In-situ Amplitude-Phase Measurements with a Mobile Hearing Aid Prototype for the Mitigation of Comb Filter Effects by Manipulation of the Gain Table

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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. Due to the superposition of attenuated direct sound and processed sound at the eardrum of the listener comb filtering effects can occur. 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.

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 access the critical amplitude-phase values which would lead to a cancellation of the sound. Hence, an in-situ psychoacoustic measurement is performed in which the listener's task is to find the relative amplitude-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 [1].

From the outcome, the frequency-dependent amplitude-phase combinations can be used to realize a manipulation of the gain table of the device as a compensation method. Possible strategies are proposed in the discussion section. Besides compensation for hearing devices, this approach could also be beneficial for a so-called smart hearing protection.

Methods

Mobile hearing aid prototype

The mobile hearing aid prototype was built from mostly commodity hardware. Binaural earphones were connected via an audio-shield to a Raspberry Pi microcontroller. For the necessary recording gain, a pre-amplifier was used. Power was supplied by a power bank. Implementations of hearing aid algorithms, such as, e.g., multiband dynamic compression, were provided by openMHA [2]. A pre-configured SD-card image with openMHA is available for download [3]. For mobile usage and convenience, a Bluetooth controller can be used to control the prototype such as, e.g., for self-fitting. Calibration methods for the prototype are documented by Buhl et al. (2019) [4].

The direct sound signal was 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.

Comb filter effects

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, which can result in audible signal distortions. Figure 1 shows the paths of the two signals. The superposition of a

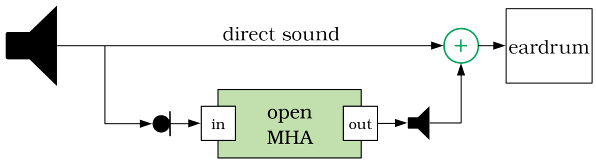

Figure 1: Signal paths of the direct sound and the processed sound from sound source to superposition at the eardrum.

sound source with its delayed and coherent counterpart leads to comb-filtering. Dependent on this time delay Δt_{proc} a frequency-dependent phase shift φ_{proc} for the openMHA path can be determined:

$$\varphi_{proc}(f) = 2\pi f \Delta t_{proc}. \qquad \text{[radians]} \qquad (1)$$

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). The critical positions of the comb filter can be determined as follows:

$$f_c = \frac{1}{\Delta t_{proc}} \cdot n \qquad \text{[Hz]} \qquad (2)$$

$$f_d = \frac{1}{\Delta t_{proc}} \cdot (n - \frac{1}{2}) \qquad \text{[Hz]} \qquad (3)$$

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 integers n.

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

$$s_{ds}(t) = a_{ds} \cdot sin(2\pi ft + \varphi_{ds}), \qquad (4)$$

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

$$s_{proc}(t) = a_{proc} \cdot sin(2\pi ft + \varphi_{proc}), \qquad (5)$$

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

$$s(t) = s_{ds}(t) + s_{proc}(t) = A \cdot sin(2\pi ft + \varphi), \qquad (6)$$

with a superimposed amplitude

$$A = \sqrt{a_{ds}^2 + a_{proc}^2 + 2adsa_{proc}cos(\varphi_{ds} - \varphi_{proc})}, \qquad (7)$$

and a superimposed phase shift

$$tanh\varphi = \frac{a_{ds} \cdot sin(\varphi_{ds}) + a_{proc} \cdot sin(\varphi_{proc})}{a_{ds} \cdot cos(\varphi_{ds}) + a_{proc} \cdot cos(\varphi_{proc})}. \qquad (8)$$

Because only the relative phase is of interest, it is assumed that $\varphi_{ds} = 0$. Furthermore, the amplitude of the resulting wave can be expressed as a comb filter gain G:

$$G = 20 \cdot log_{10}\left(\frac{A}{a_{ds}}\right) \qquad \text{[dB]} \qquad (9)$$

with combined amplitude A from Eq. 6 and amplitude of the direct sound a_{ds}. The strongest effect of the comb filter in terms of change in sound energy will occur for the destructive case which can be expressed as follows:

$$G_d = 20 \cdot log_{10}\left(1 - 10^{-\frac{|\Delta L|}{20}}\right), \qquad \text{[dB]} \qquad (10)$$

with the level difference ΔL between direct and processed sound. Figure 2 shows the amplification level for different phase shifts. The red curve represents the destructive case as described by Eq. 10.

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) [5] 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.

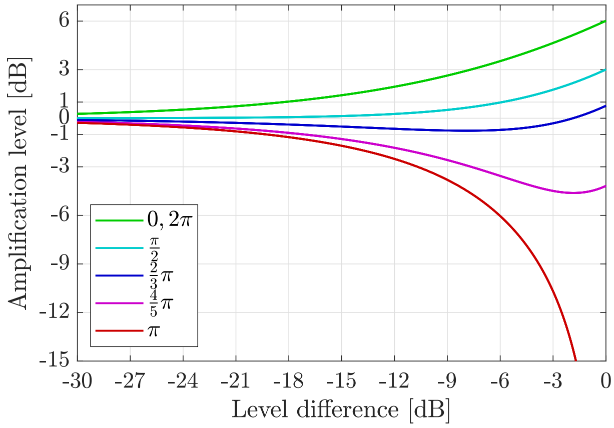

Figure 2: Amplification level dependent on the level difference between the direct and processed sound for different phases.

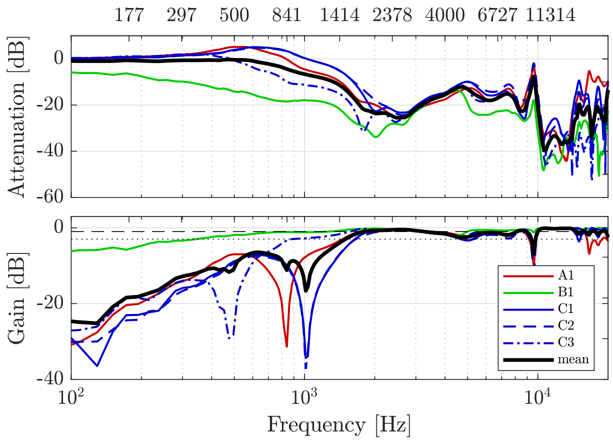

Figure 3: Attenuation of the earphones (upper panel) in the left ear for a signal coming directly from the left (S_{270}) for different earphones of the same kind (A, B, C) and repeated insertions of the same earphone (C1, C2, C3). The average of the whole set of measurements is represented in bold black. Resulting comb filter gain (lower panel) for the destructive case of $\varphi = \pi$.

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 (upper panel) shows the frequency-dependent attenuation for several pairs of earphones, which was measured in an anechoic chamber with a dummy head [6].

From the attenuation a critical comb filter gain can be determined as seen in the lower panel of Figure 3. The graphs show the worst case scenario, if it is assumed that all frequencies are 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 Eq. 10. The highest gain occurs for frequencies up to 2 kHz. For higher frequencies the gain tends to to stay between -3 and 0 dB. 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 represent a crossing of the 0 dB attenuation line or values close to 0 dB.

In-situ amplitude-phase measurement

An in-situ amplitude-phase measurement was conducted in which the subject's task was to adjust the controls of an Octave GUI 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 4 shows the schematic of the psychoacoustical measurement.

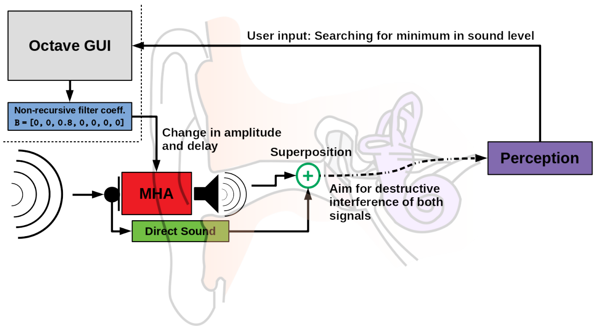

Figure 4: Implementation of the in-situ amplitude-phase measurement.

Nine subjects were placed in front of a desk in a soundproof cabin operating an Octave GUI (Figure 5) on a computer. Pure tones at the center frequencies of the prototype were played back around 70 dB SPL from a single loudspeaker to the left of the subject in line with the left ear. The subject was wearing both earphones of the prototype with the left earphone operating normally and the right earphone playing back a white noise signal to mask the sound from the loudspeaker. In this way the tones would only be perceived on the left side.

The task of the measurement was for the subjects to adjust the phase and amplitude until the pure tone on the left side was not perceived anymore. By clicking inside a box, where the horizontal axis controlled the phase shift, and the vertical axis the gain, the subjects were instructed to first adjust the phase until a minimum in sound level of the pure tone was achieved. Secondly, keeping the phase constant, the amplitude was chosen until the tone ideally was canceled out completely. After saving the specific amplitude-phase combinations the next frequency was chosen and the process repeated.

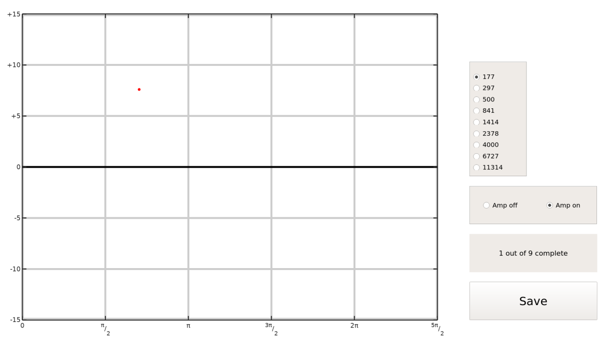

Figure 5: Octave GUI with a horizontal axis for the phase adjustment and a vertical axis for the amplitude adjustment. The code for the GUI is available in the project repository [7].

For the manipulation of the processed signal, a FIR-filter design was used. The non-recursive filter coefficients were 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

$$d = \frac{\varphi_{MHA}}{2\pi} \cdot \frac{f_s}{f} \tag{11}$$

with frequency f, sampling frequency f_s, and φ_{MHA} as the phase adjustment by the subject. The amplification level is set through the amplitude adjustment a_{MHA} and is inserted at the previously determined digital delay d:

$$b[n] = [0\,0\,10^{\frac{a_{MHA}}{20}}\;0\;\ldots]. \tag{12}$$

A discrete convolution between the recorded signal of the incident sound wave and the FIR-filter coefficients is performed. By this operation, the signal is shifted in time according to the adjusted phase and amplified according to the adjusted amplitude.

Results

Figure 6 shows the results of the amplitude-phase measurement. The phase axis ranges 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 are marked with large squares and error bars visualize the standard deviation.

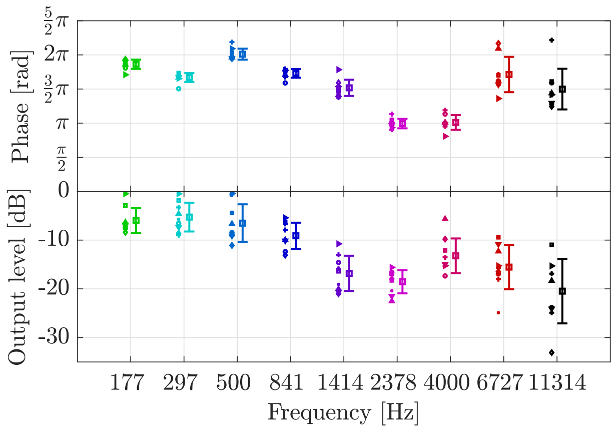

Figure 6: Measurement data for the phase and amplitude across all measured frequencies. Standard deviations are shown as error bars from the mean.

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. The standard deviations for the phase are rather small for frequencies up to 4000 Hz. Above, the standard deviation exceeds the value of $\frac{\pi}{4}$. For the amplitude the standard deviations are very similar across all frequencies with, again, an increase to larger frequencies. Low- and mid-range frequencies show a standard deviation of around 3 dB.

From the amplitude values the attenuation of the earphones and the resulting potential destructive gain can be directly determined and compared with the preexisting data from the measurement on the dummy head as seen in Figure 7. The attenuation is depicted in the upper panel and the destructive gain in the lower panel. It can be seen that most of the data points lie within one standard deviation from the mean.

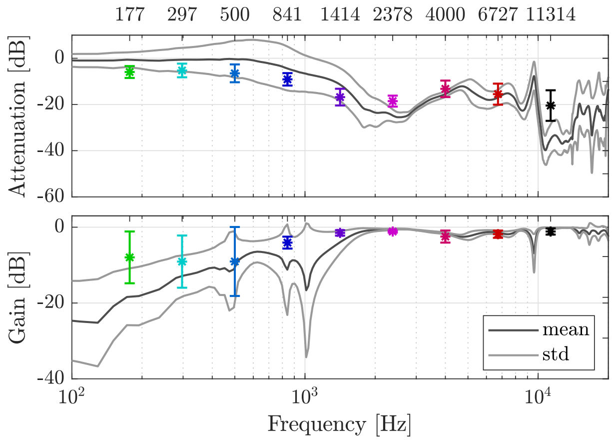

Figure 7: Experimental attenuation (upper panel) and the corresponding destructive gain (lower panel) in comparison to the mean values from the dummy head measurement.

A compensation approach was implemented which mutes the channels with center frequencies that could result in a change in gain by more than -6 dB due to comb-filtering. In pilot listening experiments with noise and music stimuli the original and manipulated settings were distinguishable. However, none of the settings were immediately recognizable as more natural.

Discussion and Conclusions

The results for the amplitude-phase measurement show consistent data across all subjects which indicates that the proposed measurement procedure produced reliable data. Sound cancellation by destructive interference of sinusoidal waves at the eardrum can be exploited for measurements of relative transfer functions.

Small deviations for the phase suggest a conformity of the individual phase relations across all subjects and could make the implementation of a single fit possible. Large deviations for high frequencies can be explained by the limited resolution of the phase adjustment dependent on the sampling rate of the prototype. The deviations for the amplitude (never below 1 dB) might be explained by a potential masking of the pure tones by the contralateral white noise. Also, high frequencies suffer from the limited phase resolution, as amplitude and phase are dependent on each other.

The attenuation data from the measurement and the dummy head showed a similar trend. Deviations could be explained by the differences in anatomy of the human ear and the dummy ear. Furthermore, variability across different earphones due to the manufacturing process may have caused the slight offset from the mean of the dummy head. Generally, at frequencies where the difference in sound levels between direct sound and playback is pronounced, e.g., at high frequencies in our experiment, the effect of the comb filter can possibly be neglected. The main artifacts due to comb filtering were located by the measurements at frequencies below 1 kHz, as could be expected for the employed earphones from the attenuation measured with the dummy head. However, if the gain table would be configured to provide negative gains, similar levels for direct sound and playback, and hence comb filter artifacts will be the result.

A proposed gain table manipulation for an unamplified case avoids the critical areas of the comb filter at the threshold of 6 dB negative gain. Low frequencies need to be attenuated, while high frequencies are not critical. Regarding hearing loss, the manipulation has to be individually considered for the respective severity of the hearing loss, because if the gain table implements compression amplification, the relative levels of direct sound and playback sound also depend on the input level.

For a smart hearing protection based on a platform similar to the hearing aid prototype, the maximum achievable stable attenuation is the passive attenuation of the earphone, which is the case when the playback is muted. For listeners with impaired hearing, attenuation is only desired at high input levels, while at low input levels amplification is beneficial for them. Hence, the point of equal levels between direct sound and playback has to be crossed when the input level changes from low to high. For this application, a mitigation of the comb filter effect would be especially interesting.

In the future, repeated measurements could be used to evaluate the reliability of the measurement. The proposed method to manipulate the gain table should be evaluated in listening experiments with the aim to test if the perceived sound quality improves.

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