Augmenting Room Acoustics and System Interaction for Intentional Control of Audio Feedback

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ABSTRACT

This paper presents the interactive enhancement of audio feedback through context-based control, leading to the generation of desired sonic behaviors by augmenting the effects of physical space in the feedback sound. Our prototype maps approximations of room reverberation to tempo-scale characteristics of the audio feedback. These characteristics are generated by a combination of adaptive amplification control and a digital variable delay line in the feedback loop. Room reverberation is inferred from the feedback sound by real-time cross-correlation of input with output signals, which is used to guide the variable delay line. This variation, coupled with an adaptive gain control, determines room-dependent tempo effects.

1. INTRODUCTION

Audio feedback is an acoustic phenomenon that occurs when sound played by a loudspeaker is received by a microphone to create a persistent loop through a sound system. While audio feedback is generally regarded as an undesired situation, for example when a PA system manifests an unpleasant howling tone, there have been numerous artistic examples and compositions that make use of its tone-generating nature. Jimi Hendrix is an oft-cited example of how electric guitar players create feedback-based tones by holding their instruments close to the amplifiers, and Steve Reich's *Pendulum Music* (1968) features phasing feedback tones generated by suspending microphones above loudspeakers.

A modern approach to audio feedback in experimental improvisation and compositional works utilizes computer-based control over sound generation and organization: examples are introduced in [1]. Di Scipio's *Audio Ecosystems* [2] is a prominent example in which a self-feeding feedback loop interconnects a digital system with its acoustic environment.

By inserting a network of low-level components, represented by a chain of transducers and other acoustic components inserted into the loop, these audio feedback systems can lead to nonlinear behaviours that Di Scipio refers to as the emergence of high-level behaviours [2], since specific performances cannot be accurately predicted in advance. Many audio feedback systems have focused on design of the low-level relations to generate and organize the feedback sounds while paying less attention to control over the overall sonic shape [3].

We previously proposed a new concept of audio feedback systems [4] that supports intentional control through tendencies while preserving attractive nonlinear features, which could open up new possibilities of musical applications that combine nonlinearity and interactivity. In this paper, we explore this concept further by taking account of the relation between system and room acoustics.

Room acoustics are an essential yet under-examined factor in the shaping of audio feedback. Our work is designed to augment the interaction between system and room acoustics. This supports the intentional control of audio feedback through the generation of long-term sonic behaviours that respond appropriately to the acoustics of the environment. Our prototype maps inferences of room reverberation to tempo characteristics of audio feedback. Reverberation is a key recognizable characteristic of the acoustic environment, while tempo characteristics are a rarely considered property of audio feedback. Cross-correlation of input with output is used to infer the amount of reverberation in the space, which in turn is used to vary the length of a delay line. This variation, coupled with an adaptive gain control, determines tempo characteristics through the occurrence rates of Larsen tones. We have explored two mappings which enable both fast/slow tempo in strongly/weakly reverberant spaces.

2. RELATED WORK

In audio feedback systems, a complex of computational signal-processing components and a physical space are connected through sound, typically mediated by microphones and loudspeakers. The generated sound diffuses in the room, reflected by walls and other objects, and re-enters the computer via the microphones. No emitted sound will re-enter the system unmodified, and in addition environmental noise will be included in the input signal to further stimulate the system. The physical, acoustic part of the system can be regarded as

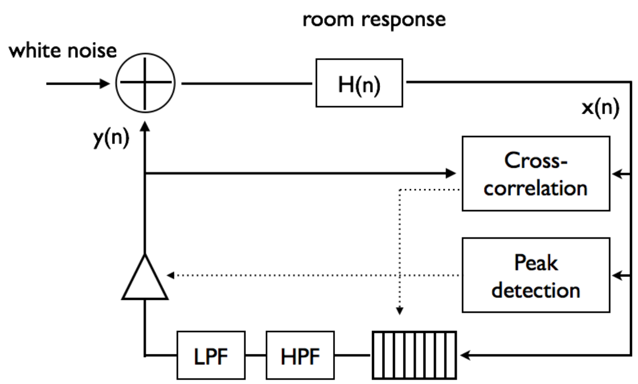

Figure 1: Overview of the system for simulating room reverberation-based tempo control in audio feedback, which mainly consists of an adaptive gain control and delay control based on cross-correlation between input and output.

a medium for relationships between sonic agents (including circular links), or it can be considered as another agent in the network.

The acoustic characteristics of a room, such as resonance and reverberation, influence all sound received. The long-term dynamics of Larsen tones (e.g., stable frequencies) are known to be strongly determined by the resonant modes of the chamber in which they are placed as well as the placement of microphones and speakers.

Our system extends such dependency in a system by teasing out specificities in sonic feedback systems and mapping them to control parameters, generating long-term sonic behaviours that significantly respond to the acoustics of the environment. The structure is similar to that of the *Audible Ecosystems* (also 'Control' Information Rate in [1]): features extracted from the received sound are used as parameters for sound synthesis. In our system, however, the selected feature is explicitly designed to be sensitive to information regarding the acoustic environment, in order to augment the specific effects of any particular physical space.

Syntaxis [5] is also a feedback system sensitive to the acoustic environment. It uses a genetic algorithm to gradually evolve bandpass filter banks toward the resonant peaks in acoustic feedback, and thus the total system adapts to the acoustic characteristics of an physical space. Our system focuses on control of feedback sounds following a composer's intention, by using the received features to generate desired tendencies of sonic behaviour.

3. SYSTEM DESIGN & VALIDATION

Figure 1 shows the overall diagram for simulating tempo effects of audio feedback and validating the mapping using computer software. The amplifier is adaptively controlled based on the peak amplitude of the input signal. Convolution of an impulse response simulates the room acoustic response, which is shaped by propagation and reflections in physical

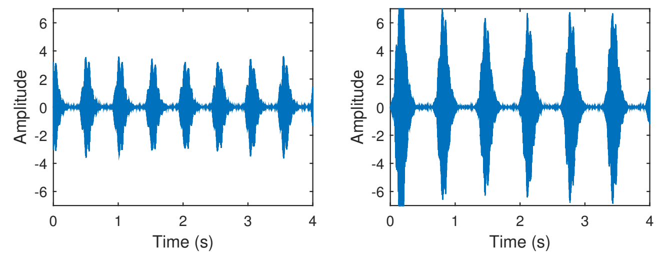

Figure 2: Simulated tempo effects of feedback sounds when length of the delay line is (a) 22000 samples (2 Hz) and (b) 28500 samples (1.55 Hz)

space. We used impulse response data from Fokke van Saane [1] and Aleksey Vaneev [2]. A noise function simulates room ambient noise, used as an excitation energy source in the feedback loop. We used a high-pass filter with a cutoff frequency below 80Hz to remove excessive amplification of low frequency feedback, which is uncommon in a real acoustic environment. The low-pass filter is used similarly to the low-pass damping loop filter of a waveguide model, to reduce the howling that can be introduced by excessive amplification.

3.1 Tempo effects of feedback sounds

Tempo effects in the feedback sounds emerge from a combination of the adaptive gain control and the long delay line in the feedback loop. The synthesis of Larsen tones requires positive amplification of feedback, and once established, negative feedback to prevent saturation. This is achieved in our system by increasing/decreasing the signal amplitude by one percent in each period (3ms). Amplification or de-amplification is determined by comparison to high/low threshold values. When the peak amplitude of the signal over a three millisecond window exceeds the high threshold value (0.7), de-amplification is applied, otherwise the signal is amplified when the peak amplitude does not exceed the low threshold value (0.3). When the length of a delay line set to more than approximately 5000 samples (e.g., 100ms), this range is more relevant to tempo-scale periodic occurrences of Larsen tones (see Figure 2).

3.2 Measurement of reverberation level and mapping with tempo (delay)

It is inevitable that we can attain only approximate information about the acoustic environment from the real-time audio input, since complete real-time segregation of ambient noise and acoustic reflection information from the received feedback sound is practically impossible. Accepting this limitation, we have designed our system to infer what properties it can, and use them to augment the specificity of the result.

The cross-correlation of input $x[n]$ with delayed output $y[n]$ is used to derive an approximate measure of the amount of re-

[1] http://fokkie.home.xs4all.nl/IR.htm
[2] http://www.voxengo.com/impulses

Room Types	RT60 (s)	Xcorr	L1	L2
Livingroom	0.28	0.0312	23446	46991
Bathroom	0.58	0.1287	30267	42633
Church	0.97	0.3253	44426	30691
Long Echo Hall	3.07	0.4198	54570	25302

Table 1: Comparison of the reverberant characteristics measured by RT60 (the reverberation time over a 60dB decay range), the cross-correlation values when the delay line is set to 28000 samples, and average delay line lengths using the proportional (L1) and reflected (L2) mappings.

verberation in the physical space. Cross-correlation measures the similarity between two signals as a function of time-lag, and is defined as:

$$\hat{r}_{xy}(l) = \frac{1}{N}\sum_{n=0}^{N-1} \bar{x}(n)y(n+l), \qquad (1)$$
$$l = D_1, D_1+1, D_1+2, ..., D_2.$$

If a sound causes strong reverberation, the reflected sound is also large and cross-correlation is expected to be high. In order to exclude direct (non-reflected) sound propagation, the cross-correlation value is taken as the mean over time-lags from 5000(D_1) to 18000(D_2) samples (i.e. from 113 to 408 ms). This value is then divided by the maximum value over time-lags from zero to 18000 samples to minimize effect from amplitude of the feedback sound from the system. (Without this division, a louder feedback sound would also increase the cross-correlation value).

The resulting approximation of reverberation level is mapped to the length of a delay line in the feedback loop, over a range of approximately 20000 to 55000 samples (450 to 1250ms). In this experiment, two opposite mappings were investigated for fast/slow tempo in strongly/weakly reverberant spaces, defined as:

$$L_1 = a_1 * r + c_1$$
$$L_2 = -a_2 * r + c_2 \qquad (2)$$

where L_1 and L_2 mean delay line length and r means cross-correlation value. The constants were chosen to achieve delay times of approximately 20000 to 55000 samples, such as a_1 = 80000, c_1 = 20000, a_2 = 60000, c_2 = 50000. Figure 3 evaluates the two mappings and Table 1 compares the average delay line lengths and the reverberation characteristics (RT60 and the cross-correlation values), using the impulse response data in several locations. These results affirm that the mapping results in intentional control of tendencies in audio feedback according to inferences of room reverberation.

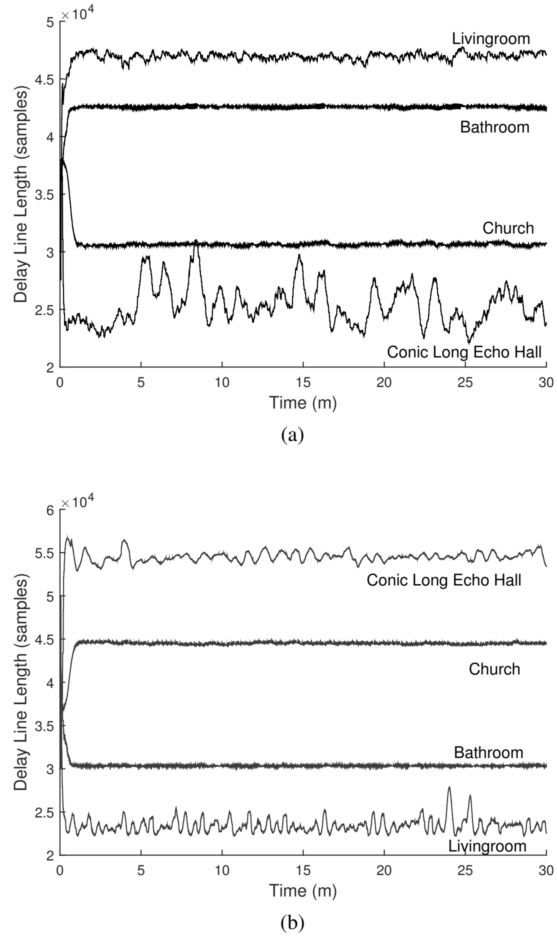

Figure 3: Delay line length curves simulated using impulse response data in several locations with different reverberant characteristics, using the (a) proportional (L_1) and (b) reflected (L_2) mappings .

4. EXPERIMENTS

The designed dependencies were implemented as software authored using Max/MSP and openFrameworks (Figure 4), and investigated by experiments in rooms with different reverberation characteristics; a 4x3m office room and a 2x2m restroom. Tests were performed with external audio devices [3] : they were centered in the room and the distance between a microphone and a loudspeaker was identical. In each location, an impulse response was simply measured using a sine sweep signal (the method is demonstrated in [6]), and RT20 (the reverberation time over a 20dB decay range) values are approximately measured as 0.11 and 0.14 seconds.

Fig. 5 evaluates the two mappings to control delay line length from the cross-correlation value. Although the differences are not as clearly distinguishable compared to those from the simulated rooms, we still observe differences in the long-term averages as expected, both when using the propor-

[3] M-Audio BX5a and Apogee MIC

tional mapping (10860 and 10260 in high and low reverberant places respectively), and also with the reflected mapping (16790 and 17370 in high and low reverberant places respectively).

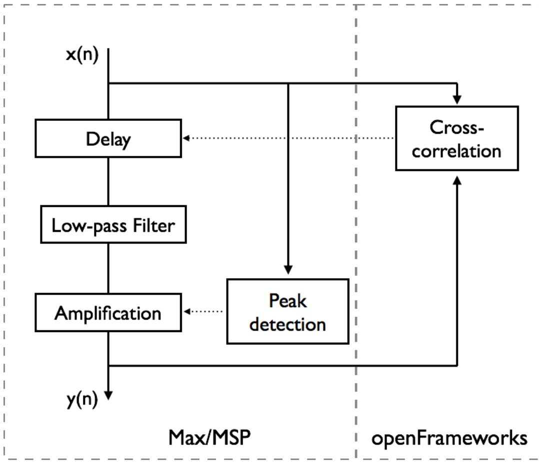

Figure 4: Overview of the system implemented as software using Max/MSP and openFrameworks, which was evaluated in rooms with different sonic characteristics

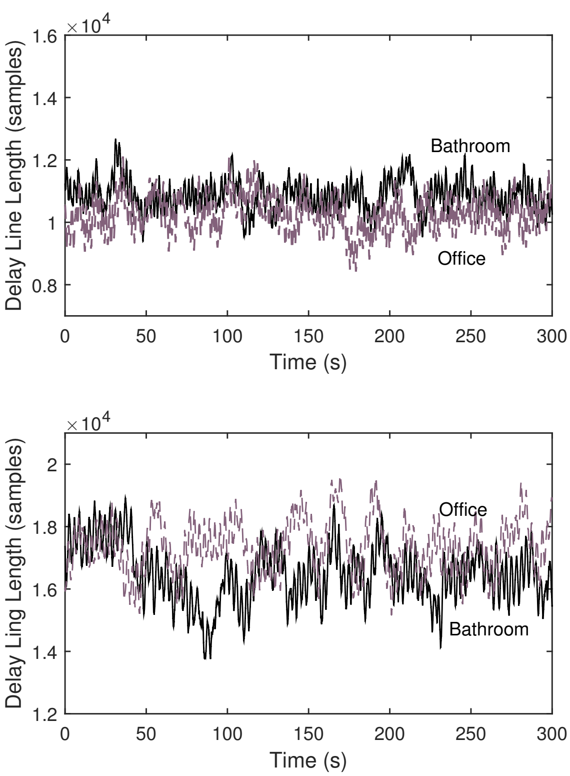

Figure 5: Delay line length curves measured in the strongly reverberant place (black solid curves) and weakly reverberant places (brown dash curves) using the (a) proportional and (b) reflected mappings.

5. CONCLUSION

In this paper we presented research progress augmenting composed audio feedback interactions within acoustic environments; supporting sound generation that strikes a balance between unpredictable short-term behavior and intentional long-term tendencies, considered as viability conditions for natural processes to follow. We designed the long-term tendencies in terms of tempo characteristics depending upon reverberant properties inferred automatically from the environment, and measured this design through simulations as well as acoustic experiments.

Beyond regarding the environment as a filter and source of disturbance, it can also be considered a site of discovery. The composed system attempts to differentiate and affirm itself through reflections, yet by doing so augments or exaggerates the specificity of the environment. This duality is also evident in the analysis: comparing input and output signals cannot fully segregate external and internal influence, as the feedback sounds depend upon parameters within the system, which in turn depend on the analysis. We are satisfied that affirming specificity was achieved, but we believe this is only an initial step in developing truly adaptive, self-augmenting responsive sonic environments.

6. REFERENCES

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