Corrective Impulse Response Generation and Real Time Convolution for Spatial Reconstruction

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Abstract

Applying a target hall impulse response directly in a live venue does not reproduce that hall consistently because the processed sound is filtered again by the current venue and sound reinforcement path. This paper presents a corrective impulse response method that estimates a regularized inverse filter from measured current venue and target hall responses. The implementation aligns impulse responses with GCC-PHAT, gates the relevant response, limits correction to 125 Hz to 4 kHz, smooths the complex response, and constrains gain and peak level before exporting a channel-specific finite impulse response filter. A six-channel Max/MSP system applies these filters to the initial response, while Convology XT adds the target hall reverberation tail from approximately 200 ms onward. The system was tested in the DIMA Jiseong Hall using the Sir Jack Lyons Concert Hall as the target. At the fourth-row center measurement position, the mean spectral error across six surround channels decreased from 2.69 dB to 2.24 dB within the initial 0 to 40 ms window and the 125 Hz to 4 kHz band. This first-stage comparison does not demonstrate physical sound-field replication across the audience or perceptual equivalence to the target hall.

Keywords impulse response room equalization inverse filtering spatial audio convolution live sound

1 Introduction and Motivation

The acoustic impression of a performance depends on more than the signal radiated by the loudspeakers. Reverberation time, reflection pattern, spectral balance, and spatial distribution all shape what listeners perceive. A convolution reverb can reproduce the measured response of a room in a studio signal path, but applying the same target impulse response in different live venues does not produce the same result. The convolved signal must still travel through the installed loudspeaker system and the receiving room, so the current venue response is superimposed on the target response.

This problem was observed while designing live performances in venues with different sizes and reverberation characteristics. The research question was therefore narrower than complete room reproduction: can a live venue be corrected so that its initial response and reverberant impression move closer to those of another hall? The study treats the venue, loudspeaker, microphone position, and signal path as one measured system. Rather than sending the target hall impulse response directly to every channel, it derives a corrective finite impulse response filter for each measured path.

The current venue was the approximately 200 seat DIMA Jiseong Hall, measured at a 500 Hz T30 of 0.39 s. The target was the approximately 350 seat Sir Jack Lyons Concert Hall at the University of York, whose OpenAIR data report a 500 Hz T30 of 1.89 s [10]. The contrast provided a practical test of whether a short and comparatively dry live band venue could acquire selected early response and reverberant characteristics of a longer classical hall without claiming complete physical replication.

This English paper consolidates the formal method description and the field implementation account that were originally prepared in Korean [12][13].

2 Previous Work

2.1 Convolution and active acoustics

Impulse response measurement with swept sine excitation provides a practical basis for characterizing a room and separating linear response from distortion [1]. Convolution then allows a captured room response to be applied to another signal. Room acoustics research shows, however, that reverberation, clarity, reflection timing, and spatial distribution interact with the listener and source positions [2][3][4]. A single target impulse response therefore cannot cancel the acoustic contribution of an arbitrary receiving venue.

Installed active acoustics systems such as Yamaha AFC Enhance extend or reshape reverberation through microphones, processing, and distributed loudspeakers [11]. These systems motivated the live deployment considered here, but the present work focuses on a compact research implementation that derives path specific correction filters from measured impulse responses and combines them with a separately applied target hall tail.

2.2 Inverse filtering and time alignment

Room equalization can be formulated as an inverse problem, but direct spectral division is unstable when the measured response contains deep notches, noise, spatial modes, or measurement error. Regularized inverse filter design limits the energy required to correct poorly conditioned frequencies and reduces ringing [6][7]. Accurate temporal registration is equally important because a direct arrival offset appears as a phase difference. The generalized cross correlation with phase transform method provides a robust delay estimate for this alignment task [5].

HISSTools provides multichannel convolution objects suitable for real-time artistic systems [9]. The gap addressed in this project lies in combining these established methods into a venue-specific six-channel workflow that separates initial response correction from the target hall reverberation tail and that can be operated during a live performance.

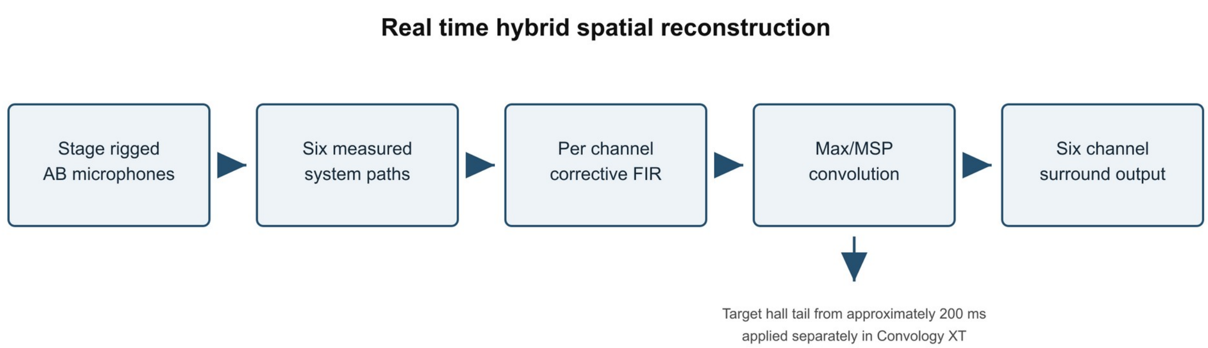

Figure 1 Processing architecture from the stage microphone pair to six surround outputs

3 Proposed Research and Methodology

3.1 Research scope and acoustic spaces

The study defines spatial reconstruction as reinforcement of selected target hall characteristics. It does not attempt to reconstruct the target hall sound field at every seat. The primary comparison uses a measurement position centered on the fourth audience row, which was also the reference position for system alignment. The target hall data were obtained from the University of York AudioLab OpenAIR library [10].

Space	Capacity	500 Hz T30	Role in study
DIMA Jiseong Hall	about 200	0.39 s	Current venue and measured system paths
Sir Jack Lyons Concert Hall	about 350	1.89 s	Target impulse response from OpenAIR

Table 1 Acoustic spaces used in the study

Figure 2 DIMA Jiseong Hall during system preparation author photograph

3.2 Measurement and channel configuration

Six surround loudspeakers were placed at the left, right, and rear audience areas. Four d&b MAX loudspeakers served the side positions, two Behringer B112 loudspeakers served the rear positions, and Turbosound IQ12 and IQ18 loudspeakers formed the main system. The main PA retained the direct sound and primary mix balance. The surround system supplied the corrected initial response and target hall reverberant energy.

A RØDE NT5 matched pair was rigged above the stage in an AB configuration. A test signal was reproduced from each surround loudspeaker in turn and captured by the microphone pair. These six measurements represent the loudspeaker-to-microphone paths through the current venue. A separate corrective FIR was generated for SLF, SRF, SLC, SRC, SLB, and SRB because the loudspeakers differed in position and model.

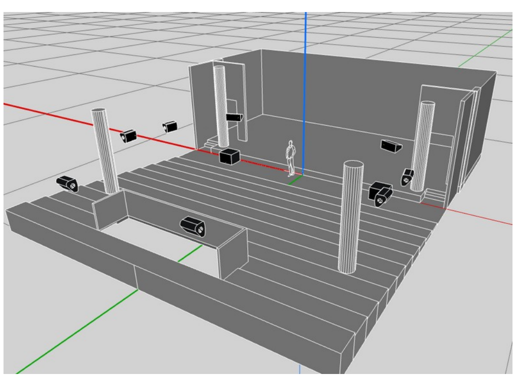

Figure 3 Preproduction model of the six surround loudspeaker positions

Figure 4 Stage rigged AB microphone pair used as the live system input author photograph

3.3 Corrective FIR generation

Let a(t) denote the target hall impulse response, b(t) the current measured system path, and f(t) the corrective filter. The desired relationship is that convolution of b(t) and f(t) approaches a(t). Directly dividing A by B in the frequency domain can produce excessive gain at notches, so the implementation uses the regularized inverse shown in Figure 5. The regularization term is referenced to the maximum magnitude of the current response; the default reg_db setting is minus 32 dB.

$$b(t) \times f(t) \approx a(t)$$

$$F(\omega) \approx \frac{A(\omega)\, B^*(\omega)}{|B(\omega)|^2 + \lambda}$$

Figure 5 Corrective filter objective and regularized frequency domain estimate

Step	Process	Implementation
1	Input and resampling	Load current and target IRs and resample the target when sample rates differ
2	Time alignment	Estimate delay with GCC-PHAT and align direct arrivals
3	Gating	Select the response region used to derive the corrective filter
4	Regularized inversion	Estimate a stable inverse instead of unconstrained spectral division
5	Band limitation	Concentrate correction between 125 Hz and 4 kHz
6	Stability constraints	Apply complex smoothing passband masking gain clipping and peak limiting
7	Export and verification	Export a channel specific WAV FIR and compare the convolved result with the target

Table 2 Python pipeline used to generate each corrective FIR

3.4 Hybrid early and late response processing

The corrective FIR is designed around the initial response rather than the complete decay. Approximately the first 40 ms are used as the principal correction region. The target hall impulse response from approximately 200 ms onward is applied separately as a reverberation tail in Convology XT. The interval from 40 ms to 200 ms is treated as a transition between the residual corrective filter response and the target hall tail, not as an independently validated inverse correction region.

This hybrid structure reduces the risk of attempting to invert the entire reverberant field. It also keeps the live system operationally interpretable: the channel specific FIRs reshape the initial response, while the target hall tail supplies later reverberant energy. The system therefore reinforces an acoustic impression instead of claiming to reproduce the target hall's complete sound field.

3.5 Real time implementation

The six corrective FIRs were loaded into a Max 9.0.7 patch using HISSTools multiconvolve tilde. The AB microphone input was distributed to six processing paths, convolved with the corresponding channel filter, combined with the separately processed Convology XT tail, and routed to the surround outputs. The field computer also generated updated FIR files when measurements changed. Figure 6 shows the processing patch, and Figure 7 shows the on-site correction workflow.

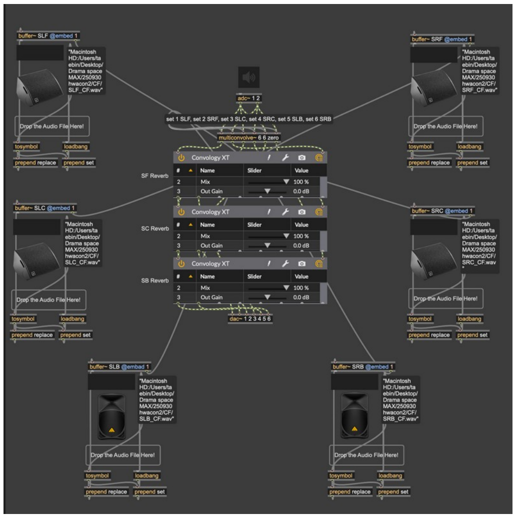

Figure 6 Max/MSP patch for six-channel corrective convolution and separate reverberation tail processing

Figure 7 Corrective FIR generation and system operation at front of house author photograph

3.6 Evaluation protocol and results

For each channel, the measured current system path was convolved with its corrective FIR and aligned with the target response. The comparison used the initial 0 to 40 ms window, one third octave smoothing, and the 125 Hz to 4 kHz correction band. Mean absolute spectral error in decibels was calculated relative to the target response. A lower value indicates that the evaluated initial spectrum is closer to the target under this definition.

All six channels showed lower error after correction. The largest reductions occurred at SRC, from 3.17 dB to 2.17 dB, and SLF, from 2.27 dB to 1.47 dB. Across all channels, the mean error decreased from 2.69 dB to 2.24 dB, a reduction of 0.45 dB. Figure 8 and Table 3 report the channel values.

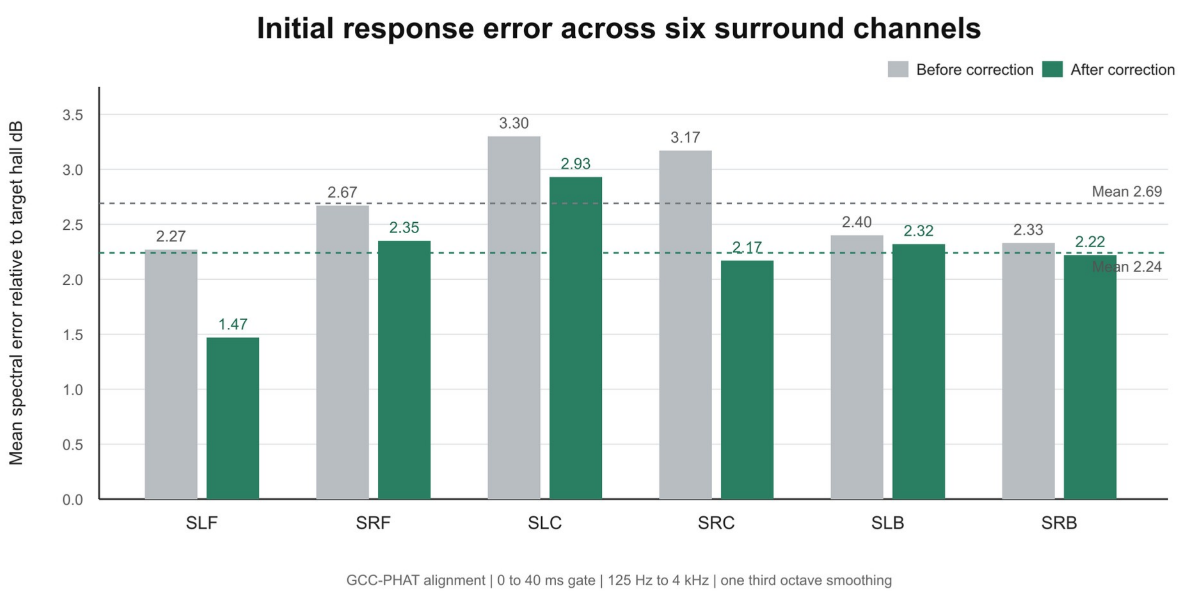

Figure 8 Mean spectral error before and after correction in the defined initial response analysis

Channel	Before dB	After dB	Change dB
SLF	2.27	1.47	-0.81
SRF	2.67	2.35	-0.32
SLC	3.30	2.93	-0.37
SRC	3.17	2.17	-1.00
SLB	2.40	2.32	-0.08
SRB	2.33	2.22	-0.11
Mean	**2.69**	**2.24**	**-0.45**

Table 3 Mean spectral error relative to the target hall in the initial 0 to 40 ms window and 125 Hz to 4 kHz band

3.7 Limitations and future validation

The result is a first stage numerical comparison at one measurement position using the same system path data involved in filter generation. It does not establish the final response at other audience seats, an equivalent sound field across the venue, or perceptual equivalence to the target hall. No controlled listening experiment was conducted. The installed system is also constrained by feedback margin, latency, loudspeaker headroom, acoustic coupling, audience absorption, and variation between measurement positions.

Future work should compare four matched conditions: no correction, direct target impulse response convolution, the proposed hybrid method, and the target hall reference. Evaluation should include reverberation time, early decay time, C80, D50, energy decay behavior, spatial variation across audience positions, and controlled listening tests. Independent measurement data should be reserved for validation so that filter derivation and evaluation are not performed on the same observations.

4 Contributions and Summary

This work contributes a measurement-based workflow for approximating selected characteristics of a target hall in a live venue. First, it defines the correction problem around measured venue-dependent signal paths rather than assuming that direct target impulse response convolution is sufficient. Second, it implements a stable per-channel FIR pipeline with sample rate matching, GCC-PHAT alignment, gating, regularized inversion, band limitation, smoothing, gain constraints, and peak limiting. Third, it separates initial response correction from later reverberation through a hybrid Max/MSP and Convology XT structure. Fourth, it demonstrates the complete workflow in a six-channel live performance system.

Within the stated measurement and analysis conditions, the six channel mean spectral error decreased from 2.69 dB to 2.24 dB. The result supports continued investigation of the method but remains limited to the initial 0 to 40 ms response at the fourth row center reference position. The study should therefore be understood as an operational proof of concept for live spatial reinforcement, not as evidence that the target hall was physically or perceptually reproduced.

5 References

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