Low-Rank Multi-Channel Fast Convolution for Real-Time Auralization of Church Acoustics

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***Abstract*—Interactive auralization of room acoustics is a key research tool for investigating the impact of performance space acoustics on music practice. In the context of early music, particularly plainchant and polyphony, the historically relevant performance spaces are usually churches. The interactive auralization of church acoustics is highly challenging due to the long reverberation times involved, which conflict with the real-time, low-latency requirements of interactive systems. This paper investigates the use of low-rank compression with partially invariant transformation to achieve low-latency convolution in real-time multi-channel auralization systems. In previous work, we have demonstrated that low-rank joint compression of Room Impulse Responses (RIRs) measured in the same room facilitates low-latency convolution while preserving the acoustic properties of the RIRs. This study expands on the method to achieve low-latency convolution for auralization filters derived from a space-time analysis of RIR measurements. A case study of the interactive auralization of the 16th-century Nassau Chapel in Brussels is presented to demonstrate how significant computational savings can be achieved while preserving auralization quality.**

***Index Terms*—auralization, low-rank approximation, real-time processing**

I. Introduction

Interactive auralization of room acoustics is an essential tool for researchers and practitioners in the field of music performance, as it allows them to investigate the impact of different acoustic environments on music performance practice [1]–[4]. In the context of early music, particularly plainchant and polyphony, the historically relevant performance spaces are typically churches or large chapels [5]. Real-time auralization of the acoustics of such large spaces presents unique challenges due to their long reverberation time resulting in substantially sized auralization filters that need to be processed under real-time conditions. Therefore, the development of efficient and accurate methods for real-time auralization of early music performance spaces is an important area of research.

For large auralization filters, traditional time-domain convolution is too computationally expensive to achieve real-time auralization, even with modern hardware [6]. This limitation has led to the development of alternative approaches, such as a hybrid convolution approach that combines convolutional reverberation with a Feedback Delay Network (FDN) to achieve low-latency auralization [7]. However, modeling late reverberation using FDNs can cause spectral coloration, which is undesirable for accurate auralization [8]. Another approach, using block-based Fast Fourier Transform (FFT) convolutions, incurs high processing latency due to buffering of the input signal. Partition-based convolution algorithms were proposed to address this limitation by spreading the computational load over time [9], reducing the latency of the convolution process. However, when dealing with filters of considerable length, these methods require significant memory and computational resources, rendering their application infeasible under limited hardware constraints. Another way to minimize processing latency is to truncate the auralization filters until the filtering can be performed within the desired time frame [10]. This approach, however, can lead to a loss of accuracy in the late reverberation tail of the auralized RIR.

In this paper, we investigate the use of low-rank compression with partially invariant transformation to achieve low-latency convolution in real-time multi-channel auralization systems. This approach builds on previous work that has demonstrated the effectiveness of low-rank joint compression of Room Impulse Responses (RIRs) measured in the same room for facilitating low-latency convolution while retaining the acoustic properties of the RIRs [11], [12]. In this study, we extend the method presented in [12] to achieve low-latency convolution for auralization filters derived from a space-time analysis of RIR measurements. This approach allows for the preservation of the spatial and temporal characteristics of the acoustic environment, which is essential to achieve accurate auralization while heavily reducing the computational complexity compared to time-domain convolution without introducing latency. To illustrate the effectiveness of this approach, we present a case study on the interactive auralization of the 16th-century Nassau Chapel in Brussels.

The paper is structured as follows: Section II provides a brief overview of the method used to achieve low-latency time-domain convolution in multi-channel RIRs. Section III presents the use of low-rank multi-channel fast convolution for auralization filters derived from a space-time analysis of RIR measurements. Section IV describes a case study on interactive auralization of the acoustics of the 16th-century Nassau Chapel in Brussels. Finally, Section V concludes the paper and outlines future research directions.

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II. Low-Rank Multi-Channel Compression and Fast Convolution

In this section, the method for achieving low-latency time-domain convolution using low-rank compression of multi-channel RIRs with partially invariant transformation is briefly summarized. For a more detailed description of the method, the reader is referred to [12].

A. Low-Rank Compression of Multi-Channel RIRs

The low-rank compression of RIRs is based on the observation that RIRs, measured in the same room, share common spatial and temporal characteristics. This allows for the joint compression of the RIRs using low-rank approximation techniques.

Consider a discrete-time RIR $h(k)$, where $k = 1, 2, \ldots, n_h$ is the time index. The RIR can be reshaped into a matrix $\mathbf{H} \in \mathbb{R}^{r \times c}$ as,

$$\mathbf{H} = \begin{bmatrix} h(1) & h(r+1) & \ldots & h((c-1)r+1) \\ \vdots & \vdots & & \vdots \\ h(r) & h(2r) & \ldots & h(cr) \end{bmatrix}, \quad (1)$$

where it is assumed that $n_h = rc$ and $r, c \in \mathbb{N}$. It is well established that matrix $\mathbf{H}$ possesses a low-rank structure [13], [14]. This property can be leveraged for compressing the RIRs using low-rank approximation techniques [15], as well as for efficient time-domain convolution [11], [16]. The low-rank approximation of a set of matricized RIRs $\mathbf{H}_j \in \mathbb{R}^{r \times c}$ for $j = 1, 2, \ldots, n$ can be found by applying the Generalized Low-Rank Approximation of Matrices (GLRAM) [17], for which the low-rank approximation is obtained by solving the following optimization problem,

$$\begin{aligned} \min_{\mathbf{L},\mathbf{R},\mathbf{D}_j} \quad & \sum_{j=1}^{n} \|\mathbf{H}_j - \mathbf{L}\mathbf{D}_j\mathbf{R}^T\|_F^2, \quad (2) \\ \text{s.t.} \quad & \mathbf{L}^T\mathbf{L} = \mathbf{I}_{\ell_1}, \mathbf{R}^T\mathbf{R} = \mathbf{I}_{\ell_2} \end{aligned}$$

where $\mathbf{L} \in \mathbb{R}^{r \times \ell_1}$, $\mathbf{R} \in \mathbb{R}^{c \times \ell_2}$, and $\mathbf{D}_j \in \mathbb{R}^{\ell_1 \times \ell_2}$. The low-rank approximation of each RIR $\mathbf{H}_j$ is then given by $\hat{\mathbf{H}}_j = \mathbf{L}\mathbf{D}_j\mathbf{R}^T$, where matrices $\mathbf{L}$ and $\mathbf{R}$ are common to all RIRs in the set. An algorithm for solving this optimization problem for multi-channel RIRs is presented in [12].

B. Fast Convolution with Partially Invariant Transformation

Consider a set of closely spaced RIRs, $\mathcal{H}$, where the cardinality of the set is denoted by $|\mathcal{H}| = n$. When the low-rank approximation is performed on individual reshaped RIRs $\mathbf{H}_j$ for $j = 1, 2, \ldots, n$, the low-rank approximated filter can be formed from $\mathbf{L}_j$, $\mathbf{D}_j$, and $\mathbf{R}_j$ as $\hat{\mathbf{H}}_j = \mathbf{L}_j\mathbf{D}_j\mathbf{R}_j^T$. In previous work [12], it was observed that the low-rank structure appears to be partially invariant across all low-rank approximations of reshaped RIRs in the set $\mathcal{H}$. It could be seen that matrices $\mathbf{R}_j$ show a high degree of similarity, whereas matrices $\mathbf{L}_j$ and $\mathbf{D}_j$ exhibit a high degree of variability across the low-rank approximated RIRs in the set $\mathcal{H}$. This observation can be leveraged for achieving low-latency time-domain convolution by first extracting the matrix $\mathbf{R}$ common to all low-rank approximated RIRs in the set $\mathcal{H}$, and calculating the matrix $\mathbf{W}_j$ unique to each $\hat{\mathbf{H}}_j$ that captures the variability of $\mathbf{L}_j$ and $\mathbf{D}_j$. This matrix $\mathbf{W}_j$ is calculated as

$$\mathbf{W}_j = \mathbf{L}_j\mathbf{D}_j = \mathbf{L}_j\mathbf{L}_j^T\mathbf{H}_j\mathbf{R}, \quad (3)$$

where $\mathbf{L}_j$ is computed from matrix $\mathbf{R}$ and the reshaped RIR $\mathbf{H}_j$ [12]. The low-rank approximated RIRs $\hat{\mathbf{H}}_j$ can then be expressed as

$$\hat{\mathbf{H}}_j = \mathbf{W}_j\mathbf{R}^T. \quad (4)$$

The values ℓ_1 and ℓ_2 determine how much information is retained in the compressed RIRs. These values are chosen based on the desired trade-off between computational storage and accuracy of the low-rank approximation. We can define the compression rate for the set of low-rank approximated RIRs $\hat{\mathcal{H}}$, referring to the reduction in computational storage as compared to the original set of RIRs $\mathcal{H}$, as

$$C(\hat{\mathcal{H}}) = 1 - \frac{nr\ell_2 + c\ell_2}{nn_h}. \quad (5)$$

The multi-channel convolution of an input signal $\mathbf{x} \in \mathbb{R}^{n_x}$ with a set of RIRs $\mathbf{h}_j \in \mathbb{R}^{n_h}$ for $j = 1, 2, \ldots, n$, produces the output signals $\mathbf{y}_j \in \mathbb{R}^{n_y}$, where $n_y = n_x + n_h - 1$. When n_h and n are large, time-domain convolution can become computationally expensive, leading to high processing latency. However, when the filters $\mathbf{h}_j$ exhibit low-rank properties, it is possible to approximate the convolution as a set of filtering operations with considerably shorter filters. These shorter filters correspond to the columns of $\mathbf{W}_j$ and $\mathbf{R}$. Moreover, as $\mathbf{R}$ is common among all RIRs in the set $\mathcal{H}$, the latter filtering operation needs to be computed only once for all RIRs [12], resulting in a significant reduction in computational complexity.

We can compare the computational complexity of traditional time-domain convolution to low-rank approximated convolution by quantifying the required number of multiply-accumulate (MAC) operations for both methods. Traditional time-domain convolution requires a total of nn_yn_h MAC operations, whereas the proposed low-rank approximated convolution requires $nn_yr\ell_2 + (n_x + (c-1)r)\ell_2(\lfloor \frac{n_x}{r} \rfloor + 1)$ MAC operations. For $n \gg 1$ and $n_h \geq n_x$, this yields a computational complexity reduction by a factor of $\approx \frac{c}{\ell_2}$ [12]. The assumption $n_h \geq n_x$ is reasonable in the context of real-time convolution, as the input signal $\mathbf{x}$ is typically processed online in frames of n_x samples. To ensure low-latency processing, n_x is chosen such that the processing time lies within the desired time window n_x/f_s, where f_s represents the sampling rate of the system in Hz. In powerful systems, $n_x = 1$ can be chosen, which corresponds to processing the input signal on a sample-by-sample basis, incurring no additional latency.

III. Low-Rank Multi-Channel Fast Convolution for Auralization Filters

In this section, low-rank multi-channel fast convolution is applied to auralization filters derived from a space-time analysis of RIR measurements to enable the real-time reproduction of acoustics in multi-channel auralization systems.

A. Real-Time Interactive Auralization Systems

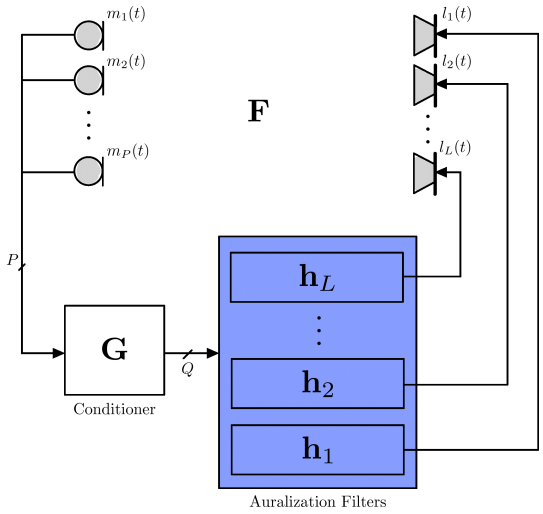


Fig. 1. Block diagram of a real-time interactive auralization system. The microphone signals are pre-processed and convolved with a set of auralization filters $\mathcal{H}_{\text{aur}}$. The filtered signals are played back through the loudspeakers in real-time, creating an immersive reproduction of the target acoustics. For simplicity, this diagram does not include acoustic feedback cancellation.

Real-time interactive auralization systems are designed to provide an immersive and interactive reproduction of the acoustics of a target space in real-time. Consider a multi-channel auralization system with L loudspeakers, where the input signal is captured using P microphones. The input signals are first conditioned using a pre-processing stage $\mathbf{G}$, which may include different weighting or equalization for each microphone signal [18]. The conditioned signals are then convolved with a set of auralization filters $\mathcal{H}_{\text{aur}}$, one for each loudspeaker in the reproduction setup, i.e., $|\mathcal{H}_{\text{aur}}| = L$. The filtered signals are played back through the loudspeakers in real-time, creating an immersive reproduction of the acoustics. Fig. 1 shows a simplified block diagram of the system.

B. Auralization Filters

The set of auralization filters $\mathcal{H}_{\text{aur}}$ are derived by performing a space-time analysis on measured multi-channel RIRs of the target space. One such space-time analysis method is the Spatial Decomposition Method (SDM) [19], which decomposes measured multi-channel RIRs into a set of image sources localized in both 3D space and time. The auralization filters are derived by mapping each localized image source to a single loudspeaker in the reproduction setup. In this paper, this mapping is done based on the closest direction between each image source position and loudspeaker. This is known as Nearest Neighbor Synthesis [20]. Each resulting auralization filter $h_l \in \mathcal{H}_{\text{aur}}$ consists of a collection of image sources that are spread out over time. Moreover, the number of image sources in each auralization filter is generally much smaller than the number of filter taps required to represent the auralization filters in the time domain. These properties make auralization filters good candidates for low-rank approximation.

C. Low-Rank Fast Convolution of Auralization Filters

The low-rank approximation of auralization filters is performed by reshaping each filter h_l into a matrix $\mathbf{H}_l \in \mathbb{R}^{r \times c}$ and applying the multi-channel low-rank compression method detailed in Section II to the entire set of filters. This compression is done offline, resulting in a set of low-rank compressed auralization filters, denoted $\hat{\mathcal{H}}_{\text{aur}}$, which are then stored in memory. By controlling the parameters ℓ_1 and ℓ_2, as discussed in Section II, the computational complexity of time-domain convolution between an input signal and $\hat{\mathcal{H}}_{\text{aur}}$ can be significantly reduced.

IV. Case Study: Auralization of the Nassau Chapel's Acoustics

This section presents a case study of the real-time auralization of the acoustics of the Nassau Chapel, a historic venue located in Brussels, Belgium. We compare the effect of low-rank compressed auralization filters on the auralization performance by evaluating objective acoustic parameters of the original and low-rank compressed auralization filters. We also evaluate the performance of the real-time auralization system in terms of computational complexity and auralization quality.

The real-time interactive auralization system that was used in this paper aims to provide early music performers with an interactive tool to experience the acoustics of various historically relevant venues in real time, allowing them to explore the relationship between early music performances and the acoustics for which the composition was intended. The auralization system consists of 28 loudspeakers and four microphones arranged in a 3D configuration. The reproduction room is acoustically treated with two soundproof drapes, resulting in a measured T_{30} of 0.55 seconds. The setup is further explained in detail in [21], and acoustic measurements of the system are available in [22].

A. Nassau Chapel Measurements

The acoustics of the Nassau Chapel were measured using a Vector Intensity Probe GRAS 50VI-1, a microphone-array consisting of six omnidirectional microphones arranged in a 3D Cartesian grid with a 25 mm spacing. The measurement setup consisted of a Genelec 8030C loudspeaker positioned on the chapel's choir loft and the above-mentioned microphone array on the chapel floor. The RIRs were measured using the Exponential Sine Sweep method [25]. The loudspeaker excited the room with a 15-second-long sine sweep, followed by a 10-second silence period. Upon completion of the measurements, the sweeps were deconvolved to obtain the multi-channel RIRs. The measurements were carried out at a sampling rate of 192 kHz. After deconvolution, the RIRs were downsampled to 48 kHz. The auralization filters $\mathbf{h}_l$, one for every loudspeaker $l = 1, 2, \dots L$ in the reproduction setup, were obtained from SDM analysis of the measured multi-channel RIRs.

The Nassau Chapel exhibits an unusually long reverberation time, as seen in Table I, resulting in substantially long auralization filters. This makes auralization filtering computationally expensive and difficult to achieve under real-time conditions. It

TABLE I

OBJECTIVE ACOUSTIC PARAMETERS OF THE NASSAU CHAPEL, REFERENCE AURALIZATION FILTERS, AND LOW-RANK COMPRESSED AURALIZATION FILTERS. PARAMETERS T_{30}, EDT, C_{80}, AND D_{50} WERE AVERAGED IN THE 500 HZ AND 1 KHZ OCTAVE BANDS FOLLOWING ISO:3382 [23], [24].

Measurement	$\ell_1 = \ell_2$	Compression Rate $C(\hat{\mathcal{H}})(\%)$	Computational Reduction (%)	T_{30} (s)	EDT (s)	C_{80} (dB)	D_{50}
Nassau Chapel measurement				6.23	6.10	-6.66	0.12
Reference Auralization filters				6.21	6.42	-6.87	0.12
Low Rank Auralization filters	550	-3.57	-0.01	6.21	6.42	-6.88	0.12
	500	5.84	9.09	6.21	6.42	-6.88	0.12
	450	15.26	18.18	6.21	6.43	-7.13	0.11
	400	24.68	27.27	6.19	6.44	-7.35	0.10
	300	43.51	45.45	5.91	6.45	-7.44	0.10
	200	62.34	63.62	5.02	6.36	-7.15	0.11
	100	81.17	82.82	4.46	5.64	-6.71	0.12
	50	90.58	90.91	4.50	3.84	-5.16	0.16
Just Noticeable Difference (JND)					5%	1	0.05

is therefore necessary to reduce the computational complexity of the auralization filtering while preserving the acoustic properties of the auralization.

B. Low-Rank Compression of Auralization Filters

The number of filter taps n_h required to represent each auralization filter was computed from the reverberation time T_{30}. The Nassau Chapel has a reverberation time of 6.23 seconds, which requires 299,040 filter taps to represent at a sampling rate of $f_s = 48$ kHz. We therefore chose $r = c = 550$ to represent each matricized auralization filter $\mathbf{H}_l$.

The low-rank compression of the auralization filters was done by varying the parameters ℓ_1 and ℓ_2. It was shown in [12] that the choice of ℓ_1 and ℓ_2 can be optimized to achieve the best tradeoff between compression and approximation accuracy. However, for simplicity, we consider $\ell_1 = \ell_2$ and vary the compression by varying the value of $\ell_1 = \ell_2$. The low-rank compression of the auralization filters was performed by varying $\ell_1 = \ell_2$ from 550, 500, 450, 400, 300, 200, 100, and 50 to achieve a reduction in computational complexity of -0.01%, 9.09%, 18.18%, 27.27%, 45.45%, 63.62%, 82.82%, and 90.91%, respectively.

C. Performance Evaluation

1) Objective Acoustic Parameters: To measure the objective acoustic parameters of the low-rank compressed auralization filters, we positioned an omnidirectional microphone in the center of the reproduction setup. The auralization filters were convolved with an Exponential Sine Sweep [25] and played back simultaneously through the loudspeakers. The microphone captured the auralized signals obtained with the original auralization filters, and with the compressed auralization filters for different compression rates. By deconvolving the captured signal, the RIR that models the acoustics reproduced by the auralization filters, including the acoustics of the reproduction room, was obtained.

The objective acoustic parameters of the original and low-rank compressed auralization filters were evaluated in terms of T_{30}, EDT, C_{80}, and D_{50}, and are shown in Table I. The objective acoustic parameters were averaged in the 500 Hz and 1 kHz octave bands according to ISO:3382 [23], [24]. It can be seen that the low-rank compressed auralization filters are able to reproduce the acoustics of the Nassau Chapel with a high degree of accuracy. For a compression rate that results in a computational reduction of 63.62 %, which corresponds to $\ell_1 = \ell_2 = 200$, the parameters EDT, C_{80}, and D_{50} fall within one Just Noticeable Difference (JND) [23]. For compression rates that result in computational reductions higher than 80%, the auralization quality starts to degrade. This is to be expected, since there is not enough information in the low-rank compressed auralization filters to accurately reproduce the acoustics of the Nassau Chapel.

2) Auralization Quality: The auralization quality for the low-rank compressed filters was evaluated by filtering band-pass filtered signals with both the original and low-rank compressed auralization filters. The Normalized Mean Square Error (NMSE) was then calculated to compare the signals filtered with the original and low-rank compressed auralization filters. The NMSE is defined as

$$\text{NMSE} = \frac{1}{L}\sum_{l=1}^{L}\frac{\|\hat{\mathbf{h}}_l * \mathbf{s} - \mathbf{h}_l * \mathbf{s}\|_2^2}{\|\mathbf{h}_l * \mathbf{s}\|_2^2}, \tag{6}$$

where $\mathbf{s}$ is a band-pass filtered signal, $\mathbf{h}_l$ and $\hat{\mathbf{h}}_l$ are the original and low-rank compressed auralization filters, respectively, and L is the number of auralization filters.

The NMSE was computed for various compression rates of the low-rank approximated auralization filters. The signal $\mathbf{s}$ was band-pass filtered into frequency bands corresponding to bass (20 - 250 Hz), midrange (250 - 2 kHz), upper midrange (2 - 4 kHz), and high-frequency (4 - 20 kHz) content. The results are shown in Fig. 2. It can be seen that the NMSE is low for small compression rates, indicating agreement between the signals filtered with the original and low-rank compressed auralization filters. However, as the compression rate increases, the NMSE also increases, indicating that the spectra of the reproduced acoustics are affected by low-rank compression. This is particularly noticeable for high-frequency content, which seems to be more affected by low-rank compression than low-frequency content.

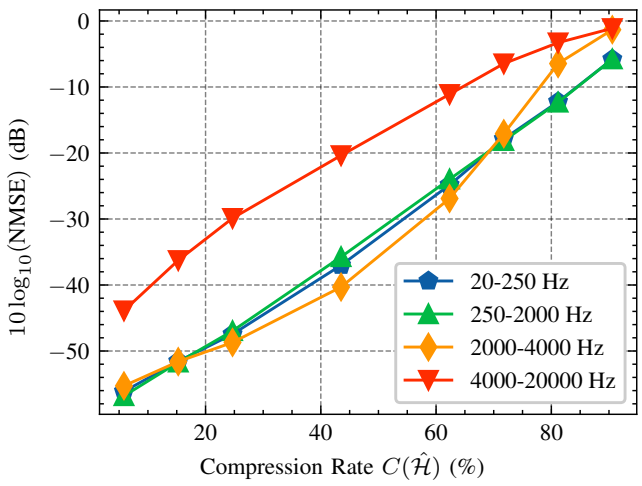


Fig. 2. Normalized Mean Square Error (NMSE) for various compression rates of the low-rank compressed auralization filters. The NMSE was computed for band-pass filtered signals of frequency bands corresponding to bass (20 - 250 Hz), midrange (250 - 2 kHz), upper midrange (2 - 4 kHz), and high-frequency (4 - 20 kHz) content.

V. Conclusions

This paper presented an adaptation of the low-rank multi-channel fast convolution method to achieve low-latency convolution in auralization filters derived from a space-time analysis of multi-channel RIR measurements. The method was applied to real-time interactive auralization of room acoustics, and demonstrated through a case study on interactive auralization of the 16th-century Nassau Chapel in Brussels, Belgium. The study assessed the auralization quality at different compression rates by analyzing objective acoustic parameters and by calculating the NMSE between auralized signals for different frequency bands. The findings showed a significant reduction in computational complexity of up to 63.62%, while maintaining a JND of less than one for key acoustic parameters. The case study revealed that low-rank compression of auralization filters has a greater negative impact on the auralization quality of high-frequency content than on low-frequency content, suggesting a need for further investigation into the perceived quality of auralization using compressed filters. Future research could focus on performing perceptual evaluations of real-time auralization with compressed auralization filters. This study highlights the potential of multi-channel low-rank time-domain convolution for enabling real-time auralization of large-room acoustics, which has important implications for research and practice in the field of music performance and archaeoacoustics.

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