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TIME-VARYING FILTER REPRESENTATION OF ACOUSTIC TRANSMISSION THROUGH PANELS UNDER MOVING SOURCE EXCITATION

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ABSTRACT

This paper provides a filter design approach for efficient time-domain synthesis of transmitted sounds from moving sources through infinite panels. The moving source problem can be found in vehicle pass-by noise which is a major source of broadband noise in urban environments (indoor and outdoor). The audio rendering of the filtering effect from insulating panels on exterior noise is an outstanding challenge, specially in the case of moving sources, such as vehicles. The present work is structured as follows. First, a minimum phase filter is designed to represent measured or simulated sound transmission through a panel. Then, a moving source signal is synthesized from static multi-microphone array measurements of a vehicle in operational conditions in a semi-anechoic room. The resulting receiver signal is obtained from a time-varying transfer path approach. Psychoacoustic evaluation can be performed in order to assess the insulation effect from a perceptual perspective, with a focus on early design stages. The methodology is applied to a commercial vehicle engine noise and a comparison of the measured and analytical formulation with different panels is discussed.

1. INTRODUCTION

The goal of this paper is to build digital audio filters for efficiently representing the sound pressure transmitted through an insulation panel from a moving source excitation. The filters are designed from measured reduction index curves and from analytical models. Moreover, the effect of the angle-dependent directivity of the sound transmission loss is investigated. The technique presented here allows for the correlation of the insulation material properties with the annoyance metrics perceived by a listener. In previous work [1], a correlation between material properties and perceived annoyance was observed for static sources.

First, an introduction of the fundamental aspects of airborne sound insulation by panels is presented; then, the filters are designed for the sound propagation and for the panel insulation considering both a diffuse and the angle-dependent case.

2. METHODOLOGY

2.1 Transmission loss filter

A single solid panel freely suspended is investigated both from measurements and from an analytical perspective. The acoustic transmission is written in decibel scale as

$$\Delta L_{TL}(\theta, \omega) = -10 \log \tau(\theta, \omega), \quad (1)$$

where $\tau(\theta, \omega)$ is the acoustic transmission loss coefficient for an angle of incidence θ in radians and ω is the circular frequency in rad/s.

The values of $\tau(\theta, \omega)$ can be computed from the ratio of the transmitted acoustic power to the acoustic incident power. In this paper, the analytical formulation follows [2] while the measured reduction index obtained from Table B.2 in the ISO 12354-1 [3] is used as a reference measurement. The façade considered here is simplified as a finite single panel with no flanking elements.

The filters are designed from the measured or modeled reduction index magnitudes by first converting them into a minimum-phase frequency response following the cepstral method [4]. To account for the angle of incidence in the transmission loss, the transmission loss is estimated for 19 discrete points corresponding to an array of target microphone positions. The resulting filter is a superposition of the angle-dependent transmission loss with the diffuse field transmission loss, for each position.

2.2 Propagation filter

According to the ISO 362-3 [5], the noise transfer functions are measured at discretized target microphone positions 7.5 m away from the source. The procedure consists of first propagating the source signal using measured noise transfer functions (NTFs), then combining the propagated signals using the time-varying IIR filters and then applying the transmission loss filter. The decomposition of the NTF is given by

$$H(j\omega) = |H_{\min}(j\omega)H_{ap}(j\omega)|e^{j\phi_{\min}(j\omega)}e^{j\phi_{ap}(j\omega)}, \quad (2)$$

where $\phi_{\min}(j\omega)$ denotes the minimum phase, $\phi_{ap}(j\omega)$ denotes the excess phase, $|H_{\min}(j\omega)|$ is the minimum-phase

magnitude and the $|H_{ap}(j\omega)| = 1$ is the all-pass magnitude.

The IIR filter with time-varying coefficients is implemented by solving the recursive expression

$$y(n) = \sum_{i=0}^M b_i(n)x(n-i) - \sum_{k=1}^N a_k(n)y(n-k), \quad (3)$$

where M is the feedforward filter order and N is the feedback filter order. The b_i and a_j coefficients are linearly interpolated from the estimated IIR filter coefficients. The final step in the synthesis is the inclusion of the Doppler effect as a propagation time delay, which can be inferred from the slope of the all-pass excess phase.

3. RESULTS

In this section, the filters designed from the measured reduction index and analytical transmission loss are shown. The source signal in this example comes from a motorcycle noise in constant speed, which is driven by a slick chassis dyno in a semi-anechoic chamber.

Figure 1 shows the filter obtained from the measured reduction index and analytical transmission loss for the concrete panel with $h = 120$ mm and $h = 260$ mm. The filters are constructed with an 8^{th} order scheme.

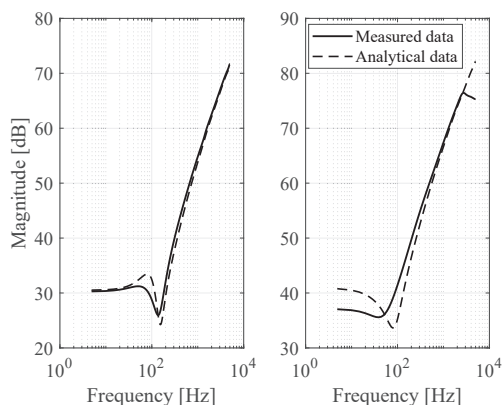


Figure 1. IIR filter obtained from the measured reduction index and from the analytical calculations of the diffuse field transmission loss coefficient for (left) $h = 120$ mm and (right) $h = 260$ mm.

Both filters show a good match, specifically for frequencies higher than 200 Hz. For frequencies close to the coincidence dip, the two filters deviate within a range of 2 dB for $h = 120$ mm and 5 dB for $h = 260$ mm case.

Figure 2 shows the sound pressure level (SPL) of the filtered pass-by noise source for both measured and analytical filters. As expected, the attenuation for the large thickness façade is around 15 dB higher than for the thinner façade due to the additional mass.

4. CONCLUSION

This paper proposed an audio synthesis technique which includes a moving source and a transmission loss panel.

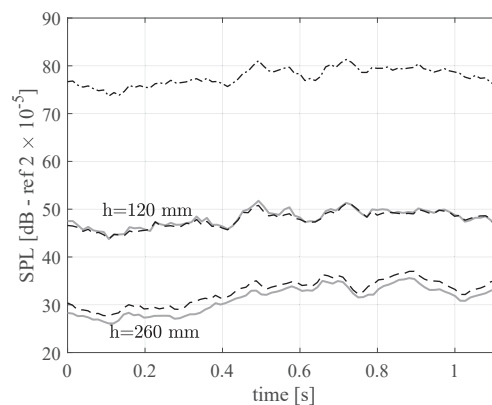


Figure 2. Sound pressure level of the propagated output in dash lines (---) and the response after filtering by the concrete panel with $h = 120$ mm and $h = 260$ mm, the (---) represent the filtered signal by the analytical transmission loss and the gray lines (—) represent the SPL filtered by the measured reduction index curves.

The filters were designed and implemented following analytical models and measured data. Results showed that both filtering strategies can produce realistic results and accurately capture the reduce sound pressure level. The angle dependent transmission loss filter using the same methodology can be performed and showed promising results.

5. ACKNOWLEDGMENTS

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