity). This observation together with the aforementioned problem that HAs often either provide not enough amplification to soft sounds or make loud sounds uncomfortably loud, suggests that the compression systems in HAs should be improved for music listening. Whereas the loudness related problems seem to refer to the amount of compression that is applied, some of the sound quality related problems may be addressed with changes in the frequency-dependent amplification. The aim of the present study was therefore to design a listening experiment that would help to better understand the effect of the frequency-dependent behaviour of WDRC on a HA user's satisfaction with music listening at realistic playback level. Hence, there is a clear separation between the physical stimulus arriving at the HA microphones and the resulting (aided) signal provided by the HA to the listener's ears. The focus of the study was on the second aspect, but assuming realistic (subject-specific) physical levels. The use of a simulated HA ensured the implementation of the signal processing manipulations that were necessary to address those objectives. This also presented with the advantage that characteristics of the signal (e.g., sound level) could be easily calculated at any point in the signal flow. The listening experiment was designed to simulate the scenario of recorded music played through stereo loudspeakers in a living room and amplified by a HAs, as this was the most frequent listening scenario reported in the survey reported in Chapter 2. Excerpts of twelve different Classical and Jazz & Blues music pieces were selected as stimuli, as these were some of the most common genres of music reported in the same survey (see Appendix B). Only music without vocals was selected to avoid preference judgements being made on the basis of speech intelligibility. This influenced the choice of the genre of music employed for Classical and Jazz & Blues over Country & Folk or Pop & Rock (genres also commonly reported in the survey) as the two latter genres are typically characterised with the presence of a singer/lyrics.

3.3 Method

The current study was approved by the Australian Hearing and Macquarie University Human Research Ethics Committees.

3.3.1 Participants

Twenty-six bilateral HA users participated in the listening experiment (seven females and 19 males), 58 to 90 years of age (median 72 years). Sixteen participants were recruited from the participants of an online survey conducted previously (Chapter 2). Nine participants were recruited from the research volunteer database of the National Acoustic Laboratories (NAL) and one participant was recruited after

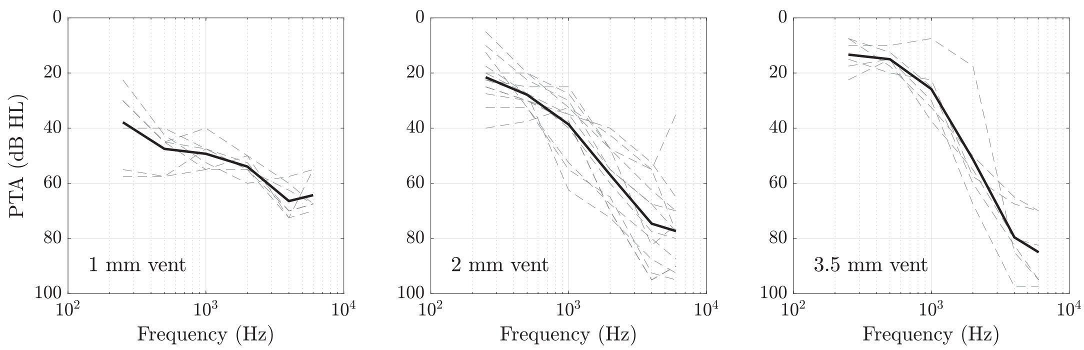

FIGURE 3.1: Audiometric thresholds of the 26 participants averaged across both ears fitted with 1 mm vents (***left panel***, $n = 7$), 2 mm vents (***middle panel***, $n = 13$) and 3.5 mm vents (***right panel***, $n = 6$). The thick solid lines show the average audiograms of each of the three groups.

a flyer advertisement of the listening experiment was displayed at Macquarie University Speech and Hearing clinic. All participants had symmetrical sensorineural hearing loss (i.e., the difference in pure tone audiometry (PTA) between the two ears was less than 15 dB HL at each frequency between 250 Hz and 6,000 Hz). Their degree of hearing loss ranged from mild to moderately severe, with 4FAHL (four frequency average hearing loss) between 30 and 62.5 dB HL (median 50.5 dB HL). Conductive hearing loss was not investigated in the current study. Participants' audiometric thresholds were measured prior to the listening experiment and the average across both ears is shown in Figure 3.1 by the dashed lines. Depending on their low-frequency PTA at 500 Hz (Keidser et al., 2007), subjects were fitted with 1 mm vents for PTA(500 Hz) $\geq$ 40 dB HL (left panel, $n = 7$), 2 mm vents for 20 dB HL < PTA(500 Hz) < 40 dB HL (middle panel, $n = 13$) and 3.5 mm vents for PTA(500 Hz) $\leq$ 20 dB HL (right panel, $n = 6$). Average thresholds for each of the three groups are shown in Figure 3.1 by the solid lines.

3.3.2 Stimuli

Music material

The music stimuli were 10.2 to 31.2 secs excerpts of different music pieces. Six stimuli were classical music pieces and the other six were jazz and blues music pieces. An additional stimulus was an excerpt of the preferred piece of music as reported by the individual participant during the recruitment phase. The list of the twelve music pieces, their genre and their dynamics category are shown in Table 3.1. The music excerpts were selected in order to have three different dynamics for both genres: two excerpts from a loud passage (e.g., forte), two excerpts from a soft passage (e.g., piano) and two excerpts containing a transition between a soft and a loud passage (e.g., crescendo, decrescendo, sforzando). The list of the individual preferred music pieces is included in Appendix C. All music pieces were purchased and downloaded from iTunes (Apple Inc.) with the best quality available as 44.1 kHz 32-bit (floating point) *.m4a* files. The bit rate of each original file is included in Appendix C. The use of commercial recordings implies that different amount of compression limiting may have been applied during the mastering process. The original music pieces were rms-equalized (root-mean-square) to -25 dB FS (relative-to-Full-Scale) prior to sampling the excerpts. This was done so that the difference in loudness between stimuli would be representative of their respective dynamics. Only the 6^{th} classical music piece (Dvořák) was rms-equalized to -31 dB FS because the passage from which the excerpt was taken was the loudest passage of a piece with a very wide dynamic range and despite being equalized to a lower rms value, the excerpt remained the loudest stimulus of the experiment. None of the stimuli included any vocals.

The dynamic range histograms of the 12 musical excerpts are shown in Figure 3.2. These histograms were derived from a short-term frequency analysis of the music excerpts using a 20-ms long rectangular window and third-octave frequency bands. The analysis was performed on each stereo channel separately and the power in each time-frequency bin was then integrated across the two channels. The music signals were level-normalized such that their RMS levels reflected the average presentation level in dB SPL across all subjects as played by the loudspeaker. The solid lines refer to the median values across third-octave bands, the light gray areas to the corresponding 5% to 95% quantiles, and the dark gray areas to the 25% to 75% quantiles. The solid lines with circles refer to the long-term RMS levels. After level normalization, the musical excerpts were processed in order to simulate the scenario of a listener using their HAs to listen to music played through a stereo system in a living room (see *subsection Input stimuli* below).

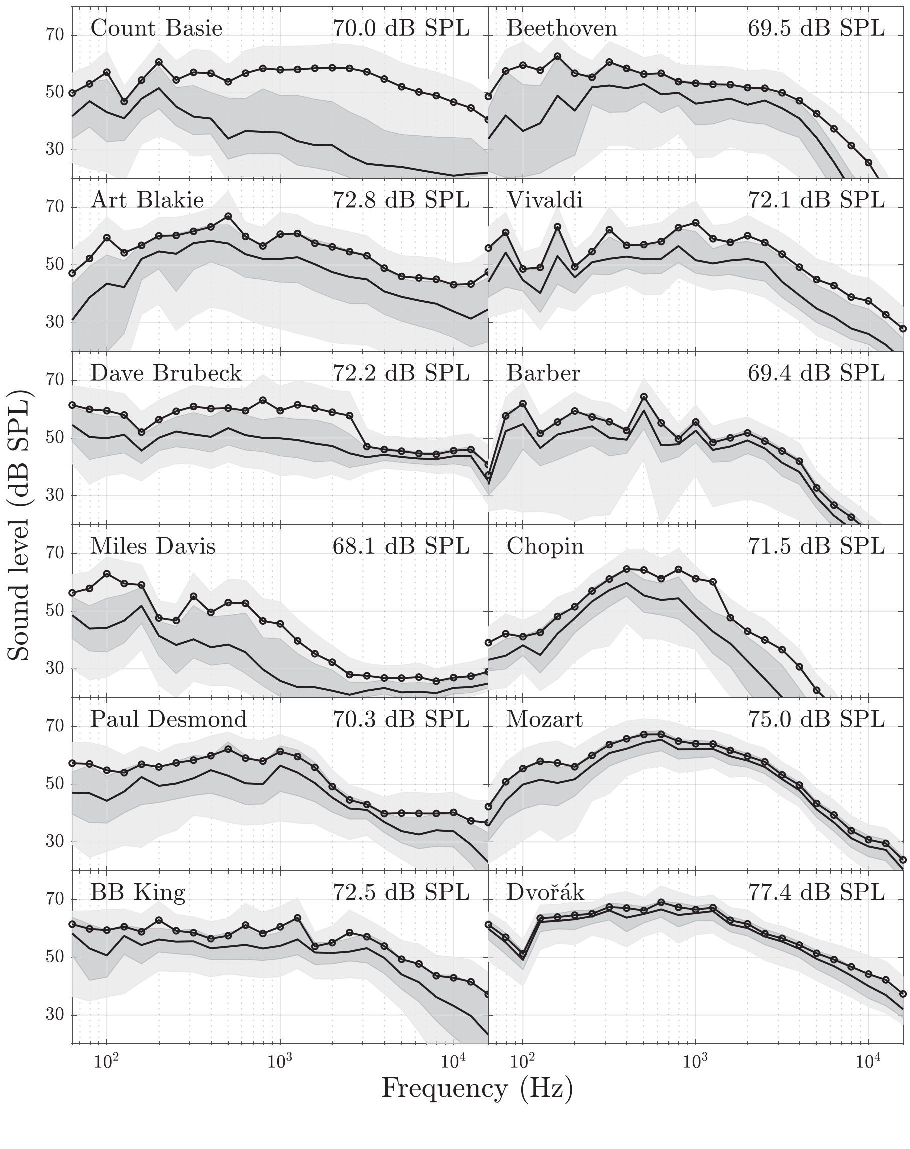

FIGURE 3.2: SPLs (Sound Pressure Levels) in third octave bands for the 12 musical excerpts. Solid lines with circles indicate the long-term RMS level in third-octave bands. Plain solid lines indicate the median level. The light grey areas indicate the corresponding range for the 5% to 95% quantiles and the dark gray areas for the 25% to 75% quantiles.

TABLE 3.1: List of music stimuli

Genre	Dynamics	Music piece
Jazz & Blues	Transition	**Count Basie**, All of Me
Jazz & Blues	Transition	**Art Blakey**, Moanin’
Jazz & Blues	Piano	The **Dave Brubeck** Quarter, Take Five
Jazz & Blues	Piano	**Miles Davis**, So What
Jazz & Blues	Forte	**Paul Desmond**, Autumn Leaves
Jazz & Blues	Forte	**BB King**, The Thrill is Gone
Classical	Transition	**Beethoven**, Symphony No. 5 in C Minor, Op. 6
Classical	Transition	**Vivaldi**, The Four Seasons, Op. 8, Spring
Classical	Piano	**Barber**, Adagio for Strings
Classical	Piano	**Chopin**, Nocturne No. 2 in E-Flat Major, O
Classical	Forte	**Mozart**, The Magic Flute, K. 620, Overture
Classical	Forte	**Dvořák**, Symphony No. 9 in E Minor, Op. 95

Input stimuli

The music stimuli described above were convolved with the room impulse responses (RIRs) of a small existing living room with a reverberation time of about 0.2 secs to simulate the effect of room acoustics on a stereo music source played with two loudspeakers positioned at $-22.5°$ and $22.5°$ in front of a listener at a distance of 1.3 m. The RIRs were measured from a Tannoy V8 loudspeaker to the in-ear microphones of a Brüel & Kjær type 4128C Head and Torso Simulator (HATS) as well as to the front microphones of two behind-the-ear (BTE) HAs placed on HATS’ ears. The music stimuli were then convolved with the RIRs from the two (stereo) loudspeakers to the BTE as well as in-ear microphones of the HATS, and for each microphone the resulting signals were added across the two reverberant (stereo) channels. The resulting signals at the BTE microphones simulated the aided signal path and provided the direct input to the HA (see *subsection Simulated hearing aid* below). The resulting signals at the in-ear microphones were afterwards low-pass filtered to simulate the acoustic signal path that circumvents the HA fitting (see *subsection Simulated hearing aid* below). There was no head motion (which could affect spatial perception) in the simulation. For further detail on the generation of the input stimuli and the acoustic path signals, see Appendix 3.A.

Simulated hearing aid

The input signals of the HA were processed in real time with a master HA developed at NAL which realized a bilateral 15-band dual-compression system that operated independently across the two ears. Dual-compression systems (Moore and Glasberg, 1988) are compression systems that combine a slow and a fast acting compressor. The output signals were presented to the subjects via Beyerdynamic DT990 PRO headphones connected to a RME Fireface UFX II audio interface. The master HA was fitted to the left and right ear separately with the gains and compression constants prescribed by the NAL-NL2 prescription formula (Keidser et al., 2011) based on the subject's individual audiograms. The actual gains and compression constants implemented in the master HA signal processing were calculated from a piecewise linear function fitted to the Input-Gain curves prescribed by NAL-NL2 for each frequency band. As a result, the applied compression constants varied slightly from the values provided by NAL-NL2 in some cases. The slow and fast compression time constants of the dual compressors were fixed across all bands with an attack time of 900 ms and 5 ms respectively and a release time of 1400ms and 30ms respectively. These values were determined from the measurement of the compression attack and release times of a recent model of a commercial HA with a dual-compression system implementation. The maximum power output of the HA was limited at 105 dB SPL.

In order to simulate **semi-open fittings** for three different vent sizes, the simulated stereo signals reaching the subject's ears via the acoustic path (see *subsection Input stimuli* above) were filtered with a 2^{nd} order low-pass filter with a selectable cut-off frequency (Dillon, 2012, p. 136-138). The output signal of the master HA was filtered with a 2^{nd} order high-pass Butterworth filter fitted to the estimated vent effect (Dillon, 2012, p. 136, 137) and added to the simulated acoustic path. The summation between the acoustic path and the output of the HA was done in real-time using the mixer of the RME audio interface. Due to the frequency-analysis and resynthesis applied in the master HA, the aided path was delayed by about 6 ms relative to the acoustic path. This delay is similar to that of most commercial HAs. The low-pass acoustic path filters and high-pass aided path filters were obtained from three different sets of measurements with vent sizes of 1 mm, 2 mm and 3.5 mm

(Dillon, 2012, p. 137-138). Participants were fitted with a different vent size depending on their audiometric threshold at 500 Hz averaged across both ears as shown in the three different panels of Figure 3.1. For further detail on the implementation of the master HA, see Appendix 3.A.

During the listening experiment, the subjects used a software interface to vary the frequency specific gain (spectral tilt) and compression ratio (CR) within the WDRC of the master HA relative to the baseline prescription provided by NAL-NL2 while roughly keeping the overall loudness constant.

The frequency-specific gain was varied by a single parameter that controlled the **spectral tilt**, which was done by increasing the prescribed insertion gains at frequency bands below the tilt cut-off frequency and decreasing the insertion gains at frequency bands above the tilt cut-off frequency, or vice versa. The tilt cut-off frequency was determined by the frequency band at the center of the HA bandwidth.[1] Thus, the cut-off frequencies were 1.5 kHz, 2 kHz and 2.5 kHz for a HA fitted with a vent of 1 mm, 2 mm and 3.5 mm, respectively. An example of the spectral tilt variation is shown in the left panel of Figure 3.3 for the case of a of a gently sloping hearing loss and a 2 mm vent. The solid line refers to the gain prescribed by NAL-NL2, and the other two curves refer to a low-frequency emphasis of -3 dB/octave (dashed-dotted curve) and a high-frequency emphasis of $+3$ dB/octave (dotted curve), respectively. This variation of the frequency-specific gain was implemented to minimise loudness variation between different conditions (Keidser et al., 2005) and was evaluated during pilot tests. Step sizes of ± 3 dB/octave were used.

The **compression ratio** was varied simultaneously across all frequency bands. The linear gain was adjusted to have the new static input-output level curve of the HA compressor meeting the baseline curve for input signal levels 20 dB above the (frequency-dependent) CT as illustrated in the right panel of Figure 3.3 for a given baseline CR at a given frequency-band. This simultaneous variation of the insertion gain and the CR was implemented in order to minimise loudness variation between different compression conditions and was evaluated during pilot tests. Compression thresholds were kept constant across all conditions at the values prescribed by

[1] The higher cut-off frequency was 10 kHz and the lower cut-off frequencies were 263, 437, and 867 Hz for 1, 2 and 3.5 mm vents, respectively.

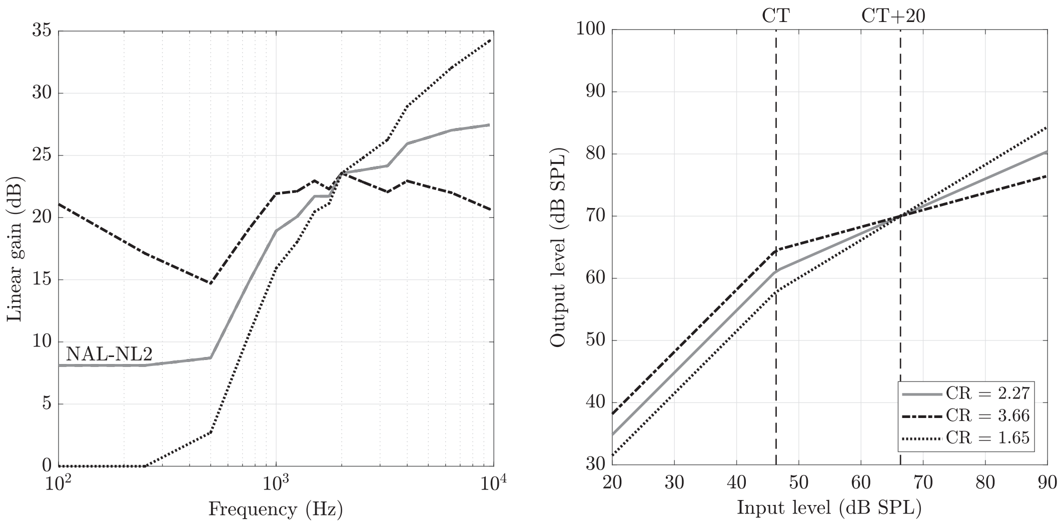

FIGURE 3.3: **Example of spectral tilt variations of insertion gains** (left panel): The grey solid curve shows the baseline gains prescribed by NAL-NL2 for a gently sloping hearing loss and a 2 mm vent. The dash-dotted and the dotted curves show spectral tilt variation with a -3 dB/octave low-frequency emphasis and a +3 dB/octave high-frequency emphasis, respectively. **Example of compression ratio (CR) variations** (right panel): The grey solid curve shows the static input-output level curve for the gain prescribed by NAL-NL2 for a given CR at a given frequency-band. The dash-dotted and dotted curves show the effect of an increase and decrease of CR, respectively. The vertical dashed lines show the compression threshold (CT) input value and the intersection of all static curves 20 dB above the CT.
Note: The maximum power output of the HA was limited to an upper knee-point at 105 dB SPL in all conditions.

NAL-NL2. Step sizes of $\Delta\frac{1}{CR} = \pm\frac{5}{30}$ were used so that for an input signal with 30 dB dynamic range a CR change would result in a 5 dB increase or decrease of the dynamic range at the output.

Instrumentation

The left and right ear signals for the aided and acoustic path for all music samples were generated prior to the experiment and stored on a test computer as 4-channel, 24-bit, 44.1 kHz *.wav* files. The input signals of the HA were processed in real time by the master HA implemented on the test computer and compiled in Matlab 2018a using Microsoft Visual C++ 2010. The output signals of the HAs at the left and right ear were summed with the corresponding acoustic path signals. The resulting stimuli were presented to the participants with a pair of Beyerdynamic DT 990 PRO semi-open headphones. The input and output of the master HA, the acoustic path signal levels and the output of the headphones were calibrated on HATS with a pink noise source signal in free field for frontal incidence.

3.3.3 Procedure

A graphical interface (see Figure 3.4) was displayed on a monitor with a 5-by-5 grid of push buttons, each one representing a combination of CR and spectral tilt variations. The centre button provided the reference or baseline setting, and corresponded to the CRs and frequency-specific gains (and thus the spectral tilt) prescribed by NAL-NL2. Variations of the spectral tilt were made for buttons along the horizontal axis with two steps of emphasis of low frequencies on the left of the baseline button and two steps of emphasis of high frequencies on the right of the baseline button. Variations of CRs were made for buttons along the vertical axis with two steps of decreased CR above the baseline button and two steps of increased CR below the baseline button. During a trial, a music stimulus was played on a loop and processed in real-time by the master HA. Participants were asked to listen to the different settings by clicking on the different push buttons and then to provide a preference rating compared to the baseline button on a 7-point Likert scale from 'Much worse' (-3) to 'Much better' (+3) using a slider. The participants were asked to rate the different settings until they found their preferred setting(s). Each time they would provide a rating of a given setting the rating they provided was displayed on the corresponding button to help them navigate the grid. They also had the possibility to listen back to previously rated conditions and revise their ratings if necessary. Subjects had to provide ratings for at least half of the buttons (i.e., 12 variations from baseline) before they could go on to the next task.

Once the participants found their preferred setting(s), they clicked on a 'Next' button. An additional window panel then opened (see Appendix E), in which they were then asked to rate the importance of five attributes in the preference ratings they gave. The five attributes were 'Sharpness', 'Excessive loudness', 'Distortion', 'Tonal quality' and 'Clarity'. During this task, participants had the possibility to listen again to all the settings for which they had provided a preference rating, but they could not modify their rating. The importance ratings were given with a 5-point scale from 'Not at all' (0) to 'Extremely important' (+4) using a slider.

After clicking a second 'Next' button, a new interface opened (see Appendix E) and the participants were asked to provide feedback on how satisfied they were with each of the five above-mentioned attributes for their preferred setting(s). During this task, participants were presented with the stimuli processed by the setting for which they had provided their highest preference rating. If different settings were given the maximum rating, the participant had the possibility to listen to each of them to help them provide their satisfaction rating.

This procedure was repeated for every music stimulus in a randomized order. At the beginning of the experiment a practice trial was done with one of four music pieces with a "transition" in their dynamics (see Table 3.1) that were randomly selected. Prior to the task at the beginning of the practice trial, the participants were asked to adjust the sound level of the 4-channel stimulus so that the louder passage would not be uncomfortably loud while keeping the softer passage audible to them. This was done with NAL-NL2 amplification. If necessary, participants were allowed to re-adjust the sound level during the run of the experiment, at the beginning of a trial. This rarely happened and was done to ensure that loudness was comfortable at all time. Participants were given the option to take the test using a computer mouse or to use the touch screen of the monitor. The instructions provided to the participants are included in Appendix D.

3.3.4 Statistical analyses

The preference ratings obtained during the first task were given a score between -3 (Much worse) and +3 (Much better). These scores were represented on a 2-dimensional plane with the spectral tilt steps along the x-axis and the CR steps along the y-axis. The coordinates of x and y ranged from -2 to +2, with the origin coordinates (0,0) representing no variation from the baseline. One step size of a change in spectral tilt or CR was represented by a change of 1 on the x- or y-axis, respectively. Negative values along the x-axis represented conditions with low-frequency emphasis and positive values conditions with high-frequency emphasis. Negative values along the y-axis represented conditions with increased CR compared to NAL-NL2 prescription and positive values conditions with decreased CR compared to NAL-NL2. The preference rating scores were used to fit polynomials of 2^{nd} order for both axis

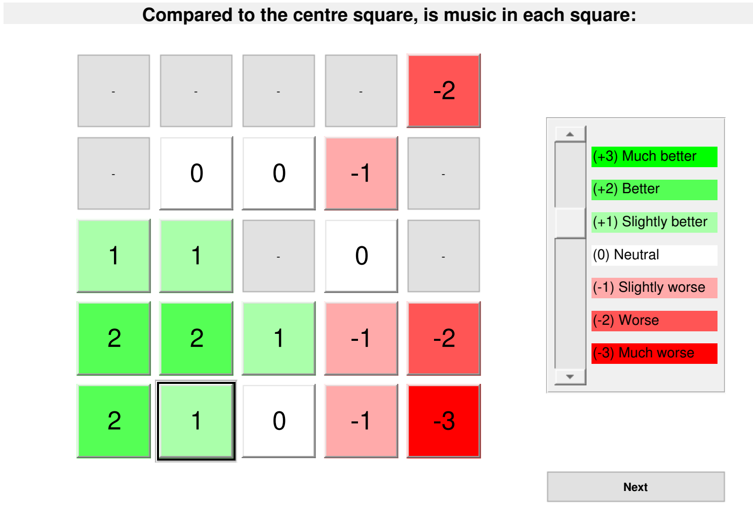

FIGURE 3.4: Graphical user interface used in the experiment to collect preference ratings with a 5-by-5 grid of push buttons, each one representing a combination of spectral tilt (along the x-axis) and CR (along the y-axis) variation with the reference at the centre and a rating slider on the right side of the grid. As seen on the figure as an example, for a condition that have been rated, the rating provided was displayed on the corresponding button.

coordinates with the least-square error method. The fitted surface contours were visually inspected to ensure that the convergence of the fit did not lead to an aberration. Additionally, different fitting models were also investigated (e.g., 2-dimensional Gaussian with rotation factor) and provided similar results for the fitted surface and the location of the extracted local maximum. Thus, the polynomial fitting was retained for the current analysis as it required fewer fitting factors than the other models. The coordinates of a local maximum (within $(x, y) \in [-2, +2]$) was extracted to obtain the estimated combination of preferred spectral tilt and CR variations. The estimated combination of preferred parameters were calculated separately (1) across all music pieces and across all participants, (2) for a given music piece across all participants, (3) for a given participant across all music pieces, as well as (4) for each individual stimulus and participant (i.e., for every trial).

Group differences for coordinates of an estimated combination of preferred parameters, chosen listening levels or rating scores were assessed with Kruskal-Wallis one-way non-parametric analysis of variance tests. To investigate potential relationships between variables (e.g., RMS level of the input signal to the HA) and coordinates of an estimated combination of preferred parameters, Kendall's τ non-parametric correlation coefficients were calculated. All tests used the value $p \leq 0.05$

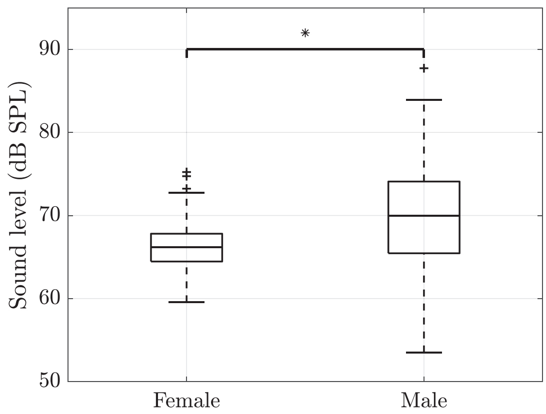

FIGURE 3.5: Box plot representation of RMS levels at the input of the hearing aids for females and males for every trial of the experiment, including training and preferred pieces of music. On each box, the central mark indicates the median, and the bottom and top edges of the box indicate the 25^{th} and 75^{th} percentiles, respectively. The whiskers extend to the most extreme data points not considered outliers, and the outliers are plotted individually using the "+" symbol.

for statistical significance. Cohen's standard was used to evaluate the strength of the relationships, where coefficients between .10 and .29 represent a weak association, coefficients between .30 and .49 represent a medium association and coefficients above .50 represent a strong association (Cohen, 1992).

3.4 Results

3.4.1 Listening levels

Chosen listening levels (CLLs) ranged from 53.5 to 87.7 dB SPL (median 68.0, SD = 5.5) at the input of the HA across all trials, including training and preferred stimuli. A Kruskal-Wallis one-way non-parametric analysis of variance showed significant group differences in CLLs with the gender of the participants, $\chi^2 = 32.83$, $df = 1$, $p < .01$. Focused comparisons of the mean ranks between the two gender groups showed that lower CLLs were observed for females (median 66.2 dB SPL) than for males (median 70.0 dB SPL). This group difference in CLLs is illustrated in Figure 3.5. No effect on CLLs of other demographic variables, such as degree of hearing loss or age, was found.

3.4.2 Spectral tilt and CR preferences

The polynomials surface fitted to the entire dataset across every piece of music and every participant (Goodness-of-fit statistics: $R^2 = 0.28$, adjusted $R^2 = 0.28$ and RMS-error $= 1.28$) is shown in Figure 3.6 with spectral tilt variation on the horizontal axis (low-frequency emphasis to the left and high-frequency emphasis to the right) and CR variation along the vertical axis (lower CR than baseline on the top and higher CR towards the bottom). The black circle indicates the coordinates of the local maximum of the surface fitted to the preference ratings, i.e., the coordinates of the estimated preferred combination of spectral tilt and CR. Thus, the preferred spectral tilt variation was found for the maximum amount of low-frequency emphasis (-2 steps of spectral tilt) while the preferred CR variation was found at almost the maximum amount of compression (-1.72 steps of CR variation).

The left panel of Figure 3.7 is a scatter plot of the estimated preferred combinations of parameters for each individual stimulus from fittings across all participants. Each dot corresponds to an individual stimulus, and refers to the coordinates of the local maximum of the surface fitted to the given preference data (similar to the location of the black circle in Figure 3.6). For every stimulus, the preferred spectral tilt was found for the maximum amount of low-frequency emphasis (-2 steps of spectral tilt) and the preferred CR condition varied between +0.02 and -2 step of CR variation (mean -1.43, SD = 0.68). The Kendall's τ correlation coefficient between the coordinates of the preferred CRs and the CCLs averaged across participants was calculated. This revealed a strong positive correlation ($\tau = 0.66, p = .005$) which indicates that the quieter a music stimulus was the more amount of compression was preferred. It is worth noting that the preferred CR for the piece of music with the highest sound level (with an average RMS level of 76.0 dB SPL at the input of the HAs across all participants) was close to the baseline CR (+0.02 step of CR).

The middle panel of Figure 3.7 is a scatter plot of the estimated preferred combinations of parameters for each individual participant from fittings across all stimuli. Each dot corresponds to an individual participant. Apart from five participants, the preferred spectral tilt was found for the maximum amount of low-frequency emphasis (-2 steps of spectral tilt, mean -1.74, SD = 0.57). The preferred CR condition

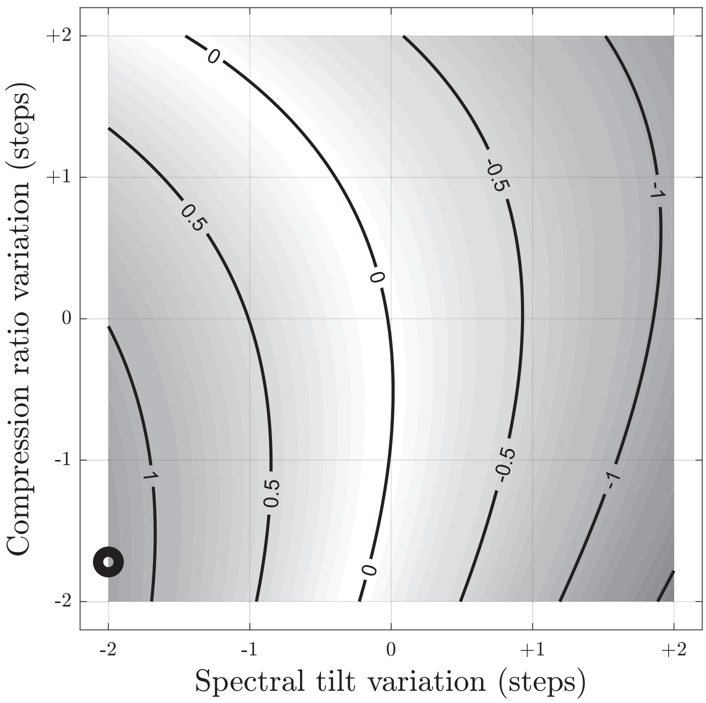

FIGURE 3.6: Surface fitted to preference ratings across all music pieces and all participants using polynomials of 2^{nd} order of tilt variations and 2^{nd} order of CR variations. On the horizontal axis are shown the 5 conditions of spectral tilt with step sizes of ± 3 dB/octave along the baseline amplification. On the vertical axis are shown the five conditions of CR variation with step sizes of $\Delta\frac{1}{CR} = \pm\frac{5}{30}$ along the baseline compression. The origin (0,0) corresponds to no variation from the baseline. The solid black lines indicate the surface contours of the polynomial model fitted to the preference ratings. The numerical values on the contour lines indicate the fitted preference ratings from a 7-point Likert scale ranging from 'Much worse' (-3) to 'Much better' (+3). The black circle indicates the location of the local maximum of the fitted surface.

varied between +2 and -2 steps of CR variation (mean -0.70, SD = 1.41).

The right panel of Figure 3.7 is a scatter plot of the estimated preferred combinations of parameters for each individual music piece and participant. Each dot corresponds to an individual stimulus for an individual participant. The preferred spectral tilt condition varied between +2 and -2 steps of spectral tilt variation (mean -1.47, SD = 1.12). Similarly to the previous fitted models, the majority of the preferred spectral tilt corresponded to the maximum amount of low-frequency emphasis. The preferred CR varied between +2 and -2 steps of CR variation (mean -0.70, SD = 1.54). A Kruskal-Wallis one-way non-parametric analysis of variance showed significant group differences in the coordinates of the preferred CRs with the gender of the participants ($\chi^2 = 5.35$, $df = 1$, $p = .021$). Focused comparisons of the mean ranks between the two gender groups showed that less amount of compression was preferred by females (mean -0.33 step, SD = 1.70, $n = 91$) than by males (mean -0.84 step, SD = 1.47, $n = 247$). A Kruskal-Wallis one-way non-parametric analysis of variance showed significant group differences in the coordinates of the preferred CRs with the vent size participants were fitted with ($\chi^2 = 6.21$, $df = 2$, $p = .045$). Focused comparisons of the mean ranks between the three vent size groups did not reveal which of the three groups had CRs significantly different from the other groups. However, participants fitted with 2 mm vents tended to prefer more compression (mean -0.92 step, SD = 1.47, $n = 169$) than participants fitted with 1 mm vents (mean -0.57 step, SD = 1.51, $n = 91$) or 3.5 mm vents (mean -0.39 step, SD = 1.68, $n = 78$). A Kruskal-Wallis one-way non-parametric analysis of variance showed significant group differences in the coordinates of the preferred spectral tilt with the vent size participants were fitted with ($\chi^2 = 6.72$, $df = 2$, $p < .001$). Focused comparisons of the mean ranks between the three vent size groups showed that less amount of low-frequency emphasis was preferred by participants fitted with 3.5 mm vents (mean -0.8 step, SD = 1.47, $n = 169$) than for participants fitted with 1 mm vents (mean -1.72 step, SD = 0.85, $n = 91$) or 2 mm vents (mean -1.63 step, SD = 0.94, $n = 78$). It is worth noting that the significance of the vent diameter, as mentioned above, is confounded by the hearing loss since the vent diameter was based on the low-frequency PTA at 500 Hz. Kruskal-Wallis one-way non-parametric analyses of variance did not show group differences in the coordinates of the preferred spectral tilt and CRs with the genre of the stimuli.

3.4.3 Sound quality attributes, importance and satisfaction

Importance ratings of the five different attributes in regards with the preference judgements are shown in Figure 3.8 (left panel) for all stimuli, preferred pieces of music included. The attributes "Clarity", "Tonal quality", "Sharpness" and "Distortion" obtained a median importance rating of 3, indicating that these attributes are on average considered to be ’very important’ for preference judgements. "Excessive loudness" obtained a median importance rating of 2, indicating that this attribute

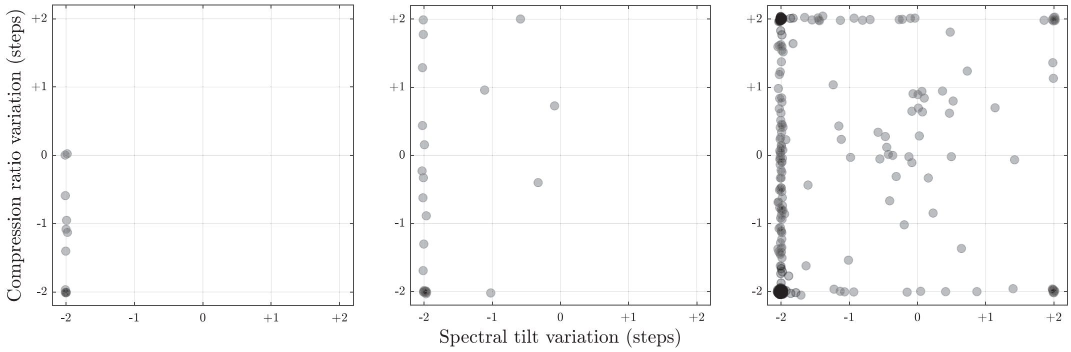

FIGURE 3.7: ***Left panel:*** Scatter-plot of the estimated preferred combinations of parameters for each individual stimulus from fittings across all participants, training and preferred piece of music excluded. Each dot corresponds to an individual stimulus. ***Middle panel:*** Scatter-plot of the estimated preferred combination of parameters for each individual participant from fittings across all stimuli, preferred piece of music included. Each dot corresponds to an individual participant. ***Right panel:*** Scatter-plot of the estimated preferred combination of parameters for each stimulus and participant. Each dot corresponds to an individual stimulus for an individual participant. In order to improve the plot readability of all three panels, a normally distributed random jitter was applied in both dimensions with a mean of 0 and a standard deviation of 0.02.

is on average considered to be 'important' for preference judgements. A Kruskal-Wallis one-way non-parametric analysis of variance showed significant group differences in importance ratings between the different attributes ($\chi^2 = 91.95$, $df = 4$, $p < .001$). Focused comparisons of the mean ranks between the five attributes showed that "Excessive loudness" (mean 2.29, SD = 1.24, $n = 337$) was given lower importance ratings than any other attributes, whereas "Distortion" (mean 2.67, SD = 1.11, $n = 337$) and "Sharpness" (mean 2.72, SD = 0.85, $n = 337$) did not have significantly different mean importance ratings nor "Tonal quality" (mean 2.92, SD = 0.92, $n = 337$) and "Clarity" (mean 3.07, SD = 0.89, $n = 337$). Significant group differences in importance ratings between the attributes are indicated with the horizontal lines above the box plots in Figure 3.8 (left panel). A Kruskal-Wallis one-way non-parametric analyses of variance did not show group differences in importance ratings with gender nor with the vent size participants were fitted with.

Satisfaction ratings with the preferred combination of parameters for the five sound quality attributes are shown in Figure 3.8 (right panel). All five attributes obtained a median satisfaction rating of 2, indicating that participants were 'satisfied' with these sound quality attributes while listening to their preferred combination of parameters. A Kruskal-Wallis one-way non-parametric analysis of variance showed significant group differences in satisfaction ratings between the different attributes ($\chi^2 = 23.26$, $df = 4$, $p < .001$). Focused comparisons of the mean ranks between the five attributes showed that "Distortion" (mean 1.62, SD = 1.17, $n = 301$) was given lower satisfaction ratings than "Clarity" (mean 2.02, SD = 0.85, $n = 301$) and "Tonal quality" (mean 1.93, SD = 0.99, $n = 301$); and "Excessive loudness" (mean 1.59, SD = 1.33, $n = 301$) was given lower satisfaction ratings than "Clarity". Satisfaction with "Sharpness" (mean 1.93, SD = 0.85, $n = 301$) did not differ significantly from any of the other attributes. Group differences in satisfaction ratings between the attributes are indicated with the horizontal lines above the box plots in Figure 3.8 (right panel). A Kruskal-Wallis analysis showed significant group differences in satisfaction ratings with the vent size participants were fitted with ($\chi^2 = 7.22$, $df = 2$, $p = .03$). Focused comparisons of the mean ranks between the three vent size groups did not show which group had significantly different satisfaction ratings from the others. However, participants fitted with 2 mm vents tended to be less satisfied (mean 1.75, $SD = 1.09$, $n = 755$) than participants fitted with 1 mm vents (mean 1.87, SD = 1.07, $n = 405$) and participants fitted with 3.5 mm vents (mean 1.91, SD = 1.00, $n = 345$). It is worth noting that the significance of the vent diameter, as mentioned above, is confounded by the hearing loss since the vent diameter was based on the low-frequency PTA at 500 Hz. A Kruskal-Wallis one-way non-parametric analysis of variance did not show group differences in importance ratings with gender.

3.5 Discussion

The aim of the current study was to further understand the signal processing strategies preferred by participants when listening to music in relation to the problems highlighted by the survey conducted previously (see Chapter 2). In particular, this listening experiment aimed at identifying what changes in frequency-specific gain and CR relative to those prescribed by the NAL-NL2 standard fitting formula were preferred by HA users for music listening. Overall, higher CRs than prescribed by NAL-NL2 were preferred by participants while the preferred frequency-specific

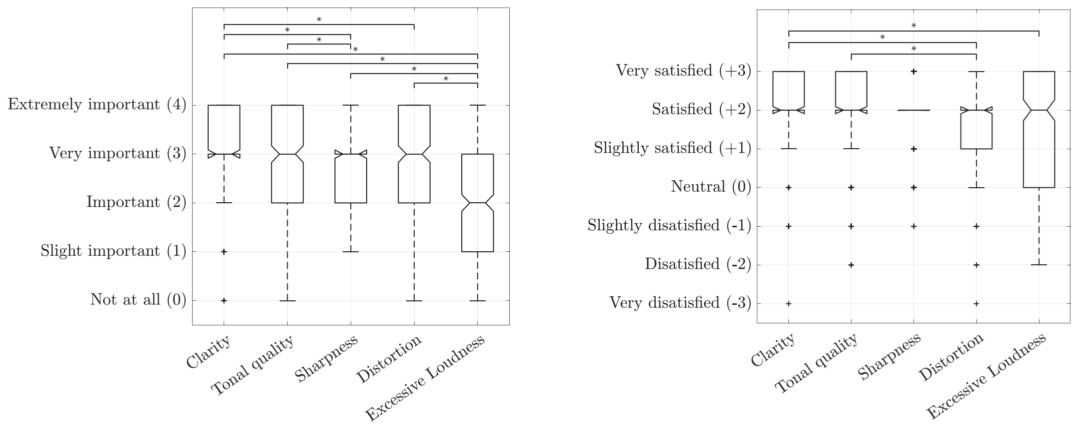

FIGURE 3.8: Box plot representations of importance (left panel) and satisfaction (right panel) ratings of five sound quality attributes for every stimulus of the experiment, including preferred pieces of music. The horizontal lines marked with an asterisk above each box indicate significantly different means between two attributes. ***Left panel:*** Importance ratings of the five attributes in regards with the preference judgements previously provided. ***Right panel:*** Satisfaction ratings with the five attributes for the preferred combination of parameters.

gain condition was in most cases the condition with the maximum amount of low-frequency emphasis provided in the experiment. More specifically, the preferred CRs estimated for each stimulus (from fittings across all participants) were found to be strongly correlated ($\tau = 0.66$) with the corresponding CLLs at the input of the master HA (averaged across participants). The quieter a music stimulus was, the higher CRs tended to be preferred. Variations in preferred spectral tilt were affected by the vent size participants were fitted with: less amount of low-frequency emphasis was preferred by participants fitted with 3.5 mm vents than for participants fitted with 1 mm or 2 mm vents. All five sound quality attributes obtained high importance ratings with the highest importance ratings given to "Clarity" and "Tonal quality" followed by "Distortion" and "Sharpness" (all four attributes with a 'Very important' median score) and "Excessive loudness" obtained the lowest importance ratings (median 'Important'). All five attributes obtained a median satisfaction score of 'Satisfied' for the preferred combination of parameters.

The listening levels chosen by participants ranged between 53.5 to 87.7 dB SPL (median 68.0) at the input of the master HA. These CLLs are consistent with Croghan et al. (2016) who reported average CLLs of 69 dBA at the entrance of the ear canal when using BTE HAs. The group difference in CLLs of 4 dB found between females and males adds to the difference of insertion gains prescribed by NAL-NL2 for females and males, i.e., for the same hearing loss profile, NAL-NL2 prescribes 2 dB more of insertion gain across all frequency bands for males than for females. This suggests that the difference in loudness preferences between females and males is even greater for music than it is for speech signals. However, to quantify more accurately the difference in preferences between females and males, data collection from a larger sample size may be required. The group differences in preferred CRs found between females and males (i.e., females preferring lower CRs than males) may indicate that their preference judgements were driven by the perceived loudness. The lower CRs preferred by females may be the result of them also preferring lower CLLs. For soft passages, raising the CRs (as most participants have done) results in a bigger increase in output level as input level decreases. Thus, the lower input levels chosen by females may have contributed to them not raising the CRs as much as males did.

In all conditions the maximum amount of low-frequency emphasis was preferred. In the context of speech, increasing low-frequency amplification could result in a decrease in clarity of speech (e.g., due to upward spread of masking). The current findings indicate that this does not generalise to the perceived clarity of music signals. However, the meaning of clarity may be ambiguous as it could be interpreted as the ability to follow the melodic lines or to focus on individual instruments. Using the open source master HA, Vaisberg et al. (2018) found that a low-frequency boost (from 100 to 800 Hz) of the insertion gains prescribed by DSL (Scollie et al., 2005) was preferred by HA users when listening to music. Franks (1982) found that increased low-frequency bandwidth was preferred over a higher cut-off frequency. Moore and Tan (2003) also manipulated low-frequency bandwidth and found that increasing the cut-off frequency was detrimental for perceived naturalness of music stimuli. It would be of interest to identify if even further low-frequency emphasis than presented in this study would be preferred by HA users.

While the findings of Franks (1982) and Moore and Tan (2003) support the importance of low-frequency amplification in the context of music listening, providing a listener with extended low-frequencies may not be achievable by all HAs. The highest naturalness rating in Moore and Tan (2003) was given to a broadband signal with a lower cut-off as low as 55 Hz while the typical lower cut-off for the amplified path in a HA with a small vent is about 200 Hz (Dillon, 2012). The amount of bass amplification that a HA receiver can provide is not only dependent on the prescribed gain but is greatly influenced by the type of fitting. While semi-open fittings are the most common fitting with current HAs the amount of gain that can be provided at low frequencies is very limited and low-frequency stimulation is generally achieved acoustically from the vent path without passing through the HA amplification.

The current study is supporting evidence that achievable low-frequency amplification is also preferable while listening to music with semi-open HAs. However, less amount of low-frequency emphasis was preferred by participants fitted with the most open vents (3.5 mm) than for participants fitted with smaller vents (1 mm or 2 mm). One potential explanation of their slight preference for a smaller amount of low-frequency emphasis is the higher cut-off frequency in the simulated acoustic path signal allowed by the most open vent ($f_c = 867$ Hz) than allowed by the 1 mm and 2 mm vents ($f_c = 263$ and 437 Hz, respectively). Since the simulated acoustic path was added to the aided path with a 0 dB insertion gain, participants fitted with a 3.5 mm vent might have preferred less low-frequency emphasis to preserve the spectral balance of the summed signal. Since their low-frequency audiometric thresholds were better than those of the two other groups, another explanation could simply be that they did not need any further amplification in this frequency region.

The music being too loud and difficulties hearing soft passages of music were some of the most frequent problems reported in Chapter 2 as well as in Madsen and Moore (2014b). However, HAs seemed to have opposite effects on each of these problems. In majority, HAs were reported to be helpful to hear the soft passages (65% of the responses) while making the music too loud (50.6%). This suggests that the HAs may be providing enough amplification at low levels to allow soft sounds to be audible but may not provide adequate gain compression, thus making the higher sound levels uncomfortably loud. These findings are consistent with the overall preferences for higher CRs than prescribed by NAL-NL2 in the current study. The strong positive relationship found between preferred CRs and CLLs in the current study is also supporting these conclusions (i.e., the quieter a music stimulus was the higher CRs were preferred). However, most studies investigating the effects of compression on music stimuli tend to argue that the use of compression in HAs is detrimental for sound quality when compared to linear amplification (e.g., VanBuuren et al., 1999; Croghan et al., 2014; Madsen et al., 2015). While the conclusion of these studies appears to contradict the observation that higher CRs were preferred over CRs prescribed by NAL-NL2 in the current study, there are potential explanations that could reconcile both findings. Firstly, the correlation between CLLs and the preferred CRs suggest that for input signals higher than the current study's CLLs, lower CRs than prescribed by NAL-NL2 could be preferred.

It would be interesting to reproduce the current study with input levels higher than the CLLs to confirm this. For example, Madsen et al. (2015) presented all their music stimuli with an average level of 70 dB SPL at the HA input (which is about 3 dB above the median CLLs of the current study) and found no effect of compression speed on clarity judgements,[2] while Moore et al. (2011) found a decrease in pleasantness with increasing compression speed with input levels of 80 dB SPL but no effect was found with lower input levels (i.e., 50 and 65 dB SPL). Croghan et al. (2014) performed acoustical analysis on music stimuli processed by WDRC and showed that increasing the compression speed reduced the distribution of samples within a smaller range of amplitudes. They concluded that compression was detrimental to music sound quality, with slow WDRC preferred over fast WDRC. The reduced distribution of samples within a smaller range of amplitudes is also expected to be an effect of increasing the CR or lowering the CT. In the current study, the CT was fixed in the simulation. Thus, at higher input intensities more of the input signal lies above the CT and the greater the amount of non-linear distortion resulting from the dynamic envelope gain changes imposed by the compression. Kates et al., 2018 provided an example of distortion for speech as a function of input level. Thus, we can conclude from Moore et al. (2011), Madsen et al. (2015) and the current study that the use of compression might be more critical for sound quality at high input levels than for low input levels. Irrespective of input levels and compression speed, Madsen et al. (2015) found that lower clarity judgements were given to WDRC than

[2]In Madsen et al. (2015), the attack/release times were 50/3000ms for the slow compression condition and 10/100ms for the fast compression condition.

for linear amplification. The current study used only one set of CTs and dual time constants. It would be interesting to investigate in a follow-up study the interaction between CRs with other compression characteristics and the genre of music.

However, in the current study, "Clarity" was given the highest importance ratings in the preference choices for variations of CR and spectral tilt among all five sound quality attributes. Also, high satisfaction ratings in "Clarity" were given to the preferred combination of CR and spectral tilt. This suggests that under certain conditions increasing the CRs may improve the clarity of the music. For instance, if the spectral tilt is set to the maximum amount of low-frequency emphasis, the reduction in insertion gains of high-frequency bands could result in these frequency regions becoming inaudible for the listener (see Figure 3.3, left panel). In this case, increasing the CR would provide some additional gain to the high-frequency region of the input signal (see Figure 3.3, right panel). This is supported by "Tonal quality" being the second most important attribute for preference judgements and its high satisfaction ratings with the preferred combination of parameters.

In Chapter 2, "the music is not clear enough", "the tonal quality is poor", "distortions occur" and "the music is too sharp/shrill" were the problems that most drove the satisfaction rating for music listening with HAs. The current study showed that it is possible to manipulate the signal processing of HAs, namely compression and frequency-specific gains, and to obtain good satisfaction outcomes for instrumental music in relation to the problems highlighted by the survey. In particular, this was achieved with insertion gains emphasising the low frequencies, reducing high-frequency amplification, and increasing the CRs above those prescribed by NAL-NL2. This highlights the limitations of most studies investigating HAs and music, increasing compression alone can appear detrimental in some cases but can be preferred when combined with other manipulations. However, it would be of interest to use a commercially available HA and a more direct paired-comparison or MUSHRA (MUltiple Stimuli with Hidden Reference and Anchor, ITU-R, 2015) like method to confirm that such changes of CRs and spectral tilt relative to NAL-NL2 are improving satisfaction of the five sound quality attributes.

Additionally, it should be highlighted that the current findings are based on the sound quality of instrumental music. Future research should investigate music containing vocals to determine whether or not preferences differ from instrumental music. For some listeners, understanding lyrics is an important aspect of the listening experience, while some other listeners may regard vocals as any another instrument and may not pay attention to lyrics at all. In the one case intelligibility should be prioritised whereas in the other case sound quality would be the most important aspect. When addressing those two aspects there may be a fundamental trade-off: understanding lyrics requires sufficient high-frequency amplification to make consonants audible, while improved sound quality requires greater low-frequency amplification. Therefore, it may not be possible to satisfy both requirements at the same time. In order to further investigate what HA signal processing strategies are preferred for music listening, future research should investigate variations of frequency-specific gain and compression of different frequency regions independently. Future research should also extend investigations on preferences with input levels above CLL as this could have implications in other contexts such as live music where the HA users have no control over the sound level likely to be well above CLLs.

3.6 Conclusions

The current study showed that variations of the frequency-specific gains and compression relative to NAL-NL2 are preferred by HA users when listening to instrumental music played via loudspeakers. In particular, participants consistently preferred amplification with an increased gain at low-frequency and reduced gain at high-frequency relative to NAL-NL2. However, the preferred amount of low-frequency emphasis was less for the participants with the most open fitting than for participants fitted with smaller vents. In terms of compression, higher CRs relative to NAL-NL2 were preferred. The extent to how much more compression was preferred was influenced by the level of the music stimuli at the input of the HAs, with higher CRs preferred for softer stimuli. Additionally, the selected input levels and preferred compression highlighted the difference in preferred listening levels between females and males, with females preferring music stimuli being 4 dB softer than males on average. This resulted in females preferring slightly less compression than males. Thus, future research intending to derive amplification prescription schemes specifically targeted at optimizing the enjoyment of music with HAs should give particular attention to gender differences, the effect of HA fitting and the sound levels of the music signals at the input of the HAs.

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The study was supported by the HEARing CRC, established and supported under the Cooperative Research Centres Program – an initiative of the Australian Government, as well as by a PhD scholarship provided by Macquarie University, Australia. Parts of this work were presented at the International Hearing Aid Research Conference (IHCON), Tahoe City, USA, August 2018.

3.A Supporting information: Simulation of the most common music listening scenario

Chapter 3 described a listening experiment that relied on the simulation of the signals received by an aided hearing-impaired listener when listening to music played by a stereo loudspeaker system inside a living room. As illustrated in Figure 3.9, the corresponding music stimuli contained four signal channels to represent the input signals to two behind-the-ear (BTE) HAs fitted to the left and right ear of the listener as well as the acoustic signals that circumvent the HAs. The BTE signals for the left and right ear were processed by the master HA and added to the corresponding acoustic path signals in real-time. The resulting signals were then presented via headphones to the listener, who was interacting with a Graphical User Interface (GUI). At the beginning of the experiment the master HA was fitted to the individual subject using the NAL-NL2 prescription formula (Keidser et al., 2011). Each time a listener selected a new combination of parameters in the GUI, a set of new compression ratios (CRs) and insertions gains were calculated and the HA processing was updated in real-time. Details on the music stimulus generation and the master HA platform are given in the following sections.

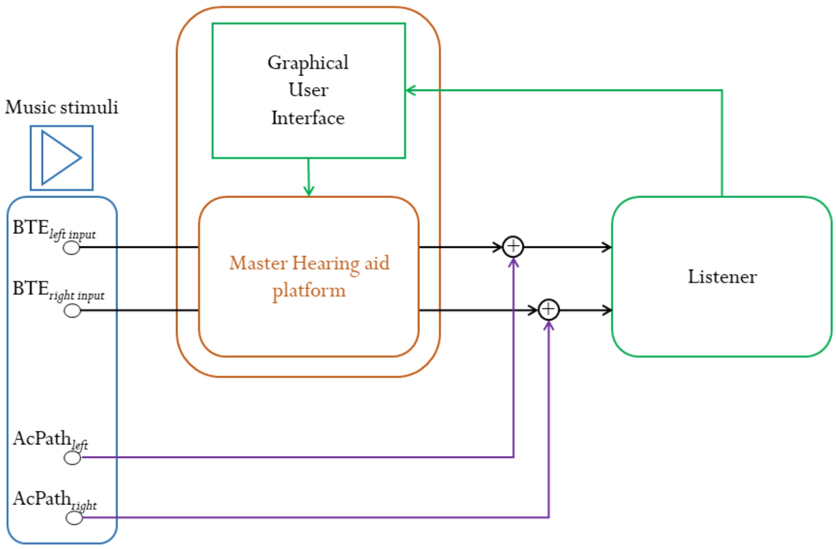

FIGURE 3.9: Signal flow diagram of the real-time processing of the master hearing aid processing. AcPath: acoustic path signals; BTE: BTE HA input signals.

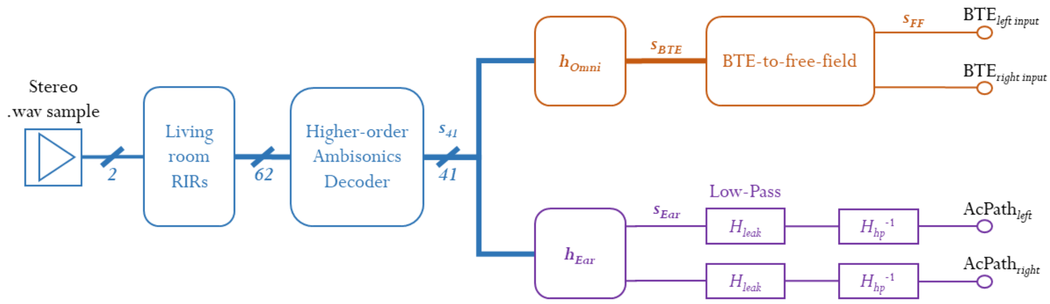

FIGURE 3.10: Signal flow diagram of the listening experiment.

3.A.1 Simulated stereo playback in a living room

In the survey described in Chapter 2, the most common music listening scenario with HAs reported by the participants was to listen to music "at home". To replicate this scenario, the listening experiment described in Chapter 3 involved a simulation of music played from a stereo system inside a living room to an aided HA listener. Figure 3.10 shows the acoustic signal pathway involved in this simulation, which also took into account the effects of venting induced by semi-open fitting of BTE HAs.

In order to simulate the playback of a stereo music recording inside a living room, first the sound field of such scenario was reproduced with the 3D loudspeaker array located inside the anechoic chamber of the Australian Hearing Hub (Macquarie University, Australia), using the methods described by Weisser et al. (in press). A small existing living room with a reverberation time of about 0.2 secs was reproduced with the two stereo loudspeakers positioned at $-22.5°$ and $22.5°$ in front of a listener at a distance of 1.3 m. For the two stereo loudspeaker locations, separate RIRs were measured from a Tannoy V8 loudspeaker to a small 62-channel hard-sphere microphone array located at the listener position inside the living room. The two 62-channel RIRs were then decoded into 41-channel RIRs for playback via the 3D loudspeaker array using the higher-order Ambisonics method. The two RIRs were finally convolved with the two stereo channels of the selected music recording and combined (by summation) into a single 41-channel signal, s_{41}. The sound field of the desired listening scenario was then created by playing back this signal via the 3D loudspeaker array.

In order to generate the 4-channel signals presented to the listeners, a Brüel &

Kjær type 4128C head and torso simulator (HATS) was placed inside the 3D loudspeaker array with two BTE HA dummies fitted to the left and right ear. The HA dummies had cables connected from their microphones to the sound card of a measurement computer, which also received input from the HATS in-ear microphones. To simulate the playback of the listening scenario, impulse responses (IRs) were measured from each of the 41 playback loudspeakers to the left and right in-ear microphones of the HATS as well as to the front microphones of the two BTE HAs. Within the measurements, each of the 41 loudspeakers was individually calibrated and free-field equalized, including a low-frequency boost to compensate for the low-frequency roll-off below 100 Hz of the applied Tannoy V8 loudspeakers. Finally, the reverberant 41-channel music signals, s_{41}, were then convolved channel-by-channel with these IRs and summed up separately for each microphone to create a 4-channel music signal.

The two signal channels for the front BTE microphones, s_{BTE}, were then transformed into free-field equivalent signals, s_{FF}, which provided the input signals to the master HA (see Figure 3.12). To realize this transformation, transfer functions (TFs) were measured in an anechoic chamber from a loudspeaker in front of HATS to the two (frontal) BTE microphones (placed on HATS) as well as to an omnidirectional measurement microphone (1/4" Type 46BL G.R.A.S.) at the location of the HATS with the HATS removed. The transformation was then realized by a minimum-phase filter that approximated the TF for the omnidirectional microphone divided by the average TF for the two BTE microphones (considering only absolute values of the TFs). The two signal channels for the in-ear microphones s_{Ear} were filtered with a 2^{nd} order infinite impulse response (IIR) low-pass filter to account for the vent transmitted gain (H_{leak}) that is applied to the acoustic signal when passing through the vent of the HA fitting. The low-frequency filters were fitted to the transmission gains from three different sets of measurements with vent sizes of 1 mm, 2 mm and 3.5 mm (Dillon, 2012, p. 137-138) as shown in Figure 3.11 (left panel). The solid lines show the measured gains and the dashed lines show the fitted filters for each vent size. The low-pass filtered signals were then equalized with the transfer function for headphones equalisation (H_{hp}^{-1}, measured on HATS) in order to compensate for the frequency response of the pair of Beyerdynamic DT 990 PRO semi-open headphones used in the experiment to present the stimuli to the participant. The resulting signals provided the acoustic path signals to be summed with the aided signals from the output of the master HA (Figure 3.12).

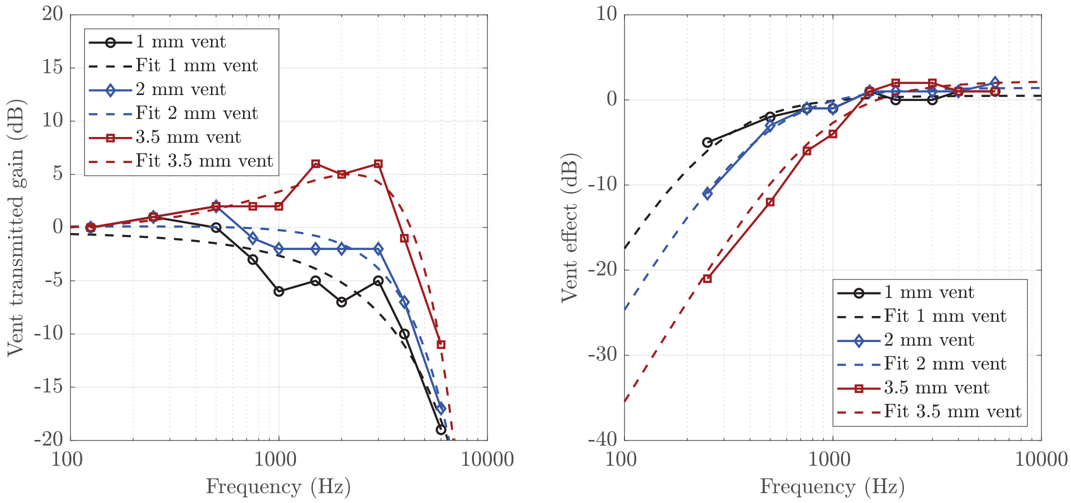

FIGURE 3.11: ***Left panel:*** Vent transmitted gains of the vent-transmitted acoustic path signals for semi-open fitting of three different vent sizes. ***Right panel:*** Effect of semi-open fitting of three different vent sizes on the amplified signal at the output of the hearing aid. Solid lines refer to measured gains and dashed lines to the frequency response of the filters that were applied to realize the measured gains.

3.A.2 Master HA processing

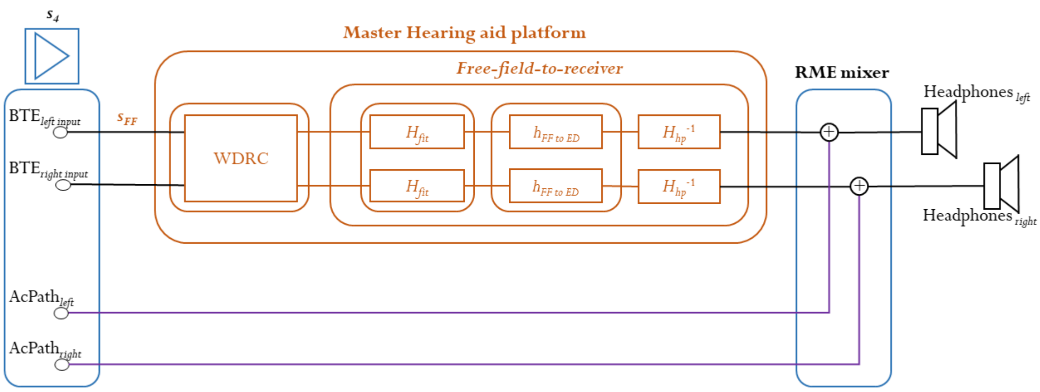

FIGURE 3.12: Signal flow diagram of the master hearing aid processing.

The free-field equivalent signals received by the BTE microphones s_{FF} were processed in real-time by the master HA, as shown in orange in Figure 3.12. The main signal processing of the master HA realized independent bilateral WDRC, which was implemented in 15 separate frequency channels and using dual compression (see section 3.3.2 for details). The output signal of the WDRC stage was filtered with pre-calculated receiver configuration filter gains to realize a *free-field-to-receiver* mapping. This mapping transformed the free-field equivalent input signals to ear-drum equivalent signals, while also considering realistic output level constraints of a HA receiver fitted with a semi-open fitting. The filtering included a high-pass filter simulating the gains introduced by the HA vent (H_{fit}), the free-field-to-ear-drum transfer functions (h_{FFtoED}) and the transfer function for headphones equalisation (H_{hp}^{-1}). The resulting signal provided the output of the master HA and realized the aided path signal.

In order to simulate different vent sizes, a 2^{nd} order high-pass Butterworth filter was fitted to the estimated vent-related gain (Dillon, 2012, p. 136, 137) as shown in Figure 3.11 (right panel). The solid lines show the measured gains and the dashed lines show the fitted filters for each vent size. The *free-field-to-ear-drum* signal transformation was realized in the same way as the *BTE-to-free-field* transformation described above, except that the TFs were measured to the in-ear (instead of BTE) microphones of the HATS for frontal sound incidence and the FIR filter was fitted to the TF to the in-ear microphones divided by the TF to the omnidirectional microphone. Hence, this transformation partly inverted the *BTE-to-free-field* transformation inherent in the input signal to the master HA, and introduced natural head shadow and pinna effects for frontal sound incidence as well as the ear-canal resonance. The headphones equalization filters were the same as applied in the acoustic path signals circumventing the HAs (H_{hp}^{-1}, see Figure 3.10).

The aided path and the acoustic path signals were then added in real-time using the mixer of a RME Fireface UFX II audio interface connected to the master HA computer. Due to the frequency-analysis and re-synthesis applied in the master HA, the aided path was delayed by about 6 ms relative to the acoustic path. This delay is similar to that of most commercial HAs. The resulting stimuli were presented to the participants with a pair of Beyerdynamic DT 990 PRO semi-open headphones.

In order to illustrate the effect of the WDRC provided by the master HA for an example music stimulus, Figure 3.13 shows the dynamic range histogram of the 3^{rd} classical music stimulus (Beethoven, Table 3.1) played at the input (left panel) and

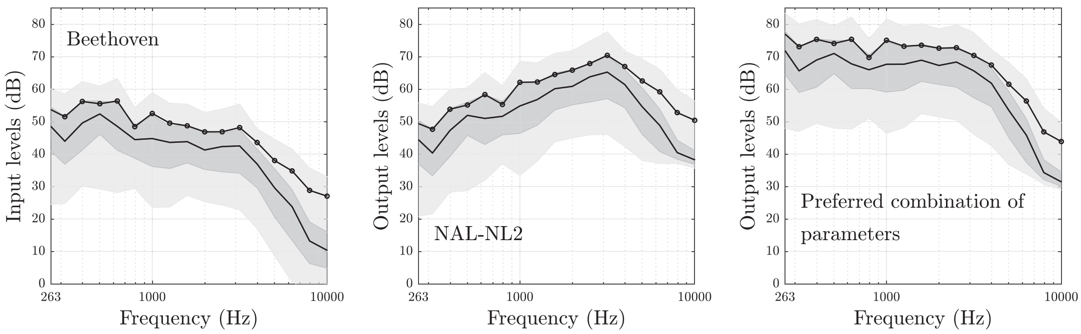

FIGURE 3.13: SPLs (Sound Pressure Levels) in third octave bands of a classical music stimulus at the HA input (***left panel***) and the direct output of the master HA with the gains prescribed by NAL-NL2 (middle panel) as well as the preferred gains derived across all music stimuli and subjects (right panel). Solid lines with circles indicate the long-term RMS level in third-octave bands. Plain solid lines indicate the median level. The light grey areas indicate the corresponding range for the 5% to 95% quantiles and the dark gray areas for the 25% to 75% quantiles.

recorded at the output of the WDRC stage of the master HA for the gains prescribed by NAL-NL2 (middle panel) as well as for the preferred combination of variation of these parameters reported in Chapter 3 (right panel, section 3.4). These histograms were derived from a short-term frequency analysis of the input and output signals of the WDRC stage using a 20-ms long rectangular window and third-octave frequency bands. The analysis was performed only for the signals at the left ear. The input signal was level-normalized such that its RMS level reflected the average presentation level in dB SPL across all subjects (i.e., 67.4 dB SPL). The reference insertion gains and CRs variations applied by the master HA were derived from NAL-NL2 prescription for the average audiogram of the group of subjects fitted with a 2 mm vent (i.e., a gently sloping hearing loss with 4FAHL = 49.5 dB HL, see Chapter 3, Figure 3.1).

The output signal was recorded at the output of the WDRC stage prior to the filtering for the *free-field-to-receiver* mapping. The solid lines refer to the median values across third-octave bands, the light gray areas to the corresponding 5% to 95% quantiles, and the dark gray areas to the 25% to 75% quantiles. The solid lines with circles refer to the long-term RMS levels. In order to improve the readability of the plot the x-axis is limited to the master HA bandwidth (for a 2 mm vent). In this example, the most prevalent difference between the input and both output signals is the reduction in dynamic range observed at high frequencies. Also, it is worth noting the relatively similar spectral balance between the input (left panel) and output signal for the preferred combination of CRs and gains (right panel). This differ from a standard HA amplification output (middle panel) which emphasizes on the high-frequencies of the input signals when fitted to such a hearing profile.

3.A.3 Additional information

Appendix C. List of individual preferred pieces of music.

Appendix E. Graphical user interface of the listening experiment.

Chapter 4

Discussion, implications, limitations and future directions

4.1 Summary of major findings

The current thesis investigated the use of hearing aids (HAs) in the context of music listening. The main motivation for this topic was the observation that HAs are primarily designed to facilitate speech communication and the lack of consideration for the large differences that exist between speech and music signals. Previous research has suggested that a significant proportion of the HA user population is dissatisfied with their HAs when listening to music. This is possibly due to the problems that users are facing when using their HA for music listening. In particular, they experience difficulties in understanding lyrics, problems with distortion and poor sound quality in specific complex environments such as live music. However, out of these problems faced by HA users, it is hard to determine a link of causality for their low satisfaction. Previous studies that have manipulated signal processing and fitting in HAs have primarily focused on manipulating characteristics of compression. This may be due to the high tailoring of compression systems in HAs for the characteristics of speech signals. The two studies presented in this thesis sought:

1. To better understand the musical listening habits of HA users.
2. To identify the problems faced by hearing-impaired listeners when listening to music with their HAs.

3. To better understand what are the factors that disturb the enjoyment of music amplified by HAs.

4. To identify the HA signal processing strategies preferred by hearing-impaired listeners when listening to music.

5. To develop HA signal processing recommendations specifically for music.

From an extensive literature review (see **Chapter 1**), it has been concluded that conducting a survey on the use of HAs while listening to music was necessary to better address the first three objectives mentioned above. Data collected from 151 HA users who answered to this online questionnaire was presented in **Chapter 2**. The questionnaire included 42 multiple-choice and open-ended questions covering general information on the respondents, their interest in music, their use of HAs to listen to music and in which listening environment, their music preferences, their musical experience and practice, the problems they experienced while listening to music and while playing to music with their HAs and their satisfaction when listening to music with their HAs.

The responses of the participants indicated that HA users were most frequently listening to recorded music at home and generally used the default program of their HAs. Only 19% of the respondents reported that their HAs had a music program, 61% of the respondents reported that their HAs didn't have one and 20% respondents reported that they didn't know. They were most satisfied with the performance of their HAs in scenarios with recorded music, where they had control over the music presentation, such as "at home" where they can adjust the loudness of the music being played.

However, 30% of the respondents were dissatisfied with the performance of their HAs with music and reported several problems that may be responsible for their low satisfaction. The most frequent problems identified were difficulties in understanding lyrics, the soft passages of music being too soft, lack of clarity of the music, poor tonal quality, the music being too sharp/shrill and the music being too loud. In particular, most participants reported that HAs were helpful for hearing the soft passages (65%) while making the music also too loud (51%). This suggests that most

HAs provide enough amplification at low levels to allow soft sounds to be audible but may not provide adequate gain compression, thus making the higher sound levels uncomfortably loud.

To identify the problems that drove most of the HA satisfaction with music, a principal component analysis followed with a multiple regression were conducted. It revealed that satisfaction was mainly influenced by the frequency of the problems related to the 'Sound quality' provided by the HAs and, to a lesser extent, by problems associated with 'Identification' of music instruments and lyrics as well as problems associated with 'Clarity' (or 'Definition') of the music. Some of the problems related to 'Sound quality' were found to be associated with the hearing loss of the respondents. The degree of hearing loss at low-frequencies was associated with the lack of clarity of the music, distortions and the music being too sharp/shrill ($\tau = 0.24$, 0.24 and 0.20 respectively). A small association was also found between the slope of the HL and distortions as well as the music being too sharp/shrill ($\tau = -0.19$, 0.24 respectively). This suggests that those with the most sloping audiograms were most likely to complain about distortions and the music being too sharp/shrill. Based on these findings, it was concluded that standard amplification prescription schemes could be optimized specifically to improve the enjoyment of music with HAs.

To address objectives 4 and 5, a controlled listening experiment was then designed based on the outcome of the survey, which is reported in **Chapter 3**. The experiment involved the simulation of the most frequent listening scenario reported from the survey respondents: listening to music at home with HAs, using the HA default program. This was implemented with the acoustic simulation of music recordings being played by stereo loudspeakers in an existing living room. The simulation of the HA involved modifications of the National Acoustic Laboratories (NAL) master HA so that it incorporated the simulation of three different vent sizes of semi-open fittings and the ability to vary the frequency-specific gains and compression ratios relative to NAL-NL2 prescription formulae in real-time. Data was collected from 26 adult bilateral HA users, with symmetrical sensorineural mild to moderately severe hearing loss.

To replicate the "at home" scenario, the participants were asked to adjust the volume at the beginning of the experiment and an excerpt of one of their preferred pieces of music was included in the stimuli along with twelve diverse famous Classical and Jazz & Blues music excerpts (i.e., the genres mostly listened to by the survey participants). The results showed that participants consistently preferred amplification with an increased gain at low-frequency and reduced gain at high-frequency relative to those prescribed by NAL-NL2. Group differences analysis revealed that participants fitted with the most open vents (i.e., those who had near-normal hearing at low-frequency) preferred less amount of low-frequency emphasis than participants who were fitted with smaller vents. Also, higher compression ratios (CRs) relative to NAL-NL2 were preferred. The extent to how much more compression was preferred depended on the sound levels at the input of the HAs, with higher CRs preferred for softer music pieces. Additionally, the selected input levels and preferred compression highlighted the difference in preferred listening levels between females and males, with females preferring music stimuli being 4 dB softer than males on average. This resulted in females preferring slightly less compression than males.

Also, participants were asked to provide satisfaction ratings of the sound quality of the music stimuli processed with their preferred combination of parameters in regard with attributes derived from the problems associated with 'Sound quality' in the survey, i.e., "Clarity", "Tonal quality", "Sharpness" and "Distortion". The median score of 'Satisfied' obtained for all four attributes suggests that such variations of the frequency-specific gains and compression relative to NAL-NL2 can alleviate some of the problems highlighted by the survey and improve the HA sound quality, at least for instrumental music played at listening levels chosen by the HA user.

These findings have major implications for future research, design considerations with HA signal processing specifically for music listening as well as for clinicians who seek to adjust the settings of a music program for their patients. These will be discussed in the section below.

4.2 Limitations and implications

The survey presented in Chapter 2 gave insights on the music listening habits of the HA users and identified problems they encounter in these environments. However, the large number of questions was a limitation for the study as it increased considerably the duration to take the survey. Thus, none of the questions was made mandatory to proceed to a following question in order to avoid participants giving up the survey. Consequently, the number of answers collected varied across questions and only 131 respondents completed the survey and 20 answered at least half of the questions. Despite the high reliability of the demographic information provided in the research databases from which participants were drawn, no information was available concerning the type of HA fitting respondents were fitted with. The HA fitting has a great influence on the lower cut-off frequency of the HA output, but no conclusion could be drawn in regard with the influence of fitting on HA satisfaction from the survey.

Consequently, a simulation of semi-open vents was implemented in the following experiment which revealed differences in insertion gain preferences for participants fitted with different vent sizes (see Chapter 3). Overall, preferred insertion gains were obtained with emphasis on low-frequency relative to NAL-NL2. However, it is important to highlight that only instrumental music stimuli were included to assess those preferences. For instance, amplification of low-frequency could reduce the intelligibility of the lyrics for music containing vocals. Bonnel et al. (2001) suggests that the attention of a listener is divided between listening to lyrics and paying attention to the melody when listening to music. And, for some music, understanding lyrics is an important aspect of the listening experience, while in other cases, some listeners may regard vocals as any another instrument or may not pay attention to lyrics at all. Thus, preferred gains might differ from the ones identified in Chapter 3 to preserve the intelligibility of the lyrics.

Also, increased CRs relative to NAL-NL2 were preferred in the context of the music being played at the chosen listening levels (CLLs) of the participants. The input levels of individual stimuli were found to be negatively correlated with the preferred amount of compression averaged across participants. This suggests that for higher input signals, lower CRs than prescribed by NAL-NL2 could be preferred. It would be interesting to investigate preferred amount of compression with a wider range of input levels to confirm the trend towards preferences for less compression relative to NAL-NL2. Such an investigation would have implications, not only for scenarios where the HA user does not have control over the music, but also for live music. Additionally, findings from Croghan et al. (2016) suggest that CLLs may be dependant on the genre of music being played. The study presented in Chapter 3 used only one set of CTs and dual time constants. Investigating the interaction between CRs with other compression characteristics, and the genre of music would make an interesting follow-up study.

The listening experiment presented in Chapter 3 did not revealed any differences in preferred amplification between the two different genres, whereas previous research found such differences for different music genres. For example, Croghan et al. (2012) and Croghan et al. (2014) found different preferences for rock and for classical music. However, these studies included only one stimulus per genre. Thus, the differences observed by those studies may be due to differences in acoustical properties of the stimuli they employed rather than due to the genre itself. It would be interesting to investigate preferred parameters with a broader range of genres.

Finally, the findings from the study presented in Chapter 3 and the suggested variations of compression and insertion gains are based on an experiment involving a HA simulation. Despite the highly realistic simulation employed, the preferred compression and insertion gains should be implemented on commercially available HAs in order to provide results with even higher ecological validity. Moreover, the listening experiment applied two parameters (amount of compression and spectral slope) to control the complex, frequency dependent characteristics of the gain provided by the master HA. Even though these two parameters may be the most relevant parameters, they also placed some constraints on the gain manipulations that were possible. Hence, future research may allow even more user control to provide more frequency specific gain manipulations. However, this will then come with the expense of an increased testing time, and it will need to be shown if participants are actually able to reliability perform such more complicated signal manipulations. Moreover, future research should also consider upcoming HA technologies that allow significant low-frequency gain with open fittings, such as provided by Earlens^{TM} or large vented transducers as applied by some "Hearables".

The preferred variations of parameters prescribed by NAL-NL2 found in Chapter 3 and in previous research could form early recommendations towards the development of a music specific fitting scheme. Although programming successfully a music program for HA may only benefit a minority of HA users. The present survey and other research (e.g., Madsen and Moore, 2014b; Greasley et al., 2015) have shown that significant proportion of HA users do not have a music program or have a minimal interaction with their HA when listening to music. This raises the question on how to implement variations of amplification to improve the sound quality of music in HA and to make it available to a maximum number of HA users. Modern HAs now have sufficient processing power to not only recognise when music is present, but also to analyse its acoustical characteristics and to adjust the HA signal processing accordingly. Thus, if one was trying to improve music listening with HAs, a music program could be implemented such that the HAs would switch automatically from the standard prescription in the presence of music as well as at the user demand.

In summary, recommendations for HA amplification specifically for music should consider a more spectrally-balanced gain with an increased gain at low-frequencies, reduced gain at high-frequencies and increased CRs relative to those prescribed by NAL-NL2. Chapter 1 provided an overview of the potential factors influencing music enjoyment with HAs divided into four categories. When implementing the recommendations mentioned above, some of these factors need to be kept in mind as they may impact the optimal HA amplification to apply. Table 4.1 summarizes the factors that potentially could represent a fitting parameter in a music program.

4.3 Conclusion

Standard prescription formulae provide insertion gain and compression recommendations to apply in HAs and are primarily aiming at improving the clarity and comfort of speech. The work presented in this thesis contributed to the limited research that investigated the use of HAs in the context of music listening comparatively

TABLE 4.1: List of retained factors to take in consideration when manipulating HA amplification specifically for music listening.

Acoustics	HA fitting	Hearing Loss (HL)	Individual factors
- **Input levels:** increased CRs relative to NAL-NL2 at CLLs and low input levels and possibly decreased CRs at high input levels - **Genre of music** - **Presence of lyrics**	- **Vent size:** less emphasis on low-frequency for most open vents	- **Degree and slope of HL:** different CRs and insertion gains to apply - **Low-frequency HL:** type of fitting, different CRs and insertion gains to apply	- **Gender:** reduced CRs and insertion gains across all bands for females - **Interest in lyrics**

to speech understanding or communication. This thesis has provided knowledge about the music listening habits of HA users, identified some of the most prevalent problems they face when listening to music with their HAs and demonstrated that standard amplification prescription schemes in HAs should be modified to specifically optimize the sound quality of music. The methods described here were able to highlight amplification recommendations specifically for music that needs to be further developed and evaluated in commercially available HAs.

Appendix A

Overview of the full questionnaire

TABLE A.1

Detailed overview of the questions included in the survey (see Chapter 2). The section titles are indicated in bold. The question numbers are shown on the left of each question. The open-ended questions are indicated with a row filled in grey. Some questions are shown with an indentation indicating their dependency on previous questions as the questionnaire adaptive, e.g., in section 3, participants who answered positively to Question 12 were then asked additional questions about their musical practice (Questions 13 to 20).
The list of questions from sections 1, 2 and 3 is shown on page 92 and the corresponding answer options on page 93. The list of questions from sections 4 and 5 is shown on page 94 and the corresponding answer options on page 95. The list of questions from sections 6 and 7 as well as the corresponding answer options are shown on page 96.

List of questions	
Section (1) "General section"	
	First name
	Family name
	Date of birth (DD/MM/YYYY)
1	Currently, how many hours a day do you use your hearing aid(s)?
2	When did you receive your most recent hearing aid(s)?
Section (2) "Interest in music, listening conditions and preferences"	
3	How important is music to you?
4	Currently, how many hours a DAY do you listen to music on average?
5	Currently, how many hours a WEEK do you listen to music in each scenario? ***Repeated for the following items:*** At home; Watching TV; In the car; Live music; Private performance; Background music; Other ;
6	What genre(s) of music do you mostly listen to? Tick all that apply.
7	Prior to having a hearing loss, how many hours a DAY did you listen to music?
8	Prior to having a hearing loss, how many hours a WEEK did you listen to music in each scenario? ***Repeated for the following items:*** At home; Watching TV; In the car; Live music; Private performance; Background music; Other ***(if answered at Q-5)***;
9	What genre(s) of music did you mostly listen to? Tick all that apply.
10	If your music listening habits changed from before you had hearing loss, could you provide some explanation/comment?
11	Is there anything else you would like to tell us about your musical listening habits, present or past?
Section (3) "Musical experience and practice"	
12	Are you a musician (singer included) or have you EVER regularly practised music?
13	Please select the forms of musical education you have had.
14	How old were you when you started your musical training?
15	Please list which musical instrument(s) you play (including singing).
16	Please list the extent of your practice (e.g. playing in a band/ensemble, singing a choir, performing at gigs etc.).
17	Please list the extent of your musical practice other than practising a musical instrument, if applicable (e.g. recording, mixing, music teaching, composition, direction of an ensemble etc.).
18	Before you had hearing loss, how many hours a week did you practice?
19	If you currently play a musical instrument, how many hours a week do you practice?
20	Are you a professional musician?

Answer options	**Section (1) "General section"**	
	1	Never; Less than 1 hour per day; 1-2 hours per day; 2-4 hours per day; 4-6 hours per day; More than 6 hours per day;
	Section (2) "Interest in music, listening conditions and preferences"	
	3	Not at all; Slightly important; Somewhat important; Very important; Extremely important;
	4	Never; Less than 1 hour per day; 1-2 hours per day; 2-4 hours per day; 4-6 hours per day; More than 6 hours per day;
	5	Never; Less than 1 hour per week; 1-2 hours per week; 2-4 hours per week; 4-6 hours per week; More than 6 hours per week;
	6	African; Asian; Blues and Jazz; Classical; Country and Folk; Electronic; Funk and Hip Hop; Hard Rock and Metal; Latin; Pop and Rock; R&B and Soul; Other (Please specify) ;
	7	NA/ I don't remember; Never; Less than 1 hour per day; 1-2 hours per day; 2-4 hours per day; 4-6 hours per day; More than 6 hours per day;
	8	NA/ I don't remember; Never; Less than 1 hour per week; 1-2 hours per week; 2-4 hours per week; 4-6 hours per week; More than 6 hours per week;
	9	African; Asian; Blues and Jazz; Classical; Country and Folk; Electronic; Funk and Hip Hop; Hard Rock and Metal; Latin; Pop and Rock; R&B and Soul; Other ***(if answered at Q-6)***;
	Section (3) "Musical experience and practice"	
	12	Yes; No;
	13	Private lessons; Group lessons; Self-tuition; Parent tuition; Music theory lessons; Music lessons at school; Other (Please specify) ;
	18	NA/ I don't remember; Never; Less than 1 hour per week; 1-2 hours per week; 2-4 hours per week; 4-6 hours per week; More than 6 hours per week;
	19	NA; Never; Less than 1 hour per week; 1-2 hours per week; 2-4 hours per week; 4-6 hours per week; More than 6 hours per week;
	20	Yes; No;

List of questions	
Section (4) "Use of hearing aids while listening to the music"	
21	How often do you listen to music in the following conditions? ***Repeated for the following items:*** I use the normal listening setting of my hearing aid(s); I change the listening setting of my hearing aid(s) (e.g. music listening setting); I have to adjust the volume of my hearing aid(s); I use an assistive listening device if available (e.g. FM systems, audio loops, Bluetooth); I use different hearing aid(s) for music listening than the one(s) I have for general use; I remove my hearing aid(s); I use another device than my hearing aid(s) (e.g. headphones); Other;
22	In which scenario(s) do you listen to music in the following conditions? Note that you can provide multiple answers for each statement. ***Repeated only for items answered "Rarely", "Sometimes", "Often" or "Always" in Q-21***
23	Does your hearing aid(s) have a music program?
24	How often do you use your music program?
25	Does music sound better or worse with your music program than with the main hearing aid program?
26	What difference do you notice between your music program and the normal hearing aid program? Please provide some explanation / comment.
27	Would you be interested to pay for a music program in your hearing aid(s) to improve significantly your enjoyment of music with your hearing aid(s)?
28	Could you tell us what you think you would need to improve your enjoyment of music with your hearing aid(s)?
29	Is there a particular reason you are not interested to pay for a music program?
Section (5) "Experienced problems while listening to music with hearing aids"	
30	How often do you encounter each problem while listening music with your hearing aid(s)? ***Repeated for the following items:*** The music is too loud; The soft passages are difficult to hear; It is hard to understand the words in lyrics; It is hard to perceive the rhythm; It is hard to identify a familiar song; It is hard to follow the melody; It is hard to identify the musical instrument(s) being played; It is hard to focus on a specific instrument; The music seems distorted; The tonal quality of the music is poor; The music seems lacking in bass; The music has too much bass; The music is too sharp or shrill; The music is not clear enough; A whistling sound occurs (Acoustic feedback); Other;
31	How far does each problem interfere with your enjoyment of music while listening with your hearing aid(s)? ***Repeated only for items answered "Rarely", "Sometimes", "Often" or "Always" in Q-30***
32	In which listening scenario(s) did you experience each problem while listening music with your hearing aid(s)? ***Repeated only for items answered "Rarely", "Sometimes", "Often" or "Always" in Q-30***
33	For each problem, is it worse or better with your hearing aid(s)? ***Repeated only for items answered "Rarely", "Sometimes", "Often" or "Always" in Q-30***
34	What else would you like to tell us about the issues you have faced while listening to music? Please leave a comment.

Answer options	**Section (4) "Use of hearing aids while listening to the music"**	
	21	NA/ I dont know; Never; Rarely; Sometimes; Often; Always;
	22	At home; Watching TV; In the car; Live music; Private performance; Background music; ***Other (if answered at Q-5)***;
	23	Yes; No; I don't know;
	24	Never; Rarely; Sometimes; Often; Always; I don't know;
	25	Much worse; Worse; Slightly worse; Neutral / I can't hear any differences; Slightly better; Better; Much better;
	27	Yes; No;
	29	I doubt it will make significant changes; It would probably be too expensive; I am not interested in any hearing aid program; I don't use my hearing aids to listen to music;
	Section (5) "Experienced problems while listening to music with hearing aids"	
	30	Never; Rarely; Sometimes; Often; Always;
	31	Not at all; Slightly; Somewhat; Very; It makes music impossible to enjoy;
	32	At home; Watching TV; In the car; Live music; Private performance; Background music; Other (if answered at Q-5);
	33	Much worse; Worse; Slightly worse; Neutral / I can't hear any differences; Slightly better; Better; Much better;

List of questions		
	Section (6) "Experienced problems while playing music"	
	35	While playing music or singing, is your hearing aid(s) helpful?
	36	How often do you encounter each problem while practising music? ***Repeated for the following items:*** The sound of the instrument I am playing/my own voice is annoying me; It is hard to play/sing in tune; Other;
	37	How far does each problem interfere with your ability to play music using your hearing aid(s)? ***Repeated only for items answered "Rarely", "Sometimes", "Often" or "Always" in Q-36***
	38	For each problem, is it worse or better with your hearing aid(s)? ***Repeated only for items answered "Rarely", "Sometimes", "Often" or "Always" in Q-36***
	39	What else would you like to tell us about the issues you have faced while practising music? Please leave a comment.
	Section (7) "Music listening with hearing aids satisfaction"	
	40	Generally, how satisfied are you with the performance of your hearing aid(s) while listening to music?
	41	Please rate how satisfied you are with the performance of your hearing aid(s) while listening to music in different scenarios. ***Repeated for the following items:*** At home; Watching TV; In the car; Live music; Private performance; Background music; Other ***(if answered at Q-5)***;
	42	How satisfied are you with the performance of your hearing aid(s) while playing music/singing?

Answer options		
	Section (6) "Experienced problems while playing music"	
	35	Not at all; Slightly helpful; Helpful; Very helpful; I cannot play/sing without it;
	36	Never; Rarely; Sometimes; Often; Always;
	37	Not at all; Slightly; Somewhat; Very; It makes music impossible to play;
	38	Much worse; Worse; Slightly worse; Neutral / I can't hear any differences; Slightly better; Better; Much better;
	Section (7) "Music listening with hearing aids satisfaction"	
	40	Very dissatisfied; Dissatisfied; Slightly dissatisfied; Neutral; Slightly satisfied; Satisfied; Very satisfied;
	41	Very dissatisfied; Dissatisfied; Slightly dissatisfied; Neutral; Slightly satisfied; Satisfied; Very satisfied;
	42	Very dissatisfied; Dissatisfied; Slightly dissatisfied; Neutral; Slightly satisfied; Satisfied; Very satisfied;

Appendix B

Supplementary questions omitted in Chapter 2

In this section, answers for some of the questions omitted in Chapter 2 are provided.

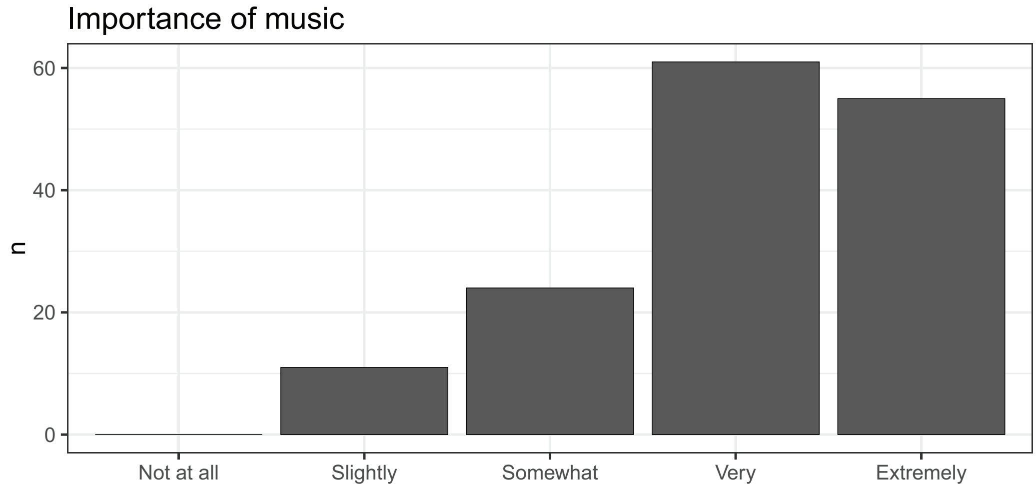

FIGURE B.1: Answers for Question 3 (section 2). "How important is music to you?"

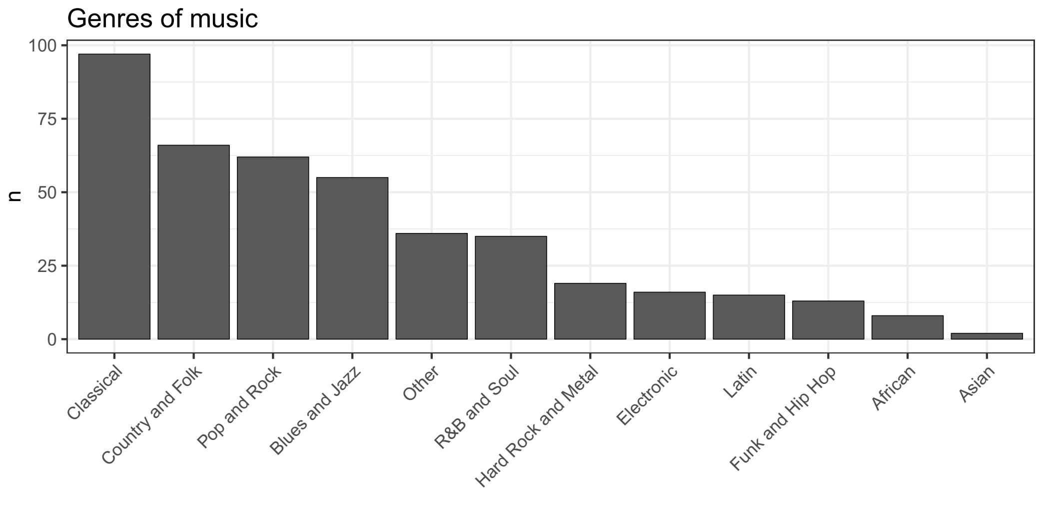

FIGURE B.2: Answers for Question 6 (section 2). "What genre(s) of music do you mostly listen to? *Tick all that apply.*"

The most popular genres participants were listening to were "Classical music" (64%), "Country and Folk" (44%), "Pop and Rock" (41%) and "Blues and Jazz (36%). The 36 participants (24%) who answered "Other" were asked to specify the other genre(s) they were listening to in an open-ended question. Many of these answers were sub-genres of the above mentioned categories (e.g., "opera", "rockabilly", "progressive rock", "swing") while some answers were sub-genres of religious music and/or choir ensembles (e.g., "gospel", "church", "Hebrew songs", "Christian songs", "1950-1970 popular"), some answers comprised military music (e.g., "military brass", "hymns") and a few answers could not be classified as any specific genre (e.g., "early music") or simply an example of a given artist (e.g "Gilbert & Sullivan").

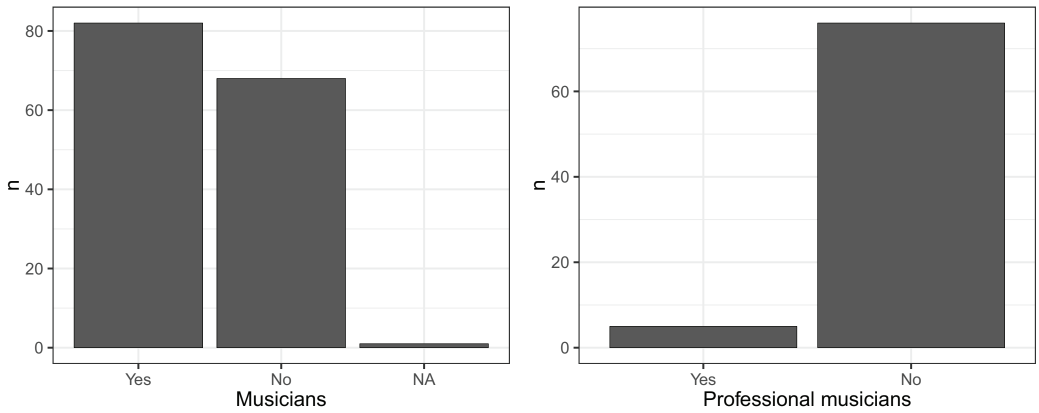

FIGURE B.3: Answers for Question 12 and 20 (section 3). ***Question 12:*** (left panel) "Are you a musician (singer included) or have you EVER regularly practised music?" ***Question 20:*** (right panel) "Are you a professional musician?"
Out of the 82 participants (54%) who reported being musician, five participants reported being professional musicians (6%).

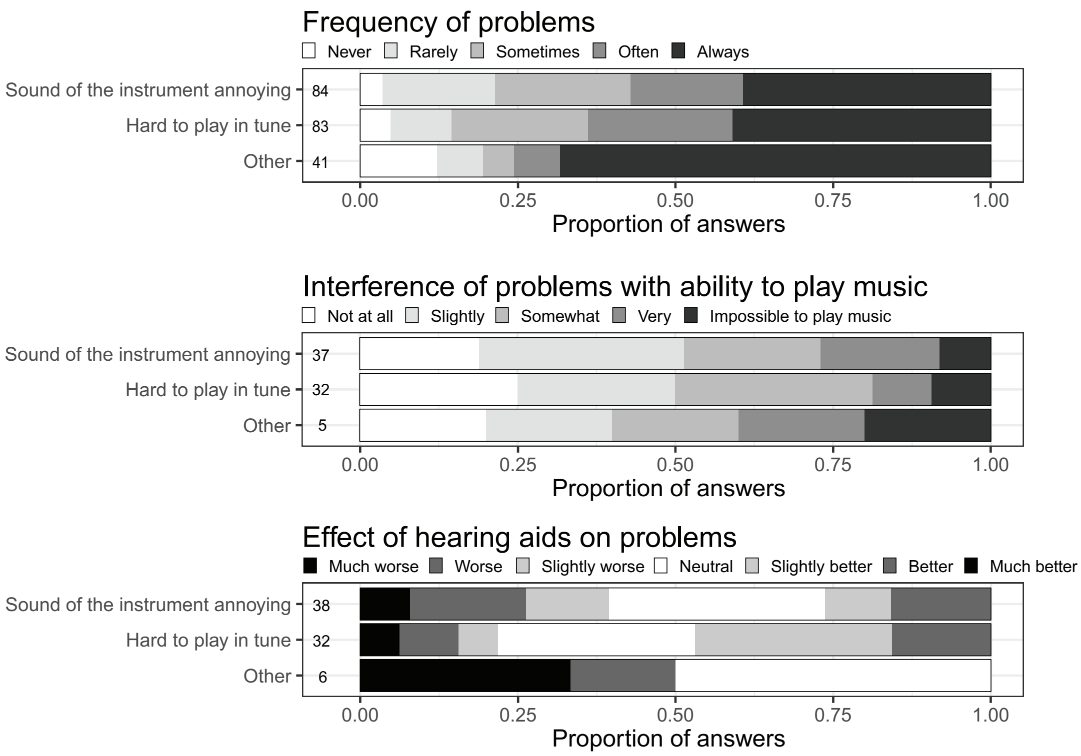

FIGURE B.4: Proportion of answers for Questions 36, 37 and 38 (section 6). In each panel, the total number of answers collected for each problem is shown on the left. ***Question 36:*** (upper panel) "How often do you encounter each problem while practising music?" ***Question 37:*** (middle panel) "How far does each problem interfere with your ability to play music using your hearing aid(s)?" ***Question 38:*** (lower panel) "For each problem, is it worse or better with your hearing aid(s)?"

Appendix C

Additional information on the music material used in the listening experiment

TABLE C.1: Bit rates in Kilobits per second (kbps) of the music material downloaded as 44.1 kHz 32-bit (floating point) *.m4a* files.

Music piece	Bit rate (kbps)
Count Basie, All of Me	276
Art Blakey, Moanin'	296
The **Dave Brubeck** Quarter, Take Five	291
Miles Davis, So What	298
Paul Desmond, Autumn Leaves	261
BB King, The Thrill is Gone	274
Beethoven, Symphony No. 5 in C Minor, Op. 6	261
Vivaldi, The Four Seasons, Op. 8, Spring	271
Barber, Adagio for Strings	271
Chopin, Nocturne No. 2 in E-Flat Major, O	261
Mozart, The Magic Flute, K. 620, Overture	265
Dvořák, Symphony No. 9 in E Minor, Op. 95	262

TABLE C.2: List of preferred pieces of music, downloaded as 44.1 kHz 32-bit (floating point) *.m4a* files.
Note: Out of the 26 participants, four of them either nominated the same preferred piece of music as a participant that had participated before or did not nominate a preferred piece of music and were provided randomly with one of the stimuli in this list during the preferred piece of music trial.

Genre	Music piece	Bit rate (kbps)
Classical	**Bach**, Notebook for Anna Magdalena, Minuet G minor, BWV Anh. 115	259
Classical	**Beethoven**, Piano Sonata No. 14 in C-sharp minor	268
Classical	**Beethoven**, Piano sonata No. 23 in F minor, Op. 57 "Appassionata": I. Allegro assai	262
Classical	**Beethoven**, Violin concerto in D major, Op.61: I. Allegro ma non troppo	253
Classical	**Bizet**, Carmen: Marche du toréador	281
Classical	**Charles Williams**, The dream of Olwen	282
Classical	**Debussy**, Children's corner, L. 119a: VI: Golliwog's cakewalk	259
Classical	**Debussy**, Suite Bergamasque: III. Clair de lune	258
Classical	**Elgar**, Pomp and Circumstance, Op. 39: No.1, March in D major	270
Classical	**Elgar**, Variations on an original theme, Op. 36, Enigma: IX. Nimrod (adagio)	261
Classical	**Gilbert & Sullivan**, The Pirates of Penzance, Act I: "Overture"	263
Classical	**Mendelssohn**, Athalie Op. 74, War march of the priests	274
Classical	**Paganini**, Violin Concerto No. 1 in D Major, Op. 6: I. Allegro maestoso	262
Classical	**Saint-Saëns**, Symphony No. 3 in C minor, Op. 78 "Organ": I. Adagio	261
Classical	**Schubert**, Adagio in E-flat major, D.897: "Notturno"	260
Classical	**Schumann**, Kinderszenen, Op. 15: VII. Träumerei	292
Classical	**Tchaikovsky**, 1812 Overture, Op 49., TH 49	261
Country & Folk	**Willie Nelson**, On the road again	272
Jazz & Blues	**The Dave Brubeck Quartet**, Flamingo (live)	303
Jazz & Blues	**Louis Armstrong**, Mack the knife	270
Pop & Rock	**Crowded House**, Weather with you	278
Pop & Rock	**Ray Conniff**, Somewhere my love	262

Appendix D

Instructions given to participants during the listening experiment

The aim of this experiment is to find the best setting for listening to music via a simulated hearing aid.

During the experiment, you will listen through headphones to some musical excerpts on a loop (that is, continually repeated). While a musical excerpt is looped you will be asked to listen to it with different settings by clicking on different squares on the screen. Each square is a different setting. You will have to compare these settings with a reference (in the middle of the screen) to find the best setting. For every setting, a slider on the right side of the screen will allow you to mark a setting as sounding better or worse than the reference setting. You might be able to find the best square without having to listen to all of them.

Once you have found the best sounding setting, click on the 'Next' button. You will then be asked to provide feedback on some attributes of the sounds that guided your decision in the 1^{st} task.

After clicking on the 'Next' button again, you will be asked to provide feedback on how satisfied you are with the best setting.

These tasks will be repeated for different musical excerpts.

Appendix E

Graphical user interface of the listening experiment

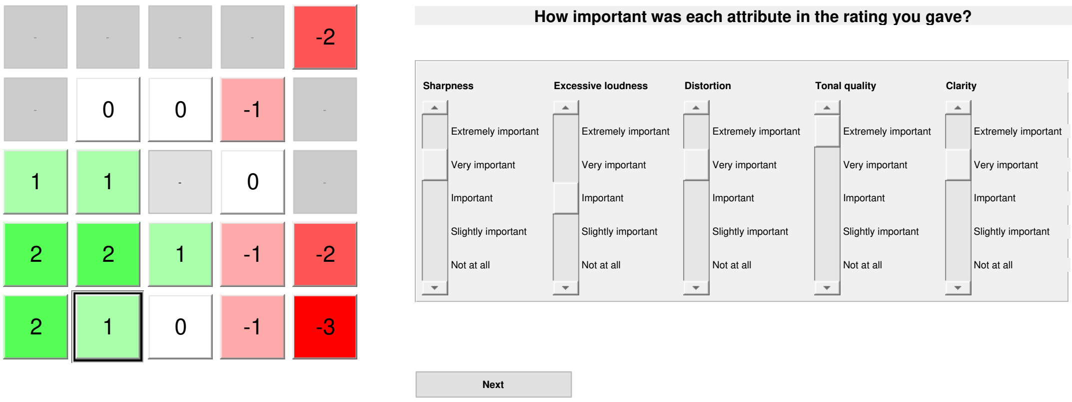

FIGURE E.1: Graphical user interface used in the experiment to collect importance ratings for the five sound quality attributes.

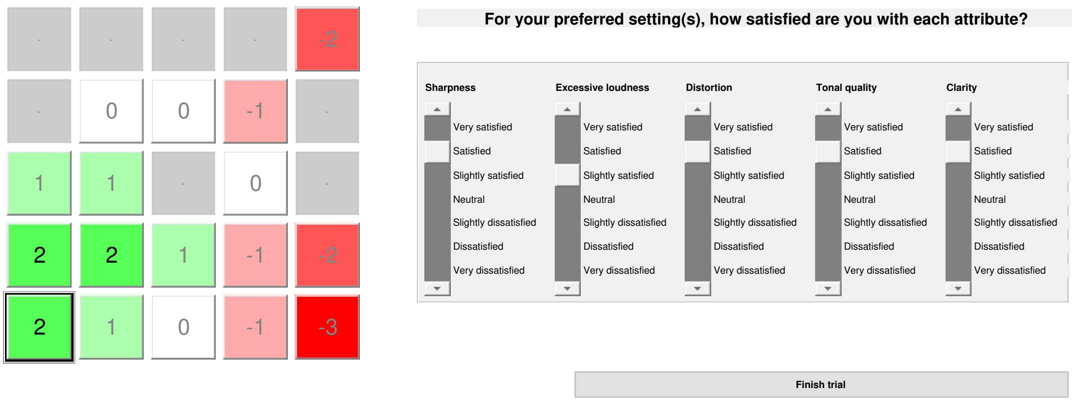

FIGURE E.2: Graphical user interface used in the experiment to collect satisfaction ratings for the preferred combination(s) of compression ratio and spectral tilt variations for the five sound quality attributes.

Appendix F

Human Research Ethics Committee Letters

Appendix F of this thesis has been removed as it may contain sensitive/confidential content

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