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Recalibration of the NASA Exterior Effects Room

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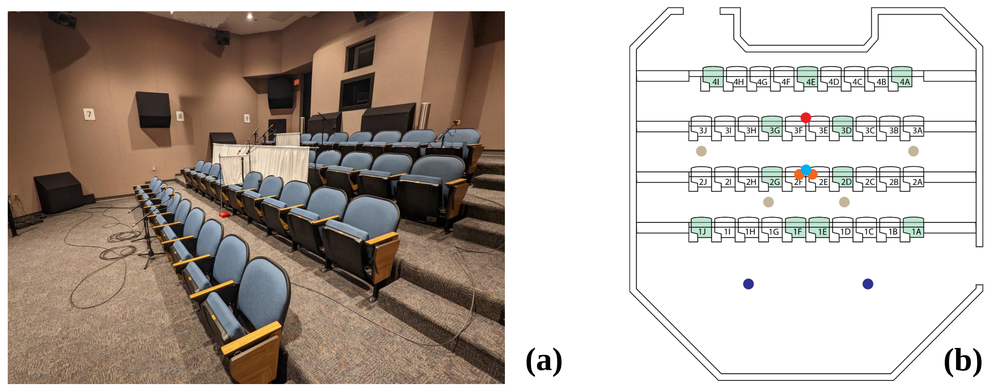
Abstract

The Exterior Effects Room (EER) is a psychoacoustic testing facility at NASA Langley Research Center which primarily focuses on testing human response to aircraft noise. The 39-seat auditorium houses a real-time spatial audio system and flyover simulation environment. The audio server utilizes an implementation of three-dimensional vector base amplitude panning (VBAP), a perceptual spatial audio technique exploiting loudspeaker triplets to place a virtual sound source at an arbitrary spatial position. Due to the irregular room geometry and nonuniform loudspeaker setup, the audio server applies equalization filtering to compensate for spectral coloration attributed to loudspeaker installation, crossover filtering, and delay/gain offsets. These filters can incorporate measurements taken at multiple listening points, allowing a more extensive listening area than filters derived from measurements taken at a single listening point. This work endeavors to update the existing equalization filter generation process by producing a new set of filters for various human subject testing scenarios.



(a)

(b)

Figure 1: (a) Photograph of EER seating area. (b) Frequency response measurements used in updated equalization framework. Positions include center location (cyan), 2Mic configuration (orange), seat locations (green), listening area positions (dark blue, beige, red).

1. Methods

The EER currently utilizes a 31-channel playback system of twenty-seven Klein & Hummel O300 and four Klein & Hummel O900 subwoofer loudspeakers (Figure 1a). The audio playback server that implements the real-time VBAP algorithm can also apply compensatory equalization filters for each speaker in the array. The primary original filter set was derived in 2011 using measurements from a two-microphone array very close to the center of the room. Since that work the EER has been used for many psychoacoustic tests in which subjects typically sit in 4 seats slightly spread from the geometric center of the room. The fact that the original filters were irrespective of this usage, combined with recent upgrades to the playback system, in addition to any effects of aging/replacement of audio equipment, make it a prudent time to resurrect the equalization process and undertake it for the system as it exists today.

Equalization for each loudspeaker is individually calculated given a measurement at a set of listening locations – one speaker's equalization is not dependent on another's. Three components are required for compensation for each loudspeaker in the EER: an equalization filter, relative gain offset, and time-of-flight delay. For each speaker, an FIR filter is generated by inverting the difference between a set of measured speaker responses and an idealized response (inclusive of a Linkwitz-Riley crossover). This can be done for multiple measurement locations at the same time using a Weiner-filtering approach. After that, an IIR approximation of the FIR filter is generated for computationally tractable real-time implementation. Gain offset compensation is generated from the same set of measurements used for the filter. The time delay is generated from a single microphone placed at the geometric center of the room.

For the current work, measurements were taken at 21 locations in the EER by considering past usages of the room. The locations measured are indicated in Figure 1b. Eleven seat location recordings were set to a center-of-head position, including the four typical subject seat locations (4Mic). Recordings were also made in the two locations used in the 2011 equalization process (2Mic).

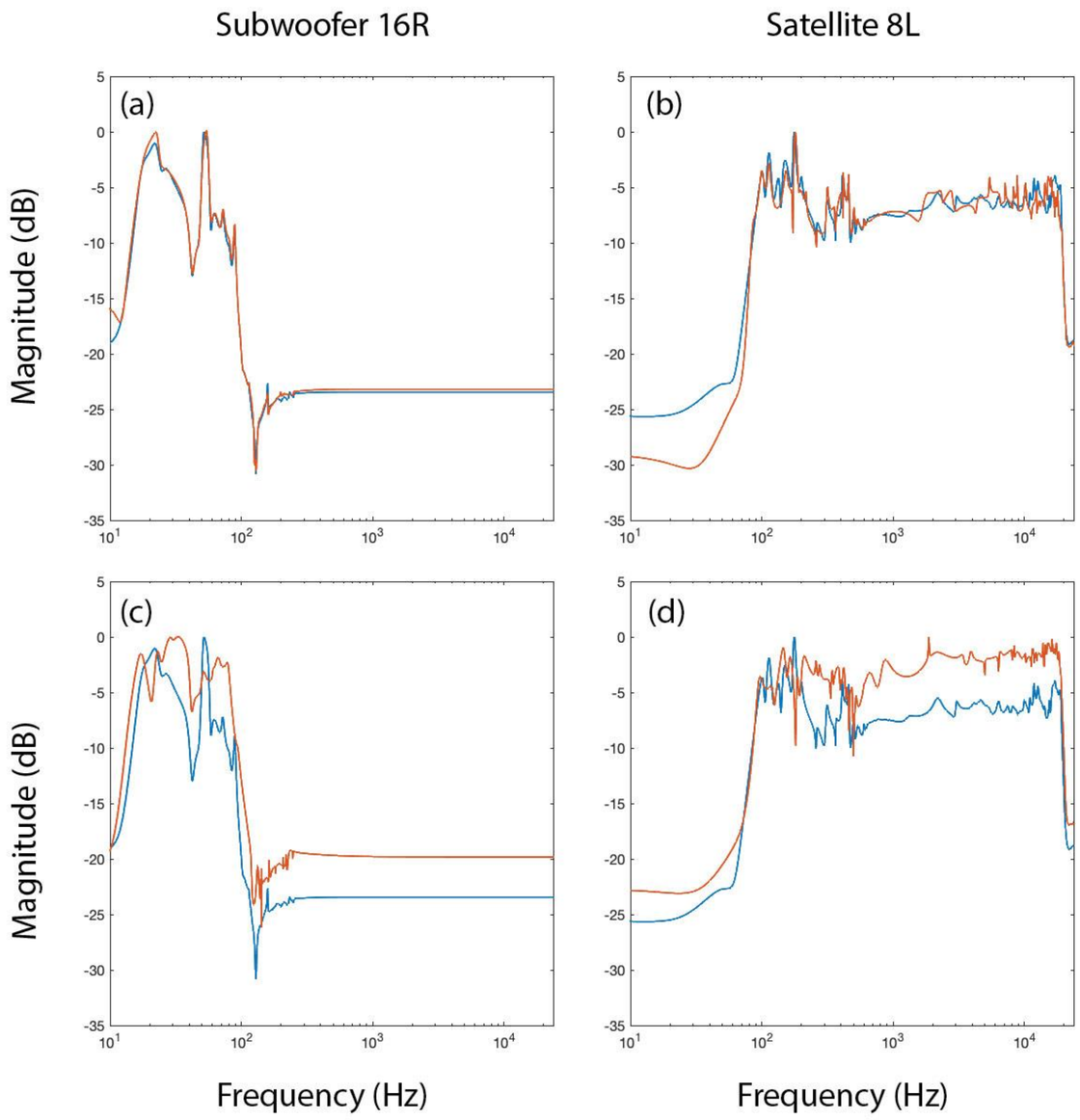


Figure 2: Comparison between old and new equalization filters: (a, c) Subwoofer 16R, (b, d) Satellite 8L. (a, b) Comparison between old 2Mic filters (blue) and 2Mic recreation with new measurements (orange). (c, d) Comparison between old 2Mic filters (blue) and new 4Mic filters derived from subject seat location measurements (orange).

The recording suite consists of two-minute pink noise responses per-speaker and per-microphone location. These recordings are processed via FFTs of 32,768 (2^{15}) samples which, at a 48 kHz sampling rate, results in a binwidth of 1.5 Hz. A 50% overlap is used to generate 240 averages. This was done for all speaker-microphone combinations – 651 responses. Choosing to include a response in the equalization process effectively extends the intended "sweet spot" of the listening area to those locations (given the constraints of the other included responses).

2. Results

Figure 2a and 2b show examples of the original 2Mic equalization filter of a satellite and subwoofer with filters derived from the new measurements that were made to replicate the old 2Mic case. This shows good agreement between the two filter sets. Although there are some minor discrepancies, especially in the higher frequencies (where these sorts of measurements are more sensitive), the trends largely match. These results are typical of the agreement found for all of the speakers between the new and old measurements.

Figures 6c and 6d compare magnitudes of the old 2Mic equalization filters used for human subject testing with the new 4Mic case derived from the measurements of the subject seat locations. As expected, the crossover filter limits are consistent while the bandpass regions show some marked differences in response. Because typical EER psychoacoustic testing positions the subjects in these four seat locations, the new 4Mic equalization filters are expected to improve the sound reproduction over the 2Mic equalization filters (new or old) for those use cases.

The time-of-flight delay compensation matches the original values to within a 0.5 ms, and gain compensations were found to match well for the 2Mic test case.