Audio Engineering Society

Convention Paper 7239

Presented at the 123rd Convention
2007 October 5–8 New York, NY, USA

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Addressing the Discrepancy Between Measured and Modeled Impulse Responses for Small Rooms

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ABSTRACT

Simple computer modeling of impulse responses for small rectangular rooms is typically based on the image source method, which results in an impulse response with very high time resolution. Image source method is easy to implement, but simulated impulse responses are often a poor match to measured impulse responses because descriptions of sources, receivers, and room surfaces are often too idealized to match real measurement conditions. In this paper, a more elaborate room impulse response computer modeling technique is developed by incorporating measured polar responses of speaker, measured polar responses of microphone, and measured reflection coefficients of room surfaces into basic image source method. Results show that compared with basic image source method, the modeled room impulse response using this method is a better match to the measured room impulse response, as predicted by standard acoustical theories and principles.

1. INTRODUCTION

The acoustic response recorded by a receiver in a room varies according to positions of the source and the receiver, the orientation of the source and the receiver, and the acoustic condition of the room surfaces. Room acoustic responses can be modeled or measured for various purposes [1]. For example, when designing new professional spaces like concert halls, the room acoustic response model can help to predict the acoustical characteristics of the finished space. In another purpose, it may be desired to produce naturally sounding studio effects for music production. A room acoustic response model can help to create a virtual studio effect without building the actual room. In yet another purpose, it may be desirable to choose the most suitable acoustical absorbers in architecture. Room acoustic response can help to compare the effect of different absorbing materials and treatments.

The room acoustic response can be separated into three segments: direct sound, early reflections, and late reverberation [2]. Direct sound is produced when sound wave propagates directly from the source to the receiver. Early reflections are sparse, discrete reflected sounds from nearby surfaces. Late reverberation is densely populated with reflected sounds.

A general approach to denote room acoustic response is based on the impulse response between the source and the receiver in the room. One straightforward method to obtain room impulse response is to measure it directly using an appropriate source signal [3]. When measuring the impulse response, the source, the room, and the microphone are treated together as a system.

An impulse response can be obtained using the dual channel method [4]. The dual channel method allows almost any broadband signal as input, but it needs to simultaneously measure and process input and output of the system. An impulse response can also be obtained by exciting the system with an impulse-like signal or a maximum length sequence (MLS) [5]–[6]. The impulse method is very simple but it usually cannot produce enough energy to give a reasonable signal to noise ratio (SNR). The MLS method can give a good SNR due to its low crest factor but it is vulnerable to nonlinear distortion and time variance of the measurement system.

The room impulse response can also be modeled using many methods. These methods are generally classified into three categories: physical models, scale models, and computer models [7]. Physical models can model wave phenomena of sound but they are very expensive to build. Scale models can reduce overall size and complexity of testing, so they are more efficient for designing large halls. For small room impulse responses, computer software models are used almost exclusively because the results can be analyzed very conveniently.

There have been decades of work devoted to modeling room impulse responses using computer software [8]. These methods are usually classified into wave-based methods and ray-based methods. Wave-based methods are based on the general solution of the wave equation and they can efficiently model the correct physics of room acoustics [9]-[10]. Since analytic solutions for the wave equation are available only for very simple cases like rectangular rooms, numerical methods must usually be applied to successfully solve different room acoustics problems. However, these wave-based methods are primarily used only for low-frequency sound because computational expense increases rapidly with increasing bandwidth. Ray-based methods are based on geometric room acoustics and assume that sound wave propagates like a plane wave, so the wavefront propagation may be defined as a ray [11]. Although this assumption is more reasonable for high-frequency sound whose wavelength is small compared to the dimension of rooms, ray-based methods are often used to solve acoustics problems over the whole audio frequency range for rooms with simple geometries.

The image source method is one of the most common ray-based methods [12]-[14]. In the image source method the reflected path from a real source is replaced by a direct path from a corresponding image source. Image source method is widely used in modeling impulse responses of small rectangular rooms since it results in an impulse response with very high time resolution. However, modeling results from the basic image source method are often found to be inaccurate because the descriptions of sources, receivers and room surfaces in the computer models are often too idealized to match real conditions. For example, the basic image source method has often assumed sound source to be an omni-directional point source for ease of implementation, but a real loudspeaker may include multiple drivers and exhibit an irregular polar response.

The principal goal of this paper is to address the discrepancies found when comparing measured and simulated impulse responses. Thus, we develop a room impulse response computer modeling technique that extends the image source method by using measured parameters of the speaker, microphone, and room surfaces. We seek to understand better why simulations and measurements may differ even for simple cases like small rectangular rooms, and therefore to obtain a better match between simulations with measurements.

The remaining sections of this paper are organized as follows. First, measurement results for the room impulse response are demonstrated for a small room with simple geometry. Next, the simple and the improved room impulse response models are described and compared to the measured impulse response. Finally, the paper concludes with a summary of results and future work.

2. MEASURING ROOM IMPULSE RESPONSES

A practical and repeatable room impulse response measurement system usually needs a signal generator, a speaker to create acoustical sound, a microphone to receive sound signal, and a recording device, as shown in Fig. 1. In our measurement system, the excitation signal is generated using a personal computer (PC) and passes through a digital to analog converter (DAC) and an audio mixer. The sound signal is then sent to a loudspeaker (Mackie HR824) and received by a microphone (DPA 4003). Both the loudspeaker and the microphone are commercially available products that are typical of common recording studio equipment. The received signal passes through the mixer and an analog to digital converter (ADC) to reach the PC.

Before making room impulse response measurements, system noise and harmonic distortion were measured to make sure that the system was sufficiently precise to measure room impulse responses. The noise of the system was found to be -80 dB compared to full scale, which is sufficient for experiments since the received signal magnitude is usually as large as -20 dB FS. There was some harmonic distortion in the recorded sound, but distortion components were 40 to 50 dB below the fundamental frequency. According to the noise and harmonic distortion measurement, the system is found to be sufficiently precise for room impulse response measurements.

As mentioned above, the goal of this work is to obtain a better match between the *modeled* room impulse response and the *measured* room impulse response. We initially had limited experience in obtaining small room impulse response measurements, so we performed several preliminary experiments to must make sure the measurements were consistent. Thus, in a preliminary experiment, the dual channel method, the impulse method, and the MLS method were used to measure room impulse responses for several speaker and microphone positions. Since the measured results using these three methods were found to be nearly identical, the impulse response measurements are deemed to be accurate and consistent, as explained next.

2.1. Measuring Room Impulse Response Using the Dual Channel Method

The speaker and the microphone were at first set at fixed positions and the room impulse response was measured using SmaartLive software [15]. SmaartLive is a software-based dual-channel audio analyzer capable of measuring the impulse response between a speaker and a microphone in a room. SmaartLive contains an internal signal generator that simplifies the measurement process by creating appropriate excitation signals for each measurement.

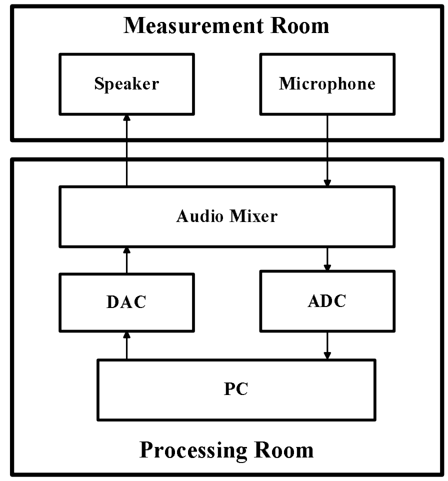

Figure 1: Room impulse response measurement system

For impulse response measurements using SmaartLive, the input signal is either an internally generated pink noise or an internally generated pink sweep. For each input signal the impulse response was measured twice. The two measurements were found to be very consistent and they were averaged to increase SNR. Thus, one averaged impulse response for every input signal was obtained. Fig. 2 shows an example room impulse response measured using SmaartLive with pink noise for the source.

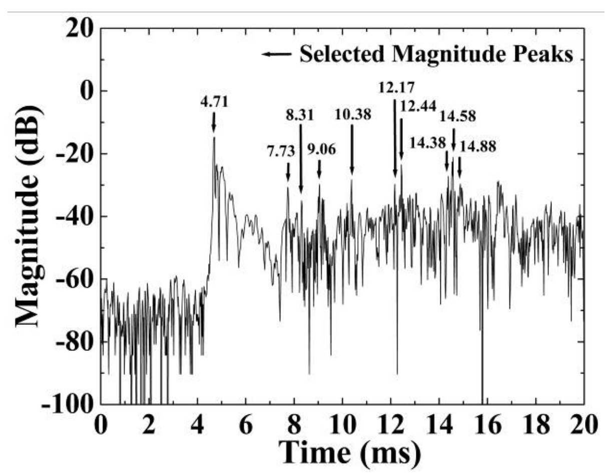

Figure 2: Measuring room impulse response using the dual channel method

To make it easier to compare measurement results with those measured using the pink sweep as the source, ten peaks which appear to be discrete reflections were chosen, as indicated with arrows in Fig. 2. The respective delay times and magnitudes of these ten peaks in every impulse response were extracted and shown in Table 1. Note that in the table the delay time for every peak is the measured value, while the magnitude of every peak is normalized by the magnitude of the first peak (strongest) peak.

The measurement results are very close for two input signals, which means that room impulse response measurements using SmaartLive are repeatable and independent of the choice of input signal.

2.2. Measuring Room Impulse Response Using the Impulse Method

The room impulse response was then measured using the impulse method. To increase the SNR, the input was fed with an impulse train with period greater than the length of the impulse response to avoid time aliasing. The recorded output of the system, i.e., the periodical impulse response, was then averaged synchronously to get a final measured impulse response. The measurement results using the impulse method are also included in Table 1. The measurement results using this method are comparable to those using the dual channel method except for a small magnitude discrepancy of some peak values.

2.3. Measuring Room Impulse Response Using the MLS Method

The room impulse response was also measured using the MLS method. In the measurement, the input was fed with MLS signals with order 15 and order 16. The output signal was then recorded. The cross correlation of the input and output signal was then used to compute the impulse response. The measured results using MLS signals are also shown in Table 1. It can be seen from the table that measurement results using the MLS method are nearly identical to those using the dual channel and the impulse methods.

2.4. Additional Tests

Three different speaker-microphone positions were utilized with the above-mentioned three methods to measure room impulse responses. Measurement results were congruent for these three methods for each position. Among these three methods, the impulse method is the simplest technique to implement, but this method needs to average many cycles of the impulse response to increase the SNR. It seems that the MLS method is the most desired method, but the result will be affected by any nonlinear or time variant parameters. One common disadvantage of the impulse method and the MLS method is that they cannot report direct transmission time between the speaker and the microphone. Consequently, these two methods cannot be used alone to get the final impulse response. Based on these practical considerations, we choose to use the dual channel method because it can measure delay time of direct transmission and other reflections accurately. Thus, in the following sections the room impulse response measurements are made with the dual channel method using pink noise as source.

Peak Number	Delay (ms)	Magnitude (db)				
		Dual Channel Method (Pink Noise)	Dual Channel Method (Pink Sweep)	Impulse Method	MLS Method (Order 15)	MLS Method (Order 16)
1	4.71	0	0	0	0	0
2	7.73	-16.0	-16.0	-16.8	-16.3	-16.4
3	8.31	-20.2	-20.8	-20.8	-21.2	-21.1
4	9.06	-15.1	-15.1	-16.2	-16.0	-16.1
5	10.38	-13.5	-14.1	-15.0	-15.3	-15.0
6	12.17	-15.0	-14.8	-15.9	-15.8	-15.7
7	12.44	-8.8	-8.6	-9.2	-9.0	-9.1
8	14.38	-12.5	-12.3	-13.5	-13.2	-13.3
9	14.58	-6.5	-6.6	-7.5	-7.7	-7.9
10	14.88	-14.9	-14.8	-15.3	-15.4	-15.7

Table 1: Delay time and magnitude of ten randomly chosen peaks of the measured room impulse responses using dual channel, impulse, and MLS methods

3. MODELING ROOM IMPULSE RESPONSE USING IMAGE SOURCE METHOD

In a second experiment, the room impulse response for a particular speaker-microphone position was measured using the dual channel method and modeled with the image source method. If the source and receiver are assumed to be omni-directional and the room surfaces are assumed to be infinite rigid boundaries, there is a large difference between modeled and measured room impulse responses, indicating that the simple model is insufficient to be relied upon in a practical design situation. If some more realistic measurement parameters—like the speaker polar response, microphone polar response, and room surface reflection coefficients—are included in the model, differences between measured and modeled responses are reduced. Consequently, it seems clear that the usefulness of the modeled room impulse response can be improved by taking into account the effects of speaker, microphone, and room surfaces. This assertion is verified by the following tests.

3.1. Simple Model

Before the room impulse response was modeled, the measurement room dimensions, the tweeter position, the woofer position, the speaker position, and the microphone position were measured. The layout of the measurement room is shown in Fig. 3. The rectangular measurement room is 3.37 meters long, 3.03 meters wide, and 2.39 meters high. The reference point for the tweeter (the tweeter position) is the center of the tweeter, the reference point for the woofer (the woofer position) is the center of the woofer, the reference point for the speaker (the speaker position) is the mid-point between the tweeter position and the woofer position, and the reference point for the microphone (the microphone position) is the center of the diaphragm. For this experiment, the tweeter position is (1.91, 1.55, 1.11) meters, the woofer position is (1.91, 1.55, 0.92) meters, the speaker position is (1.91, 1.55, 1.02) meters, and the microphone position is (2.78, 1.52, 1.02) meters. The temperature in the measurement room was 66° Fahrenheit (19 C), giving a sound velocity of 342.8 m/s.

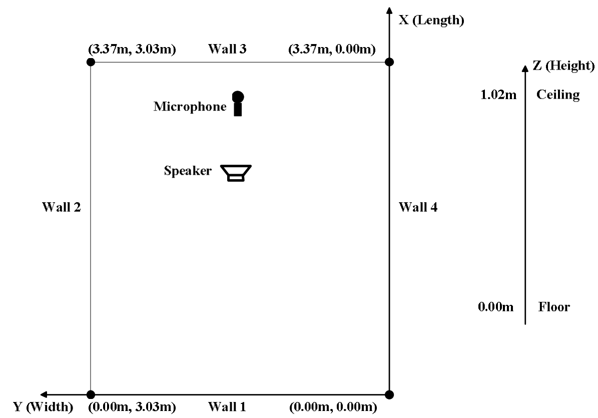

Figure 3: Layout of the measurement room

The room impulse response was measured using the dual channel method with pink noise as the source and then modeled with the image source method. The speaker and the microphone were at first assumed to be omni-directional and the room surfaces were assumed to be infinite rigid boundaries. To facilitate the analysis, only seven points were extracted from the modeled room impulse response. These seven points correspond to direct sound between the speaker and the microphone and six first-order reflections from room surfaces. The calculated pressure magnitude at these seven points was then compared with that of the corresponding points in the measured room impulse response.

Due to the measurement uncertainty (perhaps ±1.5 cm) of the room dimensions and the source and receiver positions, delay time of magnitude peaks in the modeled impulse response and that of the measured values may not be exactly the same. Thus, a satisfactory match is judged to occur if the delay time difference is within 0.10 ms. According to this rule, if the delay time of a magnitude peak in the modeled impulse response is within 0.10 ms of the measured value, the magnitude peak is verified. Otherwise, the sample in the measured impulse response having the same delay time as the modeled value will be chosen. For example, the delay time of direct sound in the modeled impulse response is 2.52 ms and that in the measured impulse response is 2.58 ms. The difference is within 0.10 ms, so the delay time is assigned to be 2.58 ms in the measured impulse response. For another example, the modeled delay time of one of the first-order reflections is assigned to be 8.42 ms but no peaks in the measured impulse response are within 0.10 ms of this point. Thus, the (non-peak) sample value at time 8.42 ms in the measured impulse response is chosen. Seven points from the measured impulse response are extracted and shown in Fig. 4.

A comparison between the modeled values and the corresponding measurements is shown in Fig. 5 and Table 2. In Table 2, the description column represents how the sound is played (sound source), where the sound is reflected (sound reflection), and what the horizontal and the vertical angles between the sound propagation path and the axis of the speaker are (sound orientation). From Table 2 it can be seen that the extracted values of the modeled and measured room impulse responses are very close for the reflected sound from wall 3, while they differ substantially for other reflected sounds. The important insight is that the reflected sound from wall 3 is nearly on the axis of the speaker while other reflections are not. Consequently, the polar responses of the speaker and the microphone that are presented in the measured response but not in the simple model must be included. Moreover, the reflection coefficients of the real room surfaces are not unity and this must also be accounted for in the modeled result.

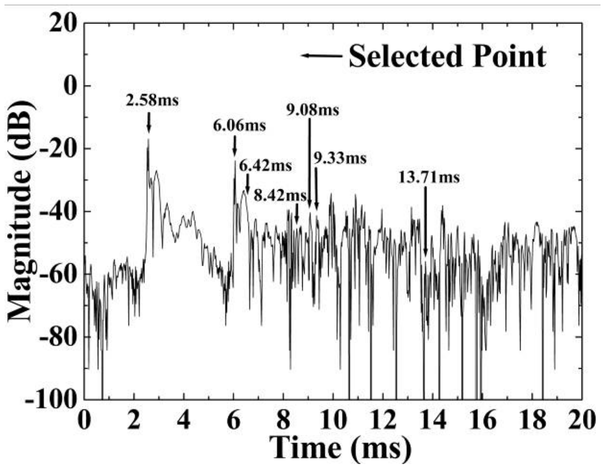

Figure 4: Direct sound and six first-order reflections of the measured room impulse response

3.2. Modified Model

Speaker polar response, microphone polar response, and room surface reflection coefficients all affect the magnitude and the phase of modeled room impulse response. These parameters were measured and then incorporated into basic image source method. In the modified model, the small room, the speaker, and the microphone were treated together as a linear time invariant (LTI) system. Since the input of this system is an impulse signal, the output of this system is the impulse response of the system, in this case the modeled room impulse response.

A MATLAB program was used to implement the modified method to model room impulse response for the above-mentioned speaker-microphone position.

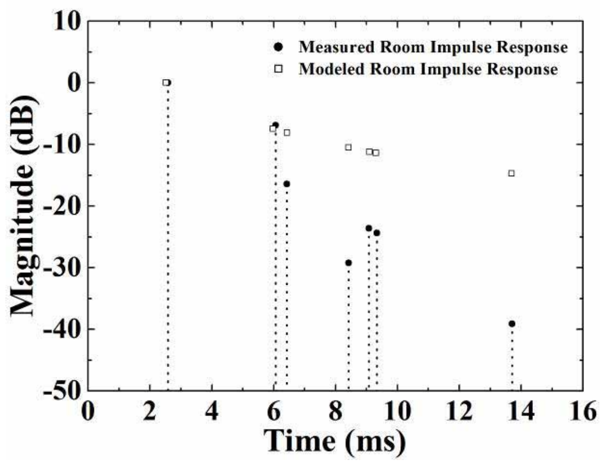

Figure 5: Comparison between the direct sound and the six first-order reflections of the simple modeled and measured room impulse response

The program is divided into the following parts: program initialization, sound source processing, speaker polar response calculation, microphone polar response calculation, room surface reflection coefficient calculation, image source processing, and finally room impulse response calculation. Fig. 6 shows the program flowchart, which is described in more detail in the following sub-sections.

3.2.1. Program initialization

In this module, values of some descriptive parameters are set. First, room dimensions, woofer position, tweeter position, and microphone position are specified. Next, sound velocity is set according to room temperature. Then, the sampling rate is set to 48 kHz, FFT length to 32 768 samples, and simulation time duration to 683 milliseconds (32 768 samples at the sampling rate 48 kHz).

Measurement		Image Source Method		Difference		Description		
Delay (ms)	Magnitude (dB)	Delay (ms)	Magnitude (dB)	Delay (ms)	Magnitude (dB)	Sound Source	Sound Reflection	Sound Orientation
2.58	0	2.52	0	-0.06	0	Played by speaker	Direct Sound	(1.9°, 90.0°)
6.06	-6.9	5.98	-7.5	-0.08	-0.6	Played by speaker	from wall 3	(0.8°, 90.0°)
6.42	-16.4	6.44	-8.1	0.02	8.3	Played by speaker	From floor	(1.9°, 23.1°)
8.42	-29.2	8.42	-10.5	0.00	18.7	Played by speaker	From ceiling	(1.9°, 162.5°)
9.08	-23.6	9.10	-11.2	0.02	12.4	Played by speaker	From wall 2	(286.1°, 90.0°)
9.33	-24.4	9.31	-11.4	-0.02	13.0	Played by speaker	From wall 4	(74.2°, 90.0°)
13.71	-39.1	13.69	-14.7	-0.02	24.4	Played by speaker	From wall 1	(179.7°, 90.0°)

Table 2: Comparison between Measured and Simple Modeled Room Impulse Response (the speaker is assumed to be one point source)

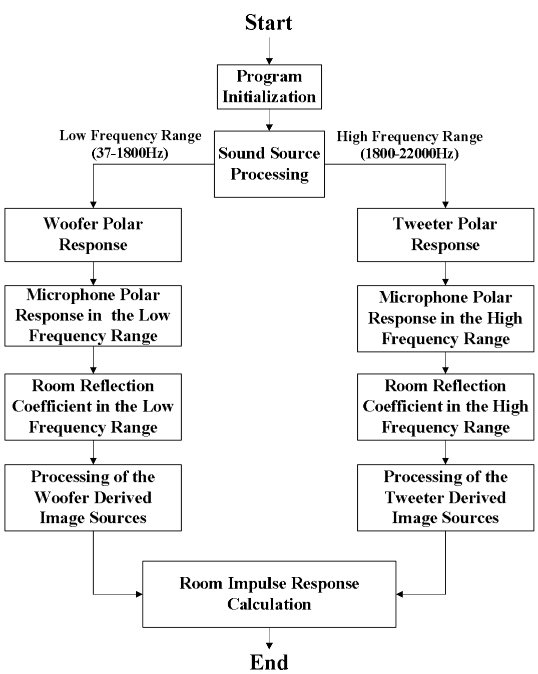

Figure 6: Program Flowchart

3.2.2. Sound source processing

A two-way speaker (woofer and tweeter) was used in this experiment. When the speaker is fed with an input signal, the crossover network directs the lower frequency part of the input signal to the woofer and the high-frequency part to the tweeter. According to the Mackie HR824 speaker specification, the woofer 3 dB bandwidth is 37 Hz to 1800 Hz while the tweeter bandwidth is 1800 Hz to 22000 Hz. In the simulation program one source was used to simulate the woofer and the other source was used to simulate the tweeter. The woofer polar response and the tweeter polar response were also individually taken into account in the simulation program.

3.2.3. Speaker polar response calculation

To measure the tweeter polar response, the driver was located in a fixed position and the reference point for the microphone was placed in different positions on three circles, centered on the tweeter. The first circle is on the horizontal plane of the tweeter (Horizontal direction), the second circle is erected perpendicular to the lateral side of the tweeter (Vertical 1 direction), while the third circle is erected perpendicular to the front side of the tweeter (Vertical 2 direction). 66 different tweeter-microphone positions were measured using a simulated bandpass signal (1.8 kHz – 22 kHz). The tweeter impulse responses for these 66 different tweeter-microphone positions were then measured using a time window short enough to include only the direct sound arrival.

A three-dimensional linear interpolation function was used to interpolate the measured tweeter polar responses for several positions, where the horizontal angles changed from 0 to 360 degrees with 3 degrees spacing and the vertical angles changed from 0 to 180 degrees with 3 degrees spacing.

Fig. 7 shows the tweeter polar response at 5 kHz in the horizontal direction, the vertical 1 direction, and the vertical 2 direction. It can be seen that the tweeter is quite directional at 5 kHz.

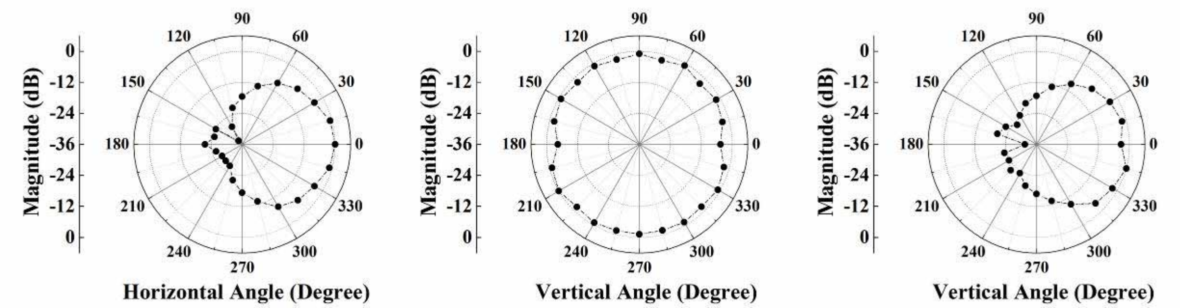

Figure 7: Tweeter Polar Response at 5 kHz (a) horizontal direction, (b) vert. 1 direction, and (c) vert. 2 direction

The interpolation in the simulation program was done in the time domain, which was very similar to that done in the frequency domain according to our test. The accuracy of the interpolation is verified as follows. The tweeter horizontal impulse response at the angles of 30, 42, and 45 degrees were at first measured and calculated. The measured horizontal impulse response at the angles of 30 and 45 degrees were then used to interpolate the horizontal impulse response of the tweeter at the angle of 42 degree. The original and the interpolated tweeter horizontal impulse responses at the angle of 42 degree were finally compared. It is shown in Fig. 8 that they have a very small discrepancy.

The woofer polar response was measured using similar procedures. The difference being that the woofer polar response was measured using a simulated bandpass signal (37 Hz – 1800 Hz) and the microphone was placed in different positions on circles centered on the woofer. Fig. 9 shows the woofer polar response at 1000 Hz in the horizontal direction, the vertical 1 direction, and the vertical 2 direction. It can be seen that the woofer is also quite directional at this frequency.

3.2.4. Microphone polar response calculation

The signal from the speaker can either go directly to the microphone or be reflected by the room surfaces and then reach the microphone. Thus, the frequency-dependent polar response of the microphone can affect the measured signal.

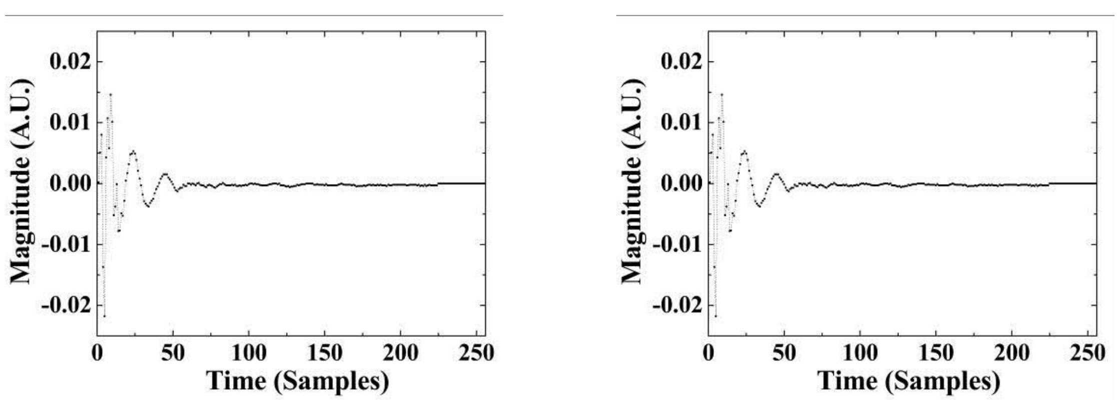

Figure 8: Comparison between (a) original and (b) interpolated tweeter horizontal impulse response at the angle of 42 degrees

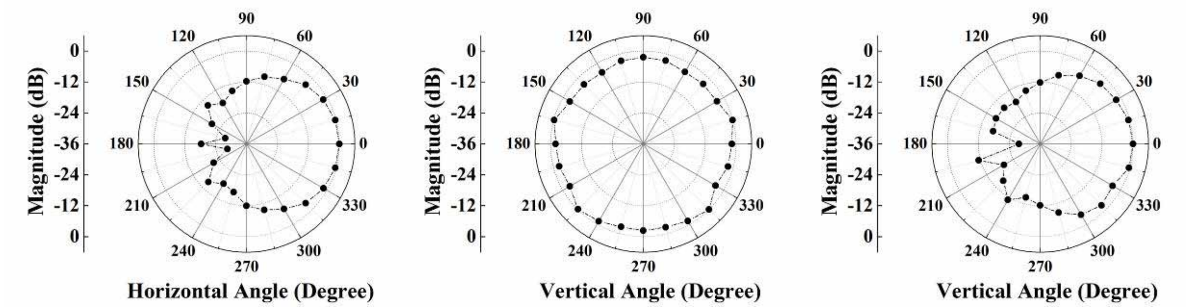

Figure 9: Woofer Polar Response at 1 kHz (a) horizontal direction, (b) vert. 1 direction, and (c) vert. 2 direction

The microphone polar response was measured for the frequency range of the tweeter, keeping the center of the tweeter and the reference point for the microphone fixed while the microphone was rotated around two circles. The first circle is erected perpendicular to the lateral side of the tweeter (Vertical 1 direction), while the second circle is erected perpendicular to the front side of the tweeter (Vertical 2 direction). Since the DPA 4003 microphone is very omnidirectional in the horizontal plane, the microphone horizontal polar response was not necessary to be considered in the simulation program and therefore was not re-measured.

The microphone polar responses for 46 different tweeter-microphone orientations were measured using a simulated bandpass signal with 3dB bandwidth from 1.8 kHz to 22 kHz. A three-dimensional linear interpolation function was used to interpolate the microphone polar response in the frequency range of the tweeter for several positions, where the horizontal angles changed from 0 to 360 degrees with 3 degrees spacing and the vertical angles changed from 0 to 180 degrees with 3 degrees spacing. The measurement of the microphone polar response in the woofer frequency range was also measured using a similar procedure.

Fig. 10 shows the microphone polar response at 1 kHz and Fig. 11 at 5 kHz in the vertical 1 direction and the vertical 2 direction. It can be seen that, as expected, the receiving microphone is less directional than the loudspeaker source, but the microphone directionality makes it a factor in modeling room impulse responses.

3.2.5. Room surface reflection coefficient calculation

The speaker output signal may be reflected by various room surfaces before reaching the microphone. The magnitude and phase of the speaker output signal will thus change due to the propagation distance and the effect of the room surface reflection coefficient. Measurements can be made to estimate the reflection coefficients of the room surfaces for use in the simulation program. The room surfaces are made of or covered with several kinds of materials. The speaker was fed with a bandpass signal with 3 dB bandwidth (37 Hz – 1800 Hz) to measure the low-frequency reflection coefficient and fed with a bandpass signal with 3 dB bandwidth (1.8 kHz – 22 kHz) to measure the reflection coefficient for the high frequencies.

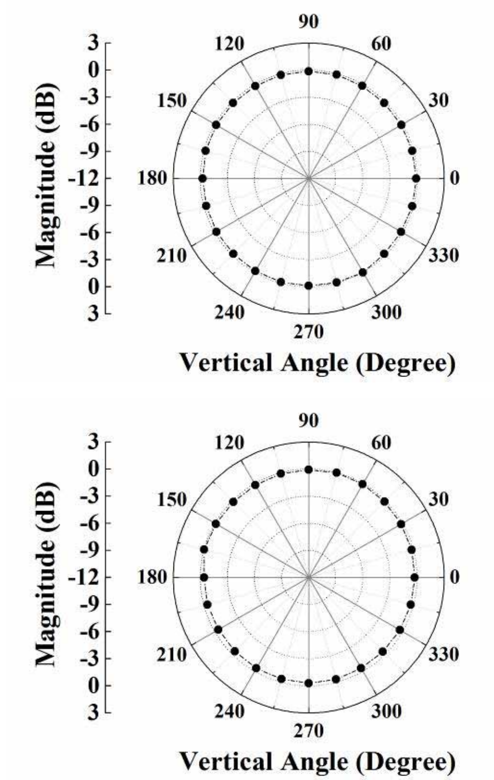

Figure 10: Microphone Polar Response at 1 kHz for (a) Vertical 1 and (b) Vertical 2 directions

In our test room the walls and ceiling are made of painted wallboard. To measure the low-frequency wall reflection coefficient, the center of the woofer was first aligned with the reference point for the microphone. The position of the woofer and the microphone must ensure that in the measured impulse response, the reflection sound from the measured room surface is the earliest reflection and may not be interfered with by the reflection from other room surfaces. The speaker was then fed with a bandpass signal with 3 dB bandwidth (37 Hz – 1800 Hz), assuming that no signal would come out of the tweeter. The woofer band limited impulse responses for 5 different woofer-microphone positions were then measured. For each position, the direct sound and the earliest reflection part were cut from the measured impulse response and transformed into the frequency domain using an FFT (length 512 samples) to calculate the frequency dependent reflection coefficients for low frequencies. The measurement method for the high-frequency wall reflection coefficient was similar, except that the center of the tweeter was aligned with the reference point for the microphone and the speaker was fed with a bandpass signal with 3 dB bandwidth (1.8 kHz – 22 kHz).

The low-frequency and high-frequency wall reflection coefficients were then combined to form the composite wall reflection coefficient for the whole audio frequency range. The processing for the reflection coefficients of the other surfaces (door and window) is similar to that for the wall. Since the walls and the ceiling of the room are made of the same material, their reflection coefficients are considered to be equal.

For convenience, measurement of the floor reflection coefficient was made when the sound path from the woofer (or tweeter) via the floor to the microphone was not on the axis of the woofer (or tweeter). The calculation of the floor reflection coefficient was similar to that of the wall reflection calculation except that the polar response of the woofer, the tweeter, and the microphone were considered.

Fig. 12 shows the magnitude of the frequency-dependent reflection coefficients of wall, door, window, and floor. Because of the small size of the window and its frame, trim, and gaskets, some part of the reflected sound from the window may be interfered with by the reflected sound from the surrounding surfaces near the window. The window reflection coefficient in some frequencies is therefore found to be greater than one.

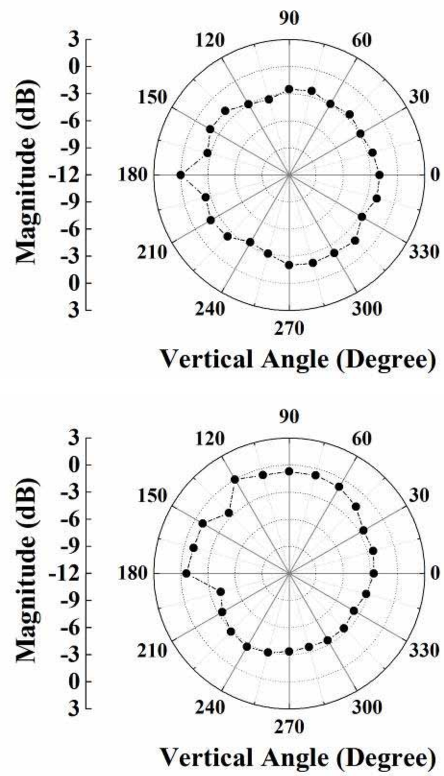

Figure 11: Microphone Polar Response at 5 kHz for (a) Vertical 1 and (b) Vertical 2 directions

3.2.6. Image source processing

When a two-way speaker is fed with an impulse or other broadband signal, the low-frequency part of the signal will be played by the woofer while the high-frequency part will be played by the tweeter. Thus, in the image method simulation program, one source was used to simulate the woofer and the other source was used to simulate the tweeter. The positions of the image sources due to the two drives were first calculated. The orientation of the reflections due to the image sources, i.e., the angles between the reflections and the real sources, were then computed. The orientation of the receiver, i.e., the angles between the reflections and the receiver, were also calculated. The intersection point between the reflection path and the room surfaces were also calculated to judge whether the intersection point was in the wall, the ceiling, the window, the door, or the floor. The amplitude loss and the phase change due to the reflections from the room surfaces were finally computed for the sound sources (the real sources and the image sources).

3.2.7. Room impulse response calculation

In the simulation program it was assumed that the sound would propagate spherically from the speaker directly to the microphone or via room surfaces and then to the microphone. The amplitude loss and the phase change from the sound sources to the receiver were computed first. The impulse responses between the receiver and sound sources were calculated by considering the amplitude and the phase change factors, which include the amplitude loss and the phase change due to the reflection coefficient of the room surfaces, the speaker polar response, the microphone polar response, and the spherical sound spreading. The impulse responses for the sound sources were then added together to form an overall impulse response.

3.2.8. Match between simulation and measurement

After measuring the speaker polar responses, the microphone polar response, and the room surface reflection coefficients, the room impulse response was re-modeled using these measured parameters. It should be noticed that when a two-way speaker is fed with a broadband signal, the low-frequency part of the signal will be played by woofer while the high-frequency part will be played by tweeter. In other words, these two sources should be considered separately. Figure 13 shows the on-axis woofer and tweeter impulse responses. It can be seen that the delay time of peaks for woofer and tweeter impulse responses is different.

Based on the above-mentioned consideration, 14 points corresponding to direct sound and six first-order reflections due to woofer as well as tweeter are selected in the modeled room impulse response. The magnitude of these fourteen points was then compared with that of the corresponding points in the measured room impulse response using the procedure designed in section 2.1 above.

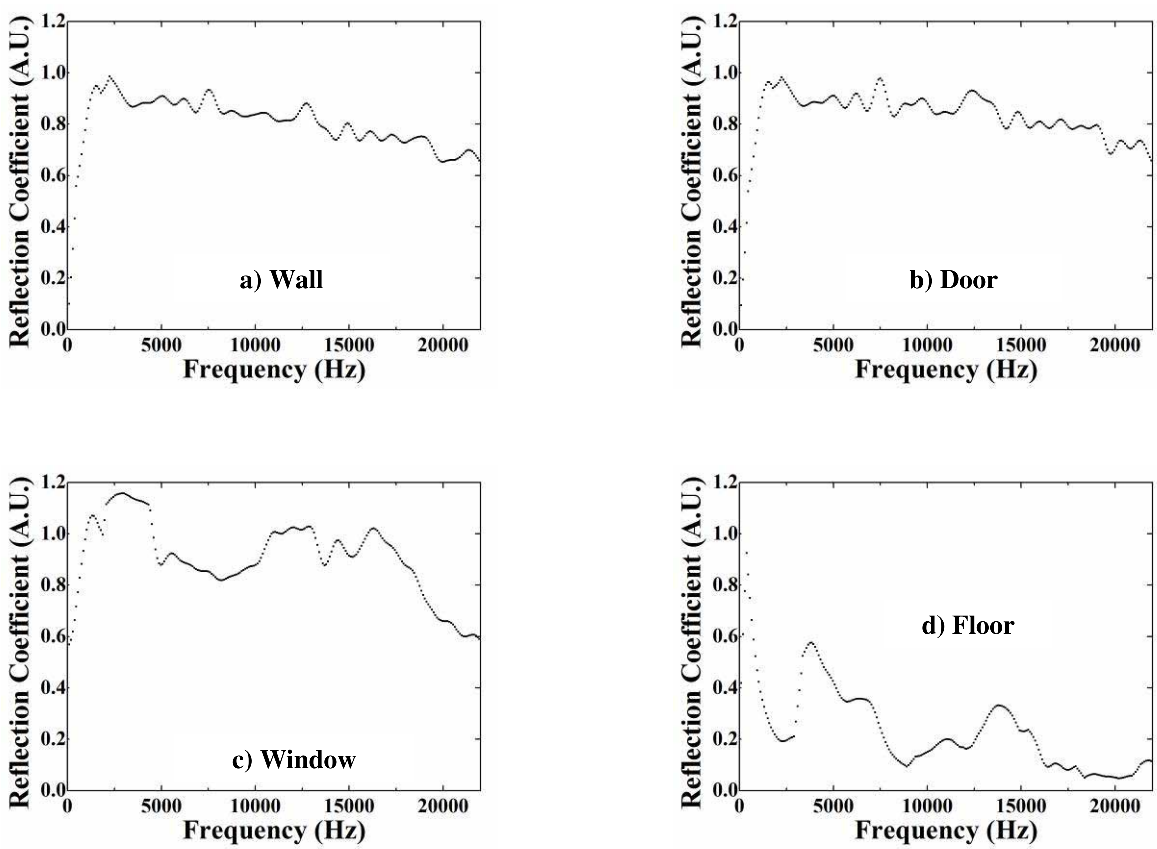

Figure 12: Measured room surface reflection coefficients as a function of frequency.

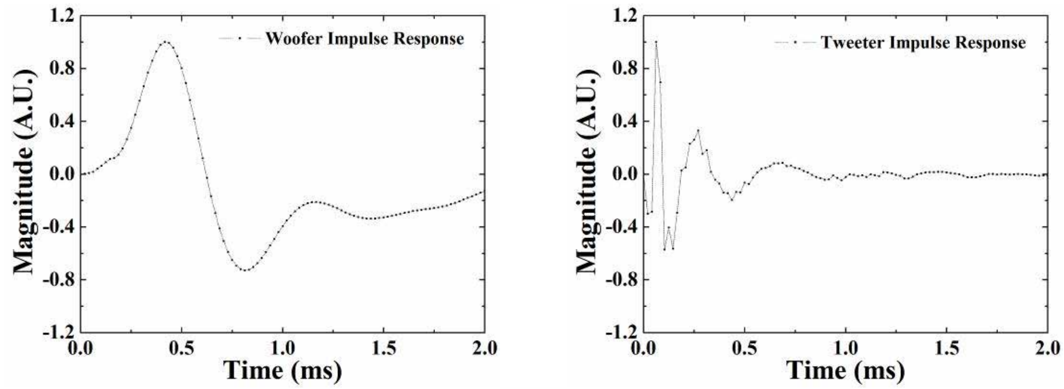

Figure 13: On-axis woofer and tweeter impulse responses.

Measurement		Image Source Method		Differences		Description		
Delay (ms)	Magnitude (dB)	Delay (ms)	Magnitude (dB)	Delay (ms)	Magnitude (dB)	Sound Source	Sound Reflection	Sound Orientation
2.58	0.0	2.63	0.0	0.05	0	Played by Tweeter	Direct Sound	(1.85°, 84.0°)
2.90	-10.0	2.94	-10.1	0.04	-0.1	Played by Woofer	Direct Sound	(1.85°, 96.0°)
6.06	-6.9	6.06	-7.1	0	-0.2	Played by Tweeter	From wall 3	(0.8°, 87.4°)
6.42	-16.4	6.38	-16.8	-0.04	-0.4	Played by Woofer	From wall 3	(0.8°, 92.6°)
6.71	-29.0	6.65	-28.9	-0.06	0.1	Played by Woofer	From floor	(1.85°, 24.1°)
6.77	-31.5	6.73	-29.8	-0.04	1.7	Played by Tweeter	From floor	(1.85°, 22.2°)
8.33	-23.6	8.33	-27.7	0	-4.1	Played by Tweeter	From ceiling	(1.85°, 161.9°)
9.00	-31.3	9.10	-29.2	0.1	2.1	Played by Woofer	From ceiling	(1.85°, 163.0°)
9.12	-23.6	9.19	-26.0	0.07	-2.4	Played by Tweeter	From wall 2	(286.1°, 88.3°)
9.71	-28.0	9.71	-29.7	0	-1.7	Played by Woofer	From wall 2	(286.1°, 91.7°)
9.33	-24.4	9.40	-23.7	0.07	0.7	Played by Tweeter	From wall 4	(74.2°, 88.4°)
9.75	-28.6	9.75	-29.2	0	-0.6	Played by Woofer	From wall 4	(74.2°, 91.6°)
13.71	-39.1	13.69	-33.3	-0.02	5.8	Played by Tweeter	From wall 1	(179.7°, 88.9°)
13.96	-30.7	13.96	-32.3	0	-1.6	Played by Woofer	From wall 1	(179.7°, 91.1°)

Table 3: Comparison between improved modeled and measured room impulse responses (the speaker is modeled with separate woofer and tweeter contributions)

The extracted values of the modeled and measured room impulse response corresponded quite well (see Fig. 14 and Table 3). The discrepancies are reduced to be within 5 dB, compared to 20 dB seen previously.

4. CONCLUSION

The principal goal of this work was to verify that the discrepancies observed between measured room impulse responses and simulations using the popular image source method are largely attributable to the simplifications used in the simulation. The results verify this goal. For a room impulse response modeled using the image method, if the source and receiver are assumed to be omni-directional and the room surfaces are assumed to be infinite rigid boundaries, there is a large discrepancy between modeled and measured room impulse responses, indicating that the simple model is insufficient. The match between modeled and measured impulse responses can be improved if we include real measurements of speaker polar response, microphone polar response, and room surface reflection coefficient in image method. Thus, the goal to obtain a better understanding of the differences between simulation and measurement in small rooms is verified. The sources of the remaining discrepancies between measurement and model are now being studied. They may be due to the scattering effect of the room surfaces, for example.

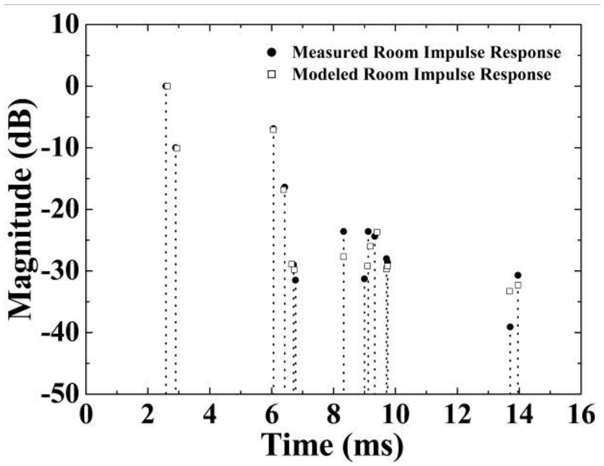

Figure 14: Comparison between direct sound and six first-order reflections, for both woofer and tweeter, of modeled and measured room impulse responses.

Further work is planned to perform similar verification experiments using frequency-domain simulations, more source-receiver, positions, and a wider variety of surface types.

Additional research is also currently being conducted to address the following three questions. First, besides the above-mentioned measured parameters, are there any other parameters that should be measured to adequately model room impulse responses? Second, given these measured parameters, is the image method a sufficiently accurate model for small room impulse responses, or would another approach such as a digital waveguide mesh be better? Third, according to the measured impulse responses, what level of objective or subjective quality of modeled impulse responses can be obtained?

5. ACKNOWLEDGEMENTS

This work was supported in part by Advanced Acoustic Concepts, Inc., and by a Benjamin Ph.D. Fellowship at Montana State University-Bozeman.

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