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Comparative Evaluation of Public Address Feedback Controllers: A Preliminary Assessment Methodology

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ABSTRACT

Anti howling (AH) systems a.k.a. acoustic feedback control allow to maintain sound quality and prevent the howling effect in public address (PA) systems. Various principles may be used for AH, each with its advantages and drawbacks. The objective of the present work is to propose a methodology for evaluating commercial or simulated AH systems as "black boxes". Following previous publications, the evaluation is based on three critical aspects : the Additional Stable Gain (ASG) which is achievable before Larsen howling occurs, the Perceptual Evaluation of Speech Quality (PESQ), a standardized speech quality criteria, and the "Reactivity" which quantifies how fast the AH device adapts to changes of the electroacoustical environment. Preliminary results are presented for six different AH systems. They illustrate that the proposed protocol allows to distinguish small differences between designs, in addition to highlight the general trends already described in the existing literature.

1 Introduction

Many audio applications include situations where a microphone may sense the acoustic pressure radiated by a nearby loudspeaker, leading to potential drawbacks. Classical examples include the local echo within a hands-free telephone set [1], audio feedback within a hearing aid [2] or acoustic feedback within a PA system used in live amplification, potentially leading to the well-known Larsen effect or "howling" [3]. Although some musical styles include the intentional use of feedback, especially for electric guitars, most situations require to prevent howling. This is primarily done using carefully placed directive microphones and speakers and adequate tuning of the PA system. This allows to cope with various situations, excepting venues with a poor acoustic behaviour [4].

Indeed, the influence of the room over sound transmission has been studied for a long time [5], including its effect on the stability of PA systems [6]. The room response may be considered as a random process, although for given locations of speaker(s) and microphone(s) and under known environmental conditions, it may be considered as a linear system combining many acoustic paths. It leads to a very irregular frequency response showing dips and peaks. This frequency response is combined with that of the microphone, speaker and processing equipment, building a closed-loop system that must adhere to classical stability criteria in order to avoid howling [7]. However, the characteristics of the acoustic paths within a room change with time and this requires to tune the PA system accordingly in order to keep its stability. Numerous solutions have been proposed to cope with automatic tuning of a PA system in order to avoid howling, and eventually to increase its stable gain without degrading much the amplified signal quality. Such systems are globally called "Anti-Howling" (AH) systems in this paper. A comprehensive and well-argumented review has been proposed by Van Waterschoot and Moonen [8], who proposed to distinguish four classes of AH methods: time-varying filters, gain reduction methods, spatial filtering methods, and room modeling methods.

These methods are implemented in various systems, each possessing distinct features and being more or less effective depending on given practical situations. It is thus difficult to rank them objectively using a single criterion: many papers have proposed comparisons within a given class of AH method, but few of them tried to give an overall assessment method allowing the comparison of a wide range of AH systems. This is the objective of the present work, which proposes a combination of previous assessment methods and their implementation using affordable tools. Section 2 gives a short description of the main AH methods proposed in the literature, allowing to understand their specificities. Their comparison is based on previous work about AH systems evaluation, presented in section 3. The proposed assessment method is then described by section 4 and first results are shown in section 5

1.1 Glossary of Abbreviations

In addressing detailed topics, this paper employs specialized abbreviations. To facilitate reader comprehension, a Glossary of the Main Abbreviations is provided in Table 1, which includes definitions and full forms of these terms.

Abbreviation	Meaning
AH	Anti Howling
AFR	Acoustic Feedback Removal
PA	Public Adress
ASG	Added Stable Gain
PESQ	Perceptual Evaluation of Speech Quality
TRI	time to recover from instability
TRI-I	TRI- at Initialisation
TRI-C	TRI- Change acoustic path
SD	Spectral Deviation
PSD	Power Density Functions
IR	Impulse Response

Table 1: Glossary of abbreviations and their meanings in the document.

2 Main anti-howling (AH) methods

Figure 1 outlines the single path (SISO) structure considered here for a sound reinforcement chain : the linear electroacoustic path [F], some potentially nonlinear processing [G] and the anti-howling system [H]. Multi-input or multi-output (MIMO) configurations are not considered in this preliminary work.

Basic principles of AH system are summarized below, following the classification already proposed in [8] but excluding the MIMO spatial filtering methods.

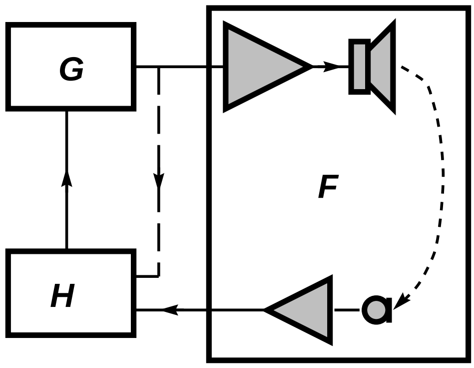

Fig. 1: Diagram of a sound reinforcement chain containing an anti-feedback system [H], a processing module [G] and an electroacoustic path [F].

2.1 Time-varying filters

In order to avoid a stable in-phase condition for the feedback path, several principles have been proposed : the first of these is frequency shifting [9]. Other time-varying filters such as phase or delay modulation have also been considered. Their drawback is a sound modulation which may be acceptable for speech, but less for music [10]. A small amount of phase modulation, however, is interesting as a means to improve other AH methods [11].

2.2 Gain reduction

The most straightforward AH method is to mimic a sound engineer: reducing the system gain, either globally [12], or by automatic equalization of critical frequency bands [13]. An alternative, widely used in commercial systems, is the so-called "Notch-filter-based Howling Suppression" (NHS) [14], which detects howling and activates notch filters tuned to the instability frequencies.

2.3 Room modelling

A very efficient AH method is based on a model of the electroacoustic loop (the $[F]$ block in figure 1). This model may be used to equalize the overall response or to substract the loudspeaker contribution from the microphone signal, a technique we call "Acoustic Feedback Removal" (AFR), as the AFC acronym is ambiguous. Compared to the acoustic echo cancellation (AEC) used in telecommunication, AFR has to cope with a high correlation between the wanted and unwanted signals, thus requiring a decorrelation mechanism. Many techiques have been proposed for this purpose, leading to various algorithms [8, 15, 16, 17, 18]. While AFR may be CPU-demanding, it has no theoretical limit to the potential improvement in stable gain.

3 Evaluation of AH systems

The comparison of AH systems require to find evaluation criteria adapted for all the principles described above, and combinations of them. Early work dealt with the investigation of feedback in PA systems, considering how equalization techniques might improve the maximum gain, the audio quality and the robustness of the system [19]; this work did not, however, consider AH systems. Later, a standard about AEC included a few evaluation criteria focused on the echo reduction [20]; it did not take into account some specificities of AH systems.

Later work did focus on the performance of AH systems. For instance, in [10], time-varying AH systems were compared, primarily in terms of the 'gain before instability' (GBI), specifically its increase. Several publications dealt specifically with AFR systems : Bispo [21] used a virtual environment, using the "misalignement" (MIS) criterion to assess the internal filter identified for AFR. Numerous publications dealt with the specific application of AH systems used in hearing aids [22]. The criteria used in these works were different for each class of AH system, and thus did not allow a generic evaluation.

A recent publication proposed a comparison of the performances of several AH system principles, based on two criteria common to all AH systems : the "added stable gain" (ASG also often denoted ΔMSG, the gradient of Maximum Stable Gain) and an objective quality criteria : "spectral distortion" (SD) [23]. This publication was followed by a review including a comparison of several AH systems, based on simulations [8]. In addition to MSG and SD, this paper proposed two criteria about "reliability" : "howling occurence probability" (HOP) and "time to recover from instability" (TRI).

The howling detection is a subject by its own, as it is both a component of NHS-based AH systems and a tool for the assessment of ASG. Specific evaluations of howling detection criteria have been proposed [24, 25, 26]. The authors compared numerous criteria for howling detection using the Receiver Operating Charasteristic, a statistical plot of the true positive rate against the false positive rate.

3.1 Challenge evaluating a commercial system

Many of the publications about AH evaluation were based on simulations of the various systems to compare. This allows the access to internal quantities (signal, filters, freeze signals, etc) which are not always available in a commercial system - especially when AH is an embedded function inside a large piece of equipment, like an audio mixer. Some authors proposed an implementation using a DSP board to speed up AH systems comparison [27]. The AH systems were, however, only AFR-based and implemented as computer codes.

To compare "black-box" commercial systems, the only way is to design a protocol using only their input and output, with suitable test signals. Such a protocol was proposed by some authors in order to evaluate commercial hearing aids [28]. A fair comparison is however difficult : the authors mentioned that measurements of the ASG for four different input signals showed that ASG depends on the input signal. Moreover, minor changes in the set-up may cause significant feedback path differences at higher frequencies.

Using physical setups has, thus, the drawback that it does not allow independent comparisons by different actors, even using standardized facilities. We therefore propose to use a virtual acoustic environment (with characteristics representative of a "typical" PA situation), which could allow to get comparable results between measurements performed in different locations or at large time intervals.

3.2 Evaluation criteria

Three criteria, selected from both existing literature and common audio practices, have been chosen because they directly address the most critical aspects of an AH system's performance.

3.2.1 Added stable gain

An important criterion for evaluating AH system is the gain (ASG) which can be added to the system when the AH system is used, compared to when it is not. Even if this criterion is less important for PA systems than for hearing aids, it is still a major one. With no access to any internal quantity, this criterion must be based on a general gain estimation.

In most experimental publications [8], ASG is determined by incrementally raising the gain of the direct path $[G]$ until instability occurs, both with and without the acoustic feedback controller (AH) active. ASG is then supposed to be the difference in gain between these two conditions, but this is not true if the AH system changes the overall gain of $[H]$, as in NHS systems. Moreover, instability onset is difficult to identify, so the estimates often depend on the operator.

A solution is to evaluate the gain as the ratio of averaged Power Density Functions (PSD) between input and output of $[H]$, estimated with and without AH. A further correction of the instability onset estimation may be performed when both $[G]$ and $[F]$ are linear and invariant, as explained in section 4.

3.2.2 Audio quality

Maintaining high audio quality is crucial in any PA system. Early works [8] introduced objective criteria, such as SD, but did not link them to the listener experience. Some authors [22] then proposed to add objective criteria based on perceived audio quality, such as PESQ for speech or PEAQ for music.

Such objective quality measure for speech and music signals can yield inconsistent results when applied together [29]. As a result, separate measures are required for speech and music evaluations. This paper focuses on PA systems mainly for speech reproduction, and thus only considers PESQ which was originally developed by the ITU-T for evaluating speech quality in the context of speech coding for telecommunications.

PESQ has since found utility in assessing the impact of various types of distortions, including acoustic feedback [30]. It compares a reference (original) speech signal to a degraded (processed) version and assigns a score between -0.5 and 4.5, with higher scores indicating better speech quality. In this paper, the reference signal is the one to be reproduced, while the test signal is the one measured as the output of the PA system (including the AH and acoustic feedback).

3.2.3 Reactivity

Previous works [8] also evaluated the robustness of AH systems, using various criteria, among which "time to recover from instability" (TRI), a reaction time quite important for practical PA systems which we call "reactivity".

Reactivity can be considered in various contexts and, in this paper, we choose to quantify TRI in terms of the howling times in two distinct relevant scenarios commonly found in the scientific literature on AFR [8, 18] that can be extended to the general evaluation of AH systems :

- During initial convergence (TRI-I), which assesses how quickly the system can adapt and suppress feedback when it first appears.
- During change of acoustic paths (TRI-C), which assesses how fast the system adapts to changing acoustic conditions, such as when the microphone or speakers are repositioned.

4 Proposed method for AH evaluation

This study delves into the realm of AH evaluation for PA systems, with the aim of providing a comprehensive methodology for assessing AH devices used in real-world PA system contexts.

4.1 Measurement environment

While the various AH devices under test may be hardware or software, a simulated environment was chosen for their measurement. Indeed, such a cost-effective environment may be varied easily, still offering accurate control and reproducibility. Moreover, it allows consistent results to be shared between many partners.

4.1.1 Principle

The measurement setup includes the AH system ([H] block of Figure 1) as a VST plug-in (for software AH systems) or as a soundcard allowing to connect an external hardware device (for hardware AH systems). A PC is used to run all other parts of the measurement environment.

This environment is depicted by Figure 2. It features a signal source (an audio player here), which represents the talker. This signal is added to the output of a convolver plug-in allowing to simulate the Impulse Response (IR) of the electroacoustic path ([F] block of of figure 1). This IR may be switched when assessing the TRI-C criterion (see section 3.2.3). The [G] block of figure 1 is a pure linear gain which is the product of an analog hardware gain K_A and a digital gain K_D.

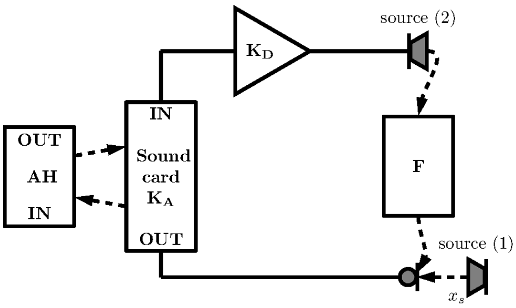

Fig. 2: Electroacoustic chain simulation diagram.

4.2 Software implementation

A VST host is used to connect all parts of the setup. We chose the open source software "Pedalboard 2" (based on the JUCE code for VST hosts), as it includes essential built-in plugins like an audio player, recorder, variable gain controls, VU meters, and channel selectors. Moreover it offers an OSC interface for seamless communication with Python, facilitating the execution of programmable action sequences.

A basic acoustic feedback simulation of Figure 2 is depicted in Figure 3 using Pedalboard 2. Gain levels, feedback, acoustic path switches and recording can be monitored via OSC commands. VU meters are employed to monitor audio levels, aiding in identifying the onset of howling even at low volumes. This implementation enables silent operation or connection to an

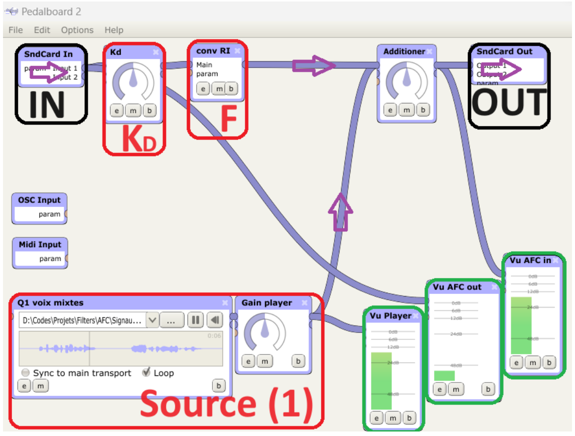

Fig. 3: Pedalboard visual of a minimal setup for simulating acoustic feedback. In red, the player and acoustic feedback F, in green, VU meters, in purple, the audio flux direction and in black, the sound card's input and output.

additional speaker, facilitating operator monitoring of simulation performance.

The electroacoustic path is simulated using the convolver VST plugin ‘R1 Convolution Rev’, whose user interface is shown in Figure 4, with the parameters corresponding to the IR used for TRI-C test (an acoustic hall).

Fig. 4: Plugin ’R1 Convolution Reverb’. The IR selected is the Hall, with a RT 60 = 1.5s.

Simulations were conducted on a portable PC (Intel i7-1255U processor / 16 GB RAM). Although the CPU usage consistently remained below 20 %, occasional audio crackling was detected when using a block size below 128. To ensure smooth operation, a block size of 256 was employed, providing a safety margin against audio disruptions. This results into a consequent 38 ms latency. Latency's influence on the results will be addressed in the future. Preliminary investigations in the same configuration as used in this paper indicate that,overall, the results' meaning remains unaffected. Note that during the evaluation of the AH systems, they were connected to a Focusrite 4i4 sound card.

4.3 Evaluation protocol: recording

The whole evaluation consists of: first, a recording session, and second, a post-processing part, where the metrics are computed. The AH evaluation recording part includes the following steps, following initial configuration and source level adjustment:

1. Keeping $K_D = 0$, the operator tunes the soundcard input gain K_A with AH bypassed, trying to adjust the gain as close as possible to the howling threshold.
2. Freezing K_A, several gains K_D are then selected, increasing from -7 dB until the value for which howling can't be prevented (ASG). For each gain:
   - Quality signals are recorded (for PESQ, see §3.2.2). The quality signal test is recorded once the AH system is stabilized (howling-free).
   - If the gain K_D is higher than zero, the TRI-I and TRI-C signals are recorded (see §3.2.3). The TRI-I test signal is recorded immediately after resetting and unbypassing the AH system, while TRI-C test signal is recorded immediately after switching the IR1 to the IR2.

Note that the input signals used for the PESQ test consist in two 8-second ITU speech signals: two sentences spoken by a female speaker and two spoken by a male speaker. The input signals used for the reactivity and the characterization are white Gaussian noise of respectively, 25 and 10 seconds.
The switched second acoustic path IR2 corresponds to the modified hall depicted in Figure 4, while the main IR1 selected is the original version (obtained by resetting the parameters in the user interface).

4.4 Evaluation protocol: post-processing

While 3.2 introduces the metrics, this section will elaborate on the specific procedures used to obtain them.

The metrics are introduced in a general manner in §3.2, while this section elaborates on the specific procedures we used to calculate them. While PESQ can be straightforwardly estimated using the ITU public code, there are multiple definitions available in the literature for ASG and reactivity.

For the gain, as stated above, the operator establishes a coarse 0 dB reference for the ASG (setting K_A factor). This is, however, not reliable enough to allow comparisons, especially between different operators. A first gain correction K_c is therefore performed, based on the IR measurement of the acoustic path ([F] block). This is adapted from [8], considering that F and $H_{bypassed}$ are linear, stationary and evaluated. The correction K_c for the 0 reference of ASG is thus given by:

$$K_c[dB] = -20\log_{10}\left[\max_{\omega\in P}\left|H_{bypassed}(\omega)F(\omega)\right|\right], \quad (1)$$

with $P = \{\omega | \angle H_{bypassed}(\omega)F(\omega) = n2\pi\}$ and $H_{bypassed}$ standing for the HR system bypassed with K_D = O dB and K_a set before the howling in order to include latent phase shift and gain from the AH system.

Then, a second gain correction, K_f, is performed because the actual gain changes may not be limited to the one adjusted by the operator (K_D factor); the AH system itself may also impact the gain. The actual gain is thus estimated from the spectra, using the following procedure: considering x_{ref} a reference signal and x_{test} the signal for another gain K_f , the following minimization is performed to estimate K_f:

$$\min \| \|To(PSD(x_{ref}))\|K_f - \|To(PSD(x_{test}))\| \|^2 \quad (2)$$

Here, To represents the signal's FFT averaged in one-third octave bands and PSD(x) is the power spectral density of x. The minimization is conducted within the frequency bands of 100 to 16000 Hz.

The actual gain K_{act} used as abscissa in the following figures is thus $K_{act} = K_f + K_c$.

When estimating the "Reactivity", both TRI are calculated by comparing the error between a howling-free reference signal and a feedback signal ($\|x_{ref} - x_{test}\| / \|x_{ref}\|$), determining the upper error envelope, defining a threshold (e.g., 10 dB above max error when there is no howling), and estimating the time at which this threshold is exceeded. This requires that the recorded signals are long enough to include this transition.

5 First results

The evaluated products include four commercial products and two prototypes. Table 2 provides an overview of the broad properties of the tested products: half of them use NHS and the other half use AFR. The NHS products are the most usual solution in commercial systems whereas only a few AFR products are commercially available yet.

Product	Feature	Type
P1	NHS	Physical
P2	NHS	Physical
P3	AFR	Physical
P4	NHS	Plugin
P5	AFR	Physical
P6	AFR	Physical

Table 2: Table of measured AH products

5.1 Sound quality versus gain increase

The results in Figure 5 illustrate the mean PESQ scores in relation to the gain increase K_{act} for all products, together with a reference pass-through configuration, considered as a reference. The ASG of each product is determined by the maximum gain reached by its curve. As stated above, O dB gain corresponds by definition to the maximum gain achievable without causing howling, when no AH system is present. The pass-through has therefore an ASG of O dB. Starting the gain scale at -7 dB provides valuable insights into how the AH systems impact the sound quality in addition to the MSG change.

It can be seen that all scores fall below 3, indicating poor speech quality according to the PESQ scale, even in the case of the pass-through. This reflects the deviation from the original audio caused by the electroacoustic path (mainly reverberation). This has already been pointed out by previous studies, which have shown that despite,the lower scores due to added reverberation, standard quality evaluation methods can still maintain the meaningfulness of the evaluation, preserving the overall trends and tendencies [31, 32, 33]. The scores given by Figure 5 should therefore be analyzed as relative values, taking the pass-through curve as a reference.

Keeping this in mind, results in Figure 5 exhibit two groups related to the AH method they employ. NHS products (P1, P2 and P4) tend to exhibit overall lower ASG (≈ 4 dB) than AFR products (P3, P5, P6), which reach ASG values above 15 dB). This is consistent with previous publications [8]. Moreover, all AFR-based products also improve the PESQ score by reducing reverberation, while NHS-based products do not change the sound quality for negative gains and tend to even degrade it for positive gains.

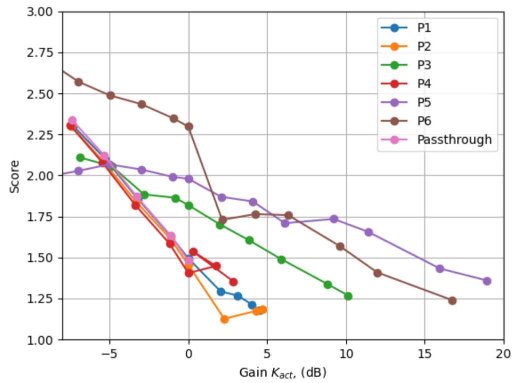

Fig. 5: Mean PESQ quality scores obtained using different AH systems versus gain increase.

5.2 Reactivity versus gain increase

The results in Figure 6 show the reactivity results for the algorithms during initial convergence (TRI-I), while Figure 7 displays the results in the case of an acoustic path change (TRI-C). Note that the abscissa gain values of Fig. 7 correspond to the initial acoustic path, which allows a higher MSG than the modified one. This explains why howling may occur even for 0 dB in this figure.

For the NHS products, Figure 6 illustrates a relatively fast convergence (below 2 seconds) only for added gain lower than 2 dB, whereas the AFR products converge even for gains of at least 8 dB. Notably, P3 demonstrates instant convergence up to 10 dB. The results in Figure 7 demonstrate a similar overall trend albeit less pronounced. NHS products do not seem robust to quick changes of the electroacoustic path, P1 being the fastest one but still needing about 10 s with a gain of 2 dB. Globally, AFR products seem more robust and show distinct TRI performances, P3 being the fastest.

In relation with Figure 5, this highlights a clear trade-off between convergence times and PESQ quality scores. Slower convergence seems to preserve speech quality more effectively.

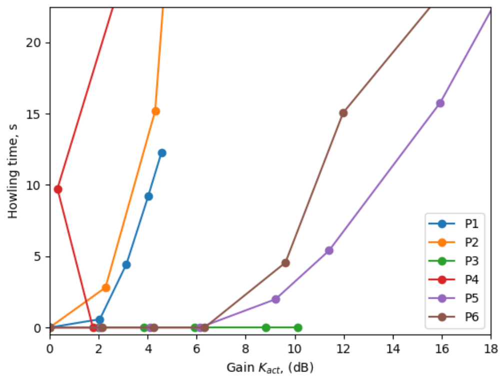

Fig. 6: Howling times during AH systems convergence versus gain increase.

6 Conclusion

The aim of this work was to bridge a gap by proposing a protocol allowing to compare different AH principles in the context of PA systems, and dealing with commercial products as well as laboratory mock-up. Moreover, we wanted to allow the comparison of evaluation results performed by various operators or teams.

The protocol described above represents a significant initial step in this direction. It has the capability to compare highly distinct systems, such as NHS- or AFR-based systems, and can highlight broad trends as well as subtle differences stemming from various design trade-offs. The primary trends observed align with previous publications, clearly illustrating the distinctions among AH principles.

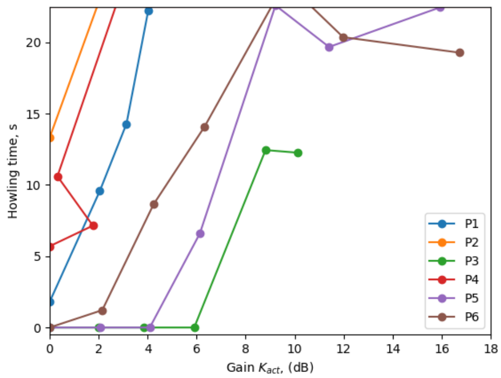

Fig. 7: Howling times during AH systems convergence after switching the acoustic path versus gain increase.

The proposed protocol is implemented using standard PC hardware and a common sound card, along with open-source and affordable software, making it easily replicable by other teams as necessary. We hope it would thus provide valuable insights for decision-making when choosing an AFC system based on specific requirements and priorities, but also for designing or tuning new products or including AH technology for new applications.

While the current evaluation method already allows to distinguish between different products in terms of a few accepted criteria, it can obviously be improved in some respects - hopefully through interactions with other interested teams.

The proposed implementation allows to easily vary the IRs used to emulate the electroacoustic path, which should be representative of a targeted application (broadly ranging from teleconference rooms to houses of worship). A suitable selection of IRs should be defined, and the corresponding results compared. The TRI-C test (time to adapt to a change of electroacoustic path) may also be more representative by using more realistic changes than switching between quite different IRs.

The influence of the implementation details (simulation and convolution latency, signal to noise ratio) should be investigated and specified in order to keep results comparable between different setups.

The quality evaluation, based only on the PESQ score, does not reflect the listener experience for musical diffusion. A complementary score could be defined, although this might lead to significant licence costs.

All these aspects are the object of current work, including the comparison of additional existing products and AH principles. Given the substantial resources required for these endeavors, we welcome any cooperation and collaboration to further enhance our efforts in this field.

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