Bachelor Thesis

Mitigation of comb filter effects by in-situ amplitude-phase measurements and gain table manipulation with a mobile hearing aid prototype

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Abstract

Modern hearing aids are optimized for processing speech, but when dealing with more tonal non-speech sources like, e.g., music, the spectral perception can be altered due to comb filtering. Depending on the internal time delay and the level differences of direct and processed sound, certain frequencies may be amplified by up to 6 dB or alternatively cancel out completely, which may also be different for individual listeners. To avoid comb-filter effects with subject-specific fittings, the actual amplitude-phase-relation at the eardrum of the listener has to be determined. An in-situ psychoacoustic measurement with a mobile hearing aid prototype is conducted in which the listener's task is to find the amplitude and phase values to cancel out the direct sound at different frequencies. From the results, which show a significant conformity for the phase across all listeners and a slightly less pronounced conformity for the amplitude, a manipulation of the gain table is proposed to mitigate the effect of the comb filter. This approach could improve music perception for listeners or musicians with mildly impaired hearing. By aiming to preserve the natural sound, e.g., of musical instruments, it might also find application in a socalled "smart hearing protection" where direct sound will be a central issue.

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1 Introduction

Modern hearing aids are optimized for processing speech, with the aim to improve the speech recognition performance of their users. But when dealing with more tonal non-speech sources such as, e.g., music, the perception can be altered in an undesirable manner.

With a hearing aid the direct sound is recorded by a microphone and played back by the receiver inside the ear canal. Also the direct sound is attenuated by the head, ear, and hearing device itself that can be perveiced by the listener. Especially for mild hearing losses, i.e., with comparatively low amplification, the processed signal interferes with the direct sound, which leads to the effect of comb-filtering. Depending on the relative time delay and the level differences of direct and processed sound, certain frequencies may be amplified by up to 6 dB or alternatively cancel out completely. The frequency dependent phase shift may be individual, and hence, the actual comb filter effect may be perceived differently by listeners. Furthermore, even small amplifications around $\pm$ 1 dB might be audible due to comb-filtering [1].

To avoid comb filter effects with subject-specific fittings, the relative amplitude and phase relation of both signal paths at the eardrum of the listener has to be determined to determine critical amplitude-phase values which would lead to a cancellation of the sound. Because the effect of perfect compensation, i.e., no energy at the eardrum, for pure tones can be perceived well, a psychoacoustic matching experiment is used to determine the relative amplification and delay that is required to compensate the direct sound.

Hence, an in-situ psychoacoustic measurement is performed in which the listener's task is to find the relative amplitude and phase values for an anti-phase signal to cancel out the direct sound at different frequencies. The method is suitable to be performed with a mobile hearing aid prototype [2] which is used to manipulate the signal using an FIR-filter in such a way, that the direct pure tone sound from a loudspeaker is cancelled by the output of the hearing aid prototype at the ear drum.

From the outcome, the frequency-dependent amplitude-phase combinations which would lead to cancellation are determined. A manipulation of the gain table of the device is used as a compensation method. More specifically, muting the output of the hearing device if an addition of the processed signal would, due to expected cancellation, result in a sound level worse than the direct sound alone.

Besides compensation for hearing devices this approach could also be benificial for a so-called smart hearing protection. There already exist several papers that are concerned with smart hearing protection for musicians. Bernier and Voix (2013) [3] use digital negative feedback to mitigate the occlusion effect and try to achieve adjustable attenuation with natural timbre. Albrecht, Jaatinen, and Lokki (2017) [4] use an analog negative feedback loop while also monitoring the instrument to achieve natural timbre. Besides the occlusion effect beeing a predominant source of distortion, comb-filter effects could lead to a complete cancellation of the incoming sound making it a critical aspect in the development of smart hearing protection for musicians.

2 Theoretical Background

When using a hearing device certain problems with acoustical phenomenons occur, which can have a major effect on the perception of speech, music, and sound in general. In this section, the fundamental principles of the underlying acoustic scenario when using a hearing device are explained and potential influences on the perception of sound are presented. This includes the acoustical paths of a hearing aid, superposition and interference of sound waves, and resulting comb filter effects.

2.1 Acoustic scenario of a hearing aid

The primary function of a hearing aid is to amplify the incoming sound to make it audible for the hearing impaired listener. In cases of severe hearing loss the amplification needs to be sufficiently high. When dealing with low to moderate hearing losses, amplification may be close to the actual direct sound. The direct sound and the processed sound by the hearing device are the two major components which have to be taken into account in order to explain some of the acoustical effects at play.

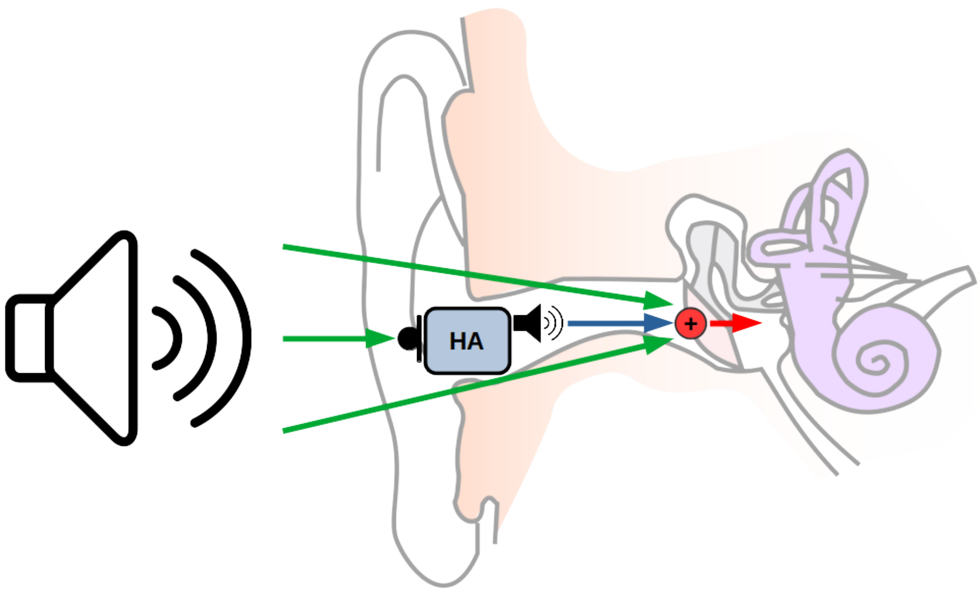

Figure 2.1: Acoustic scenario of a hearing aid. The direct sound (green arrows) reaches the microphone of the hearing aid and is also attenuated by the hearing aid and the anatomy of the ear. Both, the processed sound (blue arrow) and the attenuated direct sound are superimposed in the ear canal and reach the ear drum.

Consider the acoustic scenario in Figure 2.1. The hearing device, consisting of a microphone and a loudspeaker from a downgraded point of view, is situated at the entrance of the ear canal. Incoming sound is picked up by the microphone, processed and amplified, and played back by the loudspeaker. The sound then travels to the ear drum and is transduced from the middle to the inner ear. But not only the processed sound reaches the ear drum, also the direct sound component is picked up. It is attenuated by the anatomy of the ear and by the hearing device itself but nonetheless is still audible. Both sound waves, the direct component and the processed component, are superimposed in the ear canal and interfere with each other effectively at the ear drum. This can result in audible signal distortions as described in the following sections.

2.2 Wave superposition and interference

There are two different types of superposition and, hence, interference that can occur using a hearing device: Interference of two waves with same frequency and amplitude but different phase, and interference of two waves with same frequency but different amplitude and phase.

2.2.1 Phase dependent interference, same frequency and amplitude

With the use of the trigonometric addition theorem, the superposition of two waves with same frequency and amplitude but different phase can be computed. Consider two waves s_1 and s_2 with the same frequency ω and same amplitude a and the individual phase shifts φ_1 and φ_2:

$$s_1(t) = a \cdot \sin(\omega t + \varphi_1) \tag{2.1}$$

$$s_2(t) = a \cdot \sin(\omega t + \varphi_2) \tag{2.2}$$

The resulting superposition of these two waves is given by

$$s(t) = s_1(t) + s_2(t) = 2a\cos(\frac{\varphi_1 - \varphi_2}{2})\sin(\omega t + \frac{\varphi_1 + \varphi_2}{2}). \tag{2.3}$$

The resulting wave has the same frequency as before but the amplitude might change. It is dependent on the difference of the two initial phases. The resulting relative phase can be computed by the arithmetic mean of the initial phases. In the case of $\varphi_2 = \varphi_1$

or $\varphi_2 = n \cdot \varphi_1$, with $n \in \mathbb{N}$, the two waves interfere constructively and the resulting amplitude is doubled. In the case of $\varphi_2 = \varphi_1 + \pi$ or $\varphi_2 = n \cdot \varphi_1 + \pi$, with $n \in \mathbb{N}$, the two waves interfere destructively and completely cancel out.

2.2.2 Phase and amplitude dependent interference, same frequency

As a general case, one can consider the superposition of the two waves s_1 and s_2 with the same frequency ω but with different amplitudes a_1 and a_2 and the phases φ_1 and φ_2. With the use of complex notation and pointer arithmetic the superposition is found to be:

$$s_1 + s_2 = A \sin(\omega t + \varphi) \tag{2.4}$$

with amplitude

$$A = \sqrt{a_1^2 + a_2^2 + 2a_1 a_2 \cos(\varphi_1 - \varphi_2)} \tag{2.5}$$

and phase

$$\tan \varphi = \frac{a_1 \sin \varphi_1 + a_2 \sin \varphi_2}{a_1 \cos \varphi_1 + a_2 \cos \varphi_2}. \tag{2.6}$$

If $a_1 \neq a_2$ total destructive interference cannot be achieved because of the amplitude dependence for the resulting phase φ.

2.3 Comb-filter

The superposition of a sound source with its delayed and coherent counterpart is called comb-filtering. Depending on the frequency of both waves, which is the same, and the amplitude-phase relation between them, varying degrees of interference can be achieved. As described before total destructive or constructive intereference is achieved when the amplitude of both waves has the same magnitude and the phase shift between them is either a multiple of π or 2π, respectively.

Instead of looking at the amplitude of the resulting wave, the amplification level or comb filter gain G can be determined:

$$G = 20 \cdot log_{10}\left(\frac{A}{a_1}\right) \tag{2.7}$$

with combined amplitude A from equation 2.5 and initial amplitude a_1.

Figure 2.2 shows the resulting amplification level due to the comb filter depending on the level difference between direct and processed sound for different phase shifts. There are two critical cases for a level difference $\Delta L = 0$: At phase shifts of $\varphi = 0$ or $\varphi = 2n \cdot \pi$, for $n \in \mathbb{N}$, the two waves interfere constructively, resulting in an amplification of $+6$ dB or double the amplitude. For a phase shift of $\varphi = (2n-1) \cdot \pi$, for $n \in \mathbb{N}$, both waves interfere destructively, resulting in an amplification of $-\infty$ dB; thus, they cancel out completely.

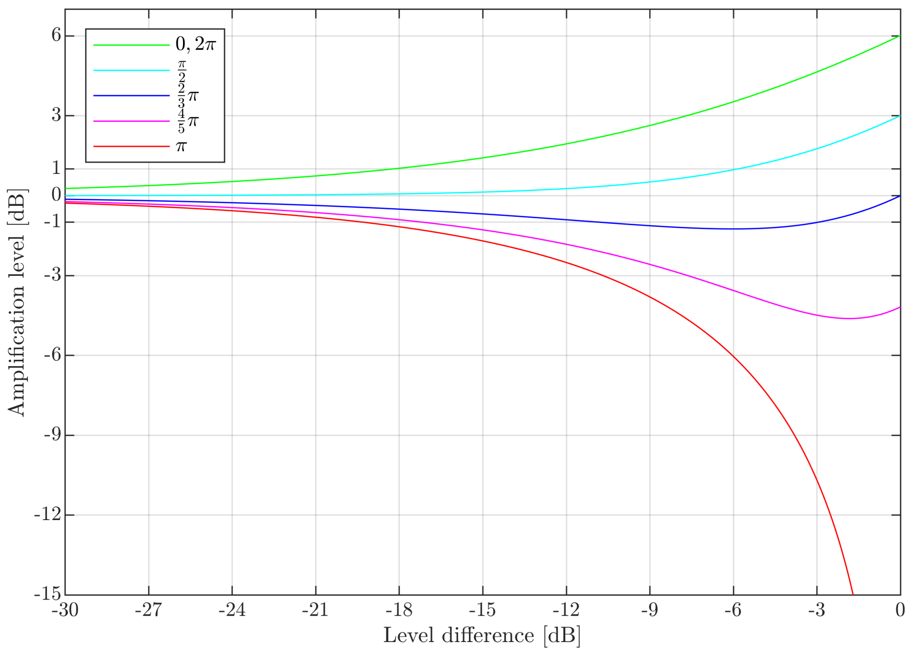

Figure 2.2: Amplification level dependent on the level difference between direct and processed sound for different phases. Extreme values for $\varphi = 0$ and $\varphi = \pi$ show an amplification of $+6$ dB and $-\infty$ dB, hence, total constructive and destructive interference.

2.4 Head-Related Transfer Function

Since individual head and ear shapes differ for people the perception of sound is different. This includes the actual amplitude and phase of the sound as well as resulting effects such as, e.g., occlusion or comb filtering. For the relation between the sound source and the sound that reaches the eardrum the so-called *head-related transfer function* (HRTF) can be determined.

A paper by Denk et al. [7] tries to achieve acoustic transparency with the use of HRTFs in a semi-open fit hearing device. By approximating the open ear transfer function acoustical effects, such as, e.g., comb filtering, are mitigated. This approach comes with a heavy task in digital signal processing and can, so far, only in theory be ultimatively achieved.

A work-around method for the representation of sound in a fixed setting, i.e., an acoustical measurement with a dummy head, can be realized when only the incident amplitude of direct and processed sound by the hearing device are considered. This simplification is later used in the actual psychoacoustical measurement described in the following section.

3 Methods

In this section the methods used for the psychoacoustical measurement are presented. These include the *openMHA prototype*, a commodity hardware hearing device that can be used to easily change the settings of a hearing aid. With the prototype comb filter effects and attenuation of the earphones are explained. Furthermore the measurement setup of the psychoacoustical experiment and an Octave GUI which controls the prototype through the use of FIR-filters is introduced. Lastly, several potential implementations for the mitigation of the comb filter effects are presented.

3.1 Mobile hearing aid prototype

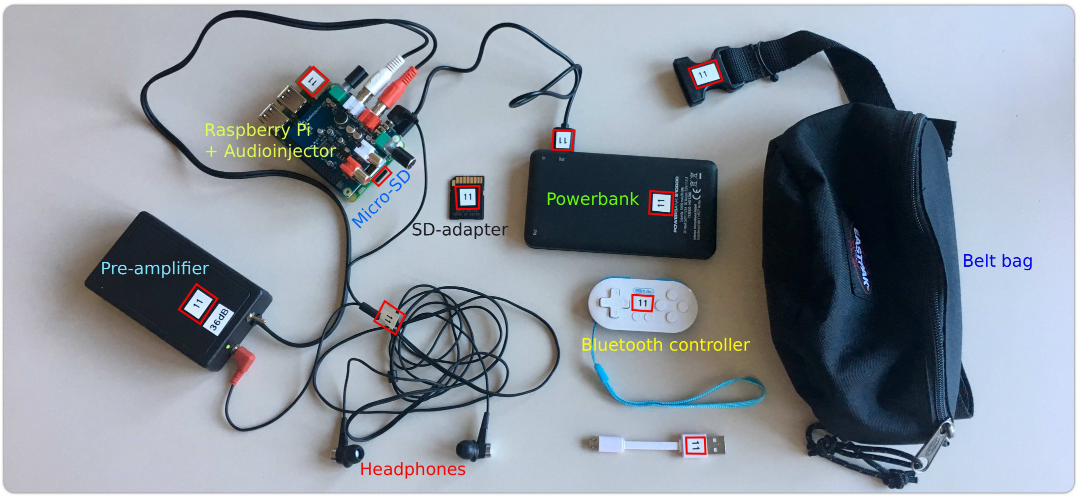

Figure 3.1: Hardware of the mobile hearing aid prototype consisting of a pair of binaural earphones, a Raspberry Pi microcontroller with an audio shield, a pre-amplifier, a powerbank and an SD-card image with openMHA algorithms. Also, the prototype can be controlled with a Bluetooth-controller.

The mobile hearing aid prototype mostly consists of commodity hardware. As seen in Figure 3.1, binaural earphones are connected via an audio-shield to a Raspberry Pi microcontroller. For the necessary recording gain, a pre-amplifier is used. Power is supplied by a power bank. Implementations of hearing aid algorithms, such as, e.g., multi-band dynamic compression, are provided by openMHA [5]. A pre-configured SD-card image with openMHA is available for download [6]. For mobile usage and convenience, a Bluetooth-controller can be used to control the prototype such as, e.g., for self-fitting.

The direct sound signal is recorded with the microphones of the binaural earphones, processed by the openMHA, and played back with the receivers of the earphones inside the ear canal. Due to the processing by the openMHA an internal time delay Δt_{proc} is caused. This delay is not constant and varies depending on the initialization of the driver. A mean time delay of 3.7 ms is usually observed.

The actual amplification of the signal is determined by the gain table. For every input level L_{in} a respective output level L_{out} is defined. The gain G, which is needed to achieve the desired output level, is specified in the gain table. The prototype operates with a filterbank of nine frequency bands for which the signal can be selectively adjusted. The center frequencies of the individual filters are 177, 297, 500, 841, 1414, 2378, 4000, 6727, and 11314 Hz. With two input channels the gain table consists in total of 18 vectors for the gain. The input levels range from -10 dB to 110 dB.

3.2 Comb filter effects with the prototype

When using the mobile hearing aid prototype, or any other conventional hearing aid, two signals, the direct sound component and the processed sound component by the hearing device itself, interfere at the eardrum, as described in section 2.1. This can result in audible signal distortions. Figure 3.2 shows the paths of the two signals.

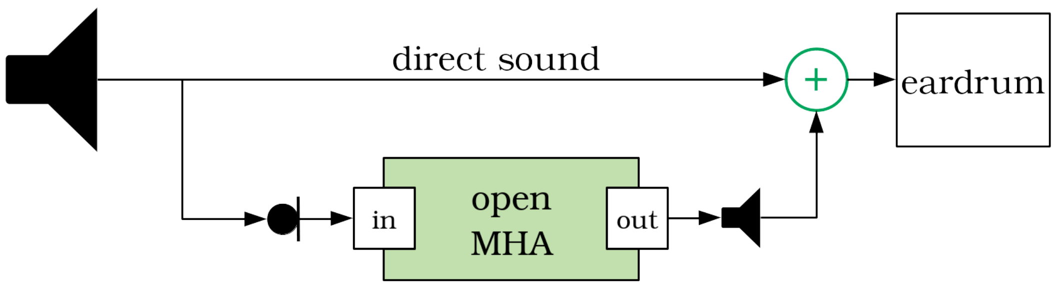

Figure 3.2: Signal paths of the direct sound and the processed sound from sound source to superposition at the eardrum. The processed sound is recorded, filtered (openMHA) and amplified. Due to the openMHA algorithm an internal time delay Δt_{proc} is caused.

The superposition of a sound source with its delayed and coherent counterpart is called comb-filtering. Dependent on this shift in time Δt_{proc} a frequency-dependent phase shift φ_{proc} for the openMHA path can be determined:

$$\varphi_{proc}(f) = 2\pi f \Delta t_{\text{proc}} \tag{3.1}$$

with frequency f.

In addition to the latency caused by the processing time of the prototype the distance between microphone and receiver of the earphones has to be taken into account. This distance was measured to be around $d_{ep} = 15$ mm. The resulting time delay Δt_{ep} is the time it takes for the sound to travel the distance from the microphone to the receiver. Thus the phase shift can be expressed as

$$\varphi_{proc}(f) = 2\pi f(\Delta t_{\text{proc}} - \Delta t_{ep}), \tag{3.2}$$

where

$$\Delta t_{ep} = \frac{d_{ep}}{c_{air}}, \tag{3.3}$$

with

$$c_{air} \approx 343\ \frac{\text{m}}{\text{s}}. \tag{3.4}$$

The time delay Δt_{ep} in this case in under $\Delta t_{ep} = 0.05$ ms. For low frequencies this additional change in overall delay time does not change the phase shift significantly and is thus neglected. Only for high frequencies the phase shift is greater than π. Table 3.1 shows the resulting phase shifts for all center frequencies.

Frequency [Hz]	φ_{ep} [rad]
177	-0.06
297	-0.09
500	-0.16
841	-0.26
1414	-0.44
2378	-0.75
4000	-1.26
6727	-2.11
11314	-3.55

Table 3.1: Phase shift for all center frequencies due to the distance ($d \approx 15\ \text{mm} \rightarrow \Delta t_{ep} \approx 0.05\ \text{ms}$) between microphone and receiver of the earphones.

Interference occurs at certain frequencies for a given time delay Δt_{proc}, or phase shift φ_{proc}, respectively, between the direct and processed sound. This interference can appear constructively (no phase shift) or destructively (phase shift of π). These critical positions of the comb filter can be determined as follows:

$$f_c = \frac{1}{\Delta t_{\text{proc}}} \cdot n \tag{3.5}$$

$$f_d = \frac{1}{\Delta t_{\text{proc}}} \cdot (n - \frac{1}{2}) \tag{3.6}$$

where f_c and f_d are the frequencies at which constructive and destructive interference, respectively, occurs. The frequencies for constructive and destructive interference are evenly spaced and can thus be computed by multiples of n, where $n \in \mathbb{N}$.

The superimposed signal $s(t)$ can be written as a summation of the direct sound signal

$$s_{ds}(t) = a_{ds} \cdot \sin{(2\pi f t + \varphi_{ds})}, \tag{3.7}$$

with amplitude a_{ds} and phase shift φ_{ds}, and the processed sound signal

$$s_{proc}(t) = a_{proc} \cdot \sin{(2\pi f t + \varphi_{proc})}, \tag{3.8}$$

with amplitude a_{ds} and phase shift φ_{ds}:

$$s(t) = s_{ds}(t) + s_{proc}(t) \tag{3.9}$$

$$= A \cdot \sin{(2\pi f t + \varphi)}, \tag{3.10}$$

with superimposed amplitude

$$A = \sqrt{a_{\text{ds}}^2 + a_{\text{proc}}^2 + 2 a_{\text{ds}} a_{\text{proc}} \cos{(\varphi_{\text{ds}} - \varphi_{\text{proc}})}}, \tag{3.11}$$

and a superimposed phase shift

$$\tanh \varphi = \frac{a_{\text{ds}} \cdot \sin{(\varphi_{\text{ds}})} + a_{\text{proc}} \cdot \sin{(\varphi_{\text{proc}})}}{a_{\text{ds}} \cdot \cos{(\varphi_{\text{ds}})} + a_{\text{proc}} \cdot \cos{(\varphi_{\text{proc}})}}. \tag{3.12}$$

This derivation is the same as described before in the theory section 2.2.

Because we are only interested in the relative phase, we assume that $\varphi_{\text{ds}} = 0$. Furthermore, the strongest effect of the comb filter in terms of change in sound energy will occur for the destructive case. Thus, we can define the resulting gain:

$$G = 20 \cdot log_{10}(1 - 10^{-\frac{|\Delta L|}{20}}), \tag{3.13}$$

with the level difference ΔL between direct and processed sound. This expression corresponds to the red curve in Figure 2.2. A level difference of $\Delta L = 0$ dB yields $-\infty$ dB gain.

Figure 3.3 shows the continuous frequency response of the expected comb filter of the prototype. The phase shift between direct and processed sound was determined by means of equation 3.1 with the mean latency of 3.7 ms due to processing.

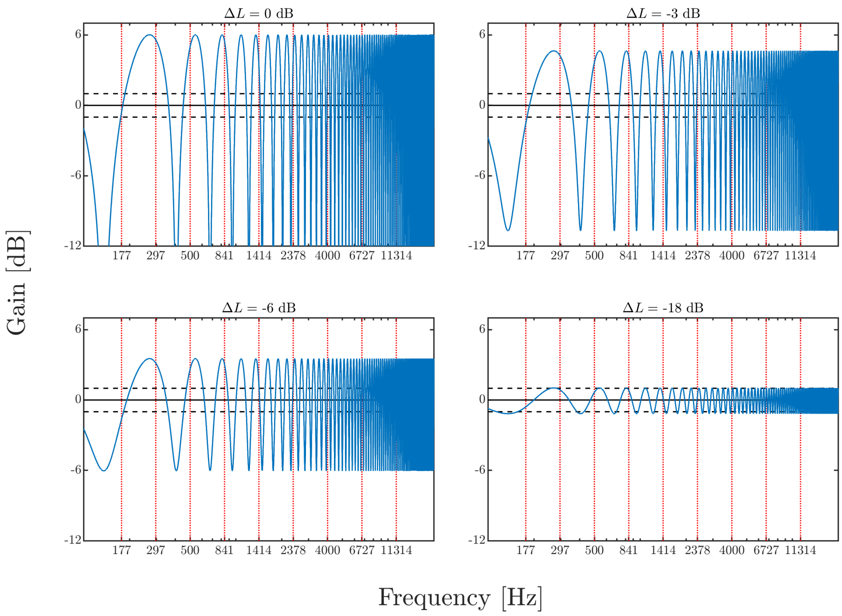

Figure 3.3: Frequency response of comb filter with $\Delta t_{\mathrm{proc}} = 3.7$ ms on a semi-logarithmic scale from 100 to 20000 Hz. The y-axis represents the resulting gain from the comb filter. Visible are the regularly spaced notches for destructive interference. For increasing level differences ΔL the resulting gain of the comb filter is reduced. A theoretical Just Noticable Difference (JND) of ± 1 dB (horizontal dashed line) is reached at a level difference of $\Delta L = -18$ dB.

The fundamental destructive frequency f_0 for $n = 1$ is given by

$$f_0 = \frac{1}{0.0037\mathrm{s}} \cdot (1 - \frac{1}{2}) \approx 135 \text{ Hz}. \tag{3.14}$$

Based on this frequency the other notches could - in theory - be easily be determined due to the periodicity of the comb filter. They are regularly spaced by $(n - \frac{1}{2})$ multiples of the fundamental frequency.

A more illustrative representation of the comb filter for the different center frequencies can be seen in Figure 3.4. The x-axis depicts the phase and the y-axis corresponds to the output level of the prototype.

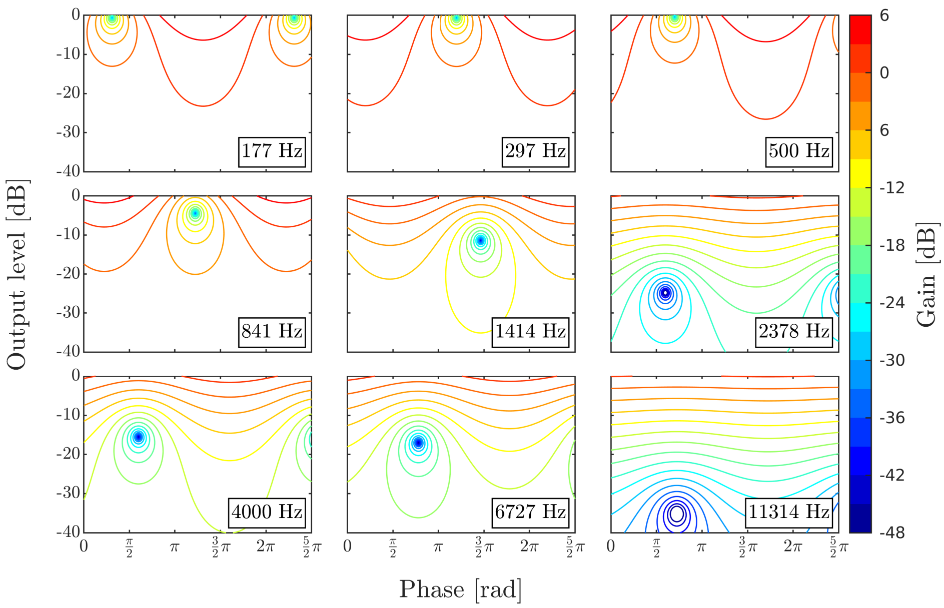

Figure 3.4: Contour plot of the comb filter for all center frequencies with a latency of $\Delta t_{algo} = 3.7$ ms. The phase is shown on the x-axis and the output level of the prototype on the y-axis. Due to the extension of the phase scale by $\frac{\pi}{2}$ there are two minima visible for certain frequencies.

It can be seen that the comb filter is symmetric along the phase axis and asymmetric along the axis for the output level L_{out} of the prototype. This means that if the minimum is approached from above changes in perceived sound level should occour more rapidly compared to an approach from below.

Overall, these comb filter effects result in spectral deviations from the source signal and are more pronounced for tonal sounds due to their periodicity and harmonic structure. More precisely, the amplitude needs to be coherent over the delay time of the processing device, i.e., 3.7 ms, for the considered mobile prototype. Especially for tonal non-speech sources like, e.g., music, this effect, also known as change in timbre, alters the spectral perception. A study by Brunner et al. (2007) [1] has shown that comb filter effects caused by reflections could still be perceived with a level difference of -18 dB between direct and reflected sound. As the effect is level-dependent, the goal is to determine the frequency-dependent levels at which it is most pronounced and mitigate it by muting the processed sound. This could possibly better preserve the natural sound of, e.g., musical instruments.

3.3 Attenuation of the earphones

A critical aspect for the effect size of the comb filter is the presence of the attenuated direct sound at the eardrum and thus the actual attenuation of the earphones. Figure 3.5 shows the frequency-dependant attenuation for several pairs of earphones. It was measured in an unechoic chamber with a dummy head [8].

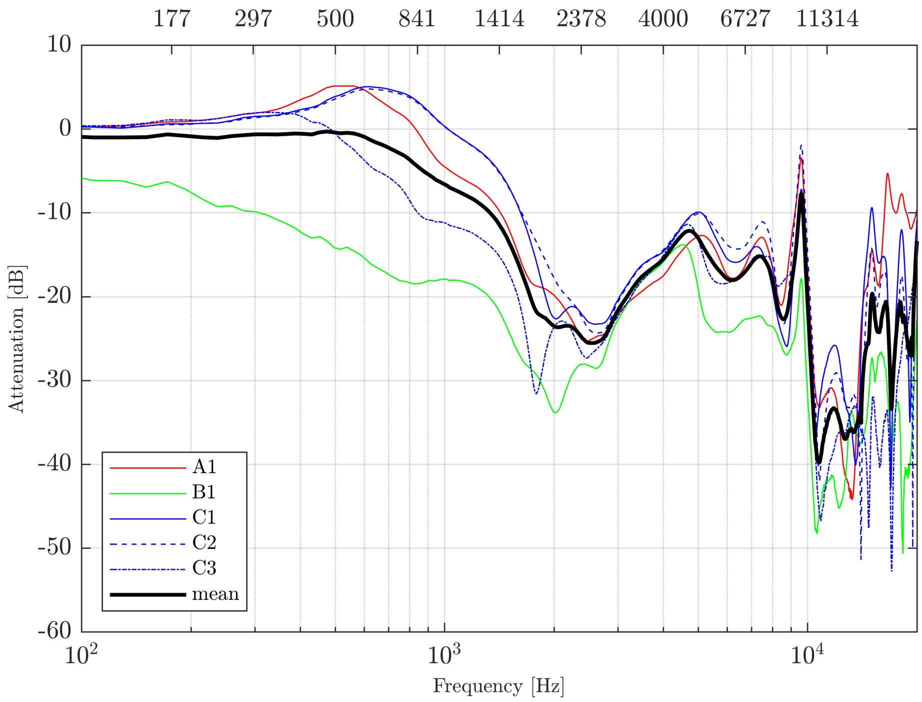

Figure 3.5: Attenuation of the earphones in the left ear for a signal coming directly from the left (S_{270}). The x-axis represents the frequency from 100 to 12000 Hz and the y-axis represents the attenuation in dB. The letters A through B mark different earphones, while the numbering 1 to 3 marks repeated insertions of the same earphone. The average of the whole set of measurements is represented in bold black. Large deviations between individual earphones can be seen.

When taking the mean it shows that for lower frequencies up to 500 Hz the attenuation is minimal and stays rather constant. For higher frequencies the attenuation tends to increase and fluctuates a lot.

From the attenuation a critical comb filter gain G_{CF} can be determined as seen in Figure 3.6. In this case the attenuation can be compared to the relative relation between attenuated direct sound and processed sound which is equivalent in amplitude to the unattenuated direct sound. This is in the case of no gain from the hearing device, because the prototype is calibrated to produce the same pressure differences at the eardrum that would be presented in an open unaided case. All frequencies are assumed to be critical with a relative phase shift of $\Delta\varphi = \pi$ Thus, the attenuation representation can be directly transferred to a representation of critical comb filter gain by means of equation 3.13. In Figure 3.6 a -1 dB threshold, corresponding to the JND, and a -3 dB threshold are visualized by a horizontal dashed and solid line, respectively.

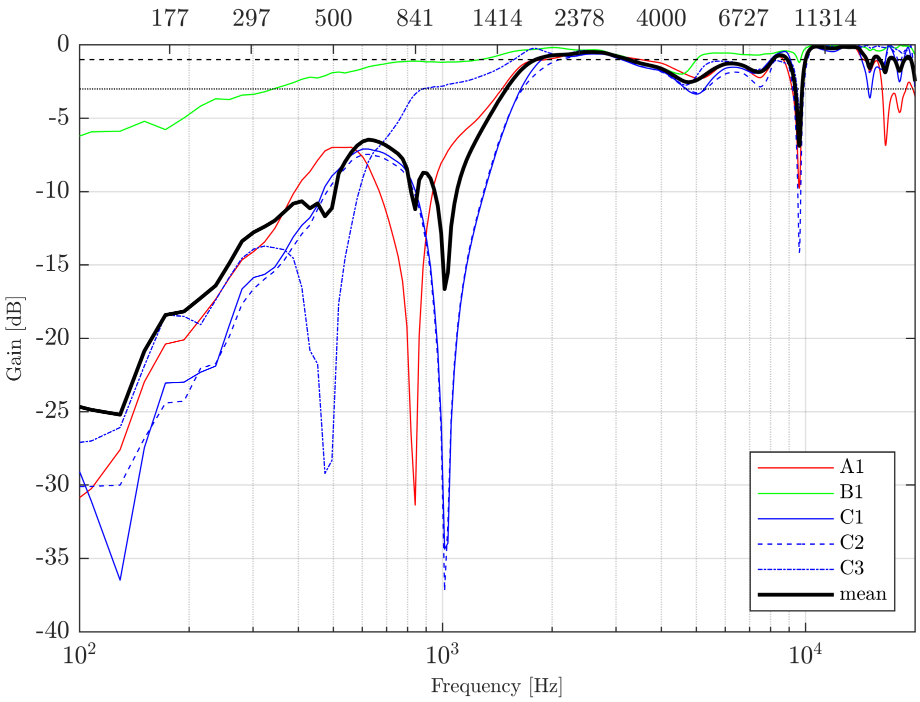

Figure 3.6: Gain of the comb filter for the $\varphi_{proc} = \pi$ case according to the attenuation in Figure 3.5. The x-axis still represents the frequency in Hz and the y-axis represents the comb filter gain in dB. The dashed black line represents the JND around -1 dB and the fine dotted line represents the 3 dB threshold.

The highest gain occurs for frequencies up to 2000 Hz. For higher frequencies the gain tends to stay between -3 and 0 dB with the exception of a dip just shy of 10 kHz. The smaller the level difference between the occluded and the open ear case, the lower the destructive gain from the comb filter. Thus, steep dips in the Figure represent a crossing of the 0 dB attenuation line or values close to 0 dB.

Table 3.2 shows the attenuation values for the center frequencies. The frequency spacing from the measurement with the dummy head was $\Delta f \approx 21.5$ Hz. For determining the values for the center frequencies the data points had to be interpolated to reach a resolution of $\Delta f_{interp.} = 1$ Hz. For the interpolation the MATLAB function `griddedInterpolant` with `spline` as interpolation method was used.

Frequency [Hz]	Attenuation [dB]
177	-0.6
297	-0.6
500	-0.4
841	-4.4
1414	-11.4
2378	-24.8
4000	-15.5
6727	-17.0
11314	-35.0

Table 3.2: Attenuation for all center frequencies.

It is to be noted that these results are directly linked to the particular HRTF of the dummy head. The characteristics of the pinna, which is the outer part of the ear (not including the ear canal), the shape of the ear canal, and also the head influence the actual sound levels present at the eardrum or rather the microphone inside the ear canal of the dummy head. Attenuation values in this work were determined for individual subjects as will be explained in the later sections. Since individual measurements of the human HRTFs would be extremely time consuming and overall unrealistic, later comparison of the results are based on the assumption that the HRTFs are very similar to each other.

3.4 In-situ amplitude-phase measurement

In this section the psychoacoustical measurement is described. This includes the overall aim of the measurement and the measurement setup. From this follow adjustments to the prototype, including a FIR-filter to manipulate amplitude and phase of the processed signal, and the use of an Octave GUI.

3.4.1 Aim of the measurement

Although theoretical frequencies for destructive interference due to comb-filtering can be determined by means of equation 3.6, the actual phase relation might be different from subject to subject. Besides the phase the amplitudes are especially important. Only if both, the attenuated direct sound and the processed sound have the same amplitude, total destructive interference, i.e. infinite negative gain, can occur.

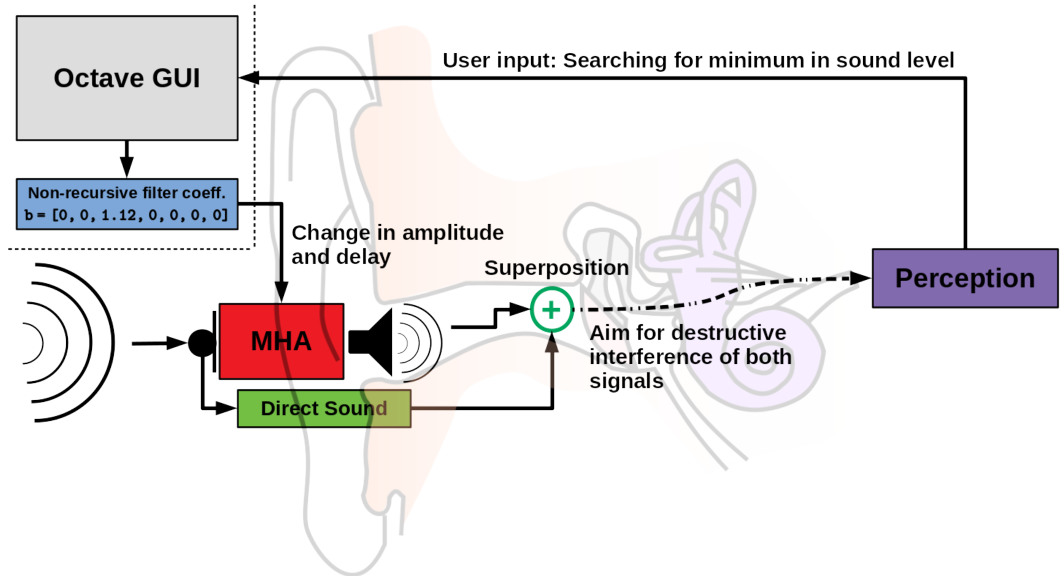

Figure 3.7: Implementation of the in-situ amplitude-phase measurement. The task of the listener is to adjust the processed signal to reach cancellation with the direct sound. The user can change the delay and amplitude via a graphical user interface (GUI) which in turn changes the filter coefficients of the FIR-filter in order to find the values which result in a compensation, i.e., destructive interference, of a pure tone.

The amplitude-phase relation might be dependent on several factors. The shape of the ear canal and the placement of the earphones can cause different resonance conditions which affect the actual amplitude present at the ear drum. But also the overall ear shape of the individual subject might cause frequency dependent distortions, as explained earlier in section 2.4, due to individual HRTFs which could lead to small variations in phase.

Another factor might be the actual latency of the prototype which also tends to fluctuate during operation and across different frequencies. The mentioned mean latency is only a rough estimate.

Thus, an in-situ amplitude-phase measurement is conducted in which the subjects have to adjust the controls of a GUI in Octave in such a way that the processed and direct sound cancel out completely in case of monaural processing by the prototype in a single ear. Figure 3.7 shows the schematic of the psychoacoustical measurement.

3.4.2 Measurement setup

The rough layout of the measurement setup can be seen in Figure 3.8. The subject was placed in front of a desk in a sound-proof cabin. On the desk a laptop running the Octave GUI was placed. A loudspeaker for the playback of the pure tones was positioned to the left of the subject, i.e., S_{270}. Its height was adjusted accordingly to the height of the subject's left ear. The distance from the loudspeaker to the ear was approximately 1 m. The subject was wearing both earphones of the prototype. The output of the right earphone was a white noise signal in order to mask the sound from the loudspeaker. This would ensure that the pure tone was only perceived in the left ear. The output of the left earphone was not changed and only later manipulated by the subject via the FIR-filter adjusted with the GUI.

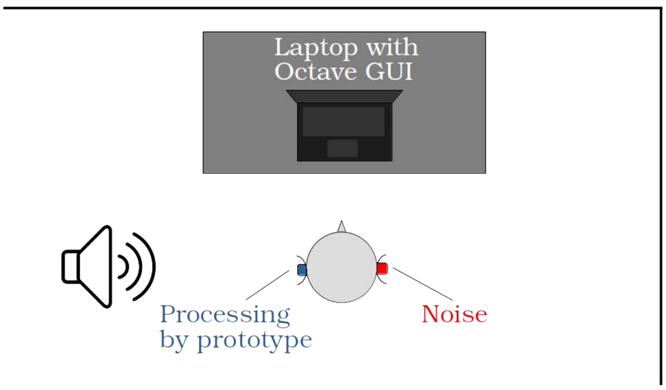

Figure 3.8: Position of the subject, GUI, and loudspeaker for the measurement.

The task of the measurement was for the subject to focus only on the sound perception of the left ear until the pure tone was not perceived anymore. Therefore the right ear had to be muted. First attempts to occlude the ear with ear plugs prooved to be not sufficient to completely mask the sound from the loudspeaker. Hence, a different method was used in which white noise was played back from the right earphones. Without proper masking the subject would perceive the sound to be attenuated but then travelling from the left ear to the right ear making it difficult to locate destructive interference only in the left ear.

From the loudspeaker pure tones were played back at around 60-70 dB SPL depending on the frequency. The set of frequencies to be measured was chosen according to the center frequencies of the prototype. Because the manipulation of the gain, described in section 3.5, is limited to the number and location of filters, the comb filter characteristic around the center frequencies is chosen to be investigated (see Figure 3.3).

The relative amplitude and phase relations, measured with the psychoacoustic experiment, reveal the relative transfer function between direct and processed sound. It can be used with additional assumptions about its smoothness to interpolate the frequencies and amplitudes where a complete cancellation would occur according to Equation (3.6).

3.4.3 Adjustments to the prototype

In order to perform the measurement with the prototype, a few changes had to be made to its configuration. These include a FIR-filter and the playback of white noise used for masking the pure tone at right ear. All important files and functions can be found in the directory `home > pi > hearingaid-prototype`. The image version used in this work was 1.2.

White noise for masking

The white noise was implemented as follows. In the file `commander.sh` a function named `thresholdnoise` is defined. Here the playback of the thresholdnoise has to be stopped for the left earphone and only played through the right earphone. Therefore lines 66 and 68 are commented out:

```
thresholdnoise() {
    local STATUS="$1"
    case "$STATUS" in
        on)
#               jack_connect thresholdnoise:output_1 abhang:input_3
            jack_connect thresholdnoise:output_2 abhang:input_4
#               jack_connect thresholdnoise:output_1 system:playback_2
            jack_connect thresholdnoise:output_2 system:playback_1
        ;;
```

Then in the file `start.sh` following two lines are added:

```
echo "initial commands"
echo feedback 3 > commandqueue
sleep 1
echo thresholdnoise on > commandqueue
```

The last step was to increase the noise level slightly. The necessary file can be found under `tools > signals`. In the file `thresholdnoise.c` the amplitude in line 15 was changed from 0.01 to 0.04:

```
15  #define AMPLITUDE 0.04
16  #define A1 0.95
```

Now, after booting, the prototype will play white noise from the right earphone at the desired level suitable for masking.

FIR-filter

For the manipulation of the processed signal a FIR-filter design is used. Beforehand, the filters have to be defined in the `openMHA.cfg` file of the prototype. Changes in the code include the following:

```
mha.transducers.mhachain.algos = [wavrec:record addsndfile:playback irrfilter:injector altplugs]

mha.transducers.mhachain.record.prefix = /dev/shm/recording
mha.transducers.mhachain.record.use_date = no
...
...
mha.transducers.mhachain.injector.A = [1.0]
mha.transducers.mhachain.injector.B = [1.0]
```

The non-recursive filter coefficients are changed according to the selected amplitude and phase values by the subject through the Octave GUI. The additional delay of the processed signal is determined by

$$d = \frac{\varphi_{\mathrm{MHA}}}{2\pi} \cdot \frac{f_s}{f}, \tag{3.15}$$

with frequency f and sampling frequency f_s. The phase φ_{MHA} is the input of the user via the GUI.

Secondly, the amplification level is determined from the input level a_{MHA} of the subject and is inserted at the previously determined digital delay d:

$$b[n] = [0\ 0\ 10^{\frac{a_{MHA}}{20}}\ 0\ \ldots]. \tag{3.16}$$

A discrete convolution between the recorded signal of the incident sound wave and the FIR-filter coefficients is performed:

$$s_{\mathrm{proc}}[n] = \sum_{i=0}^{N} b_i \cdot s_{\mathrm{rec}}[n-i], \tag{3.17}$$

where s_{proc} is the output signal, s_{rec} is the input signal, N is the filter order, and b_i the value of the i-th FIR-filter coefficient.

By this operation, the signal is shifted in time according to the adjusted phase and amplified according to the adjusted amplitude.

3.4.4 Octave GUI

The filter parameters of the digital FIR filter were calculated based on the user input via a graphical user interface implemented in GNU/Octave (Figure 3.9).

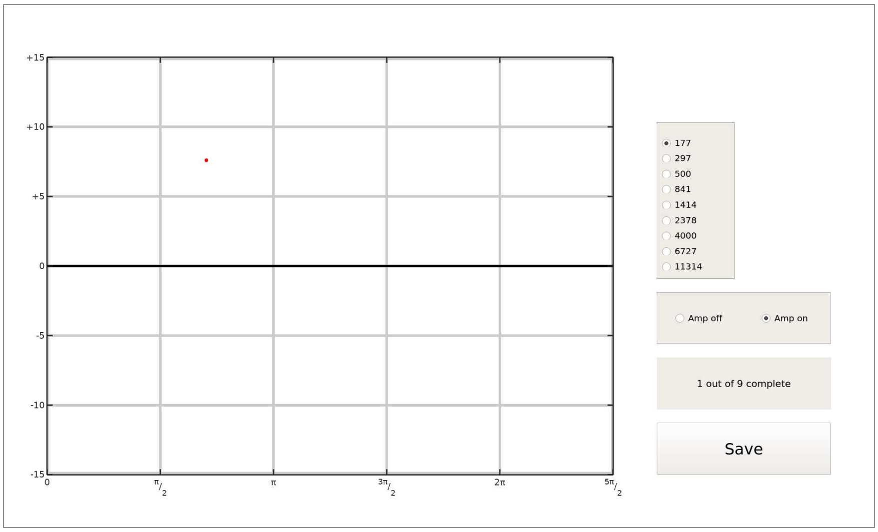

Figure 3.9: The octave GUI used in the psychoacoustic measurement. The x-axis represents the phase and the y-axis represents the amplitude. The small red dot visualizes the input from the subject.

The adjustable parameters - amplitude a_{MHA} and phase φ_{MHA} - were visualized with two-dimensional axes. The phase was presented on the x-axis and the amplitude on the y-axis. The range of the phase axis was set to values from 0 to $\frac{5}{2}\pi$ to prevent confusion if the measured value was found around 0 or, respectively, 2π. Hence, the test subject would need to jump between the outermost left and right side of the graph. Adding an additional $\frac{\pi}{2}$ after 2π this problem was avoided. Of course now for certain frequencies a minimum could be found twice on the plot but the test subject was advised accordingly. The range of the amplitude axis was limited to 30 dB and centered around the theoretical attenuation of the earphones at the individual frequencies (Fig. 3.5 and Table 3.2). As axis labels no actual values in dB were given. The center of the axis was labeled 0. The topmost label was $+15$ and the downmost label -15 with 5 dB steps in between.

All test subjects were given an introduction to the measurement alongside with a useful strategy to find the minimum. This was done to decrease the measuring time and make the measuring procedure more effective and also precise. The strategy was to first only adjust the phase with constant amplitude around 0 on the plot. By clicking in intervals of $\frac{\pi}{2}$ rad, a rough estimate for the position of the minimum could be made with additional fine tuning. After finding the minimum on the phase axis the test subject would then adjust the amplitude keeping the phase fixed. With the correct amplitude values the sound could be cancelled out completely. If this was not the case, the test subject would search around the current minimum slightly adjusting the amplitude and the phase.

After successfully locating a minimum, the test subject could play back a reference signal which consisted of only the attenuated direct sound component. For this the amplification of the prototype was turned off. The pure tone signal would now be perceived again. By clicking on *Amp on*, the amplification is turned back on and the sound is cancelled again. Finally, the test subject clicked on *Save* and the amplitude-phase values were stored for the current frequency. This procedure continued for all of the frequencies to be tested.

3.5 Comb filter mitigation

With the knowledge about the actual amplitude-phase combinations the comb filter effects can be mitigated by avoiding critical levels of gain which are determined by the gain table. Here, a manipulation can be applied. Furthermore a potential implementation of an all-pass filter design could also improve the sound perception or in combination with the gain table manipulation.

3.5.1 Gain table manipulation

The gain table of a hearing aid contains the gain values dependent on the input level. This could for instance have a compressive behaviour in which low level input is sufficiently amplified but high level input is not amplified as strongly or at all.

The determined levels from the measurement can be used to switch off or attenuate the processed sound by reducing the gain for certain input levels which could potentially result in strong comb filter effects in the corresponding frequency band. Especially for mild hearing loss, with output levels that are close to the direct sound the destructive comb filter behaviour is most pronounced. As seen in Figure 3.6 especially the low frequencies have to be dealt with as the effect size of the comb filter is most pronounced in this region. Higher frequencies can be neglected.

3.5.2 All-pass filtering

A different approach would be to manipulate the phase of the processed sound in such a way that destructive interference with the direct sound is avoided. This would mean that the relative phase between direct and processed sound does not come close to the a value of π.

This method can be realized with a so-called *all-pass filter*. As the name suggests, and in contrast to low- or highpass filters, an all-pass filter lets through all frequencies equally. The incoming signal will thus only be manipulated in phase, and not in amplitude. There could be two approaches to the desired phase shifts that change the characteristic of interference.

One option would be to have the processed sound interfere constructively across all frequencies. Thus the relative phase shift between direct and processed sound would be 0 or 2π for all frequencies. In this case the resulting signal will always be amplified positively, independent of the level difference between direct and processed sound. For a level difference of $\Delta L = 0$ dB the amplitude of the resulting signal would be twice a high. No destructive interference would occur, but it could be that even the amplification of certain frequencies results in an audible effect that has a negative influence on the perception of sound.

A second and more sophisticated option would be to set the relative phase to $\varphi = \frac{2}{3}\pi$. This would correspond to an amplification of 0 dB, thus, the sound would appear unaltered. Of course these are special cases which only hold assuming a level difference of 0 dB between direct and processed sound. For level differences larger than 0 dB the comb filter behaves differently. Although, phase shifting by $\frac{2}{3}\pi$ would, theoretically, still lead to the best outcome as it shows a minimum of just below -1 dB (see Figure 2.2).

3.5.3 Gain table manipulation and all-pass filtering

The combination of gain table manipulation and all-pass filtering is also an option. This would be necessary if the gain of -1 dB is still audible for certain frequencies and a cutoff of the playback leads to an improvement in the perception of the sound.

4 Results

In this section the results from the psychoacoustical experiment are presented. They include the amplitude-phase value pairs for the individual subjects, measured attenuation by the earphones, critical frequencies of the comb filter, and a proposed implementation of a gain table manipulation.

4.1 Amplitude-phase measurement

Overall, 10 subjects participated in the psychoacoustical measurement. For the results only values of 9 of the subjects are presented due to strong inconsistencies with one subject. Their values were neglected due to a possible error during the measurement.

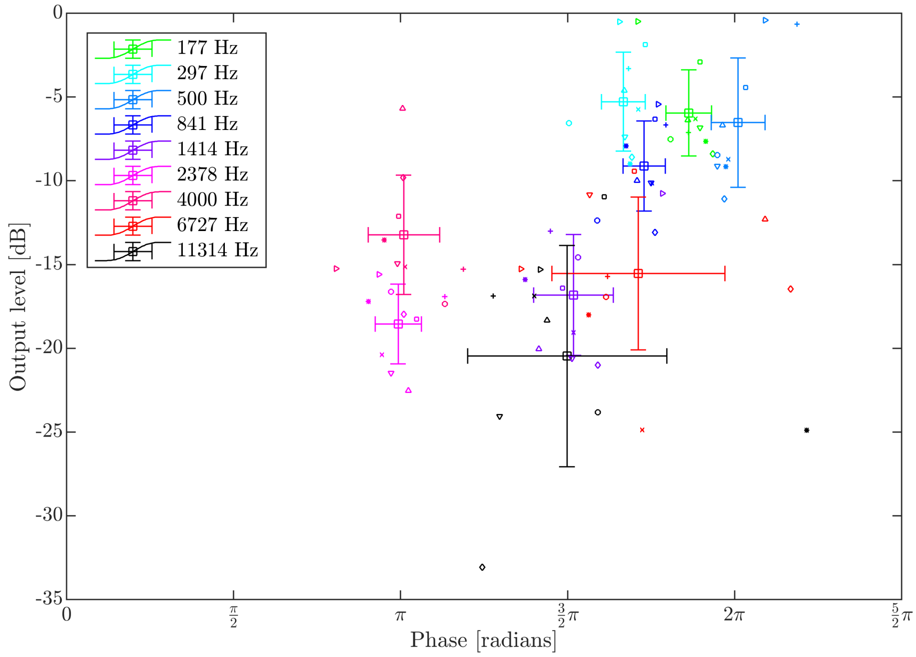

Figure 4.1: Measurement data from the amplitude-phase measurement for all subjects. Different frequencies are visualized by different colors. Data from the individual subjects is presented with individual markers. The average values for each frequency is visualized by a large square and the range of the data points is shown with bars.

Individual results for all subjects can be seen in Figure 4.1. The phase axis is shown as seen in the Octave GUI from 0 to $\frac{5}{2}\pi$. The amplitude is shown as output level in dB and the axis is fixed from -35 to 0 dB. Different colors depict different frequencies. Mean values across all subjects for phase and amplitude are marked with large squares. The errorbars visualize the standard deviation.

It can be seen that the values tend to cluster for certain frequencies and spread for others. Low frequencies appear to cluster more than high frequencies. Regarding the phase, some frequencies almost overlap with each other. This overlap can also be seen with the amplitude, especially for the lower frequencies from 177 Hz to 500 Hz.

Figure 4.2 shows the same data points from Figure 4.1 but phase and amplitude are depicted in seperate plots. The phase values are in the upper plot and amplitude values in the lower plot. The frequency is shown on the x-axis in a quasi-logarithmic manner.

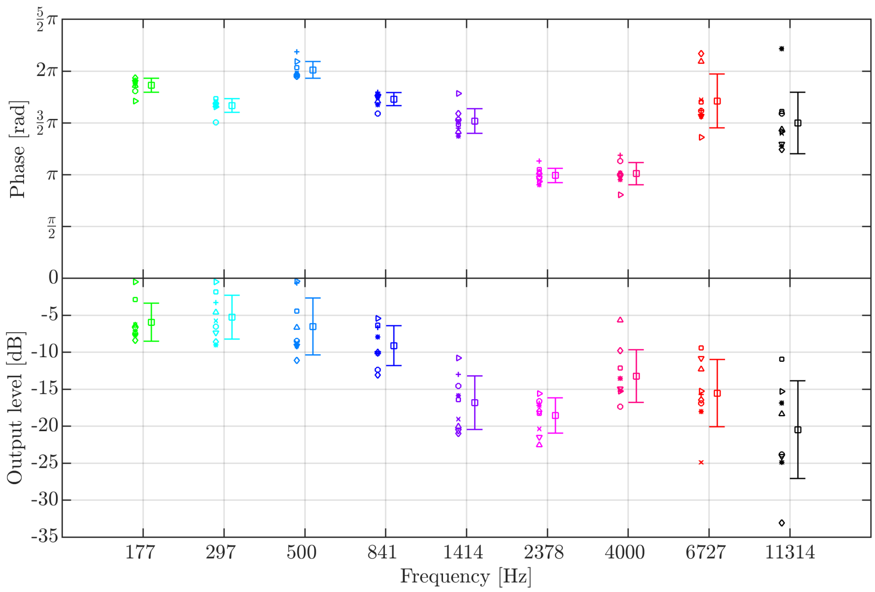

Figure 4.2: Measurement data for the phase and amplitude seperate across all measured frequencies. Standard deviations are shown as error bars from the mean. Individual data points are presented accordingly to the individual markers.

For the phase the standard deviations for frequencies up to 4000 Hz are rather small and tend to stay the same, except for a comparitively large standard deviation at 1414 Hz. Only for the frequencies 6727 Hz and 11314 Hz does the standard deviation exceed the value of $\frac{\pi}{4}$. Exact values for mean and standard deviation for the phase are shown in Table 4.1.

For the amplitude the standard deviation stays almost the same for all frequencies. Similar to the phase the standard deviations for 6727 Hz and 11314 Hz are the largest. For frequencies in the low and mid range the overall standard deviation appears to be around 3 dB. Exact values for mean and standard deviation for the amplitude are shown in Table 4.1.

f [Hz]	$\bar{\varphi}$ [rad]	σ_φ [rad]	$Range_\varphi$ [rad]	$\bar{A}$ [dB]	σ_A [dB]	$Range_A$ [dB]
177	5.85	0.21	0.71	-5.95	2.58	7.92
297	5.24	0.21	0.72	-5.28	2.96	8.50
500	6.32	0.25	0.75	-6.52	3.86	10.67
841	5.43	0.20	0.65	-9.12	2.69	7.66
1414	4.77	0.37	1.29	-16.82	3.62	10.24
2378	3.12	0.22	0.72	-18.55	2.37	6.94
4000	3.17	0.34	1.20	-13.23	3.57	11.67
6727	5.38	0.81	2.54	-15.54	4.56	15.45
11314	4.71	0.94	3.05	-20.47	6.62	22.12

Table 4.1: Mean, standard deviation, and the range of phase and amplitude across all frequencies.

4.2 Attenuation of the earphones

From the amplitude values the attenuation of the earphones can be directly determined and compared with the preexisting data from the measurements on the dummy head.

Figure 4.3 shows the mean and the standard deviation for the dummy measurements. On top of that are the mean values from the experiment. It can be seen that most of the data points lie within one standard deviation from the mean. For the lower frequencies (177 to 1414 Hz) the measured attenuation from the experiment is larger than the attenuation from the dummy measurements. For 2378 Hz and higher the experimental attenuation is smaller. It seems that for lower frequencies the offset is rather consistent. For the higher frequencies this consistency is not visible.

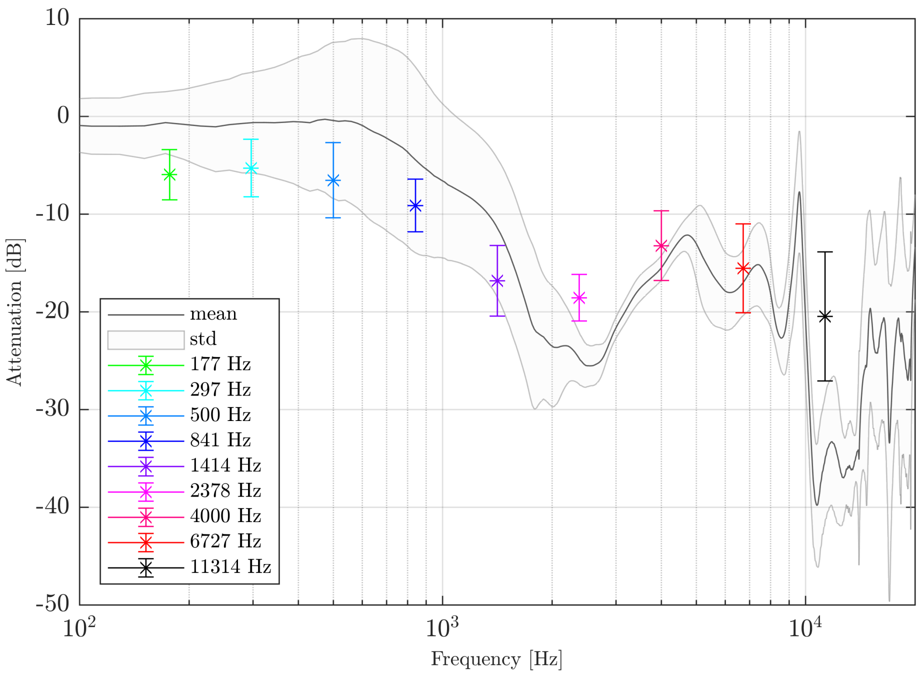

Figure 4.3: Mean values of the attenuation from the experiment on top of the mean of the attenuation from the dummy measurement. Standard deviation from the dummy measurement is visualized by the light gray shaded area.

For both the attenuation from the dummy head and the attenuation from the measurement a clear dip around roughly 2.5 kHz is visible. Around 5-6 kHz a case of resonance can be observed.

As mentioned before in section 3.2, individual HRTFs of the subjects have to be taken into account when looking at these results and the measured attenuation with the dummy head. Nontheless, a direct comparison of the results can be seen in table 4.2.

f[Hz]	A_{exp}[dB]	A_{dummy}[dB]	ΔA [dB]
177	-6.0	-0.6	-5.3
297	-5.3	-0.6	-4.6
500	-6.5	-0.4	-6.1
841	-9.1	-4.4	-4.7
1414	-16.8	-11.4	-5.4
2378	-18.6	-24.8	6.2
4000	-13.2	-15.5	2.3
6727	-15.5	-17.0	1.5
11314	-20.5	-35.0	14.5

Table 4.2: Attenuation for the psychoacoustical experiment and the dummy head across all center frequencies. All values were individually rounded and, thus, might not add up correctly here.

As seen in section 3.2, the experimental attenuation from the earphones can be converted into a destructive gain as seen in Figure 4.4. Due to the stronger measured attenuation for low frequencies the potential negative comb filter gain is less pronounced and around 7-8 dB. For higher frequencies the gain stays the same and the values are closer to the premeasured values from the dummy head. Large deviations for the low frequencies are visible whereas the high frequencies show smaller deviations. Exact values for the mean values from the measurement and the values of the dummy head are written in Table 4.3.

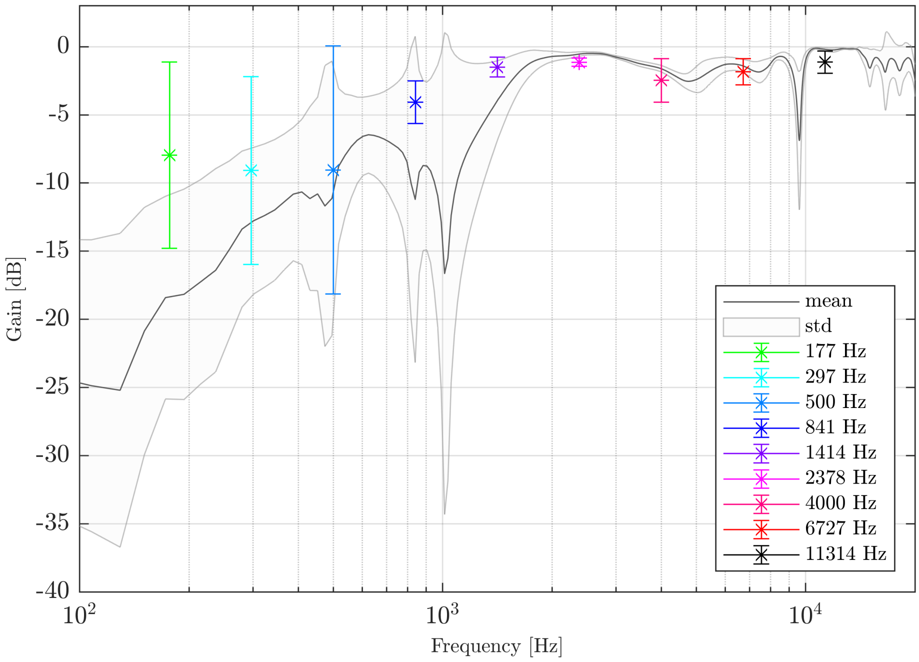

Figure 4.4: Experimental destructive gain on top of the gain calculated from the measured attenuation of the earphones on the dummy head. Large deviation for the lower frequencies and small deviation in the high er frequency range are clearly visible.

f [Hz]	G_{exp} [dB]	G_{dummy} [dB]	ΔG [dB]
177	-8.0	-18.3	10.3
297	-9.1	-12.8	3.8
500	-9.0	-10.2	1.3
841	-4.1	-10.8	6.9
1414	-1.5	-3.9	2.5
2378	-1.1	-0.5	0.6
4000	-2.5	-1.6	0.9
6727	-1.8	-1.4	0.4
11314	-1.1	-0.2	0.9

Table 4.3: Gain resulting from the attenuation for the psychoacoustical experiment and the dummy head across all center frequencies. All values were individually rounded and, thus, might not add up correctly here.

4.3 Latency and critical frequencies

From the mean of the phase values $\bar{\varphi}$ the latency of the prototype can be determined by means of equation 3.1:

$$\Delta t_{proc} = \frac{\bar{\varphi} + 2\pi n - \pi}{2\pi f} = \frac{\bar{\varphi} + \pi(2n-1)}{2\pi f} \tag{4.1}$$

with shift index $n = 0, 1, 2, \ldots$ which can be determined approximately with respect to the theoretical or given latency by the prototype:

$$n = \lfloor \frac{\varphi_{th}}{2\pi} \rfloor \tag{4.2}$$

with

$$\varphi_{th} = 2\pi f \Delta t_{th}. \tag{4.3}$$

The shift index is important in order to correctly determine the latency. Otherwise, especially for increasing higher frequency, the latency will be off by a multiple of 2π. Nontheless, this latency would still cause destructive interference to occur.

The resulting latencies are shown in Table 4.4.

f [Hz]	Δt_{algo} [ms]
177	2.4
297	4.5
500	3.0
841	4.0
1414	3.7
2378	3.4
4000	3.5
6727	3.6
11314	3.6

Table 4.4: Latency Δt_{algo} calculated from the mean values for the measured phase. The mean latency of the prototype is supposedly around 3.7 ms.

As seen in the table, the calculated latencies seem to vary a lot for the lower frequencies and are nowhere close to the theoretical value of 3.7 ms. Only the high frequencies are close to that value, which is expected due to larger changes in phase for a small change in latency.

From these latencies the critical frequencies at which destructive interference can occur are determined in the proximity of the individual center frequency or, respectively, within that frequency band. Figure 4.5 shows the resulting comb filter for the lowest six frequencies.

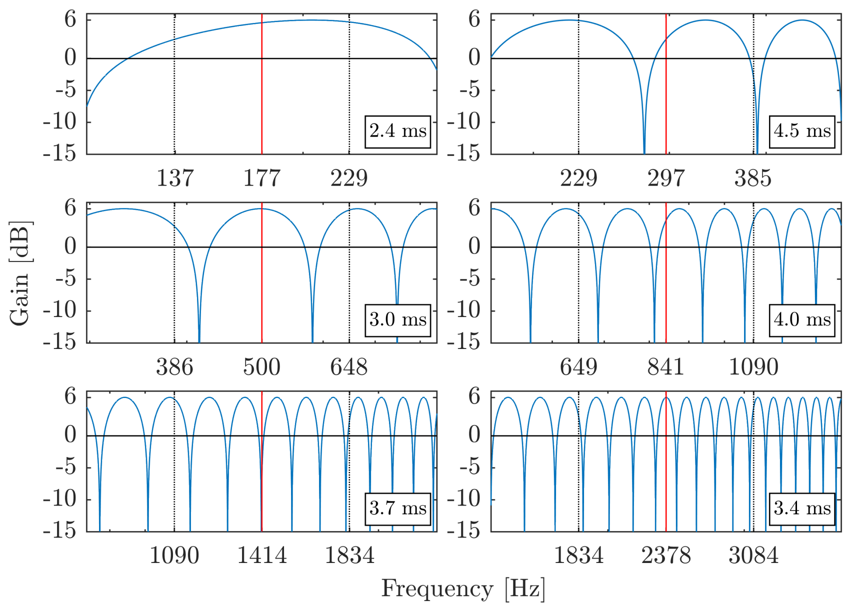

Figure 4.5: Potential comb filters for the lowest six center frequencies with their specific latencies. The red line indicates the center frequency and the dotted black lines are the respective filter borders.

It can be seen that due to the logarithmic structure of the compression filters and the corresponding center frequencies only a few critical frequencies are within one band at low frequencies whereas for higher frequencies the number of critical frequencies increases exponentially.

4.4 Gain table manipulation

According to the experimental latencies and the resulting comb filter characteristics, the gain of individual frequency bands can be adjusted, i.e., around the center frequencies of the prototype. Overall, areas with critical comb filter gain can be avoided through the manipulation of the gain table.

An example for a compressive prescription rule (*Plack2004low* [9]) can be seen in Figure 4.6. The gain is visualized as an input-output function for all center frequencies. The dotted line represents the points at which the input and output are equivalent. For these cases the destructive (and constructive) interference due to the gain of the comb filter is the strongest. Around this line, critical areas of varying resulting gain can be set. Within the gray area negative gain of 6 dB or more is possible. The light gray area marks the threshold for the 1 dB (JND) gain.

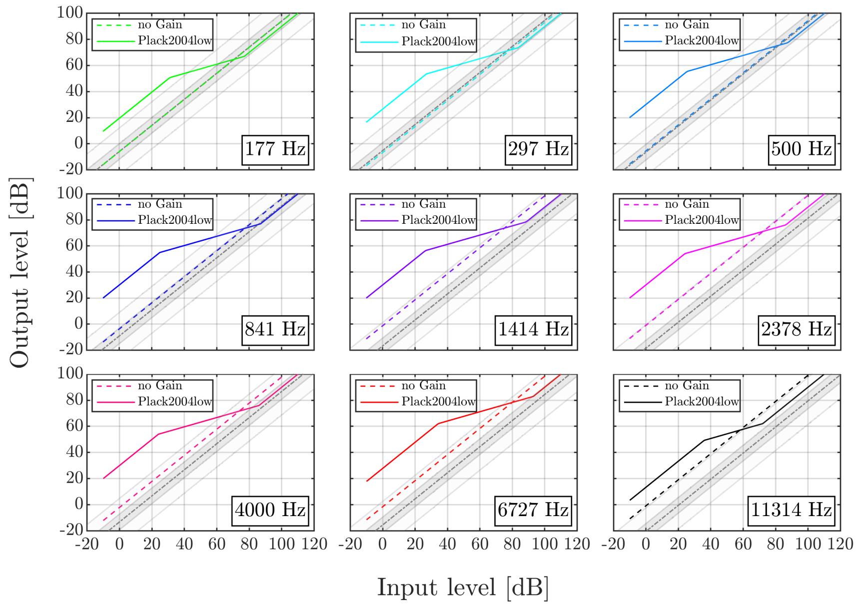

Figure 4.6: Gain table input-output relation for no amplification (dashed line) and Plack2004low (solid line) for all frequencies. Critical areas for -1 and -6 dB gain are colored in light gray and dark gray, respectively. The black dotted line represents the equivalance of input and ouput. The measured attenuation for each frequency has been taken into account.

The output level in this case is not the output of the earphones but the sum of attenuated direct sound and processed sound. Output levels which cause the attenuated direct sound and the processed sound to have the same level (dashed line in Figure 4.6) are calculated as follows:

$$L_{out} = 20 \cdot log_{10}(10^{\frac{L_{in}}{20}} - 10^{\frac{L_{in}+A}{20}}) \tag{4.4}$$

The attenuation A is taken from the mean values of the amplitude from the measurement.

The gain from the gain table is visualized as an output level which is simply determined by the summation of the input levels and the corresponding gain:

$$L_{GT} = L_{in} + G_{GT}. \tag{4.5}$$

To avoid strong interference due to the comb filter, the gray areas in Figure 4.6 should be avoided. Figure 4.7 shows a possible course of the gain in which the critical area of 6 dB negative gain is avoided. Apparently, this manipulation leads to a major jump in the input-output function. Furthermore, the compressive behaviour for certain frequencies is cut off prematurely due to the sudden jump.

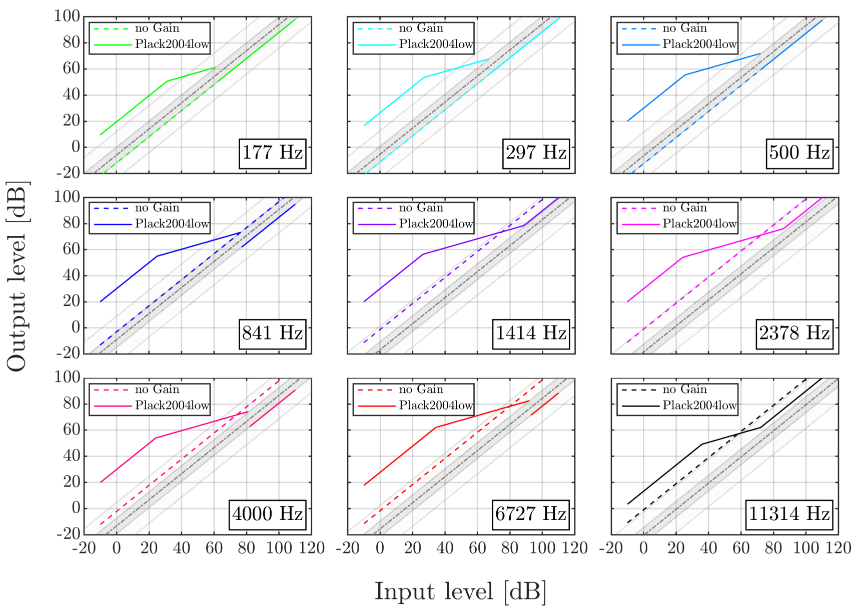

Figure 4.7: Possible gain table manipulation represented as an input-output relation. The dashed line represents the manipulated no amplification rule and the solid line represents the manipulation for the Plack2004low amplification rule. The critical area of more than -6 dB gain due to the comb filter has been avoided as compared to Figure 4.6.

It is to be noted that when determining the output level from the adjusted gain lines, the comb filter effect at that particular point has to be taken into account aswell. In other words, on top of each point in Figure 4.6 and4.7 lies another function which describes the characteristic effect of the comb filter (compare with Figure 2.2). Thus, potential adjustments and visual skips in Figure 4.7 may in reality behave differently as they appear in the graph. In this case it is only a visualization of the critical areas of the comb filter in correspondance to the applied gain from the gain table.

For 841 Hz, although the output for no amplification reached inside the critical area the resulting manipulated output was put above the attenuated direct sound level instead of below. This was done because the difference between the critical line of 6 dB and the unadjusted output inside the critical area was rather small. Thus, the resulting output level should still be tolerable as its as the difference was smaller than 1 dB. For the other frequencies where a manipulation is necessary the gain was adjusted to put the resulting output level below the critical area to avoid too high sound levels, which could potentially cause harm to the listener.

5 Discussion

In this section the previous results from the psychoacoustical measurement are discussed. Furthermore possible measurement errors are listed and potential follow-up experiments are suggested.

5.1 Amplitude-phase measurement

The results for the amplitude-phase measurement show consistent data across all subjects which indicates that the measurement overall was successful and sound cancellation by destructive interference of sinusoidal waves in the ear canal is possible.

5.1.1 Phase

Small deviations for the phase suggest a conformity of the individual phase relations across all subjects. This is especially important for the implementation of an all-pass filter or its combination with manipulation of the gain table. It can be assumed that a single fit is applicable to all listeners. Larger deviations for the high frequencies can be explained due to a poor resolution of the phase adjustment. The sample length of the FIR-filter is limited to the sampling rate f_s of the prototype. The resulting frequency dependent phase resolution is given by

$$\Delta\varphi = 2\pi \cdot \frac{f}{f_s}. \tag{5.1}$$

For frequencies above 12 kHz the resolution is larger than $\frac{\pi}{2}$ making the search for the actual minimum unreliable. In the worst case, where the subject selected a phase of, e.g., $\varphi = \frac{\pi}{4}$, the lowest possible minimum would be around -2 dB. Only for frequencies up to 4 kHz is the phase resolution sufficiently small with resulting minima below -10 dB in the worst case scenario.

5.1.2 Amplitude

Larger deviations for the amplitude might suggest a non-conformity across the amplitude; values for the standard deviation are never below 1 dB (JND). This might be explained by a potential masking of the pure tones by the contralateral white noise.

In the measurement, the minimum can be approached from above or below. As desribed in section 3.2, the comb filter shows an asymmetric characteristic along this axis. Depending on where the subject clicked on the GUI they might have located the upper or lower threshold created due to masking. Some subjects might have hit the correct spot intuitively. For certain frequencies the measurement values tend to cluster in two distinct chuncks, which could confirm the hypothesis of contralateral masking. This hypothesis could be investigated by changing the sound level of the white noise.

The strong deviations for the high frequencies can again be explained by the poor phase resolution. Only if the phase was determined correctly a strong minimum could be deteced, resulting in accurate values for the amplitude. Overall, qualitative discussion of the frequencies above 6727 Hz is questionable due to the strong deviations.

It is important to note that the individual deviations for phase and amplitude are dependent on each other. This means that a phase selection by the subject, which is slightly off, influences the selection of the correct amplitude. A clear minimum might not be found and the resulting amplitude is off. Vice versa, an over- or underestimation of the amplitude may cause the subject to not be able to find a clear minimum as well.

As the results for the phase tend to not deviate significantly an additional measurement could be conducted in which the mean phase values are locked and the subject only has to adjust the amplitude to achieve complete destructive interference. If a clear minimum can be found across all subjects the values for the incident amplitude could be measured more accurately. Overall, the task for the subject would be much more simple as they do not have to deal with adjusting the phase.

5.2 Attenuation of the earphones

The results for the amplitude show some similarities to the attenuation measured with the dummy head. The peak around 5-6 kHz and the dip between 2 and 3 kHz can be explained by resonance inside the ear canal. Open ear resonance would occur around 2-4 kHz [10]. This is due to the $\frac{\lambda}{4}$ resonator characteristic of the ear canal. When the ear is occluded by, e.g., earphones from a hearing device, the ear canal would react as a $\frac{\lambda}{2}$ resonator. This would result in resonance around 4-8 kHz and also a dip at half that frequency, which puts the dip around 2-4 kHz again. This characteristic can be seen in Figure 4.3 for both the dummy head and the subjects.

Deviations from the calculated mean attenuation could be explained by difference in the anatomy of the ear of the dummy head and the ear of a real human subject. Also, resonance conditions may vary. These deviations could be overall explained by the previously mentioned HRTFs. As individual HRTFs have not been measured a good comparison of the attenuation among the subjects and with the dummy head proves to be limited by the uncertainty of the individual effects of the HRTFs. But as the attenuation was only effectively measured for a single direction and overall consistencies in amplitude across all subjects were observed, differences in the HRTFs of the individual subjects are neglected and not further evaluated at this point.

Furthermore, the inconsistencies across different earphones, which can be seen in Figure 3.5, may have caused the slight offset from the mean. In order to test this hypothesis the attenuation of the earphones used in this experiment should be measured with the dummy head. But when comparing the results with the earphone *B1*, the measured values are always in between it and the mean value for the lower frequencies up to 2378 Hz. Hence, results from this measurement seem to be reasonable in terms of variation among the earphones themselves.

The values for the destructive gain in the critical case of a phase shift of $(2n - 1) \cdot \pi$ in the high frequencies suggest that the effect of the comb filter, which is less pronounced, might not lead to a strong audible distortion and the perception of, e.g., music might not be altered in a negative sense. This of course has to be tested in another psychoacoustical measurement with and without mitigation and also with and without the earphones. Large deviations among the values from the measurement for the lower frequencies can be explained by the negative exponential behaviour of the comb filter as seen in Figure 2.2. Small changes level difference between direct and processed sound lead to large changes in gain from the comb filter.

Concerning the lower frequencies, a possible manipulation of the gain table may lead to a reduction of the negative gain for critical frequencies. This also has to be tested in a psychoacoustical measurement with the manipulated gain table.

The attenuation property of the earphones also allows for a consideration of a hearing protection. Especially in the mid-to-high frequency range the attenuation is quite large.

5.3 Latency and critical frequencies

The results for the latency show strong fluctuations from the theoretical mean latency of the prototype. Thus, the latency might be heavily dependent on the frequency and the actual critical frequencies might not be evenly spaced. Hence, a measurement with the dummy head should be performed in which the frequency response of a sweep is measured. From the recorded dips the critical frequencies can be determined and verified in comparison to the measured frequencies in the psychoacoustical measurement.

5.4 General causes of measurement deviations

Overall, there are several sources of error in the current measurement setup which include reflections from the walls of the sound-proof room, binaural masking effects by the contralateral white noise, head movements of the subjects as well as resolution of the phase adjustement for higher frequencies, single measurements for each frequencies, and an overall inefficient measurement technique with the Octave GUI. Some of these errors could be avoided in later experiments.

5.4.1 Reflections from the wall

The positioning of the subject close to the corner of the wall might have caused unwanted reflections which could lead to deviations in the measured amplitudes for certain frequencies. Although the right ear was shielded by the white noise, reflection of the pure tone by the walls could have been audible at the left ear. To overcome this issue, a sufficiently large and anechoic chamber should be used in which the subject is placed directly in the middle.

5.4.2 Masking effects

The use of white noise at the contralateral ear might have caused a binaural masking effect. Thus, the audibility of the pure tone on the left ear could have been constrainend. This should result in higher amplitudes than originally suspected. Another solution would be to use some sort of ear protection that totally supresses the sound on the contralateral ear. A work-around with *Oropax* and sufficiently low sound levels for the pure tones might proove to be satisfactory. A comparison between the two techniques could be investigated in a later experiment.

5.4.3 Head movements

Especially for high frequencies the movement of the subject's head has a strong influence on the perception of the pure tone. Whereas translational movement along the axis of the incident sound wave has little to no effect due to the relative incidence of the two sound waves, small rotations of the head alter the sound interference substantially making it hard to impossible to cancel out the sound completely. Although the subjects had been instructed to keep their head still, slight movements can not be avoided. A solution would be to fixate the head to some sort of mount to prevent head movements by the subject.

5.4.4 Phase resolution

The resolution of the phase adjustment for high frequencies as discussed before is limited by the sampling frequency f_s, which in this case was $f_s = 48000$ Hz. For a pure tone at $f = 12$ kHz this would result in a phase resolution of $\frac{\pi}{2}$ for each sample. An accurate measurement of the correct phase relation at higher frequencies via the Octave GUI is thus not possible. For frequencies up to 4000 Hz the phase resolution is sufficient. To measure the phase relations at higher frequencies precisely, a much larger sampling frequency is necessary. But the prototype is currently constrained to 48 kHz.

Another approach would be to split the phase between two digital samples as percentages of the amplitude. For example if the phase input of the subject for a high frequency lies exactly between two samples the resulting filter coefficients would have 50 % of the amplitude in one sample and 50 % in the other sample. The effective output would be an approximation of all the energy in one sample that has been shifted in time.

5.4.5 Single measurements

Overall, the results from the measurement and calculated values for the mean and standard deviation might not be accurate enough due to a single measurements for each subject. As the process of locating the minimum has proven to be rather difficult for certain frequencies, repeating the procedure multiple times for each frequency would show how accurate a mean value could be determined across all subjects.

5.4.6 Overall measurement technique

The technique of determining the amplitude-phase values could be improved. As seen before, the measured values for the phase do not vary significantly across all subjects. This could be used to adapt the measurement accordingly. Hence, the phase would be preselected and kept constant with only the amplitude being adjusted by the subject. This approach could improve the precision and accuracy of the actual amplitude values. Besides that, a reduced latency between a click in the GUI and the adjustement of the processed sound could improve the performance of the subject.

5.5 Gain table manipulation

The proposed manipulation of the gain table for the unamplified case and the examplary aided Plack2004low case avoids the critical areas of the combfilter at the threshold of 6 dB negative gain. The resulting input-output relation shows a jump at the point where the gain would cause the output level to be within the critical area.

Thus, for the case of no amplification the gain table supplies negative gain for the low frequencies except for the case at 841 Hz, where zero gain is very close to the upper 6 db negative gain line. The resulting higher output level needs to be investigated to ensure reasonable output levels for high input levels.

For the aided Plack2004low amplification rule the premature cutoff of the compressive behaviour needs to be investigated for it may cause strong audible effects and, although comb filtering is mitigated, lead to a worsening of the overall perception of sound. As mentioned before, the true change in perceived sound level can not be directly determined due to the additional effect of the comb filter which in turn changes the output level and thus the difference between attenuated direct sound and processed sound. This again would result in a change in comb filter gain and so on. Hence, the true effect of the comb filter might only be described by a differential function. But this is subject of investigation for a later work on this topic.

Furthermore, the threshold of 6 dB negative gain is only a theoretical measure and serves as a reference point. Because comb filter effects might be audible to up to -18 dB level difference or only 1 dB negative gain, respectively, this threshold might need to be adjusted [1]. In the worst case the playback by the receivers has to be stopped completely, hence infinite negative gain which would result in only the attenuated direct sound being present at the eardrum. Psychoacoustical experiments would need to be conducted to evaluate the threshold and, thus, the extent of the manipulation of the gain table. The listener would be presented with music excerpts with and without gain table manipulation. The task would be to select the version which subjectively results in a better listening experience.

Regarding hearing loss, the manipulation of the gain table has to be individually considered for the respective severity of the hearing loss. Because hearing loss can be compared to the attenuation of the direct sound, different gains become critical. A profound hearing loss would need a higher gain which increases the level difference between direct and processed sound, thus, reducing the effect of the comb filter. On the other hand a weak hearing loss puts the output level closer to the critical area making it necessary to manipulate the signal accordingly.

Regarding a smart hearing protection for, e.g., musicians it would be optimal for the direct sound to be attenuated by more than 18 dB across all frequencies. In this case the resulting comb filter effects would not exceed a negative gain of 1 dB and the manipulation of the gain table would not be necessary. Only in the case of very high input levels, where a negative gain is applied, the attenuated direct sound and the processed sound might interfere destructively with each other. In this case switching off the processing completely might proove to be a sufficient solution. Of course for a strongly occluded ear canal occlusion effects are a major problem again, but since a lot of research is done towards mitigating this effect, a hybrid version of occlusion and comb filter effect mitigation could be possible.

6 Conclusion and Outlook

In this work a psychoacoustical experiment has been conducted with the use of a mobile hearing aid prototype. It has proven to be successful to obtain individual amplitude-phase values for the anti-phase cancellation of pure tones at different frequencies. The aim of the measurement was to retrieve the amplitude-phase relations which would lead to total destructive interference between the direct sound and the processed sound of the hearing aid. These comb filter distortions are ought to be mitigated through a possible manipulation of the gain table.

The data shows only small deviations in phase across all subjects and slightly larger deviations for the amplitude. Overall, the results suggest the possibility of the implementation of a single fit (universal setting). A manipulation of the gain table is necessary for the low frequencies, high frequencies are sufficiently attenuated to not be in the critical area of the comb filter. A possible manipulation of the gain table has been proposed and is subject to testing in another psychoacoustial experiment.

Furthermore, there are some changes that can be made to the psychoacoustical measurement. Repetition of single measurements and an extended subject group should lead to a better precision in the calculation of mean phase and amplitude and would confirm individual accuracy. The measurement presented in this work already showed that especially for the phase the values amongst the subjects did not differ significantly. But finding the actual spot of total destructive interference is quite challenging and thus increasing the amount of subjects would lead to improved values for mean and standard deviation. This would in turn lead to a better determination of the critical frequencies.

Next steps would include implementing the suggested manipulation of the gain table and conducting a psychoacoustical comparison experiment in which manipulated and unaltered sound experpts, including music for strong tonal components, are played back and evaluated in terms of sound perception by the subject. From the outcome the gain table manipulation could be asserted qualitatively. Besides the approach via the gain table an all-pass filter design could also be implemented and tested. Both approaches could prove to be critical in designing a hearing aid or hearing protection that is suitable for musiscians.

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Declaration

Hiermit versichere ich an Eides statt, dass ich diese Arbeit selbstständig verfasst und keine anderen als die angegebenen Quellen und Hilfsmittel benutzt habe. Außerdem versichere ich, dass ich die allgemeinen Prinzipien wissenschaftlicher Arbeit und Veröffentlichung, wie sie in den Leitlinien guter wissenschaftlicher Praxis der Carl von Ossietzky Universität festgelegt sind, befolgt habe.

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Oldenburg, November 7, 2019