

## Room Transfer Function Estimation and Room Equalization in Noise Environments

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### ABSTRACT

Audio quality in listening situation is degraded by indoor room reverberation. Room equalization can be used to increase the audio quality by applying the inverse transfer function to the input audio signals. In noise environments, however, it is hard to exactly measure the room transfer function. In this work, we developed the techniques to measure the room transfer function in indoor noise environments and to enhance the audio quality by room equalization. From the experimental results, we showed that the proposed techniques can be successfully used in indoor noise environments.

**Keywords** – room transfer function, room equalization, reverberation, noise environments.

### I. INTRODUCTION

Audio quality in listening situation is degraded by indoor room reverberation and can be increased by using room equalization that applies the inverse of room transfer function (RTF) to input signals. First, we used the short-time spectral masking method for RTF estimation in noise environments [1]. Next, room equalization techniques are developed for the audio quality enhancement. The audible frequency range is separated into unequal segments by one-third octave band filters. In addition, the estimated RTF is smoothed in frequency domain, and the gain parameters for the equalization are obtained by inverting the average spectral magnitude at each band and are normalized.

First, Section 2 introduces the RTF estimation method in noisy environments. Section 3 describes room equalization based on smoothed RTF. Section 4 shows the experimental results. The last Section concludes this work.

### II. ROOM TRANSFER FUNCTION ESTIMATION IN NOISE ENVIRONMENTS

For the room equalization, first, RTF should be measured. In this work, we used sine sweep signal for the accurate RTF measurement [1]. As shown in Fig.1, the sine sweep signal  $x(n)$  is convolved with  $h(n)$  that is room impulse response. The *not processed* signal  $y_{np}(n)$  is a noisy signal.

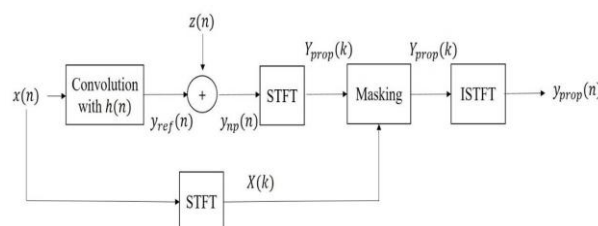


Figure 1. RTF measurement method [1]

Fig. 1 shows the process for the RTF measurement. The method divides  $y_{np}(n)$  into 50% overlap frames and applies the Hanning window. Then short-time Fourier transform (STFT) is applied to obtain  $Y_{np}(k)$ . Next,  $Y_{np}(k)$  is masked to obtain the frequency response of the sine sweep signal. The masker has a different range and location for each frame.  $K_{\max}(i)$  is the frequency bin index of largest frequency component in  $X(k)$  when the frame index is  $i$ . The masker  $M(i, k)$  is described as in Eq. (1).

$$M(i, k) = \begin{cases} 1, -\Delta_1(i) \leq K_{\max}(i) - k \leq \Delta_2(i) \\ 0, \text{otherwise} \end{cases} \quad (1)$$

The masked spectral components are expressed in Eq. (2).

$$Y_{prop}(i, k) = Y_{np}(i, k) M(i, k) \quad (2)$$

Likewise,  $Y_{prop}(i, k)$  is divided into 50% overlap frames and the Hanning window is applied. Afterward, inverse STFT is performed to obtain the noise-suppressed signal  $y_{prop}(n)$ .

### III. ROOM EQUALIZATION

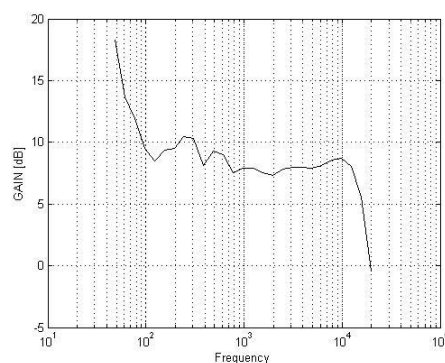
In this section, we describe the room equalization techniques based on RTF estimated by the short-time spectral masking method. When the RTFs are known a priori or are capable of being accurately estimated, this approach has been shown to achieve high inverse filtering performance [2, 3]. However, in actual acoustic environments, there are disturbances that affect the inverse filtering performance. In some audio applications (e.g., sound reproduction systems in train stations or other large spaces) where audio quality is an issue, an equalization filter is commonly used to compensate for the frequency response of the room. The equalization performance then depends on the model from which the inverse filter is derived. In our work, the equalization filter is one-third octave band filter which is composed of 27-octave bands that first frequency doubled every 3rd octave at a sampling frequency 48000Hz. Also, the center frequency of first band is about 47Hz. Filter gains are obtained by averaging the smoothed RTF. In this work, we use the complex smoothing method [4]. The complex smoothing operation may be described as a circular convolution as in Eq. (3).

$$H_{CS}(K) = H(k) \otimes W_{sm}(k) = \sum_{i=0}^{N-1} H((k-i) \bmod N) W_{sm}(i) \quad (3)$$

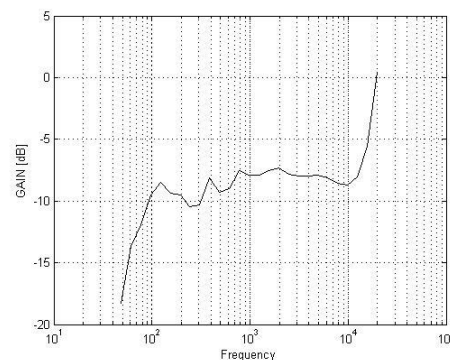
where the symbol  $\otimes$  denotes the operation of circular convolution and  $H(k)$  is transfer function,  $W_{sm}(k)$  is a spectral smoothing function having the general form of a low-pass filter.  $H_{CS}(K)$  is the result of complex smoothing of  $H(k)$ . By using the filters that have gains based on the smoothed inverse RTF, the overall RTF is equalized. audio quality, in such scenarios, is enhanced by performing the room equalization.

### IV. EXPERIMENT AND RESULTS

From the experiments, the estimated RTF and inverse of estimated RTF are shown in the Fig. 2.



(a) The estimated RTF function



(b) The inverse of estimated RTF function

Figure 2. The estimated RTF and the inverse of the RTF

The *processed* signal by the proposed room equalization becomes similar to original signal. In this experiment, we used log spectral distance (LSD) for objective assessment. The LSD is the difference between reference signals and signals obtained in the frequency region, and can be calculated as

$$LSD = \frac{1}{M} \sum_{i=1}^M \sqrt{\left( \frac{1}{N} \sum_{k=1}^N (10 \log \left( \frac{|Y_{ref}(i, k)|^2}{|Y(i, k)|^2} \right)) \right)^2} \quad (4)$$

where  $M$  is the number of frames. As shown in the Table 1, the developed room equalization techniques improved audio quality. The table shows the LSD comparison results between original signal and *not processed* or *processed* signals, respectively. Two male and two female speech signals were used for the experiments. The sampling frequency was 48 kHz and the length of signals was 4 seconds.

Table 1. LSD (dB) between original signal and not processed or processed signals

signal	not processed	processed
Male 1	29.86	25.84
Male 2	29.73	25.78
Female 1	29.12	25.08
Female 2	28.87	25.13

### V. CONCLUSION

In listening situations, audio quality can be degraded by reverberation. First, we developed the techniques to measure the room transfer function in noise environments. Then, room equalization is used to enhance the audio quality by applying the inverse transfer function to the input audio signals. From the experiments, it is shown that the room equalization techniques can improve the audio quality in reverberant indoor room environments.

### ACKNOWLEDGEMENTS

This study was supported by the Research Program funded by the Seoul National University of Science and Technology.

### REFERENCES

- [1]. Deokgyu Yun, Jonghan Joo, Seunghyon Na, Min-jeong Kim, Yeoung Hwangbo and Seung Ho Choi, "A Short-Time Spectral Masking Method for Room Transfer Function Measurement Based on Sine Sweep Signal in Noisy Environment," The 2<sup>nd</sup> International Joint Conference on Convergence, pp. 220-221, Jan. 2016.
- [2]. P. A. Nelson, F. Orduna-Bustamante, and H. Hamada, "Inverse filter design and equalization zones in multichannel sound reproduction," IEEE Trans. Speech Audio Process., vol. 3, no. 3, pp. 185 – 192, May 1995.
- [3]. Tobias Corbach, Adrian von dem Knesebeck, Kristjan Dempo, "Automated Equalization for Room Resonance Suppression," Proc. Of the Int. Conf. on Digital Audio Effects, Sep. 2009.
- [4]. Panagiotis Hatziantoniou, John Mourjopoulos, "Generalized Fractional-Octave Smoothing of Audio and Acoustic Responses," Journal of the Audio Engineering Society, Vol. 48, No. 4, pp. 259-280, Apr. 2000.