INTRUSIVE HOWLING DETECTION METHODS FOR HEARING AID EVALUATIONS

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ABSTRACT

Audio applications often suffer from the acoustic feedback problem which leads to significant sound quality degradation and howling in the worst case. Many solutions exist to minimize the effect of feedback. However, it is not trivial to evaluate these solutions. In this work, we focus on intrusive howling detection methods by comparing a test signal to a known reference signal without howling. Traditional howling detection methods are less reliable when acoustic feedback control systems make use of some decorrelation techniques, such as frequency shifting and/or probe noise injection. In this paper, we propose two new simple detection methods which are robust against these processing strategies.

Index Terms— Howling detection, acoustic feedback, hearing aids, spectral divergence, false alarm.

1. INTRODUCTION

Acoustic feedback is a phenomenon which occurs when microphones of an audio device pick up the output loudspeaker signals from the same device, and an acoustic loop is thereby created. The acoustic feedback problem often causes significant sound quality degradation in audio systems. In the worst case, the audio system becomes unstable and clearly audible howling occurs. Some typical audio devices suffering from the feedback problem are public address systems and hearing aids.

Fig. 1 illustrates the acoustic feedback problem in a simple hearing aid system; for convenience, we denoted all signals as discrete signals with the time index n. The main goal of the hearing aid is to amplify the incoming signal $x(n)$ [1–3], the output loudspeaker signal $u(n)$ is ideally the amplified version of $x(n)$, the forward path amplification is simply modelled by a time-varying finite impulse response $\mathbf{f}(n)$. The acoustic feedback problem occurs as the loudspeaker signal $u(n)$ partly returns to the microphone as the feedback signal $v(n)$ through the acoustic feedback path modelled by the finite impulse response $\mathbf{h}(n)$.

Many solutions exist to minimize the effect of the acoustic feedback problem, see, e.g., [4] and the references therein. One of the best solutions so far is by using an adaptive filter $\hat{\mathbf{h}}(n)$ in a system identification configuration [5]. The adaptive filter $\hat{\mathbf{h}}(n)$ estimates the true acoustic feedback path $\mathbf{h}(n)$ and it generates a cancellation signal $\hat{v}(n)$. Ideally, $\hat{\mathbf{h}}(n) = \mathbf{h}(n)$ and we get a perfect cancellation, i.e., $\hat{v}(n) = v(n)$ and $e(n) = x(n)$. Some of the state-of-the-art feedback control systems can be found in [6–20].

In practice, perfect cancellation can not be achieved, and another question thereby arises: *how effective is a specific feedback control system to remove feedback?* Unfortunately, there are so far no standardized evaluation methods. An evaluation of a feedback control system needs typically to cover different aspects such as feedback cancellation performance and sound quality degradation, see, e.g., [17]. The feedback cancellation performance typically relates to the highest amplification in $\mathbf{f}(n)$ before the audio system becomes unstable, an example measure of this type is the maximum stable gain [21, 22]. However, this type of evaluation is not directly related to sound quality, which is another important issue in evaluations of feedback cancellation performance. There are different types of sound quality degradation, see, e.g., [17]. In this work, we focus on methods to determine the extreme case of the sound quality degradation —howling—.

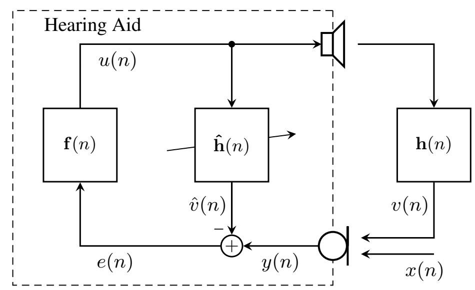

Fig. 1. A simple hearing aid system with acoustic feedback control.

There are two general types of howling detection. The first type is the so-called non-intrusive howling detection without a known reference signal. It is typically used together with a notch-filter-based howling suppression to control feedback, see, e.g., [23] and the references therein; the microphone signal $y(n)$ is typically used for the howling detection, which is often based on an analysis of the signal power of potential howling components compared to their frequency harmonics, and their neighbors in the time and frequency domains.

The second type of howling detection, the intrusive howling detection, has another goal. It is used for offline evaluation purposes only, and it performs detection by comparing a test signal to a known reference signal without howling, which is not available in the non-intrusive detections. With the access to the reference signal, this second type of howling detection is generally more reliable. We focus on this second type of howling detection in this work. A fast and reliable howling evaluation of this type is desirable to shorten the time needed to design and verify acoustic feedback control systems.

Some recently proposed evaluation criteria are power concentration ratio [24] and transfer function variation [25]. However, both criteria require that a white noise signal is used as the incoming signal $x(n)$ to the hearing aid, which provides a perfect condition for acoustic feedback cancellation systems; most (if not all) modern hearing aids handle this situation well without any howling. In contrast to this perfect condition, we are more interested in evaluating how often does a hearing aid howl in a much more challenging situation, e.g., when $x(n)$ contains tonal components such as most of music signals and alarm tones.

Modified versions of power concentration ratio are proposed in [26, 27] to allow colored incoming signals $x(n)$. Moreover, simple measures that work for colored signals $x(n)$ are presented to determine instability by comparing a feedback-free reference signal and a test signal [7, 26]. However, although these measures are generally very useful, we found that they are not reliable when certain feedback cancellation techniques are applied, such as probe noise injection and frequency shifting, more details of these techniques can, e.g., be found in [4, 17]. These signal processing techniques create additional signal components, which are not howling-related, but they can cause false howling detections.

In this work, we propose two new detection methods which work well for traditional feedback cancellation systems, and they are also robust against probe noise injection and frequency shifting. The first method is a further development of the howling detection based on the output-to-reference signal ratio [27], which determines howling over time by comparing broadband short-time signal energies between the output and reference signals. We include additional processing such as frequency range limitation and comparisons of both absolute and relative power levels. Furthermore, it detects howling in both the time and frequency domains.

The extended output-to-reference signal ratio measure turns out to be very reliable to detect howling, but it has limited ability to sound quality evaluation due to its binary decision characteristic. Therefore, we developed another method based on spectral shape divergence. The similarity between the spectra of the test and the reference signals is determined, and we declare howling when the similarity measure exceeds a certain threshold value. The similarity measure is a continuous measure, and a mapping to perceived sound quality is possible, although we consider this part as future work.

In Sec. 2 we present the proposed howling detection methods. We comment on the key elements of these two methods in Sec. 3. After that, we verify both methods by comparing their outputs to human listener detection results in Sec. 4. Finally, Sec. 5 concludes the paper.

2. HOWLING DETECTION METHODS

As mentioned in the introduction, both proposed howling detection methods compare a test signal $t(n)$ to a reference signal $r(n)$.

The test signal $t(n)$ is the processed hearing aid loudspeaker signal $u(n)$ with desired settings in $\mathbf{f}(n)$, and it is affected by the feedback signal $v(n)$ as shown in Fig. 1. On the other hand, the ideal reference signal is given by the hearing aid output signal $u(n)$ without feedback, i.e., $v(n) = 0$.

There are several methods to create the reference signal $r(n)$. In a very simple case, the reference signal $r(n)$ can be modeled as the incoming signal $x(n)$ compensated with a delay and an amplification according to $\mathbf{f}(n)$. More sophisticated methods exist such as normalizing the incoming signal $x(n)$ with the spectral shape of $\mathbf{f}(n)$, estimating the reference signal as $r(n) = \alpha \cdot u(n)$ after lowering the amplitude of $\mathbf{f}(n)$ by a factor of α to achieve $v(n) \approx 0$ etc., see more examples in, e.g., [27]. In this work, the reference signal $r(n)$ is simply obtained by lowering the amplification of $\mathbf{f}(n)$ using the factor $\alpha = 2$ and subsequently creating $r(n) = 2 \cdot u(n)$.

In the following, first we present the common part of both detection methods, then we present the remaining parts of each of them.

2.1. Preprocessing - The Common Part

In this section, we present the initial processing of the reference and test signals $r(n)$ and $t(n)$, respectively. This process includes the time-frequency (T-F) domain transformation, frequency range limitation, T-F domain smoothing, and the normalization. We perform identical processing on $r(n)$ and $t(n)$, for simplicity we only show these for $r(n)$.

First, we compute the T-F representation $R_f(m', k)$ using the short-time Fourier transform, as

$$R_f(m', k) = \left| \sum_{l=0}^{N_{DFT}-1} w(l) r(kN_D + l) e^{j2\pi l m'/N_{DFT}} \right|, \quad (1)$$

where N_{DFT} is the DFT size, N_D is the decimation factor, $m' = 0, ..., N_{DFT} - 1$ is the frequency index, and $k = 0, ...$ is the T-F domain time index, $w(l)$ with $l = 0, ..., N_{DFT} - 1$ is a window function.

Then, we discard the values of $R_f(m', k)$ at the very low and high frequencies such as $m' < m_l$ and $m' > m_h$, where $m_l < m_h$ and $m_l, m_h \in [0, N_{DFT}/2]$. We refer to the resulting matrix as $R_l(m, k)$, where $m = 0, 1, ..., M - 1$.

The next step is to smooth $R_l(m, k)$, by using a 2-D filter $b(l_m, l_k)$ with the dimension $L_m \times L_k$, as

$$R_s(m, k) = \sum_{l_m=0}^{L_m-1} \sum_{l_k=0}^{L_k-1} b(l_m, l_k) R_l(m - l_m, k - l_k). \quad (2)$$

Similarly, we obtain $T_s(m, k)$ for the test signal $t(n)$.

Finally, we normalize $R_s(m, k)$ and $T_s(m, k)$ so that a constant signal level is ensured at this stage. For the extended output-to-reference signal ratio method described in Sec. 2.2, we normalize $R_s(m, k)$ and $T_s(m, k)$ as

$$R_1(m, k) = \frac{R_s(m, k)}{\sigma_r}, \quad (3)$$

$$T_1(m, k) = \frac{T_s(m, k)}{\sigma_t}, \quad (4)$$

where σ_r and σ_t denote the standard deviation of the reference signal $r(n)$ and the test signal $t(n)$, respectively.

For the spectral divergence based method described in Sec. 2.3, we normalize $R_s(m, k)$ and $T_s(m, k)$, so that the spectrum at each time index k has unit-sum, as

$$R_2(m, k) = \frac{R_s(m, k)}{\sum_{l=0}^{M-1} R_s(l, k)}, \quad (5)$$

$$T_2(m, k) = \frac{T_s(m, k)}{\sum_{l=0}^{M-1} T_s(l, k)}. \quad (6)$$

Moreover, we define the difference ratio $D_x(m, k)$, for $x = 1, 2$, as

$$D_x(m, k) = \frac{T_x(m, k)}{R_x(m, k) + \delta}, \quad (7)$$

where δ is a small positive scalar to ensure numerical stability. The same designation of δ is also used in Sec. 2.3.

2.2. Extended Output-to-Reference Signal Ratio Method

In this method, the detection principle is based on a comparison of absolute and relative signal levels. We start the detection by declaring all T-F units as howling regions by setting $H_1(m,k) = 1$ for all m and k; then we go through different steps to assess if any unit can be declared as howling-free.

First, we declare the regions with low absolute levels, i.e., the test signal $T_1(m,k)$ is below the threshold λ_a, as howling-free,

$$H_1(m,k) = \begin{cases} 0 & \text{if } T_1(m,k) < \lambda_a, \\ H_1(m,k) & \text{otherwise.} \end{cases} \tag{8}$$

Second, we compare the test and reference signals and declare the regions with only a small amplification as howling-free regions

$$H_1(m,k) = \begin{cases} 0 & \text{if } D_1(m,k) < \lambda_r, \\ H_1(m,k) & \text{otherwise.} \end{cases} \tag{9}$$

Moreover, we can optionally declare the neighboring regions of $D_1(m,k) < \lambda_r$ as howling-free at this stage.

Finally, we correct all howling detection results $H_1(:,k)$ at each time index k by comparing the feedback to signal power $FSR(k)$ to a threshold λ_f,

$$H_1(:,k) = \begin{cases} 0 & \text{if } FSR(k) < \lambda_f, \\ H_1(:,k) & \text{otherwise,} \end{cases} \tag{10}$$

where the $FSR(k)$ is given by

$$FSR(k) = \frac{\sum_{l=0}^{M-1} \left(T_1(l,k)H_1(l,k)\right)^2}{\sum_{l=0}^{M-1} \left(T_1(l,k)(1-H_1(l,k))\right)^2}. \tag{11}$$

Should we only be interested in howling occurrence over time, we can simply ignore the frequency information by declaring howling $h_1(k)$ as

$$h_1(k) = \max_m H_1(m,k). \tag{12}$$

2.3. Spectral Divergence Based Method

This method is based on the spectral difference $R_2(m,k)$ and $T_2(m,k)$, and we use the Kullback-Leibler divergence measure $KL(k)$ [28] to express the difference, as

$$KL(k) = \sum_{m=0}^{M-1} T_2(m,k) \log 2\left(D_2(m,k) + \delta\right). \tag{13}$$

Moreover, several additional measures such as spectral flatness measure $SFM(k)$, the correlation measure $C(k)$, and the signal ratio $SR(k)$ are used to modify $KL(k)$ to make it more reliable. The spectral flatness measure $SFM(k)$ is computed as [29],

$$SFM(k) = \frac{\left(\prod_{l=0}^{M-1} D_2(l,k)\right)^{\frac{1}{M}}}{\frac{1}{M}\sum_{l=0}^{M-1} D_2(l,k) + \delta}. \tag{14}$$

The correlation measure $C(k)$ is determined by

$$C(k) = \begin{cases} 1 & \text{if } c(k) > \lambda_c, \\ 0 & \text{otherwise,} \end{cases} \tag{15}$$

where $c(k)$ is the Pearson's correlation coefficient between $D_2(:,k)$ and $T_2(:,k)$. Furthermore, the signal ratio $SR(k)$ is expressed by,

$$SR(k) = \max\left(\frac{\sum_{l=0}^{M-1} T_2^2(m,k)}{\sum_{l=0}^{M-1} R_2^2(m,k) + \delta}, 1\right). \tag{16}$$

The corrected $KL(k)$ measure $\xi(k)$ is then computed as

$$\xi(k) = (1 - SFM(k)) \cdot C(k) \cdot SR(k) \cdot KL(k). \tag{17}$$

Finally, the howling detection $h_2(k)$ is determined by comparing the measure $\xi(k)$ to a threshold λ_h, as

$$h_2(k) = \begin{cases} 1 & \text{if } \xi(k) > \lambda_h \\ 0 & \text{otherwise.} \end{cases} \tag{18}$$

3. DISCUSSIONS

In this section, we discuss the various stages presented in Sec. 2. In general, we designed both methods based on a few howling characteristics, e.g., the high energy concentration at the howling frequency(ies), leading to a peaky spectrum with single/multiple peaks.

3.1. Preprocessing

In Sec. 2.1, we created $R_l(m,k)$ from $R_f(m',k)$, because howling occurs relatively rare at the very low and high frequencies in a hearing aid system; it is due to the characteristics of typical feedback paths $\mathbf{h}(n)$ and the applied microphones and loudspeakers. Nevertheless, this step can easily be bypassed by setting $m_l = 0$ and $m_h = N_{DFT}/2$.

In (2), we performed a smoothing in the T-F domain. The smoothing effect can also be obtained by using other parameters N_{DFT} and N_D in (1). However, we prefer the smoothing to be a separate step, since it keeps the T-F resolution and the smoothing independent of each other. Moreover, the normalization in (3)-(6) ensures that signal level independent parameter values can be used for the remaining calculations.

3.2. Extended Output-to-Reference Signal Ratio Method

Equation (8) utilizes the fact that the T-F units $R(m,k)$ with very low signal energy are unlikely to be feedback howling. Furthermore, howling would typically lead to that $T_1(m,k) \gg R_1(m,k)$, which is the detection criterion in (9). Finally, we compare the energies between the frequency units declared to contain feedback howling (the numerator) and the remaining howling-free units (the denominator) at each time index k in (11); the underlying assumption was that the howling region contains much more energy.

Each stage is very simple in this method; the main challenge in using this method is to determine the appropriate threshold values λ_a, λ_r, and λ_f. We refer to Sec. 4 for more details.

3.3. Spectral Divergence Based Method

In this method, we initially use the Kullback-Leibler divergence measure $KL(k)$ to determine the difference between $R_2(m,k)$ and $T_2(m,k)$ in (13). This measure is originally known from probability and information theory. In principle, any divergence measure can be used at this stage; however, we found the $KL(k)$ measure to be reliable and it is thereby chosen.

Although howling always leads to a large numerical value of $KL(k)$ because of the significant difference in $R_2(m,k)$ and

$T_2(m,k)$, a large value of $KL(k)$ does not necessarily mean the presence of howling. This could be the case when probe noise injection or frequency shifting is applied in the forward path of an acoustic feedback cancellation system. Hence, we need to modify the measure $KL(k)$ for more reliable howling detection.

The spectral flatness measure $SFM(k) \in [0,1]$ in (14) determines how peaky the difference spectrum $D_2(m,k)$ is. If howling occurred, the difference spectrum $D_2(:,k)$ at a given time index k is peaky, i.e., there are few values in $D_2(m,k)$ which are significantly larger than the remaining values, because $T_2(m,k) \gg R_2(m,k)$ at the T-F units m and k, and $SFM(k) \to 0$. Otherwise, $D_2(m,k)$ should be relatively flat, it means that there is no big difference between its lowest and highest values and $SFM(k) \to 1$. Using $SFM(k)$ in (17) ensures that only large spectral divergence due to a peaky difference spectrum $D_2(m,k)$ is encountered.

Furthermore, the correlation measure $C(k)$ in (15) is specifically introduced to deal with probe noise injection in feedback cancellation systems. When a probe noise signal is generated using spectral masking effect, see, e.g., [16], the spectral shape of the generated probe noise in the presence of a tonal signal can be very peaky. Hence, $SFM(k)$ is not sufficient to correct the large value of $KL(k)$ due to the presence of a probe noise with peaky spectrum. Therefore, we introduced the measure $C(k)$ in (15) to correct $KL(k)$ in (17). In the case that the spectrum $T_2(m,k)$ is dominated by a tonal signal, $D_2(m,k)$ is dominated by the probe noise with a peaky but broadband spectrum, it gives thereby $C(k) = 0$.

Finally, we introduced the measure $SR(k)$ in (16); a large value of $KL(k)$ shows only a large difference between $R_2(m,k)$ and $T_2(m,k)$, but it does not take into account the fact of $T_2(m,k) \gg R_2(m,k)$ during howling. Hence, we lower the value of $KL(k)$ if the signal $R_2(m,k)$ has more energy than $T_2(m,k)$. This step is similar to (9) in the first method.

To summarize, this spectral divergence method provides a continuous measure $\xi(k)$, and we only use a single threshold λ_h to determine howling. In this method, we do not have the frequency information about howling as the first method presented in Sec. 2.2. However, it should be possible to adapt the continuous measure $\xi(k)$ to evaluate sound quality, which we consider as future work.

4. VERIFICATIONS

In total, we successfully verified both detection methods in 86 test cases with different signals and hearing aid settings, with either simulated or recorded hearing aid output signals. In the following, we present two example verification experiments.

In both cases, we use an incoming signal $x(n)$ which has a duration of 600 s. It consists of different segments including an initial white noise sequence and challenging tonal signals. Fig. 2 shows the spectrogram of the incoming signal $x(n)$ consisting of 60 s of white noise, 30 s of castanets music, 35 s of kettle whistling, 100 s of classical music, 120 s of flute music, 20 s of bird song, 20 s of cicada song, 62 s of pings (varying intervals), and 153 s of pure tones.

We perform the verification using computer simulations. The hearing aid parameters are set for a moderate hearing loss. Due to the relatively large amplification, the hearing aid system would be unstable with constant howling (at approximately 3 kHz) without a properly working feedback control system. In the first experiment, the hearing aid utilizes a frequency shifting of 10 Hz in its feedback control system, whereas in the second experiments we use probe noise injection in the hearing aid feedback control system.

In both cases, the hearing aid is kept stable due to the use of feedback control systems. However, simple howling measures would fail

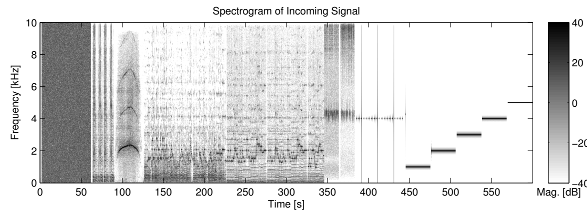

Fig. 2. The spectrogram of the incoming signal $x(n)$.

Table 1. Howling periods [s-s] when using frequency shifting.

Detection	Reference	Method 1	Method 2
Howling #1	237.3 - 237.6	237.1 - 237.9	237.3 - 237.5
Howling #2	263.5 - 263.8	263.3 - 264.2	263.6 - 263.8
Howling #3	520.8 - 521.2	520.8 - 521.4	520.9 - 521.0

Table 2. Howling periods [s-s] when using probe noise injection.

Detection	Reference	Method 1	Method 2
Howling #1	360.0 - 360.1	360.1 - 360.4	360.0 - 360.3

Table 3. False alarm periods [s-s] using a traditional evaluation.

Music Signals	Castanets	Classic	Flute
False Alarms	69.4 - 71.1	186.1 - 187.2	323.8 - 324.2

because of the artifacts introduced by the frequency shifting and the probe noise injection, although these artifacts might not be audible. Furthermore, we repeatedly modify the acoustic feedback path $\mathbf{h}(n)$ during the simulations to provoke audible howling [30].

In our simulations, we choose $N_{DFT} = 512$, $N_D = 256$, $m_l = 17$, $m_h = 240$, and $b(l_m, l_k) = \frac{1}{64}$, where $l_m = 0, ..., 15$ and $l_k = 0, ..., 3$. Furthermore, we already determined optimal parameter values at each stage in (8)-(18) in a pre-design phase, according to observed signal statistics; these values were found to be $\lambda_a = -105$ dB, $\lambda_r = 20$ dB, $\lambda_f = -16$ dB, $\lambda_c = 0.3$, and $\lambda_h = 0.2$. We used these values in the verification phase as well.

Table 1 and 2 show the detection results in the experiments with frequency shifting and probe noise injection, respectively. The detection results include both methods proposed in Sec. 2 and the howling detections determined by expert listeners, which are used as the reference. In both cases, the results indicate howling occurrences over time, and both proposed methods provide consistent results without false alarms, which are perfectly in line with the references.

In comparison, we often obtain false alarms in music and speech regions when evaluating probe noise processed signals using traditional methods based on power comparison; this is due to additional signal components (even inaudible) with significant power are introduced [17]. Table 3 shows selected false alarm examples when using a simple measure, the output-to-reference signal ratio [27], on probe noise processed music signals. Similarly, traditional measures often classify frequency shifted spectral contents as howling.

Moreover, should one be interested in determining the exact howling frequency, our extended signal ratio method is very robust (although not illustrated due to space limitations).

5. CONCLUSION

In this work, we presented two new howling detection methods for hearing aids. In contrast to existing howling detection methods, both provide reliable results for colored incoming signals $x(n)$, and even more importantly, they are reliable in the presence of feedback control system artifacts introduced by frequency shifting and probe noise. Both methods can detect howling very precisely, as we verified by simulations and with recorded hearing aid output signals.

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