Multiple-Point Equalization of Room Transfer Functions by Using Common Acoustical Poles

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***Abstract*—A multiple-point equalization filter using the common acoustical poles of room transfer functions is proposed. The common acoustical poles correspond to the resonance frequencies, which are independent of source and receiver positions. They are estimated as common autoregressive (AR) coefficients from multiple room transfer functions. The equalization is achieved with a finite impulse response (FIR) filter, which has the inverse characteristics of the common acoustical pole function. Although the proposed filter cannot recover the frequency response dips of the multiple room transfer functions, it can suppress their common peaks due to resonance; it is also less sensitive to changes in receiver position. Evaluation of the proposed equalization filter using measured room transfer functions shows that it can reduce the deviations in the frequency characteristics of multiple room transfer functions better than a conventional multiple-point inverse filter. Experiments show that the proposed filter enables 1–5 dB additional amplifier gain in a public address system without acoustic feedback at multiple receiver positions. Furthermore, the proposed filter reduces the reflected sound in room impulse responses without the pre-echo that occurs with a multiple-point inverse filter. A multiple-point equalization filter using common acoustical poles can thus equalize multiple room transfer functions by suppressing their common peaks.**

I. Introduction

THE ROOM transfer function (RTF), which is used to describe sound transmission characteristics between a source and a receiver in a room, has complex frequency characteristics that vary depending on the source and receiver positions. An equalization filter is generally used to adjust the frequency response of the RTF to eliminate the various acoustical problems. Such a filter can reduce the acoustic feedback in a public address system or suppress unnecessarily strong RTF frequency response in a sound reproduction system.

The most common equalization method uses a graphic equalizer, which is composed of multiple bandpass filters. However, considerable manual skill is needed to adjust a graphic equalizer.

Recently proposed digital equalization filters enable fine adjustment to be done more easily. In these equalization systems, a single-point equalization filter tuned for a specific RTF is affected by changes in receiver position (or "equalization point") because the frequency response of the RTF depends on both the source and receiver positions [1]. One proposed solution to this problem is to select the most suitable equalization filter from a spatial equalization filter library whenever the receiver moves [2]. The spatial equalization library is constructed using all-pole modeling of the RTF's and vector quantization. Because the all-pole model of an RTF can represent the peaks of the RTF with a smaller filter length [3], this system effectively suppresses the spectral peaks. However, this method is intrinsically a single-point equalization method, so the equalization filter cannot well equalize the responses at multiple points at the same time. Sometimes, a receiver (or a microphone) continuously moves around with a speaker in a public address system, and listeners are at different positions in a sound reproduction system. In such a case, a multiple-point equalization is required.

Two different systems have been developed for providing multiple-point equalization. One uses multiple filters and sources (loudspeakers) [4]–[6]. When the number of sources is more than the number of equalization points, this equalization system can achieve perfect equalization at the multiple points [4], where perfect equalization means equalizing not only the amplitudes but also the phases of the frequency responses. However, this equalization system needs many filters and sources.

The other system uses a single filter and a single source. While it is unable to achieve perfect equalization at multiple points, it does require less hardware. A multiple-point inverse filter with a least-square error has been proposed for such a system [7], [8]. The equalization filter coefficients are calculated to minimize the sum of the squared errors between the equalized signals at multiple points and the delayed original signals. While this least-square estimation approach is reasonable in a numerical sense, it does not reflect the physical characteristics of the RTF's. That is, while the RTF's have a physically common part and a unique part, the multiple-point inverse filter tries to equalize both parts.

We can equalize the physically common parts of the RTF's based on the resonances in the room corresponding to the different source and receiver positions. The RTF variations are due to changes in the dips (zeros of the RTF's). Since the resonance causes peaks in the frequency responses of the RTF's [9], suppressing the resonance frequencies is a powerful technique for equalizing multiple points at the same time.

We previously proposed using the common acoustical poles to model multiple RTF's [10]. In this paper, we propose a multiple-point equalization filter that uses the common acoustical poles. The common acoustical poles are estimated as common autoregressive (AR) coefficients from the multiple RTF's. When the number of estimated common acoustical

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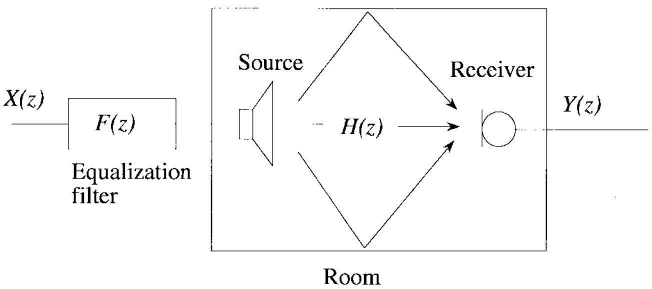


Fig. 1. Block diagram of single-point prefiltering equalization system. The equalization filter is used to adjust the frequency responses of one specific room transfer function $H(z)$.

poles is the same as the number of the resonances, each pole fits well with each resonance frequency and its Q factor. When the number of estimated common acoustical poles is less than the number of resonances, the poles correspond to the major resonance frequencies, which have a high Q factor. Since there are so many resonance frequencies over a wide frequency range, we generally estimate a smaller number of common acoustical poles. Therefore, this method does not recover the antiresonance characteristics (i.e., dips or zeros of the RTF's), but does suppress the common peaks due to major resonances of the multiple RTF's.

This paper is organized as follows. Section II reviews conventional digital equalization methods. Section III describes our proposed equalization filter. This filter is evaluated using measured room impulse responses and compared with the conventional multiple-point inverse filter in Section IV.

II. Conventional Equalization Filters for Room Transfer Functions

A. Single-Point Equalization

In a single-point prefiltering equalization system (see Fig. 1), $H(z)$ is the RTF between a source and a receiver and is expressed as a z transform; $F(z)$ is the equalization filter and $X(z)$ and $Y(z)$ are the input and output signals, respectively. Output signal $Y(z)$ is expressed as

$$Y(z) = H(z)F(z)X(z). \tag{1}$$

1) Inverse Filter: The perfect equalization filter is the inverse filter

$$F(z) = H^{-1}(z) \tag{2}$$

which equalizes not only the magnitude but also the phase of the frequency response. When (2) is substituted into (1), output signal $Y(z)$ is equal to $X(z)$. However, when RTF $H(z)$ includes nonminimum phase zeros, inverse filter $F(z)$ becomes unstable.

2) Single-Point Inverse (SPI) Filter Based on Least-Square Error: To overcome the instability, a modified inverse filter is obtained using the least-square method. In the time domain, the relationship between input signal $x(k)$ and output signal $y(k)$ is written as

$$y(k) = \sum_{n=0}^{N-1} h(n)x_{\mathrm{f}}(k-n) \tag{3}$$

where

$$x_{\mathrm{f}}(k) = \sum_{n=0}^{L-1} f_{\mathrm{SPI}}(n)x(k-n) \tag{4}$$

denotes the prefiltered input signal. Here, k and n denote discrete-time indexes, $h(n)$ is the impulse response of $H(z)$ (in Fig. 1), N is the length of the impulse response, $f_{\mathrm{SPI}}(n)$ ($n = 0, \cdots, L-1$) are the single-point equalization filter coefficients, and L is the number of taps of the equalization filter. Equalization filter coefficients $f_{\mathrm{SPI}}(n)$ are calculated to minimize the cost function

$$\varepsilon_{\mathrm{SPI}} = \sum_{k=0}^{\infty} e^2(k) = \sum_{k-0}^{\infty} [x(k-d) - y(k)]^2 \tag{5}$$

which is the square of error signal $e(k)$ between delayed original signal $x(k-d)$ and equalized signal $y(k)$. Here, d is the modeling delay, which compensates for noncausality.

3) Single-Point All-Pole (SPAP) Equalization Filter: An equalization filter using all-pole modeling of an RTF equalizes the magnitude of the frequency resonances [3]. The all-pole modeling of an RTF is expressed by the transfer function

$$H_{\mathrm{AP}}(z) = \frac{C}{A(z)} = \frac{C}{1 - \sum_{n=1}^{P} a(n)z^{-n}} \tag{6}$$

where C is an arbitrary gain constant and $a(n)$ are the AR coefficients corresponding to the poles. The single-point all-pole (SPAP) equalization filter is

$$F_{\mathrm{SPAP}}(z) = A(z) = 1 - \sum_{n=1}^{P} a(n)z^{-n}. \tag{7}$$

This filter is a moving average filter whose coefficients correspond to the poles of the RTF. It equalizes the RTF by suppressing its peaks. Since it is a minimum phase filter, it cannot achieve complete phase equalization of the actual RTF; however, it does reduce the required filter length.

B. Multiple-Point Inverse (MPI) Filter Using a Single Inverse Filter Based on Least-Square Error

Fig. 2 shows multiple-point equalization using a single inverse filter (equalization filter) $F(z)$. $H_i(z)$ and $Y_i(z)$ ($i = 1, 2, \cdots, M$) are the RTF's and output signals, respectively, for each receiver; M is the number of receivers. A perfect equalization filter cannot be achieved for all receiver positions because the RTF's have different frequency responses, especially their phase responses.

One proposed method [7], [8] for estimating the filter coefficients of a multiple-point inverse (MPI) filter using the least-square error is shown in Fig. 3. In the time domain, the

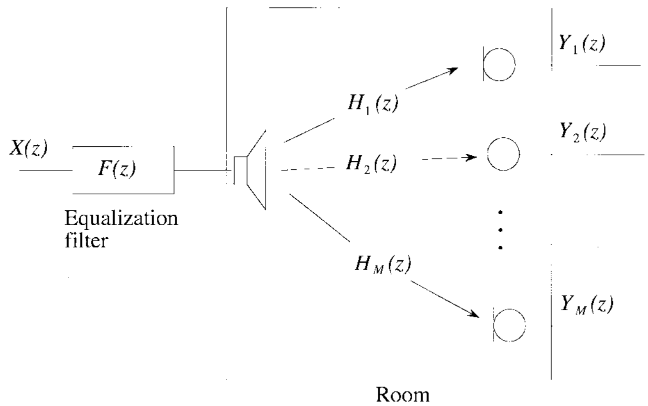


Fig. 2. Block diagram of multiple-point equalization system. The equalization filter is designed to equalize the multiple room transfer functions.

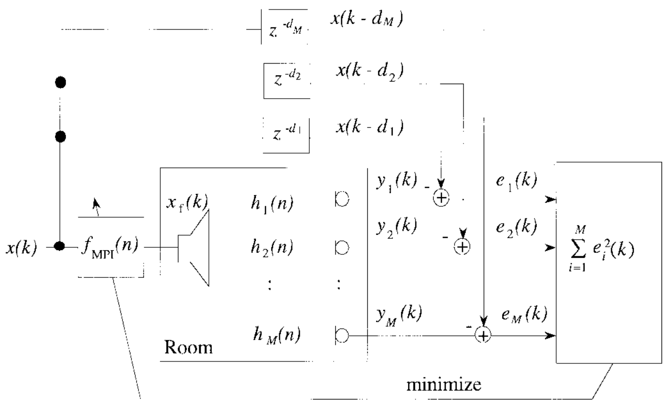


Fig. 3. Block diagram of estimation system for a conventional multiple-point inverse filter. The filter coefficients are estimated to minimize the sum of the squared errors between desired signals $x(k-d_i)$ and equalized signals $y_i(k)$ ($i = 1, 2, \cdots, M$).

relationship between input signals $x(k)$ and output signals $y_i(k)$ is written as

$$y_i(k) = \sum_{n=0}^{N-1} h_i(n) x_{\mathrm{f}}(k-n) \tag{8}$$

where

$$x_{\mathrm{f}}(k) = \sum_{n=0}^{L-1} f_{\mathrm{MPI}}(n) x(k-n). \tag{9}$$

Here, $h_i(n)$ is the impulse response of the ith RTF, $H_i(z)$ (in Fig. 2), and $f_{\mathrm{MPI}}(n)$ ($n = 0, \cdots, L-1$) are the equalization filter coefficients, which are set to minimize cost function $\varepsilon_{\mathrm{MPI}}$. This cost function is the sum of the squares of error signals $e_i(k)$ between delayed original signals $x(k-d_i)$ and output signals $y_i(k)$

$$\begin{aligned} \varepsilon_{\mathrm{MPI}} &= \sum_{i=1}^{M} \sum_{k=0}^{\infty} e_i^2(k) \\ &= \sum_{i=1}^{M} \sum_{k=0}^{\infty} [x(k-d_i) - y_i(k)]^2 \end{aligned} \tag{10}$$

where d_i is the modeling delay for the ith equalization point. The modeling delays d_i ($i = 1, \cdots, M$) are set differently reflecting the difference in the propagation times of direct sound in the impulse responses [7]. This equalization filter tries to recover the waveforms of the original source signals.

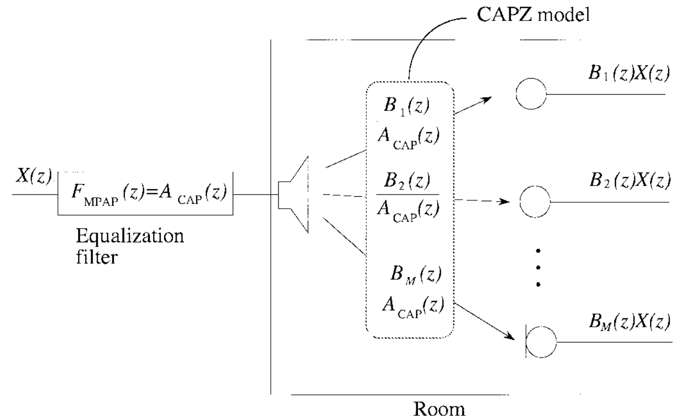


Fig. 4. Principle of multiple-point all-pole equalization filter using common acoustical poles. The common peaks of multiple room transfer functions are removed.

III. Multiple-Point Equalization Using Common Acoustical Poles

Although the RTF's are different for each source and receiver position, all RTF's in a room commonly include the resonance frequencies and their Q factors, which correspond to the damping constants [9]. Because the spectral peaks of each RTF are caused by these resonances, they are considered to be independent of the source and receiver positions. Therefore, only the dips (zeros) cause RTF variation. We, thus, propose a multiple-point equalization filter that suppresses only the common spectral peaks; it does not recover the various dips. This equalization filter performs better than the multiple-point inverse filter because it does not try to recover the individual zeros of the multiple RTF's. To obtain such an equalization filter, we used the common-acoustical-pole and zero model of RTF's [10].

A. Common-Acoustical-Pole and Zero Model

The concept of our previously proposed common-acoustical-pole and zero (CAPZ) model for RTF's is shown in Fig. 4. Each RTF, of $H_i(z)$, is expressed using a common acoustical pole (CAP) function, $A_{\mathrm{CAP}}(z)$, and a different zero function, $B_i(z)$

$$H_i(z) = \frac{B_i(z)}{A_{\mathrm{CAP}}(z)} = \frac{\displaystyle\sum_{n=0}^{Q} b_i(n) z^{-n}}{\displaystyle 1 - \sum_{n=1}^{P} a_{\mathrm{CAP}}(n) z^{-n}}. \tag{11}$$

The CAP function, $A_{\mathrm{CAP}}(z)$, does not depend on receiver position i, while the zero function, $B_i(z)$, does. Functions $A_{\mathrm{CAP}}(z)$ and $B_i(z)$ can be expressed in polynomial form by using coefficients $a_{\mathrm{CAP}}(n)$ and $b_i(n)$, as in (11). The $a_{\mathrm{CAP}}(n)$ are the common AR coefficients corresponding to the CAP's and the $b_i(n)$ are the MA coefficients; P and Q

are the orders of the poles and zeros, respectively. Note that, when poles are estimated with a single impulse response as in the conventional pole/zero model, the estimated poles are not necessarily common physical poles, because of influence of zeros in the RTF [10].

The common acoustical poles are estimated as common AR coefficients from the multiple RTF's. The estimated common acoustical poles agree well with the resonance frequencies when the estimation is done using the same number of estimated common acoustical poles as the number of resonance frequencies. When a smaller number is used, the estimated common acoustical poles represent the major resonance frequencies of the room [10].

B. Proposed Multiple-Point All-Pole (MPAP) Equalization Filter Using Common Acoustical Poles

Our proposed multiple-point equalization filter uses CAP function $A_{\mathrm{CAP}}(z)$ of the CAPZ model. This "multiple-point all-pole (MPAP) equalization filter" is defined as

$$F_{\mathrm{MPAP}}(z) = A_{\mathrm{CAP}}(z) = 1 - \sum_{n=1}^{P} a_{\mathrm{CAP}}(n) z^{-n}. \tag{12}$$

The equalization filter coefficients are calculated as common AR coefficients based on the least-squares method [11] by using the multiple RTF's. According to (11), the impulse response of the CAPZ model is expressed as

$$h_i(k) = \sum_{n=1}^{P} a_{\mathrm{CAP}}(n) h_i(k-n) + \sum_{n=0}^{Q} b_i(n)\delta(k-n) \tag{13}$$

where $\delta(k)$ is the unit pulse function [$\delta(k) = 1$ for $k = 0$, and $\delta(k) = 0$ for any other k]. Because only the common AR coefficients are required, the order of the zeros, Q, is set to zero, as follows:

$$\hat{h}_i(k) = \sum_{n=1}^{P} a_{\mathrm{CAP}}(n) \hat{h}_i(k-n). \qquad (i = 1, 2, \cdots, M). \tag{14}$$

The equation error between the actual impulse responses, $h_i(k)$, and the modeled impulse responses, $\hat{h}_i(k)$, is defined by

$$e_i(k) = h_i(k) - \sum_{n=1}^{P} a_{\mathrm{CAP}}(n) h_i(k-n). \tag{15}$$

The common AR coefficients can thus be estimated as those that minimize the least mean squares cost function $\varepsilon_{\mathrm{CAP}}$ as

$$\begin{aligned} \varepsilon_{\mathrm{CAP}} &= \sum_{i=1}^{M} \sum_{k=0}^{\infty} e_i^2(k) \\ &= \sum_{i=1}^{M} \sum_{k=0}^{\infty} \left[h_i(k) - \sum_{n=1}^{P} a_{\mathrm{CAP}}(n) h_i(k-n) \right]^2. \end{aligned} \tag{16}$$

Here, the common AR coefficients, $a_{\mathrm{CAP}}(n)$, that minimize (16) can be expressed in matrix form as

$$\mathbf{a} = (\mathbf{W}^T \mathbf{W})^{-1} \mathbf{W}^T \mathbf{v} \tag{17}$$

where we obtain (17a), shown at the bottom of the page. A similar technique that minimizes the equation error has also been studied for estimating common poles of multiple inverse filters [12].

The proposed MPAP equalization filter can be implemented as a finite impulse response filter with only a few hundred taps [13]. Output signal $Y_i(z)$ is

$$Y_i(z) = H_i(z) F_{\mathrm{MPAP}}(z) X(z) \tag{18}$$

that is,

$$\begin{aligned} Y_i(z) &= \frac{B_i(z)}{A_{\mathrm{CAP}}(z)} A_{\mathrm{CAP}}(z) X(z) \\ &= B_i(z) X(z). \end{aligned} \tag{19}$$

$$\begin{aligned} \mathbf{a} &= [a_{\mathrm{CAP}}(1), a_{\mathrm{CAP}}(2), \cdots, a_{\mathrm{CAP}}(P)]^T \\ \mathbf{W} &= [\mathbf{H}_1, \mathbf{H}_2, \cdots, \mathbf{H}_M]^T \\ \mathbf{v} &= [\mathbf{h}_1, \mathbf{h}_2, \cdots, \mathbf{h}_M]^T \\ \mathbf{h}_i &= [h_i(1), h_i(2), \cdots, h_i(N-1), 0, 0, \cdots, 0]^T \end{aligned}$$

and

$$\mathbf{H}_i = \left.\underbrace{\begin{bmatrix} h_i(0) & 0 & \cdots & 0 \\ h_i(1) & h_i(0) & \cdots & 0 \\ \vdots & \vdots & & \vdots \\ h_i(P-1) & h_i(P-2) & \cdots & h_i(0) \\ \vdots & \vdots & & \vdots \\ h_i(N-1) & h_i(N-2) & \cdots & h_i(N-P) \\ 0 & h_i(N-1) & \cdots & h_i(N-P-1) \\ \vdots & \vdots & & \vdots \\ 0 & 0 & \cdots & h_i(N-1) \end{bmatrix}}_{P}\right\} N+P-1 \tag{17a}$$

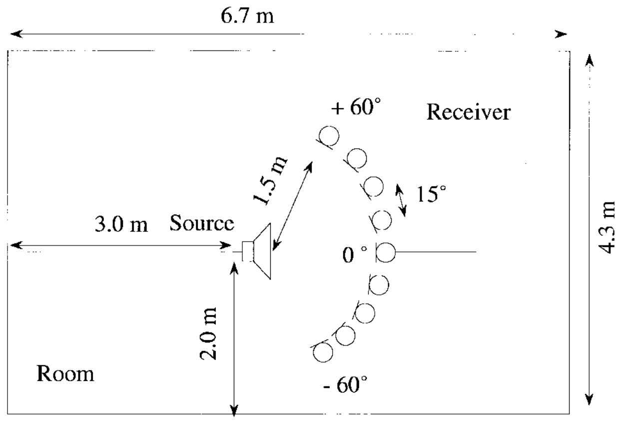


Fig. 5. Evaluation setup: the impulse responses were measured at nine receiver positions. The room was 3.0 m high and the source and receivers were 1.2 m above the floor.

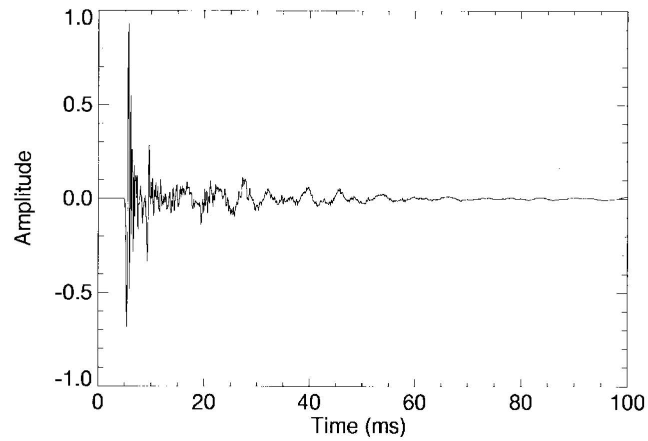


Fig. 6. Example of measured room impulse response.

This filter does not recover the zeros, $B_i(z)$, corresponding to the dips, but it does suppress the common peaks due to the resonances in the room.

IV. Performance of Proposed Equalization Filter

We evaluated the effectiveness of our proposed MPAP equalization filter using the common acoustical poles for equalizing multiple impulse responses by comparing it with the SPI filter, the SPAP equalization filter, and the MPI filter.

The impulse responses were measured at nine receiver positions, as shown in Fig. 5. The room was 6.7 × 4.3 × 3.0 m, and had a reverberation time of 0.25 s. The frequency range was set to 0.2–3.4 kHz; the sampling frequency was 8 kHz. We located the receivers on a semicircle so we could use the same modeling delay for calculating the multiple-point inverse filter. The impulse responses included the characteristics of the room acoustics and those of the source and receiver system. Fig. 6 shows an example of the impulse response.

All of the equalization filter coefficients were calculated using a filter length of 200 taps and a modeling delay of 100 samples for the SPI and MPI filters. The SPAP and MPAP filters do not need the modeling delay. All of the filters had a unit gain on average. To evaluate the equalized sound signals at the multiple receiver positions, we used the convolution results of the original impulse responses and the equalization filters (equalized RTF's) assuming the input source signal to be a unit pulse.

To assess the degree of flatness of the equalized RTF's, $Y(f)$, we introduced two evaluation criteria. One was standard deviation σ, defined as

$$\sigma = \sqrt{\frac{1}{N} \sum_{f=f_0}^{f_1} (20 \log |Y(f)| - \text{AVG})^2} \tag{20}$$

where AVG is the mean value

$$\text{AVG} = \frac{1}{N} \sum_{f=f_0}^{f_1} (20 \log |Y(f)|). \tag{21}$$

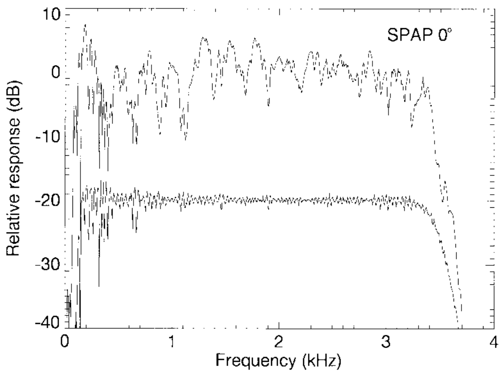
f_0 corresponds to 200 Hz, f_1 corresponds to 3.4 kHz, and N is the number of frequency samples between f_0 and f_1. The standard deviation represents the scale of the peaks and dips of the equalized RTF's. The smaller the standard deviation, the flatter the equalized RTF's.

The other evaluation criterion was the "howling onset level," which is defined as the amplifier gain at which the acoustical feedback becomes unstable in a public address system where the input signal $x(k)$ of the loudspeaker is a receiver output signal $y(k)$. An equalization filter suppresses the peaks of the frequency response, enabling a higher amplifier gain, that is, enabling a higher sound reproduction level without acoustic feedback, because the acoustic feedback becomes unstable around the highest peaks of the frequency response. The higher the howling onset level, the better the equalization filter suppresses the peaks in the frequency responses.

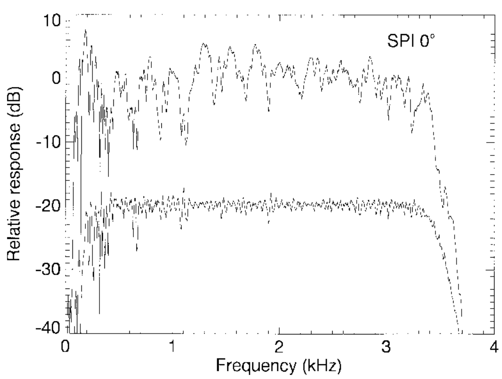
A. SPI and SPAP Filter Results

We first used the SPI and SPAP filters to equalize the multiple RTF's. These filters were calculated from the RTF corresponding to the 0° receiver position in Fig. 5. Fig. 7 shows the frequency responses before and after equalization for (a) SPAP at the 0° receiver position, (b) SPI at 0°, (c) SPAP at 30°, and (d) SPI at 30°. In Fig. 7, the upper curve in each graph shows the magnitude of the frequency response of the original RTF and the lower curve shows the magnitude of the frequency response of the equalized RTF. The average relative response levels were set to 0 dB for the original RTF's and −20 dB for the equalized RTF's. Since the equalization filters were calculated using the 0° impulse response, both filters flattened the magnitude of the frequency responses at the 0° receiver position, as shown in Fig. 7(a) and (b). At 30°, however, the deviations were not reduced, as shown in Fig. 7(c) and (d).

Fig. 8 shows the standard deviation of the original RTF's, the SPI-equalized RTF's, and the SPAP-equalized RTF's at the nine receiver positions. At and near the 0° receiver position, the standard deviations were reduced by both equalization filters. The improvement in the standard deviation at the 0° receiver position was about 2 dB for both filters. However, the further the receiver position from 0°, the less effective the equalization.



(a)



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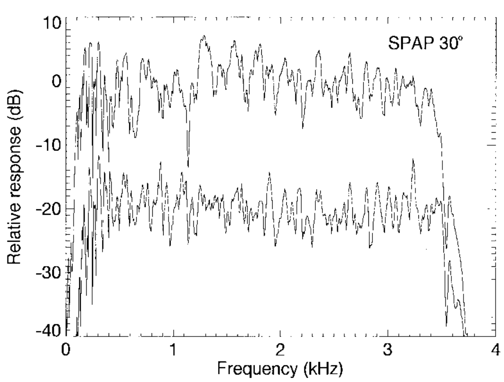

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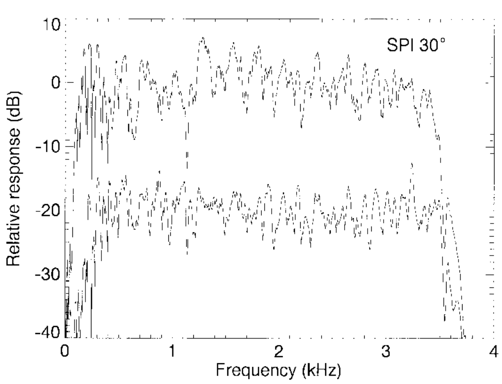


(d)

Fig. 7. Frequency responses before equalization (upper curves) and after equalization (lower curves) for (a) SPAP equalization filter at 0° receiver position, (b) SPI filter at 0°, (c) SPAP filter at 30°, and (d) SPI filter at 30°.

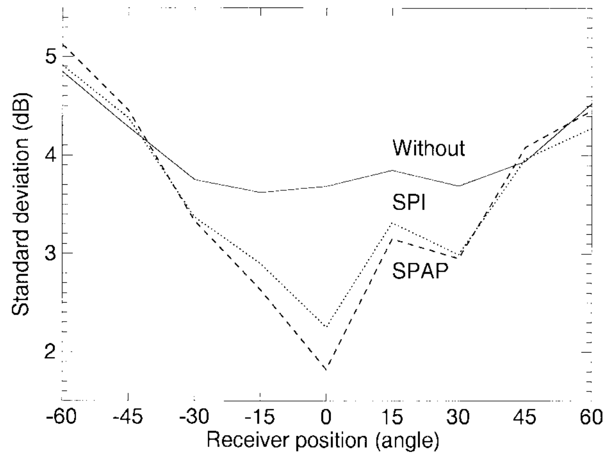


Fig. 8. Standard deviations of original (without equalization), SPI-equalized, and SPAP-equalized RTF's for the nine receiver positions.

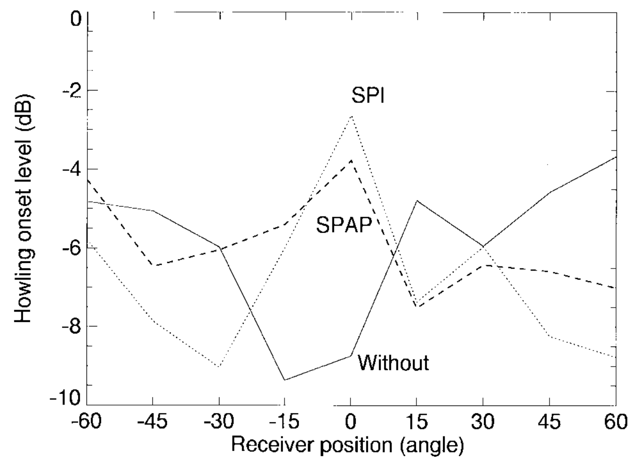


Fig. 9. Howling onset levels of original (without equalization), SPI-equalized, and SPAP-equalized RTF's for the nine receiver positions.

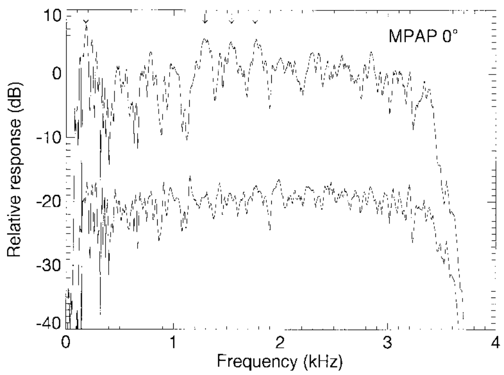
Fig. 9 shows the relative howling onset level of the original RTF's, the SPI-equalized RTF's, and the SPAP-equalized RTF's. We assume that at 0 dB, the frequency response of the equalized RTF's are completely flat. At the 0° receiver position, the relative howling onset level was 4 dB higher for both the SPAP- and SPI-equalized RTF's than the original RTF. However, when the receiver was moved to almost any other position, the relative howling onset level became lower than without equalization.

These results show that the SPI and SPAP filters, which are calculated from one impulse response, work well at a specific point but not at other points, as previously reported [1], [7]. Single-point equalization is thus sensitive to changes in receiver position.

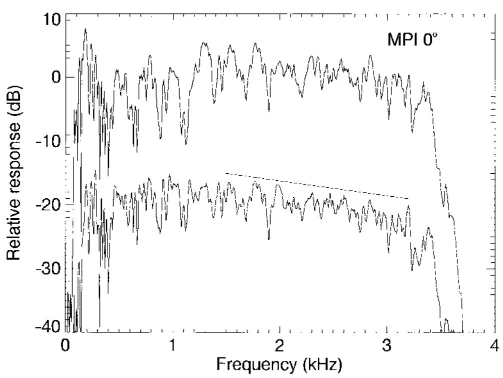
B. MPI and MPAP Filter Results

We next evaluated the MPI and MPAP filters calculated from the nine measured impulse responses. The filter length and modeling delay were the same as for the SPI and SPAP filters.

Fig. 10 shows the frequency responses before and after equalization: (a) MPAP equalization filter at the 0° receiver



(a)



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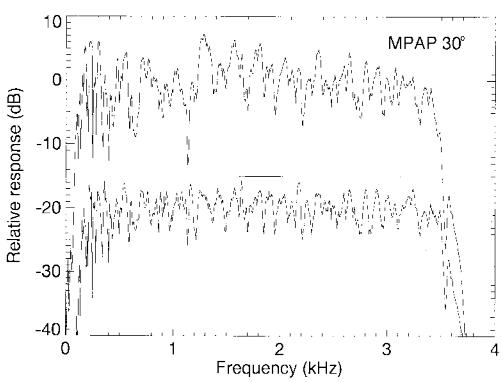

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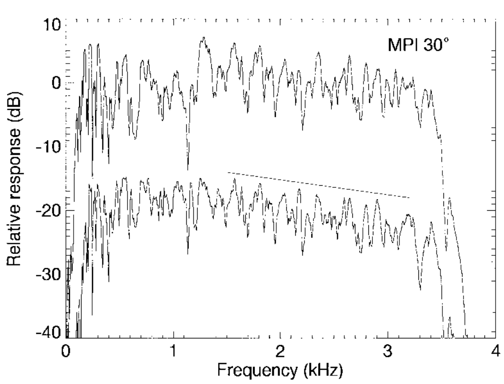

(d)

Fig. 10. Frequency responses before equalization (upper curves) and after equalization (lower curves) for (a) MPAP equalization filter at 0° receiver position, (b) MPI filter at 0°, (c) MPAP filter at 30°, and (d) MPI filter at 30°.

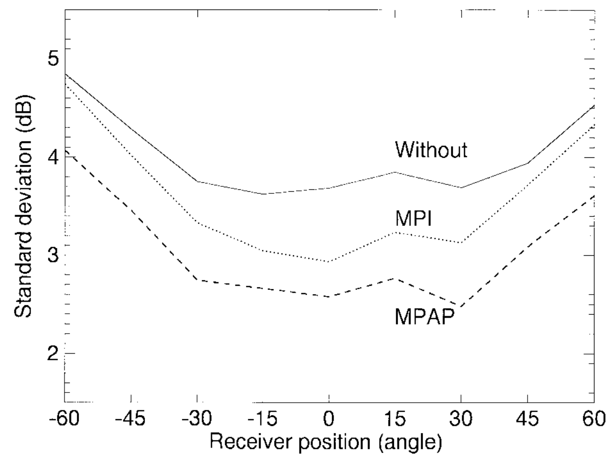

Fig. 11. Standard deviations of original (without equalization), MPI-equalized, and MPAP-equalized RTF's for the nine receiver positions.

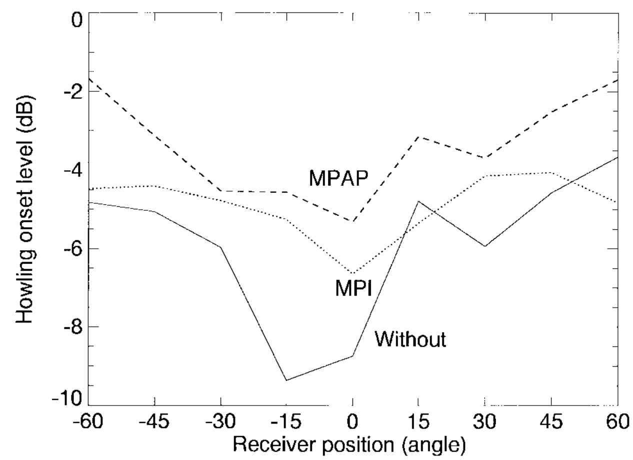

Fig. 12. Howling onset levels of original (without equalization), MPI-equalized, and MPAP equalized RTF's for the nine receiver positions.

positions, (b) MPI at 0°, (c) MPAP at 30°, and (d) MPI at 30°. Although neither filter completely flattened the RTF's, both filters suppressed the common main peaks. For example, in Fig. 10(a), the main peaks at 200, 1300, 1550, and 1800 Hz (indicated by arrows) were suppressed. This tendency was also found in the other figures. Comparing the MPI and MPAP results, while the MPAP-equalized RTF frequency response was almost flat on average, the MPI-equalized one decreased as the frequency increased, as indicated by the straight lines in Fig. 10.

Fig. 11 shows the standard deviation of the original RTF's, MPI-equalized RTF's, and MPAP-equalized RTF's at the nine receiver positions. At all receiver positions, the standard deviation of the MPAP-equalized RTF was lower than that of the MPI-equalized one. The MPAP-equalization filter is, thus, more effective than the MPI one in reducing the deviations of multiple RTF's. This is because the MPI filter equalizes not only the amplitude but also the phase, while the MPAP filter equalizes only the amplitude. The difference between standard deviation curves of the original RTF's and the MPAP-equalized RTF's was about 1.2 dB on average.

Fig. 12 shows the relative howling onset level for the original RTF's, the MPI-equalized RTF's, and the MPAP-

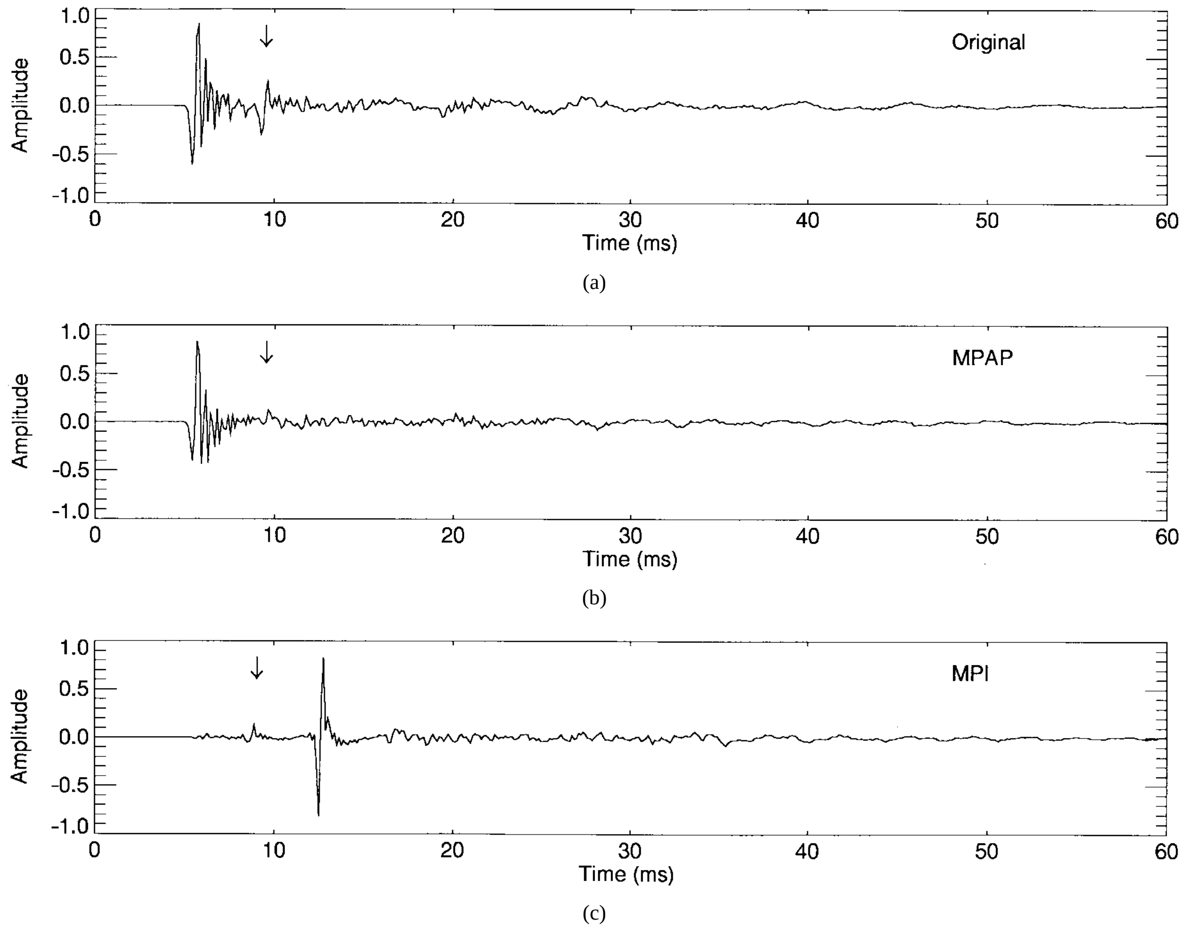

Fig. 13. Example waveforms of (a) original, (b) MPAP-equalized, and (c) MPI-equalized impulse responses.

equalized RTF's. With the MPAP filter, the howling onset level at all receiver positions was 1–5 dB higher than the original level, while with the MPI filter, the howling onset level was lower at 15 and 60°.

From Fig. 12, we can derive the safe amplifier gains when the receiver is moved between the nine positions. The safe amplifier gain is the minimum howling onset level. For example, when the amplifier gain was set to -8 dB without an equalization filter, acoustic feedback occurred at receiver positions -15 and 0°. To avoid acoustic feedback at all receiver positions, the amplifier gain had to be set below -9.5 dB based on the howling onset level at $-15°$. With the MPI and MPAP filters, the safe amplifier gains were -6.7 and -5.4 dB, respectively. Using the MPAP filter increased the safe amplifier gain by 4.1 dB from the original RTF's, while using the MPI filter increased it by 2.8 dB. An MPAP filter can, thus, effectively function as a fixed filter to prevent acoustic feedback when the receiver is moved.

Fig. 13 shows example waveforms of an original impulse response, an MPAP-equalized impulse response, and an MPI-equalized impulse response at receiver position 0°. Although the MPAP equalization filter equalized only the amplitudes of the frequency responses, it reduced the reflected sound at 10 ms, as indicated by the arrow in Fig. 13(b). Furthermore, a pre-echo was found in the MPI response but not in the MPAP response. The reason for the pre-echo in the MPI response is that the MPI filter is a double-sided filter (it uses the modeling delay to compensate for the nonminimum-phase zeros). Since the MPAP equalization filter does not consider the zeros, there is no pre-echo in its response. The MPAP equalization filter thus equalizes not only the frequency responses but also the time responses without pre-echo.

V. Conclusion

We have proposed a multiple-point all-pole equalization filter that uses the inverse characteristic of a common acoustical pole function calculated from multiple room transfer functions. The equalization is achieved with a finite impulse response filter with only a few hundred taps. Although this filter does not flatten the frequency response dips, it does suppress the resonance frequencies common to multiple RTF's.

We compared the effectiveness of the proposed equalization filter with that of a conventional multiple-point inverse single filter by using measured impulse responses with a frequency range of 0.2–3.4 kHz. The proposed filter flattened the frequency response over a wide frequency range and reduced the deviations in the frequency characteristics of multiple room transfer functions better than the conventional multiple-point inverse filter. The proposed filter thus enabled a 1–5 dB additional amplifier gain without acoustic feedback at multiple receiver positions in a public address system than with no equalization. Furthermore, the proposed equalization filter reduced the reflected sound in room impulse responses without the pre-echo that occurs with a conventional multiple-point inverse filter.

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