I. INTRODUCTION

Professional musicians are routinely subjected to sound pressure levels (SPL) in the range of 80 to 117 dB(A) (Patel, 2008). Despite the risk of noise induced hearing loss associated with those high levels, hearing protection devices (HPDs) usage statistics remain low: 6% in Finland, 15% in Denmark and about 10% in Germany (Huttunen et al., 2013). Musicians have reported the modified perception of their own voice or instrument when wearing hearing protection as one of the main reasons for disliking HPDs (Fabiocchi, 2010; Huttunen et al., 2013; Patel, 2008; Santoni and Fiorini, 2010). This perceptual discomfort is partially attributable to the non-uniform attenuation across frequencies provided by HPDs, but also to the occlusion effect (OE). While non-uniform attenuation may be addressed to some extent by passive musician’s earplug (Huttunen et al., 2013), hear-through equalization (OBrien et al., 2014) or the use of sound insulating earphones as in-ear monitors (Santucci, 2009), OE remains challenging to address effectively.

The OE is an unnatural and annoying perception of one’s own voice (Borges et al., 2014) when wearing HPDs that modifies the audio-phonation loop, the feedback system by which how one hears oneself (Bouserhal et al., 2019). It can potentially affect singers or more generally any musician whose instrument induces vibrations to the skull, such as a trumpet or violin. These vibrations reach the earcanal walls through bone conduction, causing them to vibrate and induce pressure changes on the air contained in the occluded earcanal, producing an acoustical wave that is picked up by the auditory system (Carillo et al., 2019). When the earcanal is unoccluded, its low frequency acoustic impedance is low and the volume velocity introduced by the vibrating earcanal walls does not translate into significant SPL, thus in-ear SPL is dominated by the sound wave arriving from the air conduction path between the source (e.g. vocal tract) and the ear. However, when the earcanal is occluded, its low frequency impedance is higher and the volume velocity introduced by the vibrating earcanal walls translates into high SPL (Stenfelt and Reinfeldt, 2007), thus in-ear SPL is dominated by the sound wave resulting from bone conduction since the air conduction path is blocked. The result is an augmented and unnatural "boomy" perception of ones own voice (Killion, 1988) or instrument.

Solutions have been proposed to reduce the severity of OE introduced by earplug type HPDs, including increasing the depth of insertion (Berger, 2003). While it reduces the OE that is experienced, increasing the depth of insertion can still leave residual OE in the range of 9 dB (Dean and Martin, 2000) and would benefit from further reduction. Additionally, if used excessively, deep insertion of earplugs can cause discomfort to singers, since the earplug impedes earcanal deformation caused by mandibular movement which can reach 2.3 mm (Darker et al., 2007).

Another solution that has been explored is the feedback active noise control (ANC) of in-ear SPL resulting from OE. Feedback ANC of the OE can take many forms, but can be regrouped in two large categories: fixed or adaptive. Because earcanals and eardrums vary greatly in shapes and acoustic properties across users, large inter-user occluded earcanal acoustic impedance variations must be accounted for (Goldstein et al., 2005; Liebich et al., 2019). Therefore, fixed systems must be designed with a compromise on performance to allow for sufficient safety margins to avoid instability (Bernier and Voix, 2013; Mejia et al., 2008). In addition, the specific OE reduction that is achieved on one user depends on the specific occluded earcanal acoustic impedance, its low frequency acoustic impedance is low and the volume velocity introduced by the vibrating earcanal walls does not translate into significant SPL, thus in-ear SPL is dominated by the sound wave arriving from the air conduction path between the source (e.g. vocal tract) and the ear. However, when the earcanal is occluded, its low frequency impedance is higher and the volume velocity introduced by the vibrating earcanal walls translates into high SPL (Stenfelt and Reinfeldt, 2007), thus in-ear SPL is dominated by the sound wave resulting from bone conduction since the air conduction path is blocked. The result is an augmented and unnatural "boomy" perception of ones own voice (Killion, 1988) or instrument.

Solutions have been proposed to reduce the severity of OE introduced by earplug type HPDs, including increasing the depth of insertion (Berger, 2003). While it reduces the OE that is experienced, increasing the depth of insertion can still leave residual OE in the range of 9 dB (Dean and Martin, 2000) and would benefit from further reduction. Furthermore, if used excessively, deep insertion of earplugs can cause discomfort to singers, since the earplug impedes earcanal deformation caused by mandibular movement which can reach 2.3 mm (Darker et al., 2007).

Another solution that has been explored is the feedback active noise control (ANC) of in-ear SPL resulting from OE. Feedback ANC of the OE can take many forms, but can be regrouped in two large categories: fixed or adaptive. Because earcanals and eardrums vary greatly in shapes and acoustic properties across users, large inter-user occluded earcanal acoustic impedance variations must be accounted for (Goldstein et al., 2005; Liebich et al., 2019). Therefore, fixed systems must be designed with a compromise on performance to allow for sufficient safety margins to avoid instability (Bernier and Voix, 2013; Mejia et al., 2008). In addition, the specific OE reduction that is achieved on one user depends on the specific occluded earcanal acoustic impedance,
meaning that large inter-user variation is observed in the amount of reduction. On the other hand, adaptive systems present their own challenges that depend on the principles they rely on. Some systems rely on an adaptive filter in the feedback loop that aims to continuously minimize an error criterion, such as least mean squares finite impulse response filters. By attempting to minimize an error criterion, these algorithms inherently adjust to the occluded earcanal impedance and so they do not suffer from the same performance trade-off of fixed systems. For that reason, they can reach levels of OE reduction up to 26 dB in some cases (Sunohara et al., 2014). However, they offer OE reduction that varies with time and can depend on the stability of the signal to minimize, which could sound unnatural to the user. By design, these algorithms do not try to achieve a specific target performance and rather attempt to reduce in-ear SPL as much as possible, which can lead to unpredictable results and variation in user experiences. Another interesting approach involves the use of a fixed system with an adaptive gain in the feedback loop to reduce the amount of feedback, if necessary, to ensure stability (Liebich et al., 2018). However, because this adaptive gain is a scalar and is not frequency dependent, the achieved OE reduction is still subject to inter-user variation. These variations, inherent to both fixed and adaptive systems, need to be considered when designing OE reduction systems. They were identified as the reason for outliers’ dissatisfaction with OE reduction systems in subjective studies (Liebich et al., 2018; Mejia et al., 2008).

The approaches mentioned above are sound and offer solutions that have the potential to alleviate some musicians’ complaints about OE, but none of them attempt to reach a defined target OE reduction across all users that does not vary with time. A system that guarantees a precise target OE reduction across users could allow for more control in subjective studies and potentially lead to a better understanding of how objective OE reduction relates to subjective improvement. Alternatively, a desirable target OE reduction that an adaptive system would aim to achieve could be individually calculated from measurements or obtained from user input. Such a strategy seems desirable to achieve perceived naturalness of the audio-phonation loop, since the magnitude and frequency distribution of the experienced occlusion effect varies from user to user (Dean and Martin, 2000).

One core challenging aspect of feedback ANC of the OE is the acoustic design of the earpiece that is paramount to the success of the technique. Previous research has relied on modifying existing earpieces (Liebich et al., 2018), or have not detailed the design process (Mejia et al., 2008; Sunohara et al., 2014) of a well-performing earpiece suitable for feedback ANC. There is a need to document this often overlooked aspect so that researchers can design earpieces that meet their specific needs including providing passive hearing protection.

This paper presents a different approach to an adaptive OE reduction system, one that aims to offer a given OE reduction target, within the system’s capabilities, despite inter-user occluded earcanal acoustic impedance variation. The method relies on an earpiece acoustic design that is suitable for feedback ANC, which is the foundation of such an OE reduction system, and for which a design process is described. The methodology is presented in section II, describing the theory of operation of the method, the earpiece design and modeling, along with the details of the adaptive algorithm. Validation of the system and algorithm are presented using a model of an adjustable earcanal simulator and a database from the literature in section III. Implementation and measurements on an adjustable earcanal simulator are presented in section IV, followed by a discussion and conclusions in sections V and VI respectively.

II. METHODOLOGY

A. Theory of operation

The proposed OE reduction system relies on a sound insulating earpiece enabled with feedback ANC to attenuate undesired low frequency bone conducted voice or instrument content inside the occluded earcanal. An in-ear microphone (IEM) at the inner end of the earpiece captures this undesired signal, or disturbance, and transduces it into an electrical signal. This electrical signal is then processed in such a manner that, when played back in real-time through an in-ear loudspeaker (IELS) comprised in the aforementioned earpiece inside the occluded earcanal, it interferes destructively with the original disturbance, causing an overall reduction of the magnitude of the OE.

The OE reduction performance curve $T(s)$ is characterized by equation 1 where $H(s)$ is the plant response and $G(s)$ is the compensation response. In this case, the plant response is the transfer function from the input of the IELS to the output of the IEM, while the compensation response is the transfer function of the inversion and spectral shaping of the disturbance captured by the IEM.

Because the plant response is affected differently depending on the acoustic impedance of the earcanals of a given user, the compensation must be adapted accordingly if the same target OE reduction curve is to be offered to all users. In addition, it is desirable that this adaptation occurs automatically, promptly and in-situ for OE reduction to occur shortly after inserting the earpieces.

$$T(s) = \frac{1}{1 + G(s)H(s)}$$ (1)

The proposed methodology to automatically achieve the same target OE reduction across users involves the following steps:

1. Assign a target OE reduction frequency curve, as shown in figure 1(A). While the methodology to define an ideal target is out of the scope of this paper, it could potentially result from an estimation of the actual objective occlusion effect derived from measurements involving an outer-ear microphone (OEM) and an IEM on a given user to esti-
mate the experienced occlusion effect. It could also be defined by user’s preference, or by the experimenter in the context of a subjective study.

2. Estimate the plant response binaurally on a given user in an identification phase following insertion of the earpieces containing an IEM and an IELS. Figure 1(B) shows conceptual plant responses in one ear of two users.

3. Compute the theoretical compensation response that would ensure the target OE reduction curve given the estimated plant response, according to equation 2. Figure 1(C) shows conceptual theoretical compensations calculated from the plant responses of figure 1(B).

4. Design a practical compensation that approximates the theoretical compensation using causal filters. Figure 2(A) shows a conceptual practical compensation approximating a theoretical compensation with causal filters.

5. Verify the projected stability of the closed-loop system from the projected open-loop system response by calculating gain and phase margins and ensuring that they are sufficient before proceeding.

6. Activate the practical compensation, and optionally measure the open-loop response of the system, from the input of the compensation, through the plant, to the output of the IEM to ensure compliance with projected open-loop response.

7. Close the loop, activating OE reduction. Figure 2(B) shows conceptual OE reduction curves that could be obtained on two users, matching the target reduction with the exception of some amplification above approximately 1 kHz. This type of amplification is typical to feedback ANC systems and occurs where ideal phase characteristics cannot be obtained with causal filters, as illustrated in figure 2(A).

\[
G(s) = \frac{1 - T(s)}{T(s)H(s)} \tag{2}
\]

While the maximum performance that can be achieved using the proposed strategy depends on both the plant response and the compensation, the latter can only compensate the former to some extent. Much of the potential for performance is dictated by the suitability of initial plant response for ANC, which depends on the earpiece acoustic design.

B. Electro-acoustic modeling and design

A two-port network model of the proposed earpiece is constructed to better understand the system, support the prototype earpiece design, validate the method and predict achievable target OE reduction under various conditions.

Design challenges for the proposed prototype earpiece are numerous. To be a complete solution for a musician HPD, it must not only be able to address OE, but also eventually serve as a sound insulating in-ear monitor and/or offer uniform attenuation. While outside the scope of this paper, the earpiece is designed accordingly by featuring a combination of a sound insulating eartip for passive protection and an OEM to capture external sounds to which audio processing would be applied before playback through the IELS. Additionally, since OE is capable of causing in-ear SPL exceeding 110 dB(SPL) (Killion, 1988), the earpiece should feature transducers that are capable of handling high SPL at low frequencies. Furthermore, a design target is established regarding high frequency reproduction capabilities. Indeed, during preliminary trials with professional musicians, complaints about the lack of high frequency content had been received about intermediary prototypes that were unable to produce sound energy above 12 kHz (Bernier, 2013). Therefore, to be more likely accepted by professional musicians, a design target of 16 kHz is adopted for the proposed prototype earpiece.

Given the design targets, a dual loudspeakers design with a passive crossover filter is chosen. Because the device is to function as an HPD, back-ported loudspeakers are avoided to prevent sound transmission to the earcanal through the loudspeaker’s port and its thin membrane. While the loudspeaker’s port could be coupled to an inside volume within the earpiece to isolate it from ambient sound, it would make the earpiece bigger than necessary, which is undesirable for smaller ears and would add volume inside the concha, which could potentially affect head-related transfer functions and interfere with sound localization (Brungart et al., 2003) and potentially with perceived naturalness of the provided uniform attenuation.

1. Earpiece transducers

To serve as a woofer for the earpiece, the receiver CI-22955-000 (Knowles Electronics, Itasca, Illinois, USA) is selected because it is a relatively large driver with a low resonance frequency that is able to produce a maximum SPL of 133 dB(SPL), defined by the manufacturer as the SPL above which more than 10% total harmonic distortion (THD) occurs. This woofer also provides less than 1% THD at 120 dB(SPL) between 40 and 100 Hz. To serve as a tweeter, the Knowles WBFK-30095-000 receiver is selected because of its small size that allowed positioning near the inner end of the earpiece, deep into the earcanal, to help achieve the 16 kHz frequency response requirement. Indeed, one design consideration when attempting to reproduce high frequency content inside a closed earcanal is to limit the length and/or maximize the diameter of the sound channel that couples the tweeter to the earcanal to avoid an acoustic lowpass behavior. This can be visualized if the sound channel is thought of as an acoustic mass, modeled by an inductor, and the closed earcanal as an acoustic compliance, modeled by a shunt capacitor. Simulation Program with Integrated Circuit...
Emphasis (SPICE) electro-acoustic models of both the CI-22955-000 and the WB FK-30095-000 are conveniently provided by Knowles to aid in design, making these great choices for this study.

MATLAB (MathWorks, Natick, Massachusetts, USA) software is used to carry the design process to be able to use two-port network modeling and to perform numerical operations on the frequency response curves to simulate the theory of operation discussed in section II. A. To achieve the transition from the SPICE environment to the MATLAB environment, an ABCD matrix is constructed from the Hunt parameters of the loudspeakers that are calculated using the method detailed by (Kim and Allen, 2016) with revised appendix equations by (Bernier et al., 2013) and (Kim and Allen, 2016). This method requires a minimum of three electrical impedance curves obtained when three known loads are presented to the loudspeaker. In this study, the impedance curves are simulated using PSpice Designer (Cadence Design Systems, Inc., San Jose, California, USA) when the loudspeaker model is virtually connected to lossy tubes models.

Knowles FG-23652-P16 microphones are selected for the IEM and OEM because of their uniform frequency response, small size and convenient tubular sound port that can be easily sealed to the housing of an earpiece. However, their max SPL before saturation has been measured to be around 114 dB(SPL) (Lillywhite, 2013), which may be borderline in the context of OE reduction.

2. Earcanal acoustic simulator

A custom variable earcanal acoustic simulator, based on a medical plastic syringe, is built and modeled to emulate variable lengths of earcanal. It resembles the custom adjustable earcanal simulator made by (Hiipakka et al., 2010), dubbed ADECS, except that an additional CI-22955-000 receiver is mounted on its side wall. The purpose of this additional loudspeaker is to inject sound into the earcanal simulator independently of the earpiece to mimic the OE disturbance and is therefore referred to as occlusion effect loudspeaker (OELS). The constructed ADECs with OELS is used to take measurements in section IV. The OELS also acts as a parallel Helmholtz resonator that lowers the total impedance of the simulator below 2kHz. A similar characteristic is observed in standard earcanal simulators which include parallel Helmholtz resonators to emulate the impedance of the middle ear (Briel et al., 1976). The ADECS with OELS model is used in this study to simulate representative acoustic loads to support the earpiece design process and validate the robustness of the method to variation in acoustic ear canal impedance in section III. It is modeled with T-shaped impedance blocks, which allowed tuning of the resistive elements modeling energy loss due to friction and wall absorption (Stenfelt and Reinfeldt, 2007).

Figure 3 shows the modeled impedance curves of the ADECS with OELS for various lengths, along a model of the impedance of an IEC 60318-4 compliant coupler for reference. The reference coupler is a Brüel & Kjær (Sound and Vibration Measurement A/S, Nærum, Denmark) wideband ear simulator type 4195, modeled using the electro-acoustical equivalent circuit provided by its manufacturer (Briel & Kjær, 2012).

3. Earpiece mechanical design

The earpiece mechanical design incorporates the woofer, tweeter, and both microphones in a plastic shell that mates with the earcanal through an noise insulating eartip. The selected eartip is the Comply TX-500 (Hearing Components, Inc., Oakdale, Minnesota, USA), a roll-down foam with a hollow tube through it selected for its inner tube large enough to accept the earpiece spout in which the tweeter is embedded. A connection PCB was made to hold the cross-over and achieve ease of assembly and more reliability. The proposed design of the musician earpiece is shown in figure 4. The sound channels were modeled as tubes of varying cross-section using T-shaped impedance blocks (Stenfelt and Reinfeldt, 2007) and their physical size is a compromise between desirable acoustic simulation results, space constraints, and ease of assembly. This earpiece model and that of the ADECS with OELS allow simulation of the projected acoustic response of the earpiece as it is designed and modified in SolidWorks (Waltham, Massachusetts, United States)

The modeled combined frequency response of both loudspeakers is compared to the measurement in the ADECS with OELS for a length of 16 mm in figure 5. This length is derived from average earcanal length of 25 mm (Henry and Letowski, 2007), minus an insertion depth of the earpiece of about 9 mm. Good agreement is generally observed between the model and measurements, although the high frequency simulated response of the tweeter differs slightly in the width of the resonance peak at 11kHz. This may be outside the range of the original tweeter model provided by Knowles, but might also be attributable to geometric simplification between the model and the experimental setup, or by the simplified acoustic modeling of the plunger termination the ADECS with OELS by a resistive element. The 16kHz response is about 10 dB lower relative to 1kHz, which is thought to be sufficient energy to enable practical equalization with an EQ for high frequency reproduction fidelity, if necessary.

The model enables simulating the plant frequency response of the musician earpiece, from its IELS to its IEM, which affects the performance of the ANC system in closed loop. Figure 6 compares the simulation to a measurement. This frequency response is thought to be suitable for ANC, since it appears that the transducers are capable of outputting high SPL at low frequencies while maintaining manageable deviation from flat phase. The first 180° phase shift occurs around 8 to 10kHz, although it comes very close around 5kHz, which in both cases is well above the frequency of 1kHz beyond which occlusion effect is expected to be mild or even negative (Dean and Martin, 2000).
4. Controller design algorithm

After the earpiece is inserted into the user’s ear canal, an identification phase is triggered to obtain a plant frequency response estimate, which is then used by an algorithm to design a custom compensation to achieve closed-loop ANC that matches a pre-defined target performance, as discussed in section II.

The plant response is measured by playing a broadband pink noise through the IELS and recording a measurement signal with the IEM. The pink noise is also looped back electrically and recorded to serve as a synchronised reference signal. A particular transfer function estimate, dubbed $H_3$ (Herlufsen, 1984), is then calculated from these signals. The magnitude of this transfer function estimate is then smoothed using a moving average in the frequency domain to remove measurement noise and discard high frequency detail where the controller does not need to be precise. The smoothed transfer function estimate magnitude, along with a pre-defined closed-loop target performance, are used to define an ideal compensation. A curve fitting algorithm, developed for this study and described below, attempts to fit a pre-defined number of bi-quad peak filters (PF) to the curve. The PF bi-quad in the bi-quad budget are adapted one by one to iteratively reduce the difference between the magnitude of the practical compensation and the magnitude of the ideal compensation. The principle of the curve fitting algorithm, to obtain one curve fitted bi-quad is as follow:

1. The maximum magnitude within a specified bandwidth in the magnitude of the target frequency response (MTFR) is automatically found. A peak is assumed to be centered here.

2. By scanning the slope of the MTFR using a smoothed numerical derivative, the start and end of an assumed peak are found. Zero crossings around the peak’s assumed center are searched for in the derivative in an attempt to obtain the subset of the target frequency response that can be reasonably approximated by a PF. This step yields indexes $i_{\text{start}}$ and $i_{\text{stop}}$ that correspond to the indexes in the MTFR between which a peak should be fitted.

3. The magnitude frequency response of a PF (MFRPF) centered on the assumed peak is calculated with a sharp quality factor $Q$, where $Q = 7$.

4. An error criterion is defined as the average difference between the absolute MTFR subset and the absolute MFRPF subset, as defined by equation 3.

$$e = \sum_{i = i_{\text{start}}}^{i_{\text{stop}}} \frac{|\text{MTFR}(i)| - |\text{MFRPF}(i)|}{i_{\text{stop}} - i_{\text{start}}}$$

5. If the error criterion is positive, the PF is likely too narrow and its $Q$ should be reduced. If, on the other hand, the error criterion is negative, the PF is likely too wide to be a good approximation of the MTFR subset and its $Q$ should be increased. This is done according to equation 4, unless it is determined that it would lead to a negative $Q$. In that case, equation 5 is used instead. These equations have been determined and tuned empirically.

$$Q = Q - \frac{e}{10}$$

$$Q = \frac{Q}{2}$$

6. The $Q$ of the PF is updated as well as its MFRPF and an updated error criterion is computed. This process, looping back to step 4, is repeated 40 times to obtain an optimized Q and a good MFRPF subset approximation of the MTFR subset.

The process detailed above is repeated for each bi-quad filter defined in the bi-quad budget, progressively reducing the difference between the MTFR and the magnitude of the cascaded response of all the MFRPFs. In the particular case of designing a compensation to obtain a predefined target OE reduction with the shape of low shelf filter response, such as illustrated in figure 1(A), the algorithm produces better results if the first bi-quad is fixed with a response that is the reverse of the target OE reduction. An example of fitting a target controller curve with 11 bi-quad is shown in figure 7 Practical considerations and validation of this empirical algorithm is discussed in section III.

III. VALIDATION OF THE METHOD

The method and system are validated, prior to implementation, using two techniques. First, using the models of the loudspeakers and ADECS with OELS and second, using an in-ear speech database.

To test the ability of the method and system to achieve comparable performance under various acoustical impedance loads, their combined performance on the ADECS with OELS is first simulated for various occluded lengths of 6 mm, 16 mm, 26 mm and 36 mm. The 16 mm length is derived from the average ear canal length minus the projected insertion depth of the earpiece, as discussed in section II B 3. From there, arbitrarily large steps of 1 cm are chosen to measure the ability of the method to adapt to large variation in acoustical impedance load and the plant response. Although some of these steps result in unrealistic ear canal lengths, they are used because they induce the same order of variation in the low frequency of the plant response as observed on real human subjects (Goldstein et al., 2005; Liebich et al., 2019). This is explained by variations in the compliance of real eardrums, whereas that of the ADEC is nearly rigid.

Because the projected implementation of the compensation will use a DSP selected for its low latency of 38 µs, that additional latency of 38 µs is included in the model to obtain more realistic simulation conditions.
Figure 8 compares the simulated plant response for these acoustical impedance loads, showing that variability exceeds 10 dB in the low to mid frequencies, and that greater variation exceeding 30 dB occur beyond 2 kHz.

An example target response that represents a potential desired OE reduction, based on a second order low shelf filter with a stop-band gain of -15 dB, a cutoff frequency of about 1000 Hz and a slope of 1.1, is selected. Running the controller design algorithm discussed in section II B 4 for each of these plant responses results in the closed loop responses in figure 9, with gain margins between 13.3 dB and 13.6 dB, and phase margins between 42.2 and 50.3°, which are sufficient to ensure stability. It can be seen that the projected OE reduction is very similar for all acoustic impedance loads, despite the initial variability in the plant responses. Maximum deviations from the target magnitude are below 2 dB from 30 Hz to 600 Hz, and amplification around the cutoff frequency is maintained below 6 dB. Maximum deviations between the simulated OE reductions for different lengths are within 1 dB, except on the 6 mm condition where a sharp peak deviates by about 3 dB.

A. Validation based on in-ear speech database

To further validate this empirical method in the context of active noise control of the occlusion effect on realistic plant responses across many subjects, data gathered for the in-ear speech database, dubbed SpEAR (Bousherhal et al., 2019) was re-purposed. This in-ear speech database gives access to recordings allowing one to estimate the plant response of an earpiece equipped with an IEM and IELS. The earpiece used in the SpEAR study was designed by the first author of this paper and is a variation of the earpiece design presented in this study, albeit with a different cross-over frequency that caused destructive interference between the tweeter and the woofer resulting in a pronounced notch in the frequency response magnitude and a 180° phase shift around 1.6 kHz. The earpiece also used a different IEM exhibiting a low frequency roll-off. While both these differences make the SpEAR earpiece less ideal for ANC than the earpiece presented in this study, the data is nonetheless very useful to validate the algorithms and observe the variability and accuracy of the performance that could be achieved using the proposed method on actual participants. The plant responses were truncated below 50 Hz for this analysis to ignore some of the effect of the low-frequency roll-off of the IEM, but it partially remained along with the aforementioned phase shift, and for these reasons a more attainable stop-band gain of -12 dB was selected as a target performance, rather than -15 dB.

Figure 10 shows all the calculated plant transfer function estimates on the left ear of each of the 27 subjects of the SpEAr database. It can be seen that the coherence given by the estimate drops from about 400 Hz to 2 kHz on many subjects. This is believed to be caused by the destructive interference between the tweeter and the woofer, but some coherence drop may be further caused by disturbances introduced by the test subject when the measurement was taken. The coherence drop translates into spikes and disturbances in the magnitude response that are carried over and ultimately affect the simulated closed loop response. An arbitrary performance curve based on a second order low shelf filter was selected, with a stop-band gain of -12 dB, a cutoff frequency of about 1 kHz, a slope of 1.1, and a budget of 11 bi-quad filters was allotted to the controller design algorithm. Figure 11 shows all the achievable performance curves resulting from the process, overlaid with the same performance curves with light smoothing for better visibility, along with the target performance curve. Some spread of the red curves between 400 Hz and 2 kHz is attributable to the aforementioned disturbances in the plant transfer function estimates that are carried over. Nevertheless, worst-case regeneration is contained to 6.7 dB and all users can be offered similar performance within 3 dB in the stop-band. The ability of the system to follow the target degrades below 100 Hz, where the IEM roll-off is not completely discarded. The gain and phase margins achieved for each user have been automatically calculated by the algorithm and are shown in figure 12, along with predefined safety margins threshold of 6 dB and 30°, which are surpassed for all users. These safety margins thresholds were previously derived from the literature in (Bernier, 2013) and are considered safe for a fixed system, hence even safer for this individualized system. It can be seen that participant 16 has an exceptionally low phase margin as a consequence of large disturbances in the plant transfer function estimate, confirming the importance of providing clean measurements to the compensation algorithm.

IV. IMPLEMENTATION AND RESULTS

The system was implemented with the goals of performing measurement and ANC in the ADEC with OELS off various lengths. The block diagram of the whole measurement setup is depicted in figure 13.

The proposed earpiece design physical shells illustrated in figure 4 were 3D printed using a Form 2 (Formlabs Inc., Somerville, Massachussetts, USA) stereolithography printer with clear V2 material and the earpieces were assembled by hand. A breakout box was built to connect the earpiece IEM and IELS to a fast codec with digital processing capabilities (DSP) and an OCTA-CAPTURE (Roland Corporation, Hamamatsu, Shizuoka Prefecture, Japan) USB audio interface. The identification phase is carried out with a MATLAB program that uses the USB audio interface, playing broadband noise through the DSP driving the loudspeaker, to measure the transfer function estimate from the input of the DSP to the output of the IEM, defined as the plant response. The MATLAB program then designs the compensation per the method described in section II B 4 using a budget of 11 bi-quad filters, and display the resulting predicted performance, gain and phase margins to be reviewed be-
fore proceeding. The bi-quad filters were programmed to
a DSP, the ADAU1772 (Analog Devices Inc., Norwood,
Massachusetts, USA), which was selected because of its
fast input to output latency, specified at 38 µs. Once the
DSP is programmed, the open loop compensated system
response is measured in a last validation to estimate the
gain and phase margins with the implemented compensa-
tion. Activating a switch then disconnects the output of
the USB audio interface from the input of the DSP and
connects the IEM to the input of the DSP, effectively clos-
ing the loop and engaging the ANC. To characterize the
performance of this system under various acoustical loads
and mimic in-ear speech, the ADECS with OELS was
used. Broadband noise was played through the OELS
to introduce signal to minimize inside the ADECS and
the transfer function from the input of the OELS to the
output of ear drum microphone (EDM), located at the
plunger gasket of the syringe, was estimated. The dif-
ference in this transfer function when ANC was engaged
or disengaged provided the actual reduction achieved by
the ANC.

The same target of -15 dB defined in section III
was selected and the proposed method was repeated for
lengths of 6 mm, 16 mm, 26 mm and 36 mm. The result-
ing performance is presented on Figure 14. Maximum
deviations from the target are below 2 dB from 30 Hz to
600 Hz, and amplification around 1.3 kHz is maintained
below 6 dB. Maximum deviations between the offered OE
reductions for different lengths are within 2 dB below
900 Hz and within 3 dB above that.

V. DISCUSSION
A. Method

As mentioned in section II A, the earpiece acoustic
design has one of the most noticeable effects on the plant
response measured on a given user. Along with latency
in the signal path, the earpiece design dictates the max-
imum performance magnitude limit than can be achieved
in a feedback ANC system. The proposed method of
earpiece design and the ability to model ANC behavior
under varying acoustic impedance loads before it is
implemented is advantageous in achieving an in-ear feed-
back ANC system that performs as expected. Iterating
between physical possibilities in CAD models for compo-
ent placement and sound channel sizing and their pro-
jected effects on the plant response has been valuable in
achieving all the requirements that were defined in sec-
ction II B. This method enables researchers to design their
own earpiece to address their specific challenges, such as
providing passive hearing protection, rather than rely on
modifying existing designs.

In this paper, a top-down approach was used, in
that empirical requirements were first defined, an ear-
piece that satisfied these requirements was then designed,
and its model was then used to predict its ability to of-
fer reasonable OE reduction performance under varying
acoustic impedance loads. A possible improvement to
this approach could be to reverse the process and start
with defining a maximum OE reduction performance tar-
get, set realistic boundaries for the compensation, and
derive a set of specifications to define an ideal average
plant response to guide the earpiece design process.

No formal study has been conducted to define a tar-
get reduction performance since this paper rather focused
on minimizing the variation in performance across acous-
tical loads. However, the empirical performance target
reductions of 12 and 15 dB and their general shape are
in the range of average measured occlusion effect at 250
and 500 Hz which varied between 6 and 17 dB for various
insertion depths of insert earphones (Dean and Martin,
2000).

B. Bi-quad filter budget

The method of using an adaptive compensation using
11 bi-quad filters, in conjunction with the proposed ear-
piece design, allows stable and time-invariant OE reduc-
tion of approximately 15 dB within less than 2 dB below
250 Hz despite variations in the plant response exceeding
10 dB in that frequency range, and variations exceeding
30 dB at higher frequencies. While this remaining vari-
ation may already be unnoticeable in the context of OE
reduction, steps could be taken to further reduce the per-
formance spread to obtain more precision.

Precision could be gained by increasing the bi-quad
budget. In this study, the budget of 11 bi-quads has
been derived from practical considerations. The chosen
DSP, with 4 inputs and 2 outputs, is theoretically able
to implement up to 32 bi-quads. The first envisioned
architecture of this musician HPD would use these 4 in-
puts for its 4 microphones, binaural IEMs and OEMs.
The bi-quads need to be distributed to perform feedback
loop compensation on the IEMs for both ears, equalize
an attenuated hear-through path between the OEMs and
both IELSs to provide attenuated acoustic transparency,
and mix the OEM and IEM signals to one output per
ear. Empirically balancing these requirements left 11 bi-
quads per ear for feedback control using a single instance
of this DSP for both ears, but one DSP per ear could
be used if additional precision is required. This would
help reduce smaller details in the variation of the OE
reduction such as the small peak around 1.6 kHz on fig-
ure 9. Additionally, this would allow using two outputs
per ear to manage the loudspeakers separately and offer
an alternative to the dual loudspeakers and fixed analog
cross-over strategy that was used in this study.

C. Practical difficulties

Practical difficulties explain some of the OE reduc-
tion variation observed in this study. The most signifi-
cant one was linked to the IEM output impedance and
the respective input impedance of the USB audio inter-
face and the DSP. The output impedance of the micro-
phone is rated nominally at 4.4 kΩ and can vary between
2.8 and 6.8 kΩ, while the input impedance of the USB
audio interface is rated at 740 kΩ while the DSP input impedance is tied to the specific gain setting of its preamplifier and can vary between 0.68 to 32 kΩ. Connecting the IEM output to the low impedance DSP input therefore reduces the apparent amplitude of the IEM signal, modifying the feedback loop between when the system is in identification mode and when it is engaged in ANC mode using the switch depicted on figure 13. To reduce the discrepancy between those two modes, a parallel electrical resistance in the range of 20 kΩ, which was the estimated input impedance of the DSP at the selected gain setting, was placed in the electrical path so that it is present only in the identification phase, in parallel with the input impedance of the USB audio interface. This helped bridge the discrepancies between the two conditions, but still left discrepancies in the range of 2 dB, which were further corrected by imposing a fixed digital gain in the feedback loop when it was closed. An input stage with sufficiently high input impedance would offer less variation and more control over the system.

A similar case can be made for an output stage. An unforeseen difficulty lied in the low electrical impedance of the selected tweeter which is rated at only 12.5 Ω of DC resistance. This lowered the overall electrical impedance of the earpiece transducer assembly, which was measured at 10 Ω. This has caused discrepancies in the frequency response of IELS of the earpiece between when it was driven by the USB audio interface, to validate the earpiece model, and when it was driven by the DSP’s integrated headphone amplifier, which is only rated to drive 16 Ω and higher loads. This means that the plant response differs between the two conditions, preventing the model’s projected performance of figure 9 from being directly comparable to that of the measurements of figure 14. However, despite the initial differences in plant responses, the proposed controller design algorithm is robust to these changes and is still able to offer similar performance due to its adaptive nature.

D. Dynamic range and transducer selection

While the in-depth study of dynamic range of the system has been out of focus of this article, dynamic range has a significant impact on the subjective user experience. Anecdotal testing of the proposed system on human participants yielded good results at speech levels and moderate singing, but saturation could be heard at higher levels of singing or screaming even though the transducers of the earpiece had been selected with high SPL capabilities in mind. Indeed, during another study (Bouerhal et al., 2019), it was observed that earcanal deformation resulting from jaw movement when talking caused quasi-static pressure changes inside the closed volume that is the occluded earcanal, resulting in acoustic overload of the IEM and mechanical saturation of the transducer resulting in a clipped audio signal. The problem disappeared when the IEM with a flat response was substituted with one that has a low frequency roll-off but otherwise similar sensitivity. As discussed in section III, this roll-off characteristic is detrimental to feedback ANC systems, so other solutions should be considered. One promising avenue is to use a flat response microphone, but with reduced sensitivity such as the Knowles FG45-32335-000. The FG45 series, however, requires external circuit to fully take advantage of its higher acoustic overload limit and could not be directly substituted to the FG-23652-P16 used in this study. In future work, a formal analysis of the dynamic range required for an OE reduction system would be beneficial.

VI. CONCLUSIONS

A novel method, designed to achieve a defined target OE reduction across all users regardless of their occluded earcanal acoustic impedance, is presented in this paper. It relies on an earpiece acoustic design that is suitable for feedback ANC, which can be realized using the proposed earpiece design methodology. The method uses the earpiece in an identification phase to obtain a plant transfer function estimate that, along with the defined target OE reduction, is used to calculate a target compensation to be applied in the feedback loop. This target compensation is approximated using an adaptive algorithm that attempts to curve-fit bi-quads to obtain an achievable compensation before closing the feedback loop and applying OE reduction.

The method has been validated using models of the proposed earpiece design and of a ADECS with OELS as well as with measurements from an in-ear speech database that used a similar earpiece design. The earpiece design and ADECS with OELS were implemented and used to experimentally measure the OE reduction that can be achieved using the proposed technique. The implemented system adapted to variation in the plant responses as high as 10 dB below 250 Hz and 30 dB above 2 kHz to offer 15 dB of OE reduction below 250 Hz. The variation in performance is contained to less than 2 dB below about 900 Hz and 3 dB above 32335-000. The FG45 series, however, requires external circuit to fully take advantage of its higher acoustic overload limit and could not be directly substituted to the FG-23652-P16 used in this study. In future work, a formal analysis of the dynamic range required for an OE reduction system would be beneficial.

ACKNOWLEDGMENTS

The authors would like to acknowledge the funding received from the Fonds de recherche du Québec - Nature et Technologies and the NSERC-EERS Industrial Research Chair in In-Ear Technologies (CRITIAS).


Lillywhite, S. E. (2013). “Microphone-based pressure diagnostics for boundary layer transition.”


VII. SUPPLEMENTARY MATERIAL?

In this simulation, the known loads were closed tubes taken from the Knowles SPICE library. Figure 15 shows the SPICE circuit schematics. The loudspeaker model impedance curves were calculated according to equation 6. Figure 16 shows the changes between three impedance curves when the loudspeaker is coupled to tubes with a diameter of 1.5 mm and lengths of 5, 10 and 15 mm.

\[ Z_{in} = \frac{V_{spk}}{V_{source} (1 - \frac{V_{spk}}{V_{source}})} \]  \hspace{1cm} (6)

These impedance curves are used to calculate hunt parameters \( Z_e, T_a \) and \( Z_a \), presented in figure 17 which are in turn used to obtain an ABCD matrix representation according to equation 7. Because of the numerical method used, there is in fact one corresponding 2x2 matrix for each individual frequency in the frequency vector with a resolution of 1000 points per decade.

\[
\begin{bmatrix}
E(\omega) \\
I(\omega)
\end{bmatrix} = \begin{bmatrix}
1 & Z_e(\omega) \\
0 & 1/T_a(\omega)
\end{bmatrix} \begin{bmatrix}
0 & T_a(\omega) \\
1 & 0
\end{bmatrix} \begin{bmatrix}
1 & Z_a(\omega) \\
0 & 1
\end{bmatrix} \begin{bmatrix}
P(\omega) \\
U(\omega)
\end{bmatrix}
\]  \hspace{1cm} (7)

The ABCD matrix of equation 7 can be used to aid in the earpiece acoustical design for which the CI-22955-00 is to be used as a woofer.

Figure 18 shows one typical open loop transfer function estimate of the achievable individualized controller and the individualized plant response used by the algorithm to estimate the gain and phase margins, in this case 9.8 dB and 74.7°.
FIG. 1. (A) Conceptual target OE reduction (B) Two conceptual plant responses showing the behavior of the same earpiece in two ears (C) Two conceptual target compensation responses.
FIG. 2. (A) Conceptual target and practical compensation response (B) Two conceptual achieved OE reduction compared to a conceptual target OE reduction.
FIG. 3. Impedance curves of earcanal couplers, showing the occluded impedance of a simulated IEC compliant ear coupler, along with simulated impedance curves of the ADECS with OELSL for lengths of 10 mm, 16 mm and 26 mm.
FIG. 4. The final earpiece design showing: 1. the woofer; 2. the tweeter; 3. the IEM; 4. the OEM 5. the eartip and; 6. the connection PCB. The sound channels have been highlighted with the same color as their associated transducers.
FIG. 5. Simulations results of the model (mdl) against measurements (meas) for the musician earpiece as seen by the eardrum microphone of the ADECS with OELS for a length of 16 mm.
FIG. 6. Simulations results of the model (mdl) against measurements (meas) for the musician earpiece as seen by the IEM when inserted in the ADEC with OELS for a length of 16 mm.
FIG. 7. Fitting a compensation target magnitude with individual bi-quads.

FIG. 8. Simulated plant response on the ADECS with OELSL for lengths of 6 mm, 16 mm, 2.6 mm and 3.6 mm, showing large variations in the plant response magnitude (top) and phase (bottom).
FIG. 9. Simulated achievable performance on the occluded ADECS with OELS for lengths of 6 mm, 16 mm, 26 mm and 36 mm, showing good agreement with the target performance except for amplification near the cutoff frequency starting from approximately 700 Hz to 5 kHz, which is below 6 dB.
FIG. 10. Plant transfer function estimates of the SpEAR earpiece inserted in the left ear of 27 subjects.
FIG. 11. Target and predicted performance using the individualized controller in closed loop for all SPEAR participants.

FIG. 12. Gain and phase margins obtained across all SPEAR users using the individualized controller to obtain the predefined close loop target performance.
FIG. 13. System block diagram

FIG. 14. Measured performance on the ADECS with OELSL for lengths of 6 mm, 16 mm, 26 mm and 36 mm
FIG. 15. SPICE circuit used to extract the impedance curve of the CI-22955-000 model driving known acoustical loads.

FIG. 16. CI-22955-000 SPICE model impedance curve when the loudspeaker is acoustically loaded with tube that have a diameter of 1.5mm and a length of 5, 10 and 15mm.
FIG. 17. Frequency curves of the magnitude of Hunt parameters $Z_e$, $T_a$ and $Z_a$ for the loudspeaker CI-22955-000.

FIG. 18. The open loop response of the individualized controller and Plant system, allowing to calculate gain and phase margins.