ACOUSTIC FEEDBACK CANCELLATION FOR HEARING AIDS USING A FIXED RTF-CONSTRAINED NULL-STEERING BEAMFORMER

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ABSTRACT

Recently it has been proposed to use a fixed beamformer to cancel the acoustic feedback for a custom earpiece with multiple integrated microphones and loudspeakers. By steering a spatial null in the direction of the hearing aid loudspeaker theoretically perfect feedback cancellation can be achieved. In contrast to previous approaches that constrained the beamformer coefficients in a reference microphone to a simple delay, in this paper we use a constraint based on the relative transfer function of the incoming signal aiming to perfectly preserve the incoming signal. Experimental results using measured acoustic feedback paths from a custom earpiece with three microphones show that the proposed RTF constrained null-steering beamformer allows to substantially increase the added stable gain even for unknown acoustic feedback paths, e.g., with a telephone receiver close to the ear. Furthermore, it yields a high perceptual speech quality of the incoming signal even for unknown incoming signal directions.

Index Terms— acoustic feedback cancellation, null-steering beamformer, relative transfer function, multi-microphone hearing aids

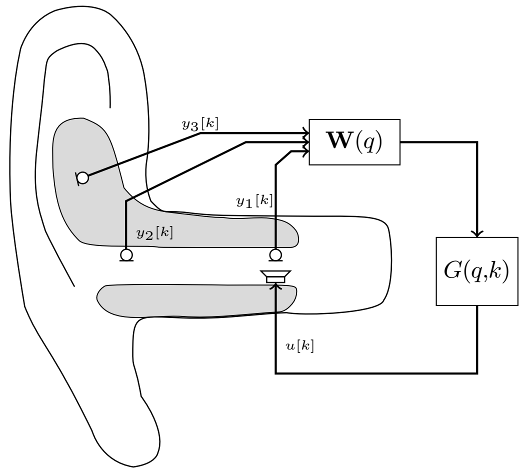

Fig. 1. Considered hearing aid setup with a single-loudspeaker three-microphone earpiece.

1. INTRODUCTION

Due to the acoustic coupling between the hearing aid loudspeaker and microphones(s), acoustic feedback is a common problem limiting the maximum applicable gain in hearing aids. Most often acoustic feedback is perceived as whistling or howling. In order to reduce the acoustic feedback and increase the maximum gain that can be applied in the hearing aid, robust feedback suppression strategies are required.

To cancel acoustic feedback an adaptive feedback cancellation (AFC) scheme is frequently used, which theoretically allows to perfectly remove the feedback component in the microphone. In an AFC scheme an adaptive filter models the acoustic feedback path(s) between the hearing aid loudspeaker and the microphone(s) [1–8]. However, due to the closed-loop acoustical system of the hearing aid, the filter adaptation is usually biased, e.g., [2, 9]. In order to reduce this bias, different approaches have been proposed, e.g., the so-called prediction-error-method [1, 2, 10], using an additional probe noise [4, 5] or using phase modulation and frequency shifting [11]. Furthermore, different approaches have been proposed that exploit multiple microphones in multi-microphone hearing aids, e.g., by adaptively removing the incoming signal in the filter adaptation [12, 13], by using a fixed null-steering beamformer to cancel the acoustic feedback [14–17], or by using a combined multi-microphone feedback cancellation and noise reduction scheme [18, 19].

In this paper we consider a custom multi-microphone hearing aid and propose to cancel the acoustic feedback using a fixed null-steering beamformer. In particular, we consider a custom earpiece [20, 21] (see Figure 1) with two closely spaced microphones and a loudspeaker in the vent and a third microphone located in the concha. In contrast to conventional behind-the-ear hearing aids, this earpiece design allows to design a fixed beamformer with a spatial null in the direction of the hearing aid loudspeaker located in the vent [14–16]. Thus, the null-steering beamformer ideally cancels all signals originating from the hearing aid receiver and does not impact the incoming (external) signal. Similarly as in [15], we propose to compute the null-steering beamformer to minimize the residual feedback power using multiple sets of acoustic feedback paths measurements. However, instead of using a constraint of a fixed delay in a reference microphone which does not directly control for any distortions of the incoming signal as in [15], in this paper we propose to incorporate a constraint based on the relative transfer function (RTF) of the incoming signal aiming to preserve the incoming signal in the beamformer output.

Experimental results using measured acoustic feedback paths show that the proposed fixed RTF-constrained null-steering beamformer outperforms the fixed delay-constrained null-steering beamformer in [15]

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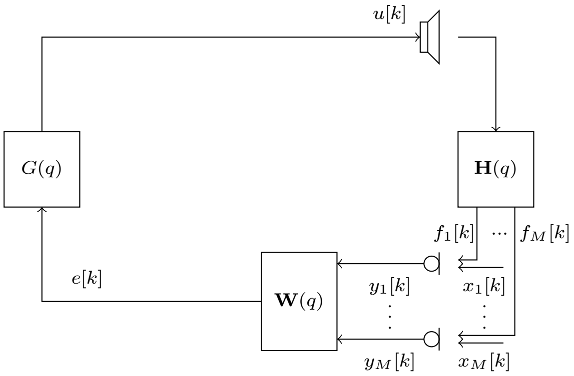

Fig. 2. Generic single-loudspeaker multi-microphone closed-loop hearing aid system.

and enables to substantially reduce the acoustic feedback. At the same time it preserves a high perceptual speech quality of the incoming signal even for changing incoming source directions. Furthermore, the fixed beamformer also enables to increase the added stable gain for changing acoustic conditions, i.e., after repositioning of the earpiece and with a telephone receiver close to the ear.

2. ACOUSTIC SCENARIO AND NOTATION

Consider a single-loudspeaker multi-microphone hearing aid system with M microphones as depicted in Figure 2. For simplicity we assume that all transfer functions are linear and time-invariant. The mth microphone signal $y_m[k]$, $m=1,...,M$, at discrete time k is the sum of the incoming signal $x_m[k]$ and the loudspeaker contribution in the mth microphone $f_m[k]$. Using vector and matrix notation this can be written as

$$\mathbf{y}[k]=\mathbf{x}[k]+\underbrace{\mathbf{H}(q)u[k]}_{\mathbf{f}[k]}, \tag{1}$$

with the M-dimensional vectors

$$\mathbf{y}[k]=\begin{bmatrix}y_1[k] & y_2[k] & ... & y_M[k]\end{bmatrix}^T, \tag{2}$$
$$\mathbf{x}[k]=\begin{bmatrix}x_1[k] & x_2[k] & ... & x_M[k]\end{bmatrix}^T, \tag{3}$$
$$\mathbf{H}(q)=\begin{bmatrix}H_1(q) & H_2(q) & ... & H_M(q)\end{bmatrix}^T, \tag{4}$$

where $[\cdot]^T$ denotes transpose operation, $u[k]$ denotes the loudspeaker signal and $H_m(q)$ denotes the acoustic feedback path between the mth microphone and the loudspeaker. We assume that the acoustic feedback path can be modelled as an L_H-dimensional polynomial in q, using a notation adopted from [22], i.e.,

$$H_m(q)=\mathbf{h}_m^T\mathbf{q}=\sum_{i=0}^{L_H-1} h_{m,i}q^{-i}, \tag{5}$$

where $\mathbf{q}$ denotes the vector containing the delay-elements of q of appropriate length and $\mathbf{h}_m$ denotes L_H-dimensional vector of the impulse response of the mth acoustic feedback path. After applying a fixed filter-and-sum beamformer to the microphone signals, the beamformer output signal $e[k]$ is obtained as

$$e[k]=\mathbf{W}^T(q)\mathbf{y}[k]=\underbrace{\mathbf{W}^T(q)\mathbf{x}[k]}_{\tilde{x}[k]}+\underbrace{\mathbf{W}^T(q)\mathbf{f}[k]}_{\tilde{f}[k]}, \tag{6}$$

where $\mathbf{W}(q)$ denotes the M-dimensional vector of the beamformer weighting functions, i.e.,

$$\mathbf{W}(q)=\begin{bmatrix}W_1(q) & ... & W_M(q)\end{bmatrix}^T, \tag{7}$$

and $\tilde{x}[k]$ and $\tilde{f}[k]$ are the residual incoming signal and residual feedback component, respectively. The L_W-dimensional beamformer coefficient vector of $W_m(q)$ for the mth microphone is defined as

$$\mathbf{w}_m=\begin{bmatrix}w_{m,0} & ... & w_{m,L_W-1}\end{bmatrix}^T, \tag{8}$$

and the ML_B-dimensional stacked vector is defined as

$$\mathbf{w}=\begin{bmatrix}\mathbf{w}_1^T & ... & \mathbf{w}_M^T\end{bmatrix}^T. \tag{9}$$

The signal $e[k]$ is then processed using the hearing aid forward path $G(q)$, yielding the loudspeaker signal $u[k]$, i.e.,

$$u[k]=G(q)e[k]. \tag{10}$$

Furthermore, we assume that the incoming signal $\mathbf{x}[k]$ is composed of a single directional speech source $s[k]$, i.e.,

$$\mathbf{x}[k]=\mathbf{D}(q)s[k], \tag{11}$$

where $\mathbf{D}(q)$ is the M-dimensional vector containing the acoustic transfer function of length L_D between the source and each of the M microphones, i.e.,

$$\mathbf{D}(q)=\begin{bmatrix}D_1(q) & ... & D_M(q)\end{bmatrix}^T. \tag{12}$$

The incoming signal $\mathbf{x}[k]$ can also be defined by using the RTF between a reference microphone m_0 and the remaining microphones, i.e.,

$$\mathbf{x}[k]=\tilde{\mathbf{D}}(q)x_{m_0}[k]=\tilde{\mathbf{D}}(q)D_{m_0}(q)s[k], \tag{13}$$

where $\tilde{\mathbf{D}}(q)$ is the M-dimensional vector containing the RTF between the microphones, i.e.,

$$\tilde{\mathbf{D}}(q)=\frac{\mathbf{D}(q)}{D_{m_0}(q)}, \tag{14}$$

with $D_{m_0}(q)$ the acoustic transfer function between the source and the reference mirophone m_0. The $L_{\tilde{D}}$-dimensional impulse response vector of the RTF for the mth microphone is defined as

$$\tilde{\mathbf{d}}_m=\begin{bmatrix}\tilde{d}_{m,0} & ... & \tilde{d}_{m,L_{\tilde{D}}-1}\end{bmatrix}^T. \tag{15}$$

3. SYSTEM ANALYSIS

In the following we briefly review the analysis of the transfer function of the hearing aid system depicted in Figure 2 as provided in [14–16]. By combining (1), (6), and (10) we can rewrite the loudspeaker signal as

$$u[k]=\frac{G(q)\mathbf{W}^T(q)}{1-G(q)\mathbf{W}^T(q)\mathbf{H}(q)}\mathbf{x}[k]. \tag{16}$$

From this expression it can observed that perfect feedback cancellation for the considered system can be achieved if the beamformer cancels the feedback contribution in the microphones, i.e.,

$$\mathbf{W}^T(q)\mathbf{H}(q)=0, \tag{17}$$

with $W_m(q)\neq 0$ for at least one $m\in[1,\dots,M]$ to avoid the trivial solution $\mathbf{W}(q)=0$.

Furthermore, if (17) holds, then from (16) and using (11) we obtain

$$u[k]=G(q)\mathbf{W}^T(q)\mathbf{D}(q)s[k]. \tag{18}$$

Note that although (17) perfectly solves the feedback cancellation problem, applying the beamformer coefficients will hence also modify the incoming signal $\mathbf{x}[k]$, possibly leading to sound quality degradation.

4. FIXED NULL-STEERING BEAMFORMER

In this section we consider the least-squares design of a fixed null-steering beamformer to minimize the residual feedback power while preserving the incoming signal in the beamformer output. While in [14–16] the trivial solution $\mathbf{w}=\mathbf{0}$ was mitigated by using a fixed delay in a reference microphone, in this paper we propose to incorporate knowledge of the incoming signal in the optimization of the null-steering beamformer. In particular, we propose to use the RTF of the incoming signal as a hard constraint aiming to perfectly preserve the incoming signal in a reference microphone m_0 in the beamformer output, i.e., $\tilde{x}[k]=x_{m_0}[k]$. In order to compute the fixed null-steering beamformer, in the following we assume knowledge of multiple (I) sets of acoustic feedback paths, e.g., by measurement. This allows to incorporate knowledge about typical variations of the acoustic feedback path in the design and hence achieve a robust design [15]. Furthermore, we assume knowledge of the acoustic transfer functions $\mathbf{D}(q)$ between the source and the earpiece microphones or their corresponding RTF $\tilde{\mathbf{D}}(q)$. Note that since the null-steering beamformer is fixed, it can be computed a priori using an offline procedure and does not need to be computed on the hearing device.

By applying the beamformer to the incoming signal $\mathbf{x}[k]$ in (11) the beamformer output for the incoming signal yields $\tilde{x}[k] = \mathbf{W}^T(q)\mathbf{D}(q)s[k]$. Similarly, this can be done for the definition of the incoming $\mathbf{x}[k]$ in (13) using the RTF where the beamformer output for the incoming signal yields $\tilde{x}[k]=\mathbf{W}^T(q)\tilde{\mathbf{D}}(q)x_{m_0}[k]$. Hence, if the beamformer output for the RTF of the incoming signal yields a unit (or an L_d samples delayed unit) response, i.e.,

$$\mathbf{W}^T(q)\tilde{\mathbf{D}}(q)=q^{-L_d}, \tag{19}$$

the incoming signal is preserved. This can be formulated using matrix and vector notation of the impulse responses (IRs) as

$$\tilde{\mathbf{D}}\mathbf{w}=\check{\mathbf{e}}_{L_d}, \tag{20}$$

with $\check{\mathbf{e}}_{L_d}$ the $(L_{\tilde{D}}+L_W-1)$-dimensional vector of zeros and the (L_d+1)th element equal to 1, and $\tilde{\mathbf{D}}$ is the $(L_{\tilde{D}}+L_W-1)\times ML_W$-dimensional matrix of concatenated $(L_{\tilde{D}}+L_W-1)\times L_W$ convolution matrices $\tilde{\mathbf{D}}_m$, i.e.,

$$\tilde{\mathbf{D}}=\begin{bmatrix}\tilde{\mathbf{D}}_1 & \dots & \tilde{\mathbf{D}}_M\end{bmatrix}. \tag{21}$$

The optimization problem to obtain the beamformer coefficients by minimizing the residual feedback power subject to the preservation of the incoming signal can thus be formulated as the following linearly constrained least-squares optimization problem

$$\min_{\mathbf{w},i} \quad \sum_{i=1}^{I}\|(\mathbf{H}^{(i)})^T\mathbf{w}\|_2^2 \tag{22a}$$

$$\text{s. t.} \quad \tilde{\mathbf{D}}\mathbf{w}=\check{\mathbf{e}}_{L_d} \tag{22b}$$

where $\mathbf{H}^{(i)}$ is the $L_W M\times(L_H+L_W-1)$-dimensional convolution matrix of the ith set of acoustic feedback path measurements similarly defined as the convolution of the RTFs $\tilde{\mathbf{D}}$ in (21).

The closed-form solution of this optimization problem is given by

$$\mathbf{w}=(\bar{\mathbf{H}}^T\bar{\mathbf{H}})^{-1}\tilde{\mathbf{D}}^T(\tilde{\mathbf{D}}(\bar{\mathbf{H}}^T\bar{\mathbf{H}})^{-1}\tilde{\mathbf{D}}^T)^{-1}\check{\mathbf{e}}_{L_d}, \tag{23}$$

where $\bar{\mathbf{H}}$ is the $(L_H+L_W-1)I\times ML_W$-dimensional matrix of stacked convolution matrices $\mathbf{H}^{(i)}$, $i=1,\dots,I$.

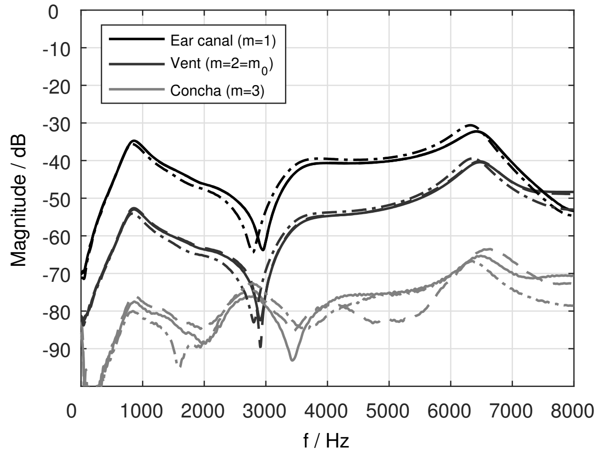

Fig. 3. Amplitude response of the measured acoustic feedback paths. Continuous lines show feedback paths in free-field, i.e., without any obstruction (used for computing the beamformer coefficients), dashed dotted lines show an exemplary responses after repositioning of the earpiece, and dashed lines show the acoustic feedback paths in the presence of a telephone receiver.

5. EXPERIMENTAL EVALUATION

In this section the performance of the proposed RTF-constrained null-steering beamformer is evaluated and compared with the existing approach in [15] that used a fixed delay in a reference microphone. In particular, we consider the ability to cancel the acoustic feedback in a challenging acoustic scenario as well as the preservation of the incoming signal for different incoming signal directions.

Acoustic feedback paths and acoustic transfer functions were measured for the three-microphone earpiece ($M=3$) as depicted in Figure 1 on a dummy head with adjustable ear canals [23]. The IRs of the acoustic feedback paths and acoustic transfer functions were sampled at $f_s=16$ kHz and truncated to length $L_H=100$ and $L_D=3000$. Measurements were performed in an acoustically treated chamber ($T_{60}\approx 300$ ms) and the distance between the external source and the dummy head was approximately 1.2 m, where the source was either positioned in front, 90 degrees to the right, in the back or 90 degress to the left of the dummy head. The RTFs of the incoming signal where computed using a regularized least-squares optimization with $L_{\tilde{D}}=8$ and $L_d=0$ and $m_0=2$. Figure 3 shows the amplitude responses of the measured acoustic feedback paths for the three different microphones and for different acoustic conditions. The forward path of the hearing aid was set to $G(q,k)=q^{-96}10^{45/20}$, corresponding to a delay of 6 ms and a broadband amplification of 45 dB. For all experiments the reference microphone $m_0=2$, i.e., the microphone located at the outer phase of the vent, was chosen since it includes most of the relevant spectral and directional cues and hence provides a natural position for sound pickup. For the optimization using the fixed delay constraint we used $L_d=L_W/2$ as suggested in [14, 15].

We evaluated the feedback cancellation performance of the null-steering beamformer using the added stable gain (ASG) [1, 25] and the perceptual quality of the signal after applying the null-steering beamformer using the perceptual quality of speech (PESQ) measure [24].

The ASG for the considered hearing aid setup is computed similarly

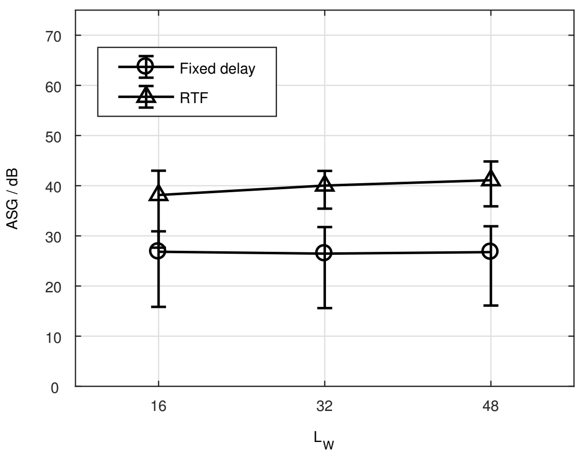

Fig. 4. Average ASG as a function of the beamformer length L_W, showing the robust performance for the proposed RTF-based constraint and the fixed delay constraint [15]. Errorbars indicate minimum and maximum ASG values.

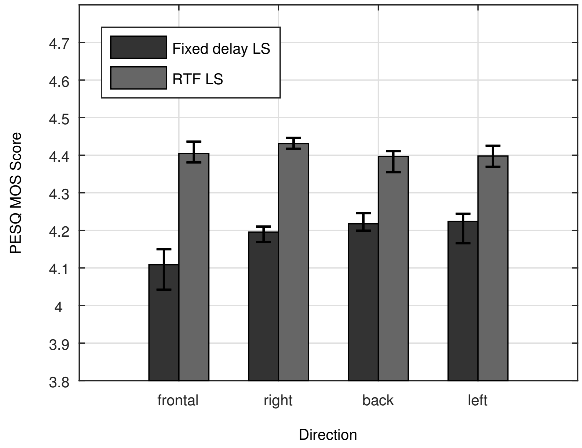

Fig. 5. Average PESQ MOS scores for different incoming signal directions using the proposed RTF-based constraint and the fixed delay constraint [15] for $L_W = 48$. Errorbars indicate minimum and maximum PESQ MOS scores.

as in [14–16] as

$$ASG = 20\log_{10}\frac{\max\limits_{\omega}|H_{m_0}(\omega)|}{\max\limits_{\omega}|\mathbf{H}^H(\omega)\mathbf{W}(\omega)|}, \tag{24}$$

with ω denoting the angular frequency. The reference signal for the PESQ measure was the incoming signal $x_{m_0}[k]$ in the reference microphone. In order to assess only the distortions of the incoming signal without considering the additional detrimental effect of the acoustic feedback signal, the test signal was the residual incoming signal $\tilde{x}[k]$ after applying the beamformer. As speech source we used a 10s long speech signal that was generated by concatenating sentences from the TIMIT database [26].

In the following we present results where we consider $I = 9$ sets of acoustic feedback paths measured in free-field and a frontal acoustic transfer function of the incoming signal in the optimization. For each of the 10 available sets of acoustic feedback paths measured in free-field a different null-steering beamformer is computed using the remaining $I = 9$ sets of acoustic feedback path measurements. To evaluate the robust performance, for each of the 10 beamformers the average ASG was computed for the set of acoustic feedback paths that was not used in the optimization, i.e., using a leave-one-out cross validation approach. However, instead of using the sets of free-field feedback path measurements for evaluation, here we used the corresponding sets of acoustic feedback paths measured with a telephone receiver in close distance. Thus, this challenging experiment includes both variations of the sound field inside the ear canal and variations of the sound field outside of the ear, which were not taken into account in the design of the fixed null-steering beamformer.

Figure 4 shows the average ASG for the null-steering beamformer using the proposed RTF-based constraint and using the fixed delay constraint proposed in [15] for different beamformers lengths L_W. As can be observed both constraints allow for a large average ASG of more than 25 dB. Furthermore, generally a slight improvement in average ASG when increasing the number of beamformer coefficients L_W can be observed. Comparing both constraints, the proposed RTF-based constraint leads to a significantly larger average ASG of up to 41 dB ($L_W = 48$) compared to the fixed delay constraint with an average ASG of up to 28 dB.

Figure 5 shows the average PESQ MOS scores for the proposed RTF-based constraint and using the fixed delay constraint proposed in [15] for different incoming signal directions. Note that the frontal direction was included in the optimization of the null-steering beamformer as a constraint. In general, both null-steering beamformers lead to high PESQ MOS scores larger than 4.1 indicating a low distortion of the incoming signal. Comparing both constraints, the RTF-based constraint aiming to perfectly preserve the incoming signal leads to significantly larger PESQ MOS scores even for incoming signal directions that were not included in the optimization. Note that a perfect score of 4.5 in PESQ MOS scores is not obtained since the true RTF is an infinite impulse response filter that cannot be perfectly modeled using a finite impulse response filter.

In summary, the results for the average ASG and average PESQ MOS scores show that using the proposed RTF-based constraint in the design of a null-steering beamformer for feedback cancellation allows not only to largely cancel the acoustic feedback but at the same time allows to almost perfectly preserve the incoming signal.

6. CONCLUSION

In this paper we proposed to use a fixed null-steering beamformer for acoustic feedback cancellation in an earpiece with multiple microphones. In contrast to previous work approach that employed a fixed delay constraint in the reference microphone, we propose to incorporate a constraint based on the RTF of the incoming signal aiming to perfectly preserve it in the beamformer output. We formulate the computation of the beamformer coefficients as a linearly constrained least-squares optimization problem aiming to reduce the residual feedback power while preserving the incoming signal. Experimental results using measured acoustic feedback paths show that the propose RTF-constrained null-steering beamformer is able to significantly outperform the previously proposed fixed delay constrained null-steering beamformer, both in feedback cancellation performance as well as incoming signal preservation. The results show that this is the case even in challenging acoustic conditions with unknown acoustic feedback paths as well as incoming signal directions. In conclusion, using the proposed RTF-constrained null-steering beamformer allows to robustly increase the ASG by up to 40 dB without distorting the incoming signal quality.

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