Placement of Sound Sources In The Stereo Field Using Measured Room Impulse Responses

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Abstract. Reverberation can be simulated by convolving dry instrument signals with physically measured impulse response data. Such reverberation effects have recently become commonplace; however, current techniques apply a single effect to an entire ensemble, and then separate individual instruments in the stereo field via panning. By measuring impulse response data from each desired instrument location, it is possible to place instruments in the stereo field using their unique early reflection and reverberation patterns without panning. A pilot study compares the perceived quality of dry signals convolved to stereo center, convolved to stereo center and panned to desired placement, and convolved with measured impulse responses to simulate placement. The results of a single blind study show a preference for location-based (as opposed to panning-based) reverberation effects.

1 Introduction

When an ensemble performs on stage before a live audience, the listening experience is undoubtedly enhanced by the physical separation of the instruments on stage. This effect does not occur by chance, as percussive instruments are often placed in the center of the stage, with bass and melodic instruments often separated to either side. The placement is formulated so as to reduce the effect of one instrument dominating the sound of another. Currently, when recording and mixing down albums, a single reverb is placed on each track, based upon either IIR filters or a convolution with a single measured impulse response. Placement is achieved using a combination of amplitude panning, pre-delays, decay times, and saturation levels in order to separate the individual instrument tracks. This method is effective, but purely artificial, providing no real psycho-acoustical clues that the instrument field is properly placed in a real acoustic space.

When an instrument is played at one location on a stage versus another, the reverberation signature is different. This effect occurs because as sound radiates from the instrument, the sound energy reflects from various walls, the floor, and ceiling, reaching one's ear at different time intervals and at different frequency-dependent amplitudes. The effect is subtle, but, in principle, recognizable. Consequently, there is a unique impulse response associated with each location on the stage (paired with each listening location in the room). In principle, if each instrument signal in an ensemble is convolved with its unique location-based impulse response, then it should enhance the psycho-acoustical illusion of the separation of the instrument field, eliminating the need for panning or other more artificial effects. We call this approach location-based reverberation. One might expect that by giving each instrument a different impulse response, the listener will find it easier to perceive each instrument individually (source separation), and this in turn might avoid the perception of one instrument overpowering the others.

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However, even convolution with multiple location-based impulse responses is only an approximation of sound radiation in a room. Acoustic instruments have frequency-dependent radiation patterns. To incorporate this level of detail, one could model impulse responses as functions of source direction as well as source location. To take advantage of this more refined approach, sound sources would need to be modeled or recorded so as to capture audio signals as a function of direction. In our simplified model, we consider only multiple source locations. The impulse responses used here incorporate the directional radiation patterns of the speakers used in the impulse response measurement process, and we expect that patterns will be somewhat different from those of acoustic instruments. Another limitation is that stereo recording does not capture the complex sound field available to the listener in an acoustic space. This is a fundamental limitation of the stereo format. Our goal in this study is only to improve the listening experience within the restrictions of the stereo format, but note that extensions to other formats are at least conceptually straightforward.

The extent to which the technique of virtual instrument placement via measured room impulse responses will improve the actual perceived quality of the performance is unknown; hence, the need for an appropriate study to evaluate the qualitative difference between current methods using single impulse responses and the proposed method using location-dependent impulse responses.

2 Previous Research

Current recording techniques are the culmination of many years of research and reasoning. Numerous studies have been conducted to evaluate the utility of current techniques in addition to considering their ability to withstand the rigors of commercial practice. Formulations of the theory can be found in Pulkki among others [15]. Regarding virtual instrument placement via location-based reverberation, not much has been studied regarding the actual quality of the effect versus current methods. The theory behind the method has been outlined on several occasions, including discussions by Reller and Griesinger [16, 9]. The Roland SRV-330 Dimensional Space Reverb uses 24 early reflections to create the impression of a 3-D acoustic space [17]. However, actual quality perception tests and implementation details are not available.

This method of location-based reverberation is related to the use of head-related transfer functions (HRTFs) to simulate spatial location. [4] HRTFs are often used to model changes in both sound source location and the orientation of the listeners head. Our method differs from HRTF models because our goal is to incorporate room reverberation into the impulse response, and we do not incorporate HRTF information in our impulse responses.

Many artificial reverberation models have been proposed [5, 13, 10]. Some of them take into account source locations, speaker locations, room geometry, and/or listener location. These systems essentially estimate the channel characteristics between each sound source and and some other room location representing the listener or a loudspeaker. As in our approach, the reverberation effect applied to each dry source depends upon the source location and the room geometry. In our approach, however, the channel characteristics from source to microphone are measured directly as impulse responses.

3 Location-Based Convolution Reverberation

The details of our reverberation system are straightforward. We will begin with an overview of current convolution-based reverberation effects and then describe the extension we have made. The principle behind convolution-based reverberation effects is that the channel from a sound source to the listener in a concert hall is linear (or at least there is a good linear approximation). Theory tells us that the channel can be modeled as the convolution of the source signal with the impulse response of the channel:

$$y(t) = x(t) * h(t) \tag{1}$$

where $y(t)$ is the signal at the listener, $x(t)$ is the source signal, and $h(t)$ is the impulse response from the source to the listener. Normally, the impulse response is estimated between a single source location on stage and a pair of microphone locations representing the left and right stereo channels. Thus, the stereo signal is computed as follows:

$$\begin{aligned} y_L(t) &= x(t) * h_L(t) \\ y_R(t) &= x(t) * h_R(t) \end{aligned} \tag{2}$$

For a typical studio recording, multiple instruments are recorded with close microphones on separate "tracks," which can be considered to be free of reverberation and acoustically isolated. The goal is to "place" these instruments in the stereo field. Ignoring the possibility of other digital audio effects, the final mix is produced as follows:

$$\begin{aligned} y_L(t) &= (\sum_i A_{i,L} x_i(t)) * h_L(t) \\ y_R(t) &= (\sum_i A_{i,R} x_i(t)) * h_R(t) \end{aligned} \tag{3}$$

where L and R stand for the left and right stereo channels, and $A_{i,L}$ and $A_{i,R}$ are scale factors to implement left-to-right panning on the i^{th} instrument. Note that a single reverberation effect is applied to a mixture of the instruments. This is computationally efficient, but treats all instruments as if they were located at the same point on stage.

It should be noted that there are other panning options in commercial implementations of convolution-based artificial reverberation. For example, the Waves IR1 plug-in omits the impulse in the response corresponding to the direct sound and separates the early reflections from the reverb tail. This gives the user the option of panning each dry instrument signal to a different simulated location and then adding a global reverb to some mixture of the dry sounds.

Location-based convolution reverb modifies the computation to incorporate location-based impulse responses. The i^{th} instrument signal is convolved with the i^{th} impulse response $h_i(t)$ Since left and right amplitudes and delays are already incorporated into the impulse responses, no additional scaling or artificial panning is necessary:

$$\begin{aligned} y_L(t) &= \sum_i (x_i(t) * h_{i,L}(t)) \\ y_R(t) &= \sum_i (x_i(t) * h_{i,R}(t)) \end{aligned} \tag{4}$$

Note that this approach requires more computation because there is a separate convolution-based reverberation effect on each source signal.

In practice, signals are of course discrete, and convolution is performed by converting blocks of the source signal to the frequency domain (using the FFT), multiplying by the frequency domain representation of the impulse response, and then using an inverse FFT to convert back to the time domain [14]. Convolution was implemented in the Nyquist programming language [6]. As indicated by Equation 4, a separate convolution is performed for each (mono) source signal and for each of two stereo channels.

Room (or hall) impulse responses (RIRs) were obtained from measurements of an acoustic space. Various techniques for measuring RIR have been studied [7, 8, 12]. The three most popular excitation signals for RIR measurement are: a Maximum Length Sequence (MLS), an impulse, and a chirp signal. For analyzing a large concert hall, however, the impulse and the MLS sequence are not good choices for a number of reasons [11]. We therefore choose the chirp signal, which contains all the frequencies required, is a linear signal so is less likely to damage the equipment and also contains a large amount of energy. Using a chirp signal longer than the RIR to be measured allows the exclusion of all harmonic distortion products, practically leaving only background noise as the limitation for the achievable SNR [7].

Our measurement system works as follows. The chirp signal is generated by a laptop computer and played to a speaker. Assuming that most RIR would not exceed 3 seconds, we use a linear chirp signal with a duration of 3 seconds and frequency sweeping from 0 to 24 kHz. At the receiver end, the output signals of a stereo microphone pair are recorded to the same laptop through a multichannel audio interface, together with the unaltered chirp to be used as the reference signal. The unaltered reference signal is important in that it eliminates the need to estimate the latency in the playback-record chain. To obtain the stereo RIR, the received signals are correlated with the reference signal. Just as in a radar processing application, this function compresses the pulse and gives rise to the room impulse response that is to be analyzed [16].

4 Methodology

Theoretically, location-based reverberation models real acoustic reverberation more faithfully than does convolution with a single stereo impulse response. However, as noted above, there are aspects of reverberation that are not modeled by either approach, including directionality and non-linearities. We wanted to evaluate location-based reverberation to determine whether it offers any subjective improvement over current techniques as judged by listeners

4.1 Experimental Design

We decided that a small pilot study would be the most appropriate initial experiment because it was unknown what, if any, differences subjects would hear. Our experimental sample, drawn from a student population, is not representative of our target demographic as a whole, but we do not believe this choice significantly altered our results.

Sample Population We used a subject pool consisting of 25 members of the Carnegie Mellon University undergraduate population. This convenient sample allowed us to quickly gather data while maintaining a well-defined reference population. The final sample demographics reflect the Carnegie Mellon undergraduate community, with an approximately 60% male and 25% minority makeup. All participants were between the ages of 18 and 23. Subjects were not screened based on other demographics such as musical background.

Sound Samples For our test, we generated three sound samples for our subjects to compare. All three were based on the same samples of a 30-second jazz excerpt consisting of drum set, contrabass, and saxophone, all recorded with close microphones to minimize cross-source contamination. The samples were chosen because we felt that a non-classical source would result in a more pronounced sonic differentiation between instruments, while the jazz idiom also requires a “live” enough feel that reverberation-based placement in a hall would be an appropriate effect.

To create our samples, we convolved hall-measured impulse response data with the dry jazz samples. These samples were then used to create three variations. The first, called *mono*, is a single-channel sample in which all three instruments are convolved with hall-center impulse response. The second, referred to as *panned*, is a stereo sample in which the three instruments are first panned such that the drums are center, the bass 80% right, and the saxophone 80% left. After panning the dry signals, the two mixed channels are convolved with the left and right channels of the hall-center impulse response, respectively (Equation 3). The final sample, called *placed*, convolves each instrument signal with a different impulse response: a center-based impulse response with the drum set, an audience-perspective right impulse response with the bass signal, and an audience perspective left impulse response with the saxophone signal (Equation 4).

At the highest granularity, the resulting sound samples are all reverberation-wet jazz performances, identical except for techniques regarding instrument placement in the stereo field. The samples were also normalized to peak at 0 dB so as to have matching volume levels. Upon initial listening by the investigators, the *placed* sample seemed to display a richness lacking in the other two samples. The pilot study would later corroborate this subjective observation.

The impulse responses themselves were recorded via a microphone array located in the audience at the center of the concert hall. The venue chosen was the 200 seat Recital Hall located at the School of Music, University of Victoria, Canada. The responses were measured using a swept sine wave through a microphone array and repeated at three locations on the stage [11]. This resulted in an array of 7 different impulse responses for each location on the stage. For our simple stereophonic setup for this experiment, we chose simply the left and right impulse responses (2 of the 7 measured responses) for each of the 3 locations, corresponding to stage right, stage left and stage center. Other measured stereo impulse responses are available for a variety of concert halls and other venues [2]; however, these measurements typically do not include multiple locations on stage, and thus cannot be used for the *placed* variation in this experiment.

Questionnaire To compare the sound samples objectively, we developed a battery of comparative questions to grade the sound samples. The three categories of comparison were "realism," as defined by the likeness of the sample to a live performance, "sound quality," and simple personal preference. The questionnaire asked the subject to listen to two sound samples consecutively, and then compare them on the three selected attributes. Each sample was paired with every other sample, making for a total of three individual listening tests. To reduce bias, the order of the sample pairings was randomized as well as the play order within a given sample pair.

Due to concerns about the ability of all subjects to distinguish between the samples, the realism and quality questions asked for a simple pair-wise comparison to indicate which of the two samples the subject preferred across the realism, quality, and overall preference metrics described above. The preference question also asked for a comparison, but also allowed for answers of "I have no preference" and "I could not tell a difference." In retrospect, listeners did not appear to have great difficulty in distinguishing the samples, with less than 6% of respondents selecting "no preference" or "no difference."

4.2 Experiment Administration

The experiment was administered over the course of a weekend to all 25 subjects. Administration of the study was not difficult due to the brevity and subject matter of the experiment. The study proceeded in a randomized single-blind fashion, on one of two reference systems[1]. Regarding volume, listeners were asked to initially adjust the volume to preference, and then leave it fixed for the duration of the listening test.

Process The study involved, first, a principal investigator providing the consent form and explaining that the study intended to compare several reverberation techniques, and that the listeners would be asked to listen to several jazz excerpts, identical except for the reverberation applied. The participants were then allowed to look over the questionnaire, but the investigator provided no interpretation as to the meaning of each question or questions regarding sample specifics.

At this point, the investigator played the first sample, identified only by a number, then the second sample. After this, the subject would record their results on the questionnaire, but the sound samples would not be replayed. The process was then repeated for the other two pairs of sound samples, the end result being that each subject would listen to each sound example twice and compare each to the others. After collecting the questionnaire, the investigators provided a brief explanation of the actual experimental intent and identified the sound samples by technique applied.

Data Analysis For a study of this size, bias due to random variation in samples is a real concern. As such, we feel that it is important to include confidence intervals along with our proportion averages so as to accurately reflect the variability of our pilot study. For this study, we considered the experimental results to be drawn from a binomial distribution, and we calculated confidence intervals based on a normal approximation of this distribution [1]. The binomial distribution assumes that each experimental trial has only two outcomes; to match this model, the preference calculations dropped "no preference" and "no difference" responses.

For example, of the 25 participants, 8 perceived *panned* as sounding more realistic than *mono*. To compute the $\alpha = .95$ confidence interval for realism, *panned* vs. *mono*, we simply used the binomial confidence interval formula for proportions:

[1] Both systems were laptop PCs, one with Sony MDR-V500 headphones, and the other with Koss UR-40 headphones.

$$CI = p \pm 1.96\sqrt{p(1-p)/N} \tag{5}$$

Here p = (8/25) = .32 and N = 25. Thus,

$$\begin{aligned} CI &= .32 \pm 1.96\sqrt{.32(1-.32)/25} \\ &= .32 \pm .182 = [.137, .503] \end{aligned} \tag{6}$$

Now we can interpret these data by saying that with 95% confidence, the true population proportion preferring *panned* to *mono* falls between 0.135 and 0.503, taking our sample size into account.

5 EXPERIMENTAL RESULTS

Our experimental results point in favor of location-based reverberation for instrument placement based on the metrics of both sound quality and personal preference. Realism does not result in as conclusive a result, but the data yields valuable insights.

Table 1. Aggregated means and confidence intervals for proportion preferring the first listed sound clip in each cell.

	Panned vs. Mono	Placed vs. Mono	Placed vs. Panned
Realism	p = .32 [.137, .503]	p = .52 [.324, .716]	p = .68 [.497, .863]
Quality	p = .72 [.497, .863]	p = .84 [.696, .984]	p = .64 [.452, .828]
Preference	p = .57 [.363, .768]	p = .70 [.508, .884]	p = .68 [.497, .863]

5.1 Realism

In this study, we defined realism as "likeness to an actual live performance." Interestingly, there does not appear to be a strong consensus that any reverberation method is most realistic. Each pair-wise comparison of realism resulted in a confidence interval that included .5, the null hypothesis that there is no perceived realism difference between the samples (see Table 1). Nevertheless, .68 rated the *mono* sample as more realistic than *panned*, and .68 rated the *placed* sample as more realistic than *panned*. This may be a reflection of a lack of realism in the *panned* sample, where the stereo spread could have been too wide to be considered realistic. Conversely, it may simply reflect a tendency of the sample population to feel that smaller stereo spreads best reflect the experience of a live performance, especially over headphones, which can exaggerate panning effects.

The other interesting observation about realism is the fact that the proportion preferring *placed* to *mono* was .52, almost exactly the null hypothesis. While the other two pairs were barely out of the 95% confidence range, it appears that our sample population could not distinguish between the two with regards to realism. We hypothesize that this indicates that the stereo spread effect is potentially a major determining factor in causing listeners to perceive a recording as realistic.

5.2 Sound Quality

In contrast to the realism judgement, our investigation found much stronger support for location-based reverberation placement with regards to "sound quality." Here, *mono* fared the worst, with .72 of the population preferring *panned*, and an extremely high .84 of the population preferring *placed*. In fact, despite the small sample size, the *placed* versus *mono* confidence interval, [.696,.984], is highly significant, and the *placed* versus *panned* interval, [.452,.828], only barely contains the .5 null hypothesis. This result suggests a larger study to determine if location-based reverberation is truly a higher-quality placement technique than panning.

One other interesting trend to note is the relationship between realism and quality for each of the three pairs. The observed relationships vary in counter-intuitive ways. Quality and realism correlate positively for *placed* versus *panned*, while they correlate negatively for *panned* versus *mono*. Finally, subjects decisively find *placed* to be of higher quality than *mono*, but seem to be unable to decide which is more realistic. With our sample size, it is entirely possible that these trends are just random variation. Their further exploration on a larger sample could prove instructive.

5.3 Personal Preference

The final metric is overall personal preference of the various sound samples. This measure shows the greatest advantage for location-based reverberation. Subjects preferred *placed*, with .70 rating it over *mono* and .68 rating it over *panned*. Even with only 25 participants, the *mono* comparison is significant at the =.95 level, and the *panned* comparison just barely misses this level of significance (see Table 1). We feel such a consistent result in favor of convolution placement is solid evidence that the technique is a viable improvement over current post-processing effects. More subjects and a larger variety of sample material would likely serve to add weight to this judgement.

In addition to these results, we find it interesting that preference seemed much more split when comparing *mono* and *panned*. Subjects preferred *panned*, but only .57 rated it over *mono*. If it really were true that the increased perception of realism in *mono* somehow cancelled out the increased sound quality with *panned*, this would prove to be another advantage for location-based reverberation placement, which seems to be able to combine the best qualities of both other methods. That said, this interpretation seems unlikely, and a much larger pool of subjects and samples would be necessary to give it much credence. The strongest indication of this pilot study is the overall preference for location-based placement over other techniques.

6 Discussion

Although the results of our pilot study are not overwhelmingly conclusive, we did observe a clear trend in favor of location-dependent reverberation (*placed*). For example, *placed* received a majority of positive ratings in all 6 comparisons to the other two methods. It should be noted that subjects listened with headphones, and the sample size was fairly small. Given the generally positive findings, a larger study is in order.

After listening to various sound examples, the authors agree with the experimental trend. Moreover, we feel that location-dependent reverberation is immediately recognizable as more realistic and natural, with a more spatial or three-dimensional quality reminiscent of live recordings with a stereo pair. This of course is exactly the sound one would expect and the sound this method is intended to produce. On the other hand, it should be noted that the authors prefer this sound and are likely to associate this sound with high quality and high realism.

This suggests an interesting interpretation of the experiment. Suppose that subjects hear a clear difference between different reverberation treatments, but disagree with respect to labels such as "quality." For example, some subjects might associate the sound of commercial pop music recordings with "high quality" even if they felt this sound was not realistic or preferable. Indeed, in the comparison of *placed* to *panned*, subjects gave slightly stronger ratings on realism and preference than to quality. The difference here could easily be due to chance, but it is interesting to consider that location-based reverberation could be a distinctive reverberation effect.

A future study might use a test based on analogies to see if subjects can actually identify location-based reverberation. We would predict a positive result. If this effect has a distinctive and recognizable sound, there are likely to be interesting artistic applications.

While our approach models the fact that the location of instruments has an effect on the reverberation, we ignore many acoustical details. One is the directional radiation pattern of sound sources. In our approach, it is assumed that the radiation pattern of instruments matches the radiation pattern of the loudspeaker used to excite the room when impulse responses are measured. Similarly, this approach does not consider directionality of the listener, e.g. we do not incorporate head-related transfer functions.

There are, however, some interesting and simple variations of our approach. First, impulse responses can be measured using intentionally directional sources. For example, a horn speaker might be used to estimate the impulse response for a brass instrument, or a small speaker array might be configured to mimic the radiation pattern of a violin. With a library of different impulse responses based on both location and directionality, each dry signal can be convolved with the appropriate impulse response, simulating both the location of the instrument on stage and the radiation pattern of that instrument.

Similarly, the microphones used to capture and estimate impulse responses can be selected according to the anticipated listening conditions. Microphones might be directional or omni-directional, stereo or multi-channel (for example, five microphones for a 5.1 recording), or mounted in a dummy head to incorporate a head-related transfer function, a common recording technique for headphone listening. Note that while one would normally estimate impulse responses using an ideally omni-directional, flat-response sound source and microphone, directional and even spectrally "colored" transducers can be used for various purposes.

Another important effect in a real acoustic environment is Doppler shift. As a musician moves toward the audience, the direct sound is shifted up in frequency, but reflections experience varying degrees of Doppler shift, with some of the radiated sound actually shifted downward. Furthermore, a moving sound source is constantly changing its location and thereby exciting different room modes and corresponding, different impulse responses [3]. These effects are not produced by our approach.

To incorporate moving sound sources, a direct approach would measure impulse responses from many locations and either switch between them or use some form of interpolation for intermediate locations. Note that linear interpolation between impulse responses A and B suggests that the instrument is radiating partially from both locations A and B, but not from some location between A and B. Another approach is to compute early reflections from a geometric model. If early reflection delays and amplitudes vary continuously, then Doppler shift will be produced as a by-product. Dense, diffuse reverb can be added to complete the reverberation effect.

7 Conclusion

Judging by our pilot study, location-based reverb is a very promising approach to high-quality artificial reverberation, and the potential impact of these techniques on the recording industry is large. Standard convolution reverberation plug-ins such as the Waves IR1 are already in use by industry. Location-based reverberation would use very similar software, but it will require a much larger pool of impulse response data. Since plug-ins of this sort already rely on hall-measured impulse-response data, the burden of measuring a larger number of instrument/listener location pairs should not be prohibitive. Thus, location-based reverberation offers a relatively inexpensive and effective post-processing technique that can be used in today's stereophonic applications to greatly enhance the psycho-acoustical experience for the listener.

The results of our single-blind pilot study clearly warrant further investigation. Within the bounds of our sample size and limited demographic, our results point in favor of location-based reverberation placement. The average listener preference to the location-based reverberation technique demonstrates not just a theoretical advantage but a subjective preference and thus a real viability in the commercial realm. We expect that larger studies will generate conclusively positive results and that location-based reverberation placement will replace current techniques for artificial reverberation and localization in stereophonic recordings.

References

1. Agresti, A.: An Introduction to Categorical Data Analysis. John Wiley & Sons, New York (1996)

2. Audio Ease: Impulse Responses. (Online at: http://www.audioease.com/IR/index.html)
3. Benade, A.: Fundamentals of musical acoustics. Second, revised edition. Oxford Press, New York (1990)
4. Cheng, C. I. and Wakefield, G. H.: Introduction to head-related transfer functions (HRTF's): Representations of HRTF's in time, frequency, and space (invited paper). In: Proceedings of the 107th Audio Engineering Society (AES) 107th Convention, New York (1999)
5. Chowning, J. M.: The simulation of moving sound sources. Computer Music Journal, Vol. 1, No. 3 (Jun., 1977) 48-52
6. Dannenberg, R.: Machine tongues XIX: Nyquist, a language for composition and sound synthesis. Comp. Music Journal, **21** (3) (Fall 1997) 50-60
7. Farina, A.: Simultaneous measurement of impulse response and distortion with a swept-sine technique. In: Proc. 108th AES Convention (2000)
8. Fausti, P., Farina, A., and Pompoli, R.: Measurements in opera houses: comparison between different techniques and equipment. In: Proc. of ICA98 - Int. Conf. on Acoustics (1998)
9. Griesinger, D.: Beyond MLS occupied hall measurement with FFT techniques. 101st Audio Eng. Society Convention, Preprint 4403 (Oct. 1996)
10. Jot, J.: Efficient models for reverberation and distance rendering in computer music and virtual audio reality. In: Proc. 1997 International Computer Music Conference (1997)
11. Li, Y., Driessen, P. F., Tzanetakis, G., Bellamy, S.: Spatial sound rendering using measured room impulse responses. Signal Processing and Information Technology, 2006 IEEE International Symposium on ISSPIT 2006 (Aug. 2006) 432-7
12. Mateljan, I.: Signal selection for the room acoustics measurement. In: Proc. 1999 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (1999)
13. Moore, F. R.: A general model for spatial processing of sounds. Computer Music Journal 7(3) (Fall 1983) 6-15
14. Oppenheim, A. V. and Schafer R. W.: Digital Signal Processing. Prentice-Hall, Englewood Cliffs, NJ (1975)
15. Pulkki, V.: Spatial sound generation and perception by amplitude panning techniques. Ph.D. Dissertation, Helsinki Univ of Technology (2001)
16. Reller, C. P. A., Jawksford, M. O. J.: Perceptually motivated processing for spatial audio microphone arrays. 115th Audio Engineering Society Convention, preprint 5933 (Oct. 2003)
17. Youngblut, C., Johnston, R., Nash, S., Wienclaw, R., and Will, C. Review of Virtual Environment Interface Technology. IDA Paper P-3186. Alexandria, VA: Inst. for Defense Analysis (IDA) (Mar. 1996) (Online at: http://www.hitl.washington.edu/scivw/scivw-ftp/publications/IDA-pdf/)