Identifying Impedances of Walls Using First and Second Order Echoes

Helena Peić Tukuljac, Hervé Lissek, Pierre Vandergheynst
École polytechnique fédérale de Lausanne (EPFL), CH-1015 Lausanne, Switzerland
{helena.peictukuljac, herve.lissek, pierre.vandergheynst}@epfl.ch

***Abstract*—Control of the sound field in a room requires detailed characterization of acoustical properties of the room, such as room shape and physical characteristics of the walls. We will focus on the estimation of the acoustic impedances of the walls in a room using room impulse response and image-source model which eliminates the limitation of rectangular room geometry as well as narrow-band low-frequency requirement imposed by many solutions. Previous solutions rely mainly on the finite difference methods that become quite cumbersome when we move to 3D.**

I. Introduction

Recently there have been multiple attempts to estimate the acoustic impedances of the walls in a room. Most of the approaches rely on the well-known finite difference time domain (FDTD) method [1], [2]. A uniform grid driven by the Courant–Friedrichs–Lewy (CFL) condition is usually introduced into the room. On the defined grid the points away from the walls are modeled by the wave equation and the points on the walls are modeled according to the Mur's boundary condition.

According to the theory of sampling the planacoustic function [3] and the theory of Courant-Friedrich-Lewy [4], for capturing the audiable part of the sound (up to $20kHz$), we need to sample with at least $40kHz$ [5] which gives a spatial step of $1.5cm$, for a uniform rectangular grid. In that case we would need almost 18 million sampling points for a small room (e.g. $3m \times 4m \times 5m$)!

Instead of focusing on the grid, we would like to exploit the sparsity that exists in the image-source model. As proposed by Dokmanić et al. [6] and optimized by Jager et al. [7], geometry of an unknown room and the position of the first and second order image sources [8] can be estimated in 2.43s. After this processing we have the position of the first and second order echoes in the echogram which we will use to further exploit the underlying data.

II. Origin of Sparsity and Data Retrieval

In order to measure the room impulse response, there are two main approaches [9]:

1) time domain: by using a Dirac pulse (usually time-stretched)
2) frequency domain: by using a white noise signal or sine sweeps (which is the best method according to [9]).

The second approach relies on the fact that the spectrum is flat (sine sweep has a spectrum similar to band-passed white noise), so the inverse Fourier transform results in a Dirac pulse. As shown in Figure 1 we can decompose our impulse response over frequencies and compute the losses for different frequencies and different points on the wall just from one RIR.

When choosing the size of the frequency subband that we analyze at once, we should exploit the fact that the absorption power in the low frequency range is much smaller than for the high frequencies.

The RIR recorder at the receiver $\mathbf{r}_j$ can be approximated as a weighted sum of delayed Dirac pulses [10]:

$$h_j(t) = \sum_k a_k \delta(t - \tau_{j,k}). \quad (1)$$

This signal is sparse and has finite number of degrees of freedom [11]: the delays $\tau_{j,k}$ depend on the distance between the receiver and the real or the image sources, and the weights a_k depend on the impedances of the walls at the points where the sound was reflected. The spreading losses are usually neglected for small size rooms.

The acoustic wall impedance at point $\mathbf{x}$ depends on the frequency of the signal as well as on the incidence angle θ:

$$Z(\mathbf{x}, \omega) = \frac{Z_{\text{air}}}{\cos(\theta(\mathbf{x}))} \left(\frac{p_{\mathbf{x}}(\omega)}{p_{\text{dir}}(\omega)} \right)^2. \quad (2)$$

III. Remote Sensing and Restrictions

In Figure 2. we see an example of covering walls of a rectangular room in 2D with 1^{st} order reflections. In the Figure 3. we illustrate the requirements for using higher order reflections as source of information. We need to enforce that the intersection of the line that connects 1^{st} order image source and one receiver, and the line that connects the 2^{nd} order image sources and some other receiver, lays on the surface of the wall defined by the wall normal. So the goal is to define a subsampling matrix that provides the greatest amount of data with the smallest number of receivers.

There exists a frequency restriction: Due to the fact that the first and second order reflections lay in the first 0.1s of room impulse response, we can not cover the low frequencies, because of the misalignment of the peaks. We need to have approximate overlap of the position of the peaks for different frequencies.

Because of the smoothness of the acoustic impedance curve, we do not have to have a dense sampling in the term of frequencies, before applying some curve fitting algorithm [12].

IV. Contribution and Future Directions

Using this approach we can easily explore the acoustic impedance of the walls for higher frequencies and for a non-rectangular room geometry, which was not the case with the FDTD method.

Future work will rely on the use of SLAM (Source Localization and Mapping) [13] and unlabeled sensing [14], which correspond to the reconstruction of the subsampling and permutation matrices. Recent papers explore the potential of these approaches by without the need for a combinatorial approach.

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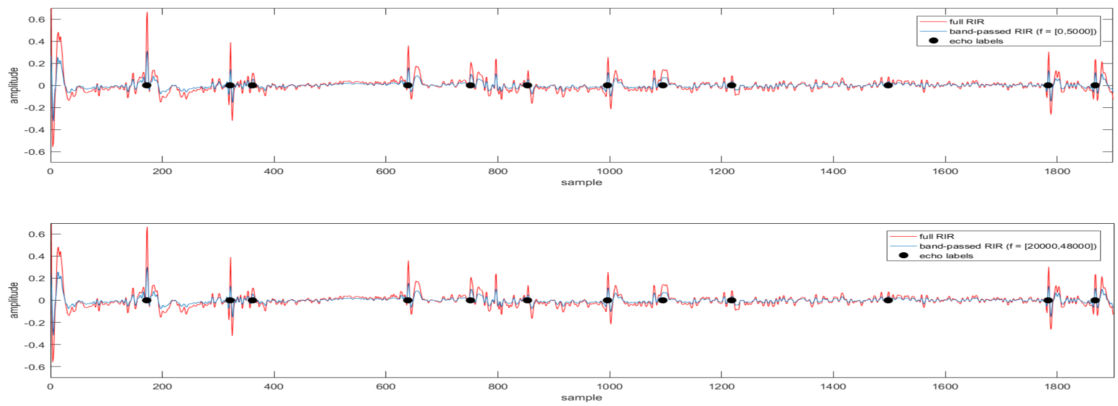

Fig. 1. A zoomed-in part of the room impulse response. By using band-pass filters we can analyze the impedance of the walls for different frequencies from the difference of the height of the peaks relative to the height of the signal from the direct path. In this example the sampling frequency was $96kHz$. This results in common support Dirac pulses, meaning that the position of the peaks for different frequencies is the same, but the heights differ, which enables us to use the available mathematical frameworks like the theory of Finite Rate of Innovation [11]. Here the first and the second order echoes are labeled.

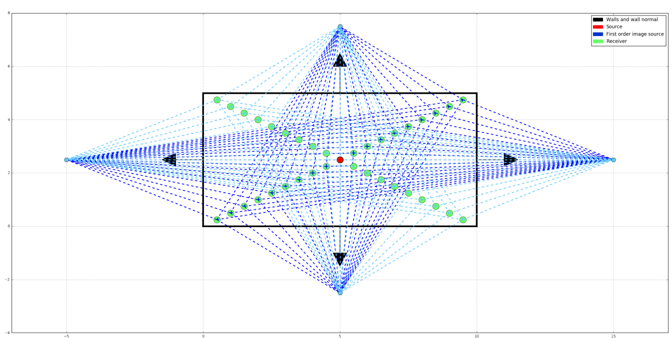

Fig. 2. By placing a receiver array along the diagonals (or by moving receivers along the same path), we have good coverage of sampling the walls.

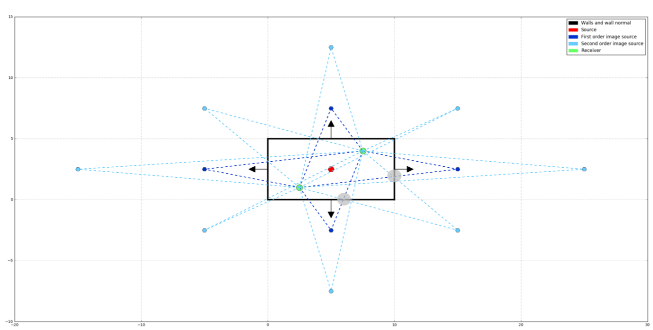

Fig. 3. In order to exploit the information form the second order reflections, we need to enforce the overlap between the first and second order reflections of different receivers (marked by grey circles).

Algorithm 1 Algorithm for Reconstructing the Acoustic Impedances of Walls

Input: set of wall normals $\mathcal{N} = \{\mathbf{n}_i\}_{i=0}^{N-1}$, position of the source $\mathbf{s}$
Output: $Z(\mathbf{x}, \omega)$ acoustic impedances for different parts of the walls

Initialisation :
compute the positions of the 1st and 2nd order image sources $\mathcal{S}' = \{\tilde{\mathbf{s}}_i\}_{i=0}^{S'-1}$ and $\mathcal{S}" = \{\tilde{\mathbf{s}}_{i,j}\}_{i,j=0}^{S"-1}$
Optimization :
In order to make 2^{nd} order reflections useful: $(\mathbf{r}_m - \tilde{\mathbf{s}}_\mathbf{i}) \cap (\mathbf{r}_n - \tilde{\mathbf{s}}_{\mathbf{i},\mathbf{j}}) \in \mathcal{W}_{\mathbf{n}_i}$, optimize the positions of the receivers $\mathcal{R} = \{\mathbf{r}_i\}_{i=0}^{R-1}$, e.g. design the subsampling matrix
for $i = 1$ to M **do**
  - record the room impulse response, label the echoes and assign the echoes to points on the walls
  - decompose the room response over frequencies with band-pass filters
  - reconstruct the impedance curves for multiple points on the wall from each impulse response, taking into account the incidence angle
end for
return $Z(\mathbf{x}, \omega)$

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