Team 43 – Inverse Filtering of Room Acoustics

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Abstract

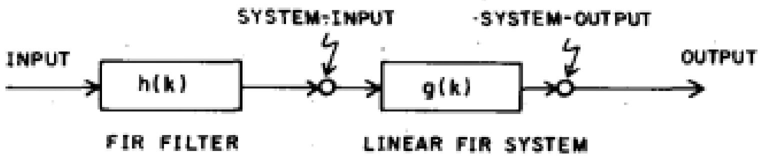
We were assigned the paper by Miyoshi and Kaneda on "Inverse Filtering of Room Acoustics", which uses N+1 inputs and N outputs to create an exact inverse filtering of the room acoustics. Acoustic signals radiated across a room are generally linearly distorted due to wall reflections. These distortions occur due to phenomena of the room like echoes and reverberation; the sound reaches the listener not only directly via the source but also by reflection off the walls, and ruins the legibility of the propagated sound, be it speech, music, and/or any other form of communication. To remove them, that is, to inverse filter them (and essentially, get rid of them), we will use both the traditional LSE methods and the novel approaches described in this paper.

There are several methods of creating an inverse filter; however, many rely on the least squares error criterion and create only an approximation of the exact inverse.

The multiple input/output theorem can be used to find the exact inverse of a linear FIR system. According to the theorem, it is possible to construct the exact inverse filter of a nonminimum-phase system. (A minimum-phase system is one whose Laplace-transform doesn't have poles or zeroes in the right half of the plane).

A short discussion regarding the existing method (LSE Method)

We have—



where, $$d(k) = \begin{cases} 1 & \text{when } k = 0 \\ 0 & \text{when } k = 1, 2, \cdots . \end{cases}$$

Here, the relationship $d(k) = g(k) * h(k) \ldots (1)$ must be satisfied [$*$ denotes discrete linear convolution].

In matrix form,

$$
\overset{L+1}{\begin{bmatrix} 1 \\ 0 \\ \vdots \\ \vdots \\ \vdots \\ 0 \end{bmatrix}} = \underset{i+1}{\begin{bmatrix} g(0) & & \\ g(1) & g(0) & 0 \\ \vdots & g(1) & \ddots \\ g(m) & \vdots & g(0) \\ & g(m) & g(1) \\ 0 & \ddots & \vdots \\ & & g(m) \end{bmatrix}} \begin{bmatrix} h(0) \\ h(1) \\ \vdots \\ \vdots \\ h(i) \end{bmatrix}
$$

OR $D = GH$ … (2),

The equation, however, doesn't have a solution since in G, $no. of\ columns\ <\ number\ of\ rows$. In this method, therefore, the FIR filter coefficients are obtained as only an approximate solution of the above equation, given by—

$H\ =\ (G^T G)^{-1} G^T D$ … (3),

where G^T is the transpose of G. Clearly, this is not an exact inverse filter; rather, it is only an approximation that we are able to obtain.

Proposed Method

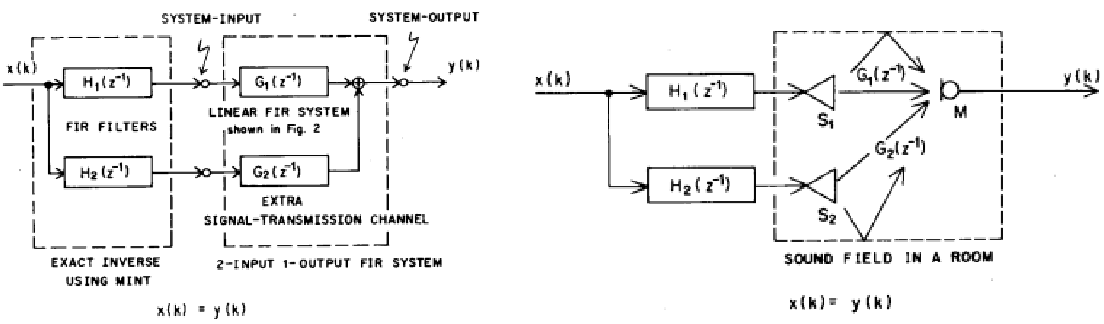
I. Principle of Proposed Inverse-Filtering Method

We use multiple-input linear FIR system by adding signal-transmission channels. Then, we can use the equation just obtained in the previous section to construct an exact inverse of the system.

We consider a two-input single-output system for ease of calculation. To construct the inverse system, the following equation must be satisfied—

$D(z^{-1}) = 1 = G_1(z^{-1})\, H_1(z^{-1}) + G_2(z^{-1})\, H_2(z^{-1})$ … (4),

where $D(z^{-1})$ is the z-transform of $d(k)$ seen earlier; G_1, G_2 are signal-transmission channels; H_1, H_2 are the respectively associated FIR filters. For the equation to hold, G_1, G_2 should not have any common zero; and the orders of H_1, H_2 must be less than those of G_2, G_1, respectively.

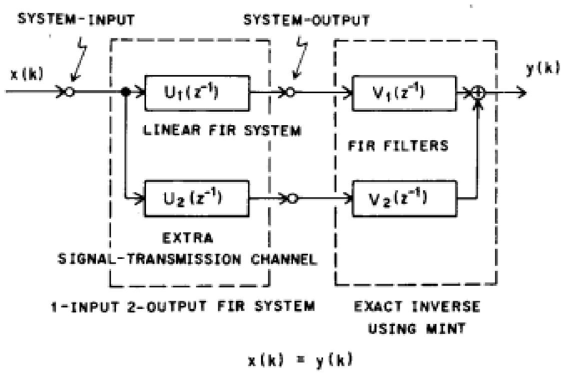


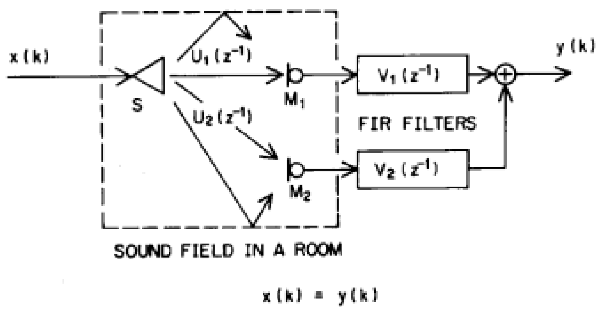
S_1, S_2 are loudspeakers; M is the receiving point.

We can also extend the principle to single-input two-output linear FIR system. The following equation must be satisfied—

$$1 = U_1(z^{-1})\, V_1(z^{-1}) + U_2(z^{-1})\, V_2(z^{-1}) \quad \ldots (5)$$

where U_1, U_2 are the signal-transmission channels; V_1, V_2 are the respectively associated FIR filters.





Here, S is the source of the sound.

The above principle can be applied to invert a multiple-input multiple-output linear FIR system. By having multiple microphones, we can achieve a “Cocktail Party Effect”, which is the observation that people, at a noisy party, can focus on a conversation by using the sound that reaches their ears and filtering out that sound which does not reach from the direction of the conversation.

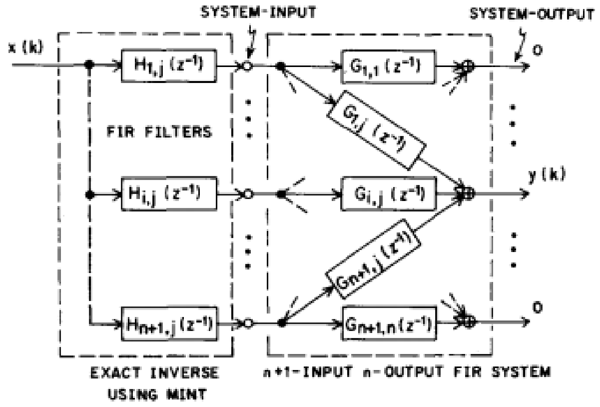
We consider an $(n+1)$-input n-output system. Let G_{ij} be the signal-transmission channel and H_{ij} be the FIR filter connected to the i^{th} output of the system. This is how the inverse filtering of the j^{th} output of the system is done—

$$n \begin{bmatrix} 0 \\ 0 \\ \vdots \\ 1 \\ \vdots \\ \vdots \\ 0 \end{bmatrix} = \underbrace{\begin{bmatrix} G_{11}(z^{-1}) & \cdots & G_{n+11}(z^{-1}) \\ G_{12}(z^{-1}) & \cdots & G_{n+12}(z^{-1}) \\ \vdots & & \vdots \\ G_{1j}(z^{-1}) & \cdots & G_{n+1j}(z^{-1}) \\ \vdots & & \vdots \\ \vdots & & \vdots \\ G_{1n}(z^{-1}) & \cdots & G_{n+1n}(z^{-1}) \end{bmatrix}}_{n+1} \begin{bmatrix} H_{1j}(z^{-1}) \\ H_{2j}(z^{-1}) \\ \vdots \\ \vdots \\ \vdots \\ \vdots \\ H_{nj}(z^{-1}) \\ H_{n+1j}(z^{-1}) \end{bmatrix} \text{ OR } R_j = GH_j \ \ldots (6)$$

Thus, it is possible to construct the exact inverse of a multiple-input multiple-output linear FIR system by the proposed MINT principle.

II. Computing FIR Filters for Exact Inverse

We now compute the FIR filters described in the earlier section. For ease of calculation, we consider a two-input one-output system as shown—



x(k) = y(k)

From equation (4), we get $d(k) = g_1(k) * h_1(k) + g_2(k) * h_2(k)$ … (7),

where g_1, g_2 are impulse responses of G_1, G_2 respectively; h_1, h_2 are the coefficients of H_1, H_2.

$$L+1 \begin{bmatrix} 1 \\ 0 \\ 0 \\ \vdots \\ 0 \end{bmatrix} =$$

$$\begin{bmatrix} g_1(0) & & g_2(0) & \\ g_1(1) & 0 & g_2(1) & 0 \\ \vdots & & \vdots & \\ \vdots & g_1(0) & g_2(n) & \\ g_1(m) & g_1(1) & & g_2(0) \\ & \vdots & & g_2(1) \\ 0 & \vdots & 0 & \vdots \\ & g_1(m) & & g_2(n) \end{bmatrix} \begin{bmatrix} h_1(0) \\ h_1(1) \\ \vdots \\ h_1(i) \\ h_2(0) \\ \vdots \\ h_2(j) \end{bmatrix}$$

$$\longleftarrow i+1 \longrightarrow \quad \longleftarrow j+1 \longrightarrow$$

OR

$$D = G_1 H_1 + G_2 H_2 = [G_1 \quad G_2] \begin{bmatrix} H_1 \\ H_2 \end{bmatrix}$$

… (8)

The coefficients of the FIR filters h_1, h_2 can be calculated by following relationship obtained from the above equation—

$$\begin{bmatrix} H_1 \\ H_2 \end{bmatrix} = [G_1 \quad G_2]^{-1} D. \quad \ldots (9)$$

III. Inverse-Filtering Experiment in a Sound Field

Because the room impulse is considered to have nonminimum phases, it has been the belief that it is impossible to devise any method to remove the distortions caused by wall reflections.

However, it is found by the paper that if the various sound-fields of the inputs from the microphones are treated as minimum-phase systems, as long as they don't have a common zero, the combined FIR system can be used to model the nonminimum phase system and find the inverse of the room impulse response. A nonminimum phase function can be represented as the product of a minimum phase function and a maximum phase function.

Our approach is to generate Gaussian noise for each microphone, treat that as the room echo, then play some known audio (in this case, a classical choral piece) to find the room impulse response, and construct the matrix required to solve the inverse filter.

Once we have obtained the $(N+1) \times N$ matrix, we can solve for the exact inverse and implement it. The matrix is basically a convolution matrix. We will use the method described in the paper to find the coefficients $h_1(k), h_2(k), \ldots$, which are then used in order to solve for $G_1, G_2, \ldots$

Simulation and Results

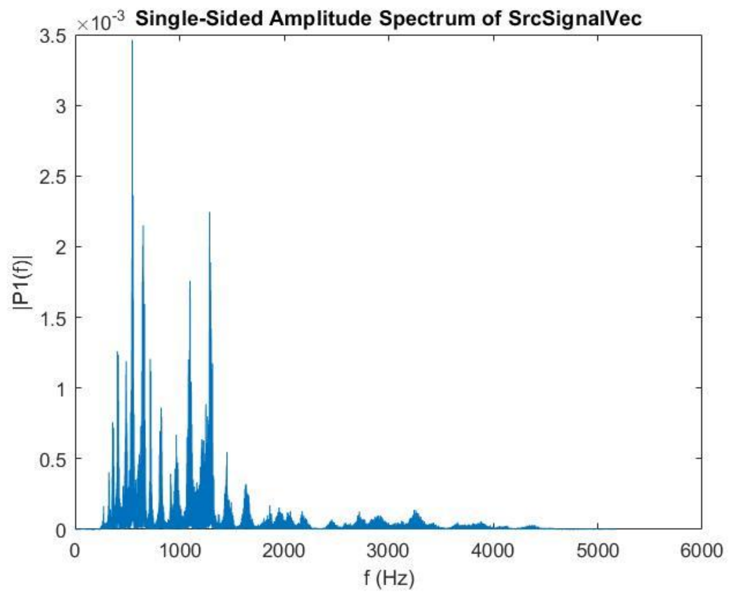


Figure 1: Original FFT

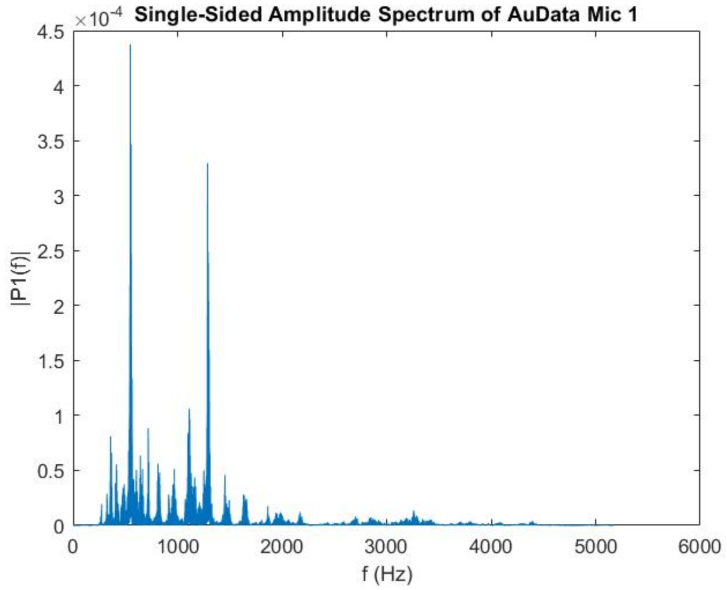


Figure 2: Mic FFT

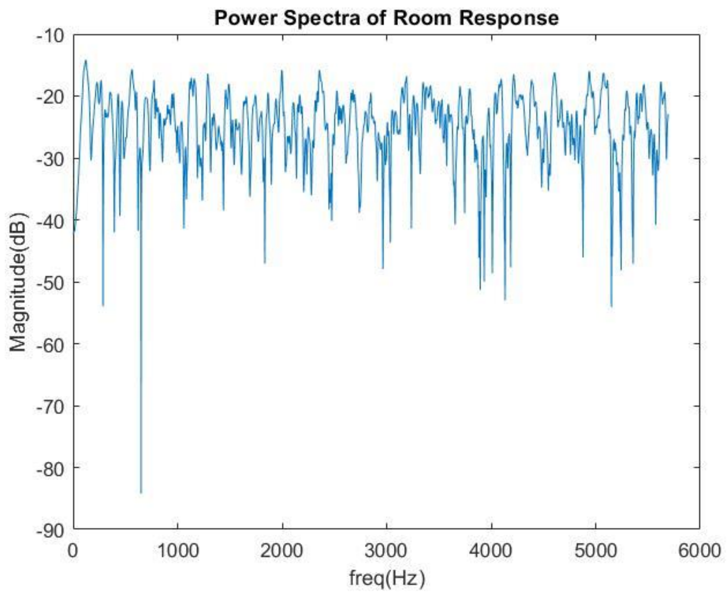


Figure 3: Power spectra of room response

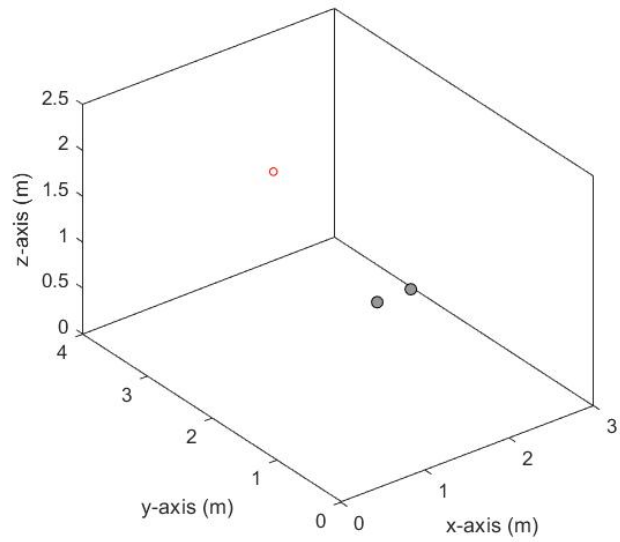


Figure 4: Room Simulation

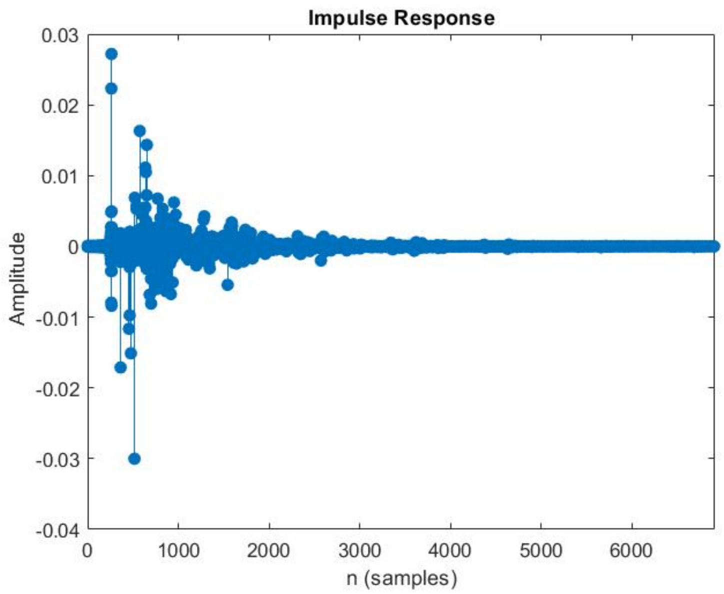


Figure 5: Room Impulse Response

Conclusion

An inverse filtering method, based on the MINT principle, was proposed to realise the exact inverse of room acoustics that have nonminimum phases. By this principle, the exact inverse is obtained using multiple FIR filters and multiple loudspeakers (or microphones). By the rules of matrix multiplication, the filter coefficients can be easily computed.

An experiment to realise inverse filtering at a point in a sound-field (in a room) was conducted. From the experiment results, it was clear that the proposed method was superior to the traditional one of employing LSE.

References

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