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Acoustic Transparency in Hearables - Technical Evaluation

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An increasing number of earphones and other hearing devices contain functionalities that are based on a so-called hear-through feature, which allows the user to hear the acoustic environment through the device. Ideally, the user would perceive the hear-through sound identical to listening with the open ear, which is referred to as acoustic transparency. In technical terms, this means that the sound transmission to the eardrum should be as similar as possible between the open ear and through the device. In this study, we evaluate the acoustic transparency of the hear-through feature of seven commercial hearables as well as two research hearing devices by means of technical measurements on a dummy head. A variety of artefacts, including frequency response deviations, comb filtering artefacts, and destruction of spatial cues, were revealed and quantified, and surprisingly large differences between current devices are noted. The corresponding subjective sound quality has been assessed in a companion study.

0 INTRODUCTION

Ear-worn technical devices for consumer applications now go far beyond earphones but include voice control and telephony, Active Noise Control (ANC), augmented reality, or assistive listening features [1–4]. Recently, the term hearables has been coined for this class of wearable hearing technology. One basic feature in many hearables is a so-called hear-through feature that allows the user to hear the external sound environment electro-acoustically through the device. Depending on the application, the hear-through sound can be modified, e.g., by amplification or attenuation, noise reduction, and custom equalization, or augmented by additional sound sources. The sound quality of the hear-through feature is thus an important criterion for the performance of hearables. In the optimum case for hear-through, referred to as acoustic transparency, the user would not perceive any difference between listening with the open ear and using the hear-through feature [5, 6]. In this work, the acoustic transparency of seven commercial hearables and two research hearing devices is assessed by means of electro-acoustic measurements on a dummy head. In a companion paper [7], the perceptual sound quality of the same devices is evaluated and discussed in the view of the physical deviations assessed in this work.

A hear-through feature is implemented by routing the signal picked up by a microphone in the ear to the headphone driver after appropriate filtering [5, 6, 8]. The design objective is usually that the transfer function to the eardrum with the device inserted is equivalent to the appropriate transfer function of the open ear, which is referred to as Head-Related Transfer Function (HRTF). Acoustic transparency in a physical sense is thus achieved if the transmission properties of the hear-through device (measured at the eardrum) are identical to that of the open ear. Different approaches to this target have been presented and evaluated, making use of generic filters for analog [8] and digital systems [6, 9], individualized filter designs [5, 10, 11], adaptive filtering techniques [12], and directionally dependent equalization [13, 14]. In the evaluation studies, the authors usually only included their own custom devices in several settings, also due to the unavailability of commercial reference devices. Typical known artefacts of hear-through processing include frequency response deviations, delay and comb-filtering artefacts, distorted spatial cues and distortions of spatial hearing, or non-linear artefacts like clipping [5, 9, 15, 16]. Furthermore, wearing hear-through devices is often perceived as disturbing when the user is speaking or chewing, due to the occlusion effect [16].

We here present a comparative technical evaluation of a set of current commercial hearables and research hearing devices. An overview is given on the general performance and differences between hear-through features of state-of-the-art devices. Following the objectives for realizing acoustic transparency, the technical evaluation presented here is mostly based on a measurement of the HRTF with the device inserted and hear-through turned on and several metrics derived from it, which are compared to the appropriate open-ear data. In addition, the linearity of the devices is assessed, which includes an assessment of the self-noise of the devices as well as their level dependence. Through a joint analysis of several devices, we expect to reveal a more comprehensive selection of artefacts that occur in hear-through devices than it would be possible by evaluating single devices. Together with the psychoacoustic results presented in a companion paper [7], the present data therefore pinpoint the most critical factors for realizing convincing acoustic transparency in hearables as a basis for future improvements.

1 METHODS

1.1 Devices

Seven commercial hearables listed in Table 1 were included in the measurements. The selection included products on the market by June 2019, when this study was initiated. In the following, these devices are anonymized and referred to as Devices A to G. Devices A to C classify themselves as hearing assistive devices; Devices D to G are wireless earphones with additional functionalities (see "Purpose" column of Table 1). All devices are in-the-ear device styles, i.e., sit in the ear canal and fill considerable parts of the cavum concha. Table 1 also shows some technical parameters of the devices. They were either cable-connected between left and right earpiece (Wireless No) or true wireless devices (Yes). The devices were controlled by their respective apps from an Android smartphone and updated to the latest firmware as of June 1st 2019.

Additionally, two research hearing devices developed at the University of Oldenburg (UOL) were included in the measurements. The first device is based on commercial earphones with integrated binaural microphones (Roland CS10-EM) connected to a RaspberryPI and referred to as UOL Commodity device [17, 18]. The second one is the so-called "acoustically transparent earpiece" (UOL Tr. Earpiece) [5, 11]. It is based on a custom electro-acoustic earmold and was connected to an RME Fireface UCX soundcard and a small form-factor Intel NUC PC. In both devices, the Oldenburg open Master Hearing Aid (openMHA [19, 20]) was used for real-time processing. For the UOL Commodity device, the hear-through feature that is default after startup was used [17]. The Tr. Earpiece was automatically calibrated for the KEMAR after insertion as described in [11]. Processing included a multi-microphone feedback reduction approach [21] as well as an equalization filter to adapt the frequency response to an estimation of the open-ear response [10]. All filters were computed based on transfer function measurements made in-situ, which required one external sound source (here over-the-ear headphones). The data shown in this paper correspond to the Tr. Earpiece In-Ear microphone (IE) condition in the companion paper [7].

Measurements were conducted for the devices set in three modes. For the Hear-Through mode, the hear-through feature was activated and all other sound processing options deactivated. The hear-through feature was available under different names for the devices, which are listed in Table 1. Additionally, measurements were performed for the devices turned off (Device Off mode) and with the devices turned on but hear-through disabled (World Off mode). In the World Off mode, ANC features were activated in devices that had them. In the other devices, the World Off mode was very similar to the Device Off mode regarding the behavior for external sound sources.

1.2 Apparatus

The devices were inserted into a G.R.A.S. KEMAR dummy head type 45BB-12 with anthropometric pinnae and low-noise ear simulators. In the following, measurements at the ear simulators are referred to as eardrum measurements. The anthropometric pinnae facilitated realistic and tight fitting of the devices in the ear. The low-noise ear simulators have a self-noise level near or below the human threshold of hearing and facilitated direct measurement of the self-noise of the devices [22, 23]. To ensure a tight fit, measurements were repeated between 10 and 20 times with reinsertion and the run with the highest and most left-right consistent attenuation of external sounds (Device Off mode) selected for further evaluation.

The KEMAR was placed in an anechoic chamber featuring a 3D array of 94 Genelec 8030 loudspeakers at a variable 2.5–3 m distance from the center point (see Fig. 1). While transfer functions from all possible incidence directions were measured, the results are presented for two subsets only. The first subset of 53 incidence directions is uniformly distributed on a sphere except for the missing loudspeaker below. It consists of 5 loudspeaker rings at elevations 0°, $\pm30^\circ$, and $\pm60^\circ$ with azimuthal resolutions of 22.5°, 30°, and 60°, respectively, as well as one loudspeaker directly above the center. Power spectrum averaging over responses from these incidence directions thus well approximate diffuse-field conditions [24]. The second subset includes all 48 loudspeakers installed in the horizontal plane with a uniform spacing of 7.5°. The loudspeakers were mounted vertically and given the distance of >2.5 m from the center, the spatial separation of woofer and tweeter (approx. 1.3°) can be neglected. The KEMAR was positioned at the array center point and rotated such that the Interaural Time Difference (ITD) for frontal incidence was below 1 sample at the used sampling rate of 44.1 kHz.

1.3 Measurements

Measurements were performed in two separate sessions. In the first session, the HRTF of the KEMAR was measured for the open-ear case as well as with the devices inserted for all directions using exponential sweeps. The sweeps covered a frequency range from 30 Hz to 22.05 kHz (half the sampling rate), were 4.2 seconds in length, and created an average presentation level of 72 dB SPL in free field. To speed up the measurements, the multiple exponential sweep

Table 1. Tested hearing devices and parameters. See text for further details.

Manufacturer	Device	Purpose	Wireless	Feature Name
Bose	Hearphones	Hearing Support	No	World 0/Neutral
NuHeara	IQBuds BOOST	Hearing Support	Yes	Neutral
Wear&Hear	BeHear NOW	Hearing Support	No	Live Music, Neutral
Bang&Olufsen	Beoplay E8	Wireless Earphone	Yes	Transparency
Bragi	TheDASH Pro	Wireless Earphone	Yes	Transparency
Jabra	Evolve 65t	Wireless Earphone	Yes	Hear-Through
Sony	WF1000X	Wireless Earphone	Yes	Environment Normal
UOL	Commodity Hearing Aid	Research Device	No	Amplification Off
UOL	Transparent Earpiece	Research Device	No	Transparent, InEar

method was used, leading to an overall duration of 32 s for all 94 incidence directions [25]. To verify a linear operation of the devices, the HRTFs for frontal and diffuse-field incidence were measured additionally using white noise and single exponential sweeps at presentation levels between 65 and 90 dB SPL. Without entering the chamber, all measurements were repeated for the three operation modes of each device. Free-field reference responses were recorded using an omnidirectional 1/8" microphone (G.R.A.S. 46DP-1) positioned at the location of the center of the KEMAR head.

In the second session, the devices were reinserted into the same KEMAR to measure their self-noise directly using the low-noise ear simulators. Great care was taken to achieve similar passive attenuation of external sounds as in the previous round of measurements to ensure a similar fit. These measurements were conducted in the same anechoic chamber with all equipment turned off to ensure a low acoustic background noise floor. Samples of 10 s of the self-noise generated by each of the devices in their hear-through mode were recorded binaurally.

1.4 Data Processing and Evaluation

In the measured impulse responses, the influence of the loudspeakers were compensated for by means of regularized spectral division by their pseudo-anechoic response (measured using the omnidirectional microphone and windowed according to [26]). The resulting Head-Related Impulse Responses (HRIRs) still contain reflections from the loudspeaker array starting approx. 13 ms after the direct sound (see also Fig. 4, top panel). However, further processing, e.g., windowing to exclude this reflection, was not performed since many of the devices include processing delays and components in this delay range of the impulse response. The reflections cause spectral ripples of less than ± 2 dB, which is small compared to the effect of hear-through devices and common to all results. All following evaluations are based on the full measured HRIR that was stored with a length of 92 ms. Further computations for data evaluation purposes are described in the appropriate subsections of Sec. 2.

2 RESULTS AND ANALYSIS

2.1 Linearity and Artefacts

Artefacts introduced by the devices in Hear-Through mode were assessed by inspecting recordings of single sweep measurements as well as by comparing the HRTFs determined with the different measurement signals and at different levels. This was not possible for Device G, since its sensors detecting the insertion into an ear did not allow for continuous operation in the dummy head. In consequence, only the multiple exponential sweep measurements could be conducted before it turned itself off. While the devices

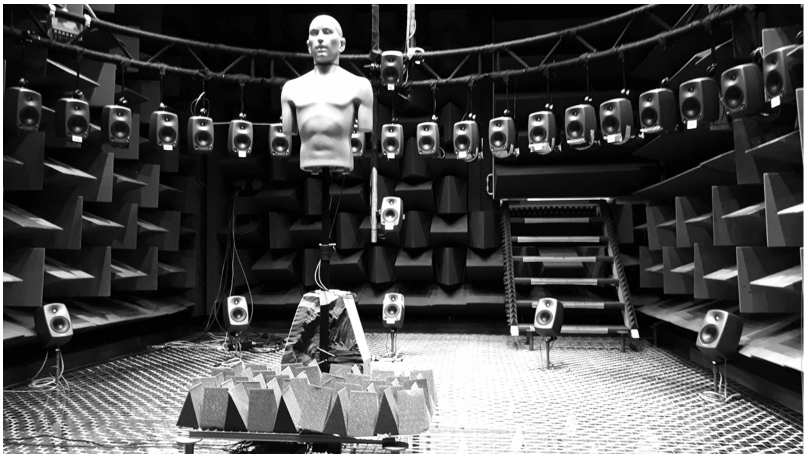

Fig. 1. Photograph of the measurement setup. The KEMAR was mounted freely in the anechoic chamber with its stand covered in absorbers.

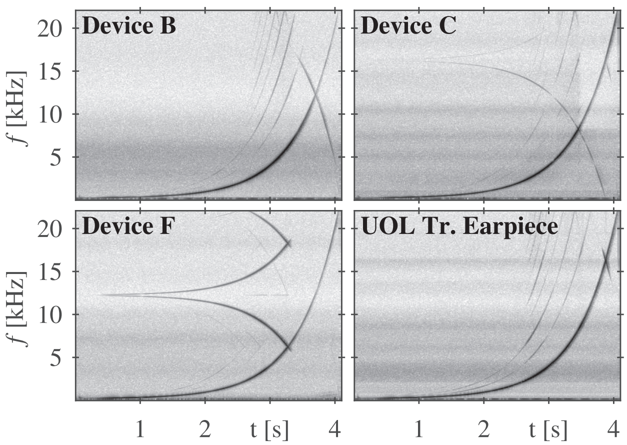

Fig. 2. Spectrograms of single sweep recordings made at the left eardrum of KEMAR with the devices (as denoted in panel title) in Hear-Through mode with 75 dB SPL free-field level. Dark colors indicate high level; individual panels are to scale.

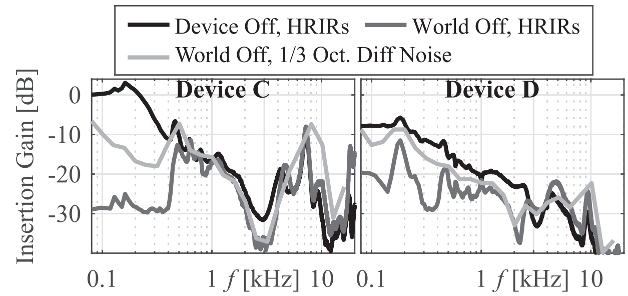

Fig. 3. Insertion gains in Device Off and World Off modes in devices that included an ANC feature, derived from HRIR measurements with multiple sweeps and a 1/3 octave band level analysis with diffuse noise.

behaved generally as expected for linear time-invariant operation, several deviations were noted:

- Weak resampling artefacts visible in the single exponential sweep recordings in Devices A, B, and C and the UOL Tr. Earpiece, as well as stronger resampling artefacts in Device F (left side only). Sample spectrograms of the sweep recordings are shown in Fig. 2; resampling artefacts are visible as downward sweeps and additional upward sweeps (e.g., starting at approx. 12 kHz in Device F).
- Weak saturation non-linearities visible in the single exponential sweep recordings of Devices A, B, and C and both UOL devices (see harmonic components for exemplary devices in Fig. 2 and [27]).
- Level reduction at frequencies > 8 kHz and presentation levels exceeding 80 dB SPL in Device C, independent of the measurement signal.
- Broadband attenuation of the listening level for presentation levels above 75 dB SPL in Device A independent of the measurement signal. For example, the insertion gain was reduced by approx. 8 dB when the presentation level was increased from 72 to 90 dB with diffuse white noise excitation.

Nevertheless, the devices in Hear-Through mode behaved approximately linear and time-invariant in the level range assessed with the multiple exponential sweep measurements. Thus, the HRTFs measured using this technique can be taken as a valid representation of the acoustic properties of the devices in regular operation conditions. However, these results also show that linear time-invariant operation and occurrence of non-linear processing should generally be verified in such measurements.

For Devices C and D, an influence of the measurement signal was evident in the World Off mode, i.e., where ANC feature is active. Fig. 3 shows the insertion gains, i.e., the ratio of pressure at eardrum with device inserted and open ear, measured in Device Off mode and World Off mode (i.e., with ANC on and Hear-Through off). The insertion gain was derived both from the multiple sweep impulse response measurements and 1/3 octave level analyses with diffuse white noise [28] with similar levels. In both devices the impact of ANC, i.e., the decreased insertion gain, is overestimated in the HRIR-based analysis as compared to the 1/3 octave level band analysis. This difference is probably caused by the very different correlations of the test signals and should be considered in further analyses. A 1/3 octave band level analysis of the multiple sweep test signal recording yielded results very similar to those based on the HRIRs. It should be noted that the responses in Hear-Through or Device Off mode were not affected by the measurement signal.

2.2 Hear-Through Impulse Responses

Fig. 4 shows the HRIRs at the eardrum for frontal incidence. The top panel shows the open-ear case, which is the reference case to be approximated for acoustic transparency. Only one peak is visible around 0 ms delay, which is identical at both sides. The main peak is shifted to the right for all devices except Device D (see last paragraph of this section for details on this device) and differs substantially in shape. The shift results from the processing delays of the devices. Estimates of the delays were computed by the temporal difference between the maxima of the Hilbert envelopes of the HRIRs with the Hear-Through and Device Off modes. The delays are given in Fig. 4 for the individual devices and sides. A large span of processing delays is observed, ranging from close to 0 (Device D) to approx. 10 ms (Device A). While in most devices the delay is identical up to 0.1 ms between left and right, considerable differences of the delay between left and right are seen for Devices E, F, and G. Especially for Device F, the left-right difference of 10 ms is striking. It should be noted that these three devices are true wireless.

Additionally to the delayed main peak, in many devices a first peak is visible at the temporal position of the open-ear HRIR. This is a sound component that acoustically leaks into the ear canal with a level that is dependent on the fit in the ear canal. A spectral analysis of the leakage component is provided in Sec. 2.3 based on measurements

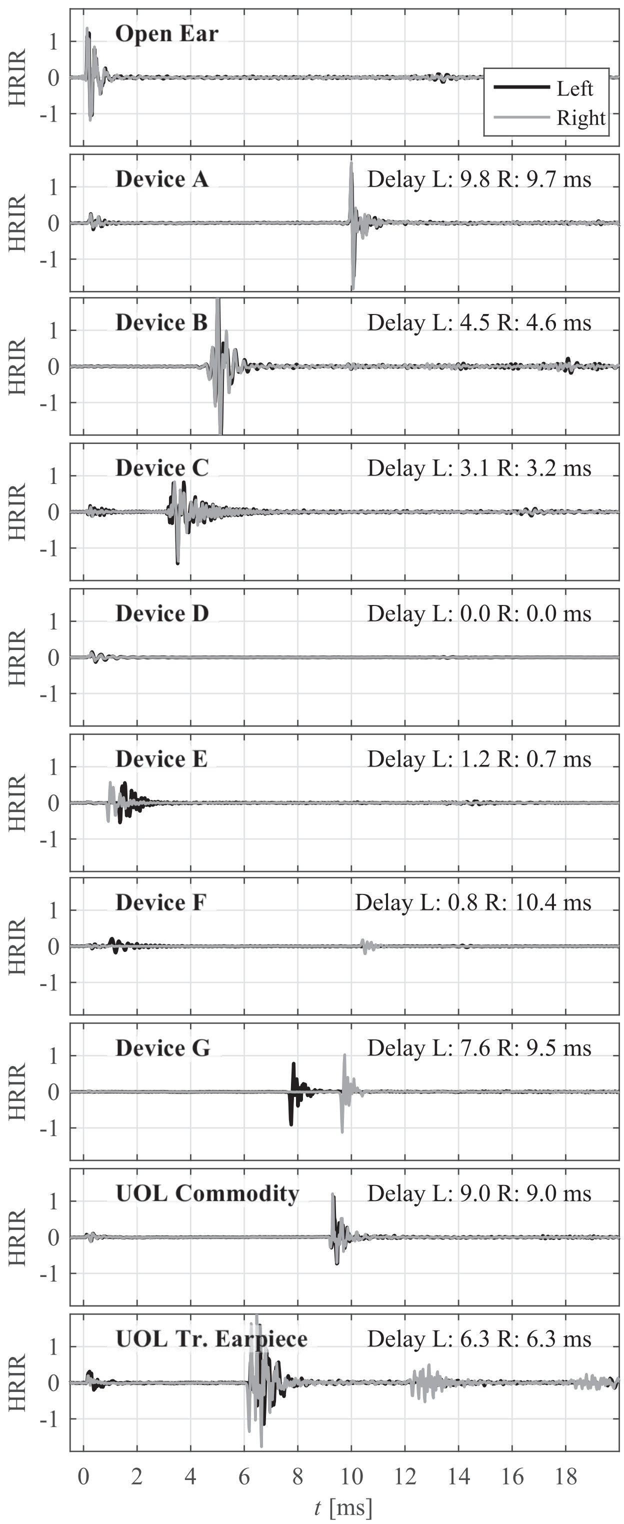

Fig. 4. HRIRs to the eardrum for frontal incidence and both ears, open ear (top panel) and with devices in hear-through mode.

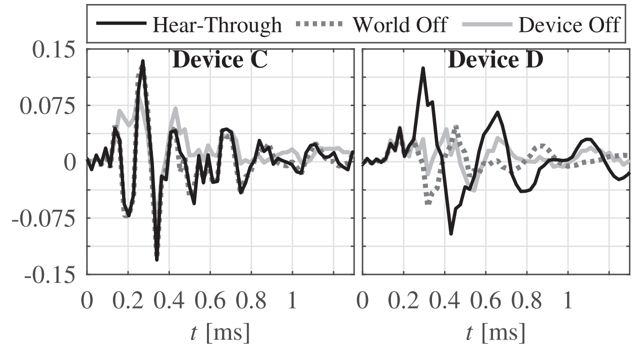

Fig. 5. Early HRIRs for devices including an ANC feature (C and D), taken from the measurements at the right ear.

of the Device Off mode. The leakage component is most pronounced in Device A and the UOL Tr. Earpiece and well visible in Devices C and F and the UOL Commodity device. The other devices provide a significant attenuation of the leakage component, at least with the fit achieved in this evaluation. In the UOL Tr. Earpiece, additional peaks are visible after the main peak (around 13 and 19 ms) due to occurrence of decaying feedback in this very open-fit device.

Only for Devices C and D, which include an ANC feature, the HRIR part in the temporal range of to the leakage component differs between the three modes of the device. Fig. 5 shows a zoom of the early part of the HRIRs from Fig. 4 for these two devices. The effect of ANC with respect to the Device Off mode is evident for both the Hear-Through and World Off modes, however differently in both devices. In Device C, the early HRIR is identical in the Hear-Through and World Off modes, both differing from the Device Off condition. In Device C, the ANC seems to be used solely to suppress the leakage component in all operation modes, and the hear-through feature is provided by a secondary signal path that has a delay of about 3.1 ms (c.f. Fig. 4). In Device D, all three conditions differ notably, showing that the hear-through feature is implemented by changing the same low-latency signal path that is used for reducing ambient noise by ANC. This is consistent with the very low delay of this device given in Fig. 4. It should be noted that the HRIR shown here does not reflect the properties of the ANC processing perfectly, as discussed in Sec. 2.1. However, we interpret differences between modes as clear evidence for differences in the low-latency output of the devices.

2.3 Frequency Responses at the Eardrum

Fig. 6 shows the responses at the eardrum with the devices inserted. All responses were smoothed over 1/24 octave prior to further calculations (power spectrum smoothing, Hann window [29]). The responses are shown for the subset of 53 incidence directions uniformly distributed on a sphere (thin gray lines) as well as diffuse-field averages (root-mean-square of amplitude values) over these incidence directions (see Sec. 1.2). For comparison, the response of the open ear (diffuse-field incidence) is given by dashed lines.

The leftmost column of Fig. 6 shows the occluded responses, i.e., the transfer function to the eardrum with the passive device inserted. Except for Device C, this response is equal to that of the leakage component visible in Fig. 4. All devices show a similar dependence of the occluded

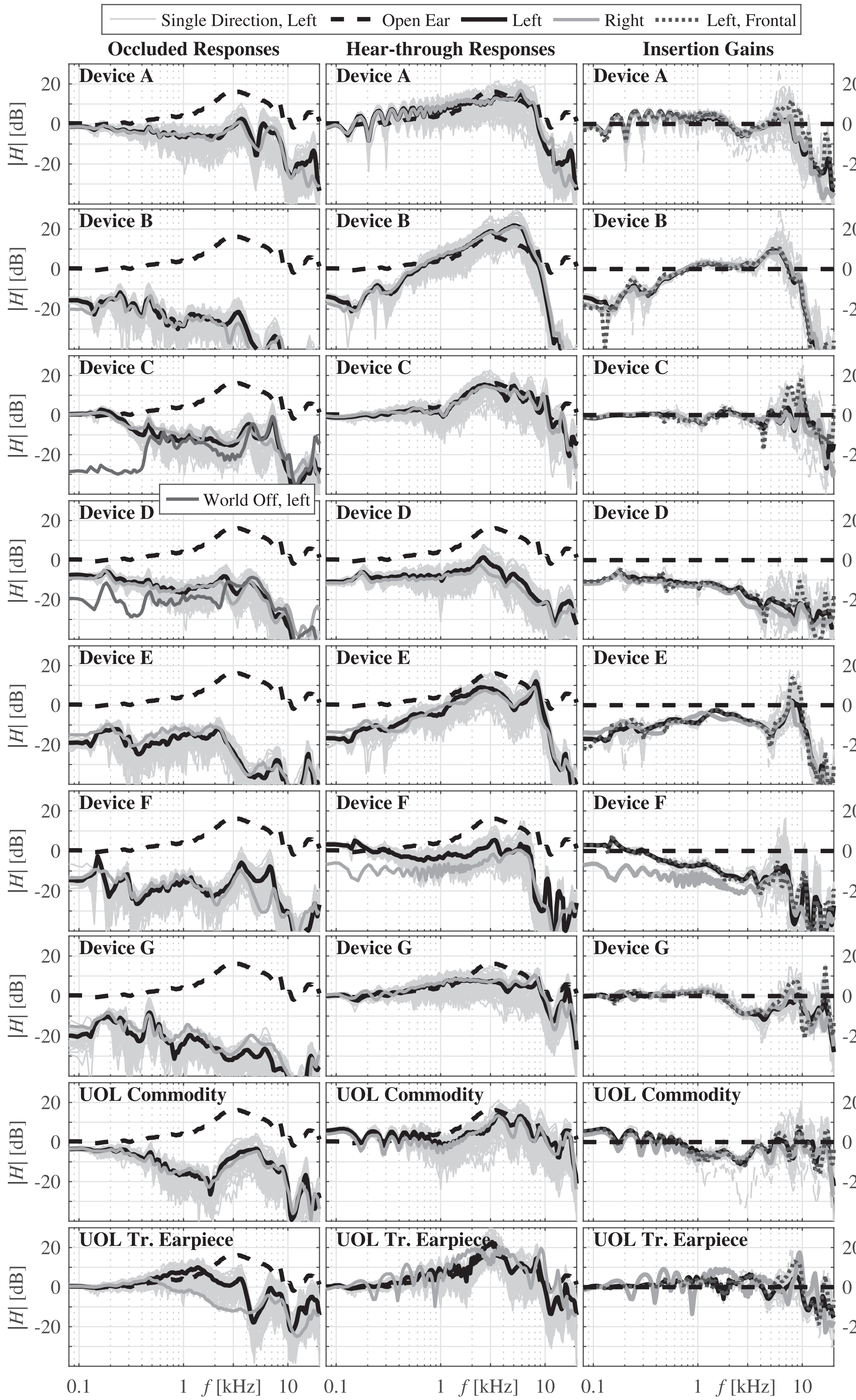

Fig. 6. HRTFs at the eardrum for the Device Off (Occluded Resonse, left column) and the hear-through case (middle column). The right column shows the hear-through insertion gains. For comparison, the open-ear responses for diffuse-field incidence are given by the dashed lines. Individual incidence directions (thin light lines) are plotted for the left ear. Thick solid lines show the diffuse-field responses for the left and right ears. For the insertion gains, the frontal incidence (dotted dark gray line) is given additionally, and the influence of ANC on the occluded response for Devices C and D (diffuse field, dark gray line).

response across incidence directions and an excellent match (mostly below 2 dB difference) between left and right. The attenuation characteristics can be roughly divided in two groups. The first group can be described as a highly attenuating fit and comprises Devices B, D, E, F, and G. Their occluded responses are characterized by a low-frequency attenuation of 10 dB (average below 1 kHz) or more with respect to the open-ear case, which increases at higher frequencies up to 30 dB and more. This is achieved by a tight sealing of the ear canal.

The other group comprises Devices A and C and both UOL research hearing devices and shows attenuation curves that are typical for loose-fit earphones or vented fits [30, 31]. In this group, an incomplete sealing of the ear canal leads to higher occluded responses as compared to the first group. In the UOL Tr. Earpiece, the attenuation of left and right devices is notably different. We verified through further measurements that this is caused by different sizes of the vents of these custom hand-made prototype devices. The vent is larger in the left device, leading to a higher occluded response only for frequencies >2 kHz. Around 1 kHz, the diffuse-field occluded response is even amplified with respect to the open-ear response, which is caused by a Helmholtz resonance of the residual ear canal and the vent opening [31]. In Devices C and D, the occluded responses in the World Off mode include an additional attenuation achieved by ANC, most pronounced in the low-frequency region. In Device D, this reduction comes additional to the larger passive attenuation with varying ANC attenuation (± 8 dB) at frequencies < 2kHz. Contrarily, in Device C the ANC mainly causes an additional attenuation in the low-frequency region < 500 Hz that is very uniform (± 2 dB).

The middle column shows the hear-through responses at the eardrum in the same manner as for the occluded responses. This response at the eardrum is a superposition of the occluded response and the output of the device. Disturbing interactions of both sound components therefore occur if the processing delay (see Fig. 4) is non-negligible and if the occluded response contributes significantly to the total hear-through response. Such superpositions lead to spectral ripples that are referred to as comb filtering effects [32, 33]. These comb filtering effects can be seen well in Devices A and F (right side only) and both UOL research hearing devices, i.e., those devices where the leakage component is only weakly attenuated. The occurrence is mostly restricted to lower frequencies (< 1 kHz), i.e., where the leakage component is most pronounced. Although Device C is loosely fitting and includes a considerable processing delay, the occurrence of comb filtering effects seems to be avoided by further attenuation of the leakage component using ANC (c.f. Sec. 2.2). We verified that the reduction of comb-filtering effects in the high frequency seen here is not an effect of the applied smoothing. In Device B, comb filtering effects occur only in the very low frequencies (notches at approx. 110 and 330 Hz, consistent with 4.5 ms delay), where the hear-through response rolls off. Apart from Device F and the UOL Tr. Earpiece, the occurrence of comb filtering effects is very symmetric across sides. In Device F, the sole occurrence at the right ear seems to be caused by the lower hear-through response at the right ear, and the larger delay at this side (10 ms vs. 1 ms). In the UOL Tr. Earpiece, the differences between the responses at the left and right ear can be explained by the differences in the attenuation properties of both sides. The less pronounced comb-filtering in the left side (which has higher leakage) is explained by independent in-situ calculation of filters at both sides, which reduces output in the low frequencies depending on the estimated leakage [10, 11].[1]

The difference between hear-through and open-ear response is plotted in the right column of Fig. 6 by means of the insertion gain, which was computed by subtracting the appropriate open-ear responses (in dB) from the hear-through responses. A perfect match is obtained if the insertion gain is zero at all frequencies and for all incidence directions. The best match of the coarse hear-through response to the open-ear response is achieved in Devices C and A and the UOL research hearing devices (maximum deviation except comb filter ripple below 8 dB up to 8 kHz). Deviations often include a main ear resonance (peak around 3 kHz in open-ear response) that is too flat or missing as in Devices D, F, and G and to weaker extent in Devices E and A, and the UOL commodity device. Only in Device B, the response around the main ear resonance is too high, and a low-frequency roll-off in the Hear-Through mode is seen already below 800 Hz. Considering the occluded responses with this device, it can be ruled out that this roll-off is caused by a poor fit. Also, in Devices D and F and to a smaller extent in E, the level of the overall hear-through response is lower than the open-ear response.[2] Generally, it can be stated that the hearing assistive devices (A–C) and research hearing devices tend to achieve a better match to the open-ear response than the earphones with additional functionalities (D–G).

The insertion gain is plotted for the frontal incidence in addition to the diffuse-field incidence for the left ear in the right column of Fig. 6. Free and diffuse-field insertion gains are largely similar up to 4 kHz, i.e., when spectral directional pinna cues play a role. For devices where both curves differ considerably (Devices A, C, E, and G), the free-field insertion gain includes an excess amplification around 8 kHz, and the devices seem to be tuned to match the open-ear response for diffuse-field incidence (c.f. [24]). The difference between free and diffuse-field insertion gain at 110 Hz in Device B is probably caused by the dominant leakage component and comb filtering effects.

2.4 Binaural Cues

Fig. 7 shows binaural cues for the open-ear and hear-through cases, namely the Interaural Time Difference (ITD), Interaural Level Difference (ILD), and interaural

[1] A new hardware platform for this device was developed and made available to the public [34], which is expected to eliminate this left/right asymmetry.

[2] The surprisingly large level offset is consistent with the subjective listening impression and assessment made in [7].

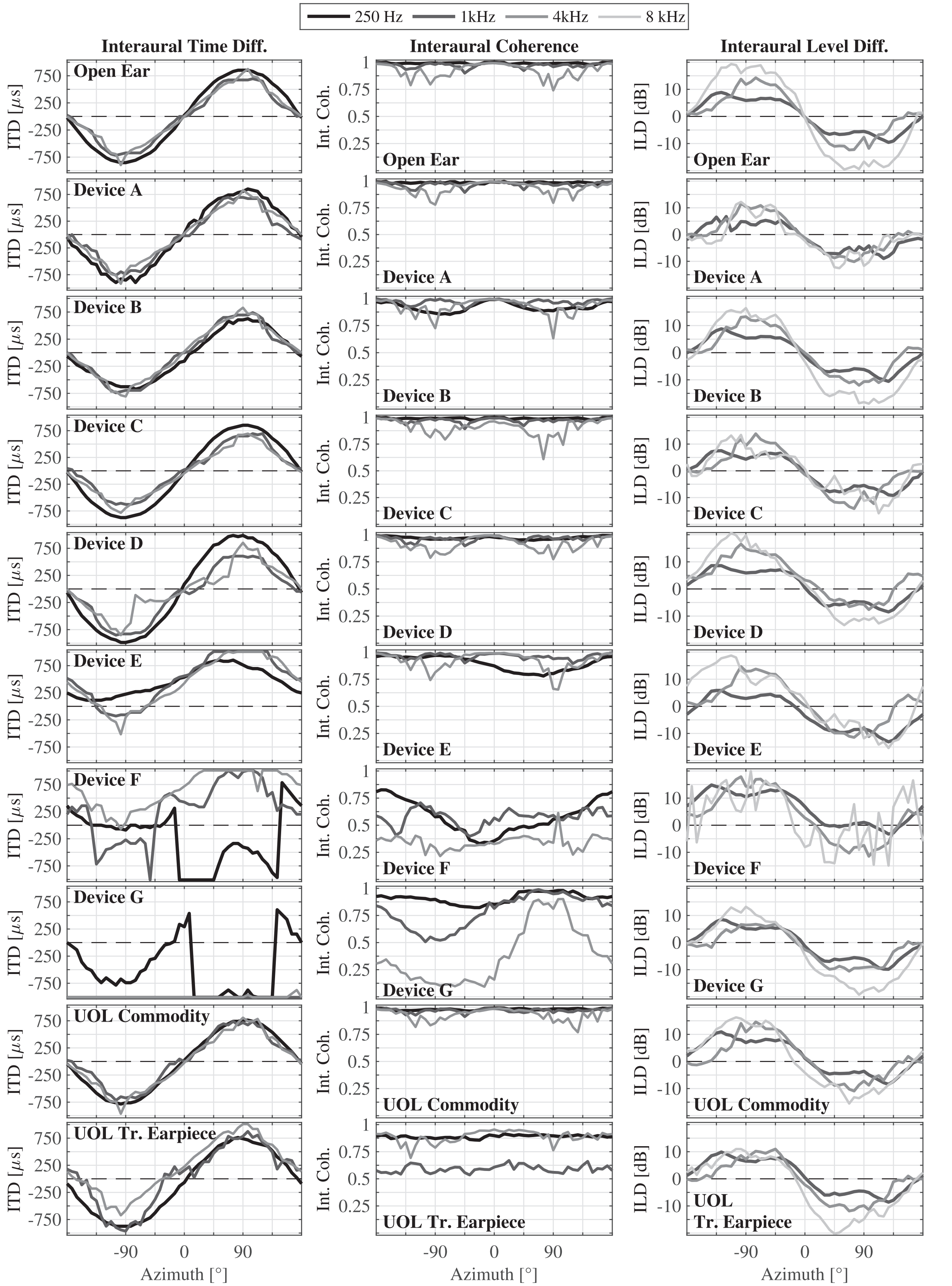

Fig. 7. Binaural cues as denoted at the top of each column with the open ear (top row) and all devices in the hear-through mode. Each panel shows the dependence of the binaural cue on the azimuth in the horizontal plane for three frequencies, where negative angles denote the left-hand hemisphere. Note that different sets of frequencies are plotted in the three columns.

coherence. The binaural cues were determined for third octave bands (third order Butterworth Bandpass) around the frequencies indicated in Fig. 7 to approximate auditory filters. The ILD was computed by taking the ratio (left/right) of the average power in the bandpass filtered HRIRs. For computation of the ITD and Interaural Coherence, a simple representation for peripheral processing was included [35]. This included half-wave rectification followed by low-pass filtering (1.4 kHz Brickwall filter) of the band-pass HRIRs. Based on the result, an interaural cross-correlation function was computed for each incidence direction and frequency band. The ITD was then determined as the position of the peak of the interaural cross correlation function within a lag of ±1 ms [36]. The interaural coherence was determined by the peak value of the same cross-correlation function [37].

The ITDs are shown in the left column of Fig. 7. In Devices A–C and the UOL Commodity device, the ITD is very similar to that of the open ear at all assessed frequencies. Some distortions, particularly for the ITD at 1 and 4 kHz, are seen in Device D and the UOL Tr. Earpiece. In Device E, the ITD somewhat follows a natural course but includes a bias of about 0.5 ms toward positive interaural delays, which is consistent with the delay difference between the left and right device (see Fig. 4). In Devices F and G, the ITD is distorted to a degree that it can be stated that this cue is completely destroyed. In these devices, the delay difference between the left and right device exceeded the 1 ms lag range that was utilized as the maximum range for the ITD calculation [35, 36]. This results in somewhat arbitrary ITD courses, although the delay difference between left and right is consistent across incidence directions.

The distortion of the ITD usually comes with a reduction of the interaural coherence shown in the middle column of Fig. 7, which is another consequence of the poor left-right synchronization, particularly in Devices F and G. An exception to this observation is the UOL Tr. Earpiece, where the ITD is reasonably well conserved but the interaural coherence is lowered, especially for the 1 kHz band. This can again be explained based on the different properties of the left and right device revealed in Fig. 6, since the leakage component dominates the hear-through response for this frequency at the left but not the right side. Neither the ITD nor the interaural coherence are distorted due to the presence of a processing delay or the occurrence of comb filtering effects alone (see esp. Device A, Figs. 4 and 6). Parameters that lead to a distortion of these interaural time cues are poor synchronization between devices even if the delay difference is constant (Devices E, F, and G) as well as variable attenuation properties between left and right (UOL Tr. Earpiece). The latter point demonstrates that a difference in fit may also induce a distortion of binaural cues that is not observed if the devices are fitted as properly as in the present data in both ears.

The conservation of ILDs as shown in the right column of Fig. 7 is generally better as compared to the interaural time cues. In most devices, the ILDs are similar to that of the open ear, especially at 1 and 4 kHz. At all devices a distortion of the ILD at 8 kHz and especially azimuths around $\pm90°$ can be seen. This can be attributed to the microphone position

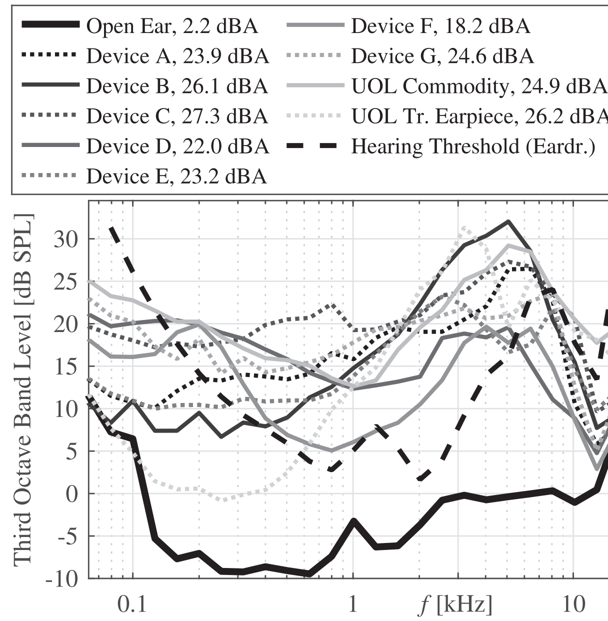

Fig. 8. Third-octave levels of the self-noise of the devices at the eardrum. Values in the legend indicate the diffuse-field equivalent A-weighted broadband level of the self-noise.

in the filled cavum conchae that alters spatial cues with respect to the open-ear case [38]. Apart from that, the ILD curves are shifted upward in Device F, which is consistent with the broadband insertion gain differences between left and right seen in Fig. 6. Similar shifts can be seen at single frequencies of some other devices (e.g., 4 kHz of Device D, 1 kHz of Device E).

2.5 Self-Noise

The frequency-resolved levels of the self-noise generated by the devices are shown in Fig. 8. The recordings were passed through a third-octave filterbank (third order Butterworth) and the Root-Mean Square (RMS) pressure of the bandpass signals computed and subsequently averaged between ears. For comparison, the equivalent open-ear recording is shown, verifying that ambient noise was at least 10 dB below the devices' self-noise in a frequency range above 150 Hz. The hearing threshold at the eardrum[3] is also plotted for comparison. The legend of Fig. 8 also denotes the appropriate diffuse-field equivalent broadband levels for each condition. To compute this value, the narrow-band RMS pressures were A-weighted and corrected by the inverse diffuse-field response of the KEMAR measured in this study and summed for frequencies starting at 80 Hz to exclude the influence of ambient acoustic noise.

In a wide frequency range (approx. 400–7 kHz), the third-octave levels of the self-noise of all devices exceeds the hearing threshold considerably, meaning that the self-noise is clearly audible in silent environments below 30–40 dBA. Large differences between the devices are noted, with dif-

[3]Free-field hearing thresholds from ISO226:2003 transformed by free-field response of KEMAR measured in this study.

ferences of more than 15 dB in some frequency bands, and a span of 9 dBA diffuse-field equivalent broadband level. It should be noted that Devices D and F, which produced the lowest noise floors, also provide a hear-through level that is lower than the open ear in most frequency regions (c.f. Fig. 6). For the other devices, the diffuse-field equivalent broadband noise level is in the range of the noise levels of current miniature electret condenser and MEMS microphones, which were most likely utilized in these devices [39].

All devices except Device F and the UOL Tr. Earpiece show a structurally similar noise spectrum, with a flat curve below 1 kHz (corresponding to a pink noise) and a broad peak around 4–5 kHz. This spectral shape can be explained by an approximately pink noise generated by the microphone, and coloration by the equalized response of the driver at the eardrum that should roughly resemble that of the open ear [6, 24]; see also Sec. 2.3. In the UOL Tr. Earpiece, the equalization included a high-pass filtering due to the effects of the vent [10] that explains the decrease of self-noise below 1 kHz.

3 DISCUSSION

Large differences regarding the acoustic transparency of the hear-through feature are noted between all tested devices. A summary of the performance parameters assessed in this study is given in Table 2. For parameters where no single number could be determined out of the measurements, qualitative summary ratings on a four-point scale are given. While some devices conserved most transmission properties of the open ear rather well, others introduced a variety of significant deviations against open-ear listening. These include:

- Deviations to the frequency response of the open ear, which could be perceived as unnatural [5].
- Processing delays of up to 10 ms, which is well in the audible range if the leakage component is of relevant level. In many devices, the superposition of both sound components resulted in visible spectral ripples or comb filtering effects. Such distortions may lead to a hollow and disturbing sound effect [32, 33].
- Temporal misalignment between left and right: In 3 out of 5 tested true wireless devices, the processing delay between left and right device differed by 0.5 ms or more with a maximum of 9.5 ms difference.
- Effects of asymmetries between devices, such as different presentation levels or a difference between comb filtering effects, at the left and right ears.
- A biased listening level, in most cases lower than normal.
- Substantial self-noise of the device that is audible in silent everyday environments.

While the variation between state-of-the-art devices and the diversity of occurring artefacts is rather surprising, it is safe to assume (due to the design of the current study) that most of the deviations compared to open-ear listening in present hear-through devices were revealed in this work. The present data are therefore a rich basis to pinpoint the perceptually most disturbing artefacts and make relations between various physical deviations and perceptual impact. A psychoacoustic evaluation of the sound quality of the devices tested here and a comparison to the deviations revealed in this work are made in a companion study [7]. Without detailing on the psychophysical results, it can already be stated here that the device that minimized the differences to the open ear (however had the highest self-noise), Device C, also received the highest subjective quality ratings.

Nevertheless, the data in this study provide insights on means to avoid such artefacts in the first place. Given the observations in the current data, a hear-through feature that approximates acoustic transparency is characterized by:

1. Synchronization and symmetry between left and right device.
2. Appropriate equalization to the open-ear response.
3. Minimization of delay and comb filtering effects.

The first point is a rather technical issue that can be solved with well-known signal processing methods and appropriate electric and electro-acoustic design of the devices. It should be noted that left-right asynchrony was only observed here in true wireless devices, where hear-through processing is probably implemented on two separate processors. Acoustic basics for appropriate equalization as well as several approaches to filter design are also well known and can be transferred to the individual devices [6, 8, 10, 24]. The present results show that it is possible to achieve satisfactory equalization at least for a median ear.

The minimization of delay and comb filtering effects, however, is a more complicated issue that is subject to several constraints and principal limitations given by the individual device. One obvious solution would be to reduce the delay to the μs range, which may not always be possible. Many devices in the present study use a completely close fit, which attenuates leakage components to a degree that interactions with a delayed device output and thus delay artefacts are avoided. Besides possible negative effects on the wearing comfort, problems with this approach may occur in practice if a good seal cannot be achieved by the end user. Rather loose or even open fits are less sensitive to such variations. Device C demonstrates how the benefits of a loose fit can be exploited while at the same time effects of the processing delay that is present in this device are minimized by the additional use of ANC to actively suppress the leakage component. Reducing the spectral effects of comb filtering effects by appropriate equalization might reduce the perceptual impact of a delay also in open-fit devices; however it is arguable whether this is (robustly) possible [4, 10].

While in Fig. 6 a substantial directional dependence is seen in the occluded and hear-through responses across a broad frequency range, the insertion gain is mostly independent (variation below $\pm$ 3 dB) on incidence direction

Table 2. Summary of results. The best result of each column is emphasized by boldface letters, the poorest by italics. Qualitative ratings are indicated by a four-point scale between "++" (very good) and "--" (very poor).

	Delay [ms] L/R	Fit	Response Match to Open Ear	Comb-Filtering Avoiding Ripples	Binaural Cues Conservation	Self-Noise [dBA DF]
Device A	*9.7*	Loose	+	-	++	23.9
Device B	4.5	Tight	-	+	++	26.1
Device C	3.1	Loose+ANC	**++**	**++**	**++**	*27.3*
Device D	**0.0**	Tight+ANC	--	++	+	22.0
Device E	1.2 / 0.7	Tight	-	++	-	23.2
Device F	0.8 / 10.4	Tight	--	-	--	**18.2**
Device G	7.6 / 9.5	Tight	+	++	--	24.6
UOL Commodity	9.0	Loose	+	-	++	24.9
UOL Tr. Earpiece	6.3	Vented	++	-	+	26.2

for frequencies up to approx. 4 kHz. Directional variations of the insertion gain in lower frequency regions are only seen if the occluded response dominates the hear-through response (e.g., Device E around 250 Hz). A principal limit for the conservation of spectral directional cues in the high-frequency region is given by biased HRTF cues that are captured by the device microphones due to filling of the cavum conchae [24, 38, 40]. These deviations would translate directly to a directionally dependent insertion gain if no directionally dependent hear-through processing is employed [14]. While we can only state for sure that a directionally independent processing was active in the UOL devices, none of the other devices eliminated these directional dependences of the insertion gain. Thus, either a non-directional processing was used or its directionality was not sufficient to generate an effect clear enough to be noticed in this evaluation. In conclusion, hear-through processing of the present devices neither improved nor impaired the spectral directional cues that can be expected from the device style and microphone position [38].

Finally, some limitations of the current study should be discussed. First, the measurements in a KEMAR cannot assess the variation over subjects originating from variable ear acoustics even in a use case without additional problems, e.g., through a non-ideal fit of the device. The ease and reliability of fit in individual users could affect the performance in practice quite drastically, as discussed above. It is not unlikely that some manufacturers have tuned their devices to match the acoustics of the KEMAR, which should be kept in mind particularly when evaluating the match to the open-ear response (c.f. Fig. 6). Second, many parameters associated with the user experience of hearing devices cannot be assessed in mannequin-based measurements. These include, for example, the perception of the own voice or activities like chewing, the wearing comfort, and other non-acoustic factors. Third, only one pair of each devices was tested, which means it cannot be ruled out that the results of single devices are caused by faults rather than their usual performance. We want to stress that we do not claim to establish a performance comparison or ranking of devices from different manufacturers but rather see this work as a basis for general improvements in achieving acoustic transparency by pinpointing deviations against open-ear listening that do occur in practical applications.

4 CONCLUSION

We evaluated the hear-through feature of 7 current commercial hearables as well as two research hearing devices. To this end, their acoustic transmission properties to the eardrum were analyzed and compared to the open-ear case based on directionally resolved impulse response measurements in a KEMAR. A surprisingly large variance between devices is noted that ranges from a very good conservation of the open-ear transmission properties to an introduction of severe artefacts including frequency response colorations, comb filtering effects, level bias, and destruction of binaural cues. Tested devices including hearing support features and the research hearing devices tended to perform better than earphones with additional functionalities. The present data in combination with a subjective sound quality evaluation of the devices presented in a companion paper [7] facilitate identification of the most crucial factors for establishing objectively and perceptually convincing transparency in hearing devices.

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