# Noise Subtraction methods for Transfer Function Measurement

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## Introduction

Usually when we are doing the transfer function measurement of a loudspeaker, the systems are disturbed by external noise, (e.g. random noise, traffic noise or machine noise); therefore the signal-to-noise ratio (SNR) is not sufficient. Generally, there are two way to get high SNR results: one is by making the excitation signals much longer or average the measurement many times; the other way is to employ more microphones and adaptive noise cancellation algorithms to reduce the noise[1]. In this paper we start from a simple noise subtraction method, employing 2 microphones to subtract the noises between the two microphones to suppress the noise.

## De-noising Method

We are using the sweep as the excitation signal. Assuming that $s(t)$ is the sweep signal generated by the loudspeaker, and $n_1(t)$ is the noise. Ch1 and Ch2 are the signals recorded by microphones, shown in (1) and (2). Here we only consider the direct sound, ignoring the reflections and diffusion fields.

$$\text{Ch1:} \quad s(t)+n_1(t) \qquad (1)$$

$$\text{Ch2:} \quad k_{21}s(t-\tau_1)+k_{22}n_1(t-\tau_2) \qquad (2)$$

Because the delays ($\tau_1$ and $\tau_2$) of the signal arrival, and the attenuation factors $k_{11}$, $k_{22}$, we can not subtract the two channels directly. We have to align the delay at first, and then subtract the two channels, as shown in (3)

$$k_{22}\mathrm{ch1}(t-\tau_2)-\mathrm{ch2}(t)=k_{22}s(t-\tau_2)-k_{21}s(t-\tau_1) \quad (3)$$

Here the delays ($\tau_1$ and $\tau_2$) are estimated by the time-difference-of arrival estimation (TDOA) technique [1], and the factor $k_{22}$ is estimated by the least mean square method which minimizes the residual noise. But after subtraction, the sweep signal are not the original sweep any more, it becomes $k_{22}s(t-\tau_2)-k_{21}s(t-\tau_1)$ . Since our measurement system is a time-invariant system, if we deconvolute it with sweep, then the impulse response will becomes

$$k_{22}h(t-\tau_2)-k_{21}h(t-\tau_1) \qquad (4)$$

Where $h$(t) is the real impulse response of the loudspeaker. Because when we calculate the transfer function from the impulse response of the loudspeaker, the only the direct sound are used, so we employ a time window to set the second term of Expression (4) to zeros and obtain the clean impulse response. Finally, we transform this clean impulse response to frequency domain to get de-noised transfer function.

## Experiment

Firstly, we do the experiment in the anechoic chamber, using one loudspeaker to generate the sweep and another loudspeaker to generate a strong white noise. The two loudspeakers and the two microphones are mounted in one straight line to avoid the $k$ factor changing with different directions at different frequencies of the loudspeaker. Secondly, we do the same measurement in a normal room

Figure 1 shows our raw data. Then we align the delay of noise and subtract the two channels. The effect of the noise reduction is frequency dependent. As shown in Figure 2, the noises between 500~4000Hz are highly suppressed, but the most of noises over the 5000 Hz are still remained.

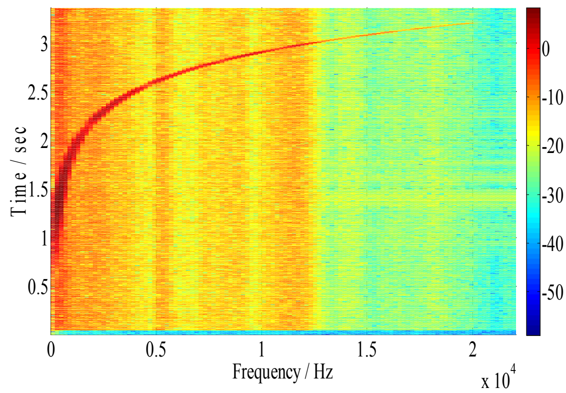


**Figure 1:** spectrogram of the original data recorded by the microphone

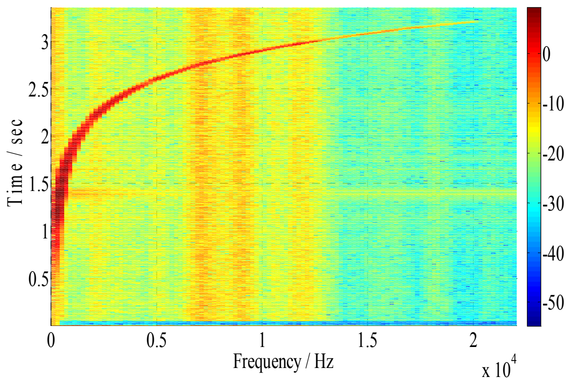


Figure 2: the spectrogram after subtraction $k_{22}$*ch1(t–$\tau_2$)-ch2(t)

If we define the improvement of the signal-to-noise ratio (SNR) as

$$SNR\ \text{improvement} = \frac{\text{noise pressure before subtraction}}{\text{noise pressure after subtraction}}$$

Then SNR improvement in the anechoic chamber is shown is Fig.3. Because no diffusion field exists here, then SNR improvement does not decrease too much with longer microphone-to-noise the distances.

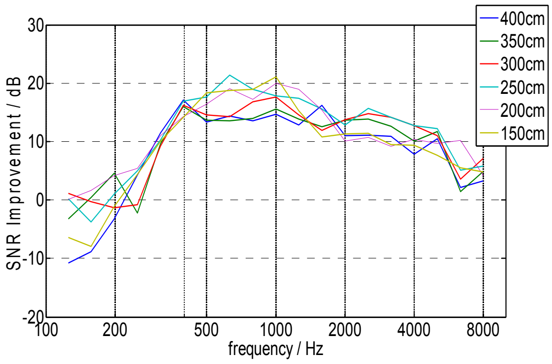


Figure 3a: the SNR improvement of subtraction methods with different noise-to-microphones and the different frequency in the anechoic chamber

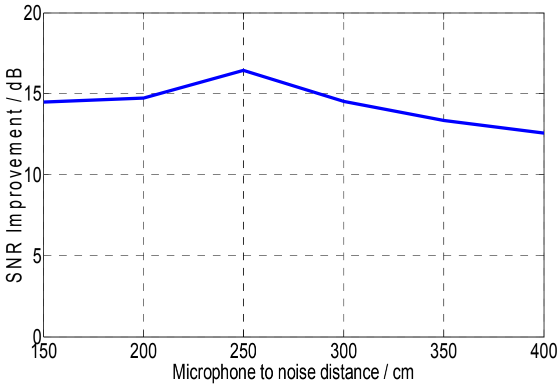


Figure 3b: average SNR improvement between 500Hz~4000Hz with the distance in the anechoic chamber.

But in a normal room, where the noise not only contains a direct sound but also contains a lot of reflections and strong diffusion field, this subtraction method does not work, because in diffusion field the delays of signal arrival are random with frequencies, and the attenuation factors $k_{21}$ and $k_{22}$ are also unpredictable with different frequencies. However, if microphones are mounted within the reverberation distance of a highly directional loudspeaker, where the direct sound are stronger than the diffusion sound, the effect of this subtraction methods are so bad.

Fig. 4 shows the SNR improvement of this subtraction method. The measurement is recorded in a normal room with size of approximately 5m × 7m × 8m (width × length × height) and with the reverberation time of 1.2 seconds. And the noise source is a directional loudspeaker. The microphones and the noise source are also mounted in one straight line, and the distance between the two microphones is fixed to 50cm, and we measure the SNR improvement with different noise-to-microphone distances. Fig. 4a show the SNR improvement of subtraction methods with different noise-to-microphones and the different frequency, Fig. 4b shows the average SNR improvement between 500Hz ~8000Hz with different noise-to-microphone distances. It decreases with larger noise-to-microphone distance. The reason is that when the direct sound field is lower than the diffusion field, we can not obtain an accurate estimation of the correct $k_{22}$ factor, and even if we can obtain an accurate $k_{22}$ factor, then diffusion field can not be suppressed by the this subtraction method. Actually if we place the microphone closer to the noise source, the SNR improvement will be higher, but original SNR will be lower, and finally the total SNR are not improved.

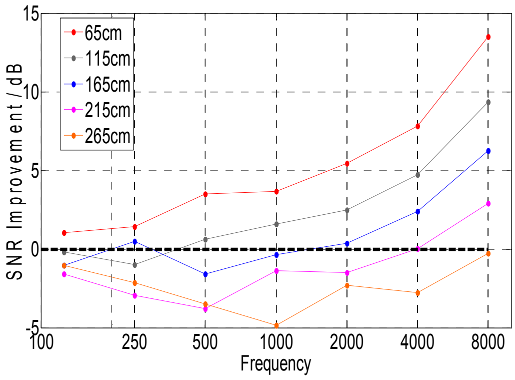


Figure 4a: the SNR improvement of subtraction methods with different noise-to-microphones and the different frequency in a normal room

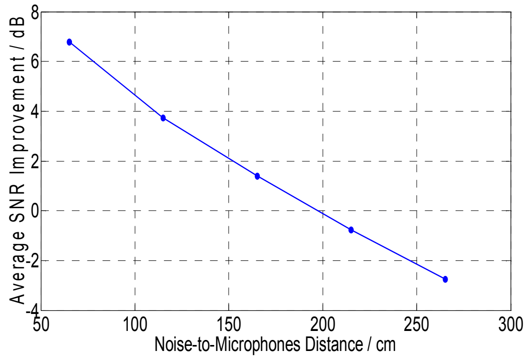


Figure 4b: average SNR improvement between 500Hz~8000Hz with the distance in a normal room

Furthermore, in this normal room, if the directional loudspeaker are replaced by an omnidirectional loudspeaker as the noise source, this subtraction methods completely can not subtract noise any more, because the omnidirectional loudspeaker generate the noise to all the directions and make diffusion field stronger comparing to the directional loudspeaker.

## Discussion

In this paper, we tried a simple subtraction method to suppress the external noise during the transfer function measurement of a loudspeaker. This method can achieve approximately 10dB SNR improvement in free field condition. It might be applied in the open air measurement, but it can not achieve a good SNR in a normal room with strong diffusion field.

## Reference

[1] Jacob Benesty, Jingdong Chen, Yiteng Huang: Microphone Array Signal Processing, Springer, 2008