REVERBERATION ENHANCMENT IN MUSIC PRACTICE ROOMS

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1 INTRODUCTION

The acoustic response of a room can have dramatic effects on its usability. Different types of music, as well as speech and drama, have different ideal acoustics. For this reason halls have been constructed with variable physical properties such as extendable acoustic absorption [1]. Another possible method is through the use of electronic reverberation enhancement. This technology was originally developed to remedy poor acoustics in halls which had insufficient reverberation time [2]. Further developments have allowed greater variety of applications such as in multi-purpose halls.

Although there have been many successful commercial implementations of reverberation enhancement, these have almost exclusively been designed for large rooms such as concert halls. The technology may also be useful in smaller spaces but this requires additional considerations due to the different acoustic properties of the room. This work examines the possibility of applying reverberation enhancement in a smaller space, specifically for application in music practice rooms. Currently available systems will be discussed and are related to the motivation for this work.

Although many possible considerations are raised by this problem, this paper will examine in detail a new method for implementing reverberation enhancement. This is the use of a simple delay in the feedback loop. A mathematical analysis of this setup will be presented along with numerical simulations and experimental results which test the accuracy of the analysis. Finally, this method will be compared to the existing alternatives.

1.1 Reverberation Enhancement

There are two main design strategies for electronic reverberation enhancement systems. Both consist of loudspeakers and microphones connected via processing with varying degrees of complexity. The difference between the two strategies is in the placement of the transducers as illustrated in Figure 1. In-line systems place the microphones as close as possible to the performers, without being intrusive. The loudspeakers are directed away from the stage to minimise feedback to the microphones. This type of system is similar in many ways to a public address system, the main difference being that its operation is hidden from the audience.

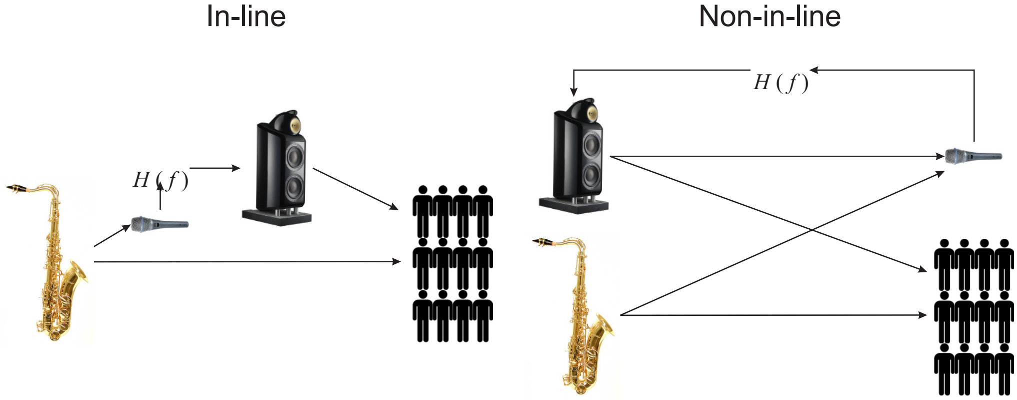

Figure 1: Comparison of in-line and non-in-line reverberation enhancement systems showing the difference in signal path.

The other design strategy is non-in-line in which the microphones are located remotely from the performers. Feedback between the loudspeakers and microphones of the system is an integral part of the operation of these systems. Non-in-line systems are designed to increase the reverberation time homogeneously so that the position of the performer and audience within the room does not significantly affect the resulting sound.

Another important aspect of the operation of a reverberation enhancement system is the processing which is applied to the microphone signal (labelled H in Figure 1). This can consist of a simple gain such as in the non-in-line MCR system [3]. More complex processing is often included to improve the performance of the system without requiring an increase in the channel count. Modern systems often include digital reverberation (LARES [4] & VRAS [5]) which allows significant increases in resultant reverberation time. As this type of processing increases the probability of instability [6] these systems also include measures to mitigate this risk. LARES includes time variant processing [7] whereas VRAS uses a bespoke all-pass reverberator [8].

1.2 Motivation

Although modern reverberation enhancement systems allow significant increases in reverberation time, they are inherently complex and based on technology which is not freely available. Simplification of the technology may allow a reduction in cost and an improvement in the ease of setup. This would allow application of reverberation enhancement to small scale projects such as music practice rooms.

One possible processing strategy is the use of delay. This has been used for in-line systems such as Ambiphony [9] and ERES [10]. However these systems are not well suited to music practice rooms because the audience is also the performer and therefore it is impractical to avoid feedback as is required for proper operation of an in-line system. LARES systems, which use an in-line configuration, have been used for music practice rooms by using directional microphones and specially constructed, semi-anechoic rooms [11]. If similar performance were possible with a non-in-line system, the convenience of installation may be preferable.

This work examines the effect of including delay in a non-in-line reverberation enhancement system. Delay is easily implemented and it will not affect the stability condition of the system as it does not alter the amplitude response within the feedback loop [6]. This type of system will be simpler than the above mentioned systems which use electronic reverberation. If this system can provide similar gains in reverberation time, then it will be well suited to application in music practice rooms. This work investigates the possible performance of this new system.

2 ANALYSIS

The behaviour of a non-in-line reverberation enhancement system including delay can be modelled using the energy balance equation. Under the assumption of a diffuse field, the dynamic energy density ε within the room can be approximated as

$$V\frac{d\varepsilon(t)}{dt}+\frac{c\alpha S}{4}\varepsilon(t)=P_s \tag{1}$$

where V is the volume of the room, t is time, c is the speed of sound, α is the average absorption coefficient of the surfaces in the room, S is the surface area and P_s is the sound power of the source [12]. The reverberation enhancement system can be included with an additional source term which is proportional to the energy density [13] which can then be suitably delayed. The equation then becomes

$$V\frac{d\varepsilon(t)}{dt}+\frac{c\alpha S}{4}\varepsilon(t)=P_s+Ng^2\varepsilon(t-\tau) \tag{2}$$

where N is the number of channels, g is the feedback gain and τ is the delay. For simplicity, it is assumed that all channels have the same feedback gain and delay.

As the reverberation time is the only quantity of interest, the source term P_s can be neglected and the homogeneous case can be solved. This is achieved by substituting a trial solution of the form

$$\varepsilon(t) = e^{-at} \tag{3}$$

with decay rate a. The reverberation time is related to the decay rate by $T = 13.8/a$. Substitution of equation (3) into equation (2) leads to the characteristic equation

$$a - \frac{c\alpha S}{4V} + \frac{Ng^2}{V} e^{a\tau} = 0\,. \tag{4}$$

Equation (4) can be simplified by substituting the unaltered decay rate of the room a_0 using the Sabine formula $T = 13.8 \times 4V/c\alpha S$ and defining a normalised feedback gain $\mu^2 = g^2\, 4/c\alpha S$ so that

$$a - a_0 + a_0 e^{a\tau} \mu^2 N = 0\,. \tag{5}$$

As this formula contains linear and exponential terms of a, the solution is not immediately obvious. However, it is possible to solve by using the Lambert W function [14]. This is a generalised function which cannot be written in closed form. Instead it is defined as the inverse of

$$we^w = x \tag{6}$$

so that w is a function of x. This function is multi-valued, but for the purposes of this analysis only real positive inputs and real values of the output will be considered. This function can be approximated numerically [15] and implementations are available in several commercial software packages such as MATLAB and Mathematica.

In order to solve equation (5), it must be rearranged so that it is in the same form as equation (6). This can be achieved by multiplying through with a factor of

$$\tau e^{\tau(a_0 - a)}\,.$$

By rearranging the resulting equation and then applying the Lambert function to both sides, it can be shown that

$$W\left[a_0 \tau e^{a_0 \tau} \mu^2 N\right] = \tau\left(a_0 - a\right) \tag{7}$$

where W denotes the Lambert function. From this it is trivial to show that

$$a = a_0 - \frac{1}{\tau} W\left[a_0 \tau e^{a_0 \tau} \mu^2 N\right] \tag{8}$$

which can then be used to predict the resultant decay rate using the unaltered decay rate, delay, feedback gain and channel count. The results of this analysis can be seen in Figure 2 showing that the inclusion of delay allows large increases in performance.

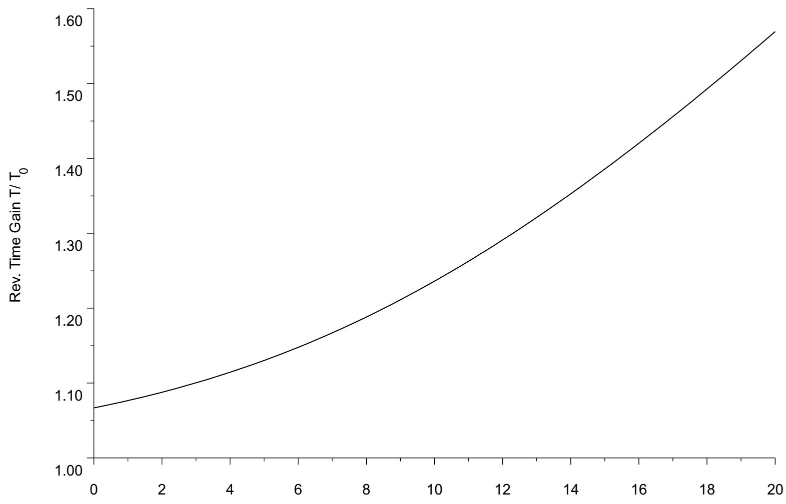

Figure 2: Reverberation time gain against delay as a percentage of unaltered reverberation time. The modelled system has a single channel and a feedback gain of −12dB.

3 SIMULATION

To test the accuracy of the analysis presented in the previous section, a numerical simulation routine has been implemented. This is a time domain simulation where each transfer function between loudspeaker and microphone is represented as an FIR filter. The room impulse response is modelled using exponentially decaying white noise as this represents the most important characteristics of a generic room response [16]. An impulse is output from a primary source and the response is “measured” at several virtual microphone positions. The reverberation time is then evaluated from the impulses recorded by these “microphones” using Schroeder’s reverse integration method [17].

One of the main advantages of this simulation method is that a variety of enclosures and systems can be modelled. Additionally, more complex simulations of the room impulse response such as an image source model [18] or ray tracing [19] could easily be included in order to investigate the effect of reverberation enhancement on a specific room. However, in the interest of brevity, only a small selection of results will be included here.

In order to validate the accuracy of the simulation itself, a system without delay has been modelled. This can then be compared to the standard diffuse field analysis, originally derived by Franssen [20], of a reverberation enhancement system without processing in the feedback loop. In this case the resultant reverberation time can be related to the unaltered room reverberation time by

$$T = \frac{T_0}{1-\mu^2 N}\,. \tag{9}$$

The simulated reverberation time of this system is shown in Figure 3 along with that predicted using equation (9). The target reverberation time of the unaltered room was 1 s but the errorbars show that there is inherent variation between the generated impulse responses. It can be seen that the predicted value is within the variations in reverberation time except in the 2 kHz octave band. This implies that the simulation is sufficiently accurate to assess the analysis presented in the previous section.

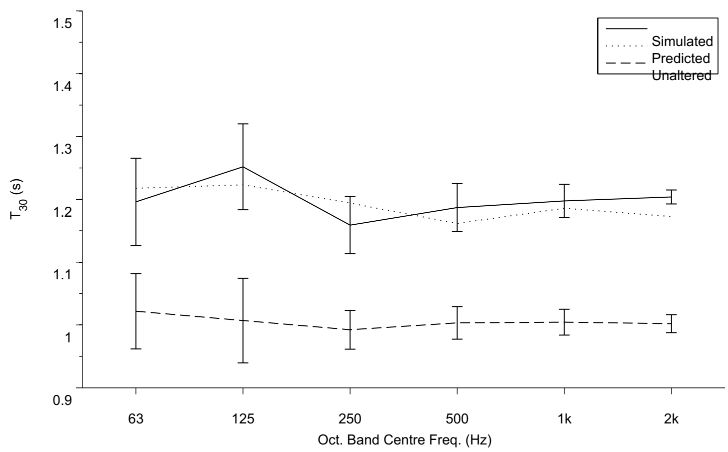

Figure 3: Reverberation times in octave bands for a room including a reverberation enhancement system with 4 channels and no delay. The errorbars show one standard deviation above and below the mean. The feedback gain is −15 dB.

The simulation of the system including delay was implemented by simply adding zeros to the start of the impulse responses which model the reverberation enhancement system. In order to display a wider range of data, the mid-frequency reverberation time T_{mid} is used. This is the mean of the reverberation times in the 500 Hz and 1 kHz octave bands.

The results of the simulation of the system including delay are shown in Figure 4. This shows a comparison between the simulated resultant reverberation times and those predicted using equation (8). From this, it can be seen that the predictions agree well with the simulated values. Three different unaltered room responses were tested and the analysis appears to work equally well for all three. This implies that the analysis is valid.

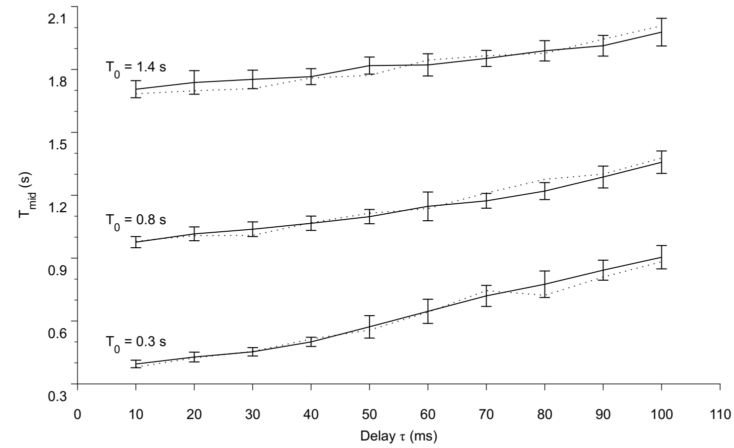

Figure 4: Mid-frequency reverberation times for a 4 channel reverberation system with variable amounts of delay. The solid line is the simulation and the dotted line is predicted. Three different unaltered reverberation times are shown. The feedback gain is −15 dB.

4 EXPERIMENT

In order to validate further the analysis from section 2, an experiment has been conducted. A single channel system was constructed using a laptop to add delay between the microphone and loudspeaker. A National Instruments data acquisition system was used to measure the feedback gain and delay within the reverberation enhancement system as well as measuring the reverberation time of the room. The reverberation time was measured using a swept sine excitation signal which was then processed to extract the impulse response of the room by deconvolution [21]. The impulse response was then used to measure the reverberation time using reverse integration [22].

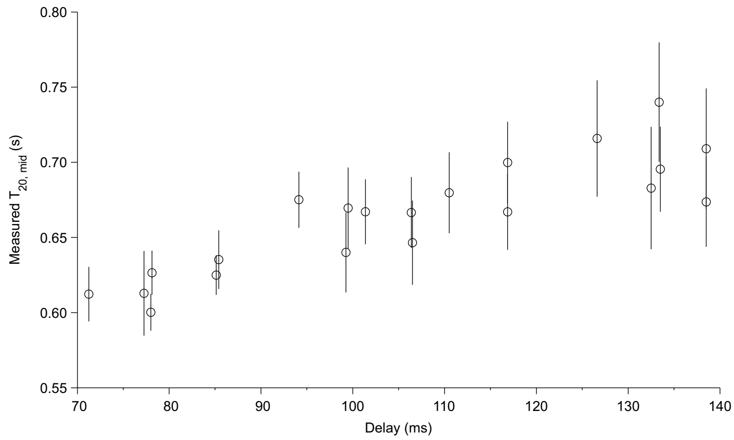

Figure 5: Measured reverberation time against delay. The errorbars show one standard deviation above and below the mean.

Due to the limitations of the available hardware, the minimum delay in this experiment was 70 ms. Values of delay up to 140 ms were tested. For each value of delay tested, the reverberation enhancement system was placed in three positions. This slightly alters the delay due to the separation of source and receiver. Additionally, small changes in feedback gain are observed. For each system position, the delay and feedback gain were measured. These values can be used, along with the measured, unaltered reverberation time of 0.53 s, to predict the resultant reverberation time using equation (8).

The results from this experiment are shown in Figure 5. This shows that the inclusion of delay allows large increases in reverberation time to be realised. The predicted values are compared to measured data in Figure 6 which shows reasonable agreement as long as the increase in reverberation time is not too large. For large increases, particularly the values of T above 0.70 s, the variations in reverberation time become larger, implying significant spatial variation in reverberation time. As a diffuse field is homogeneous, increases in spatial variation can be interpreted as a reduction in the diffusion of the sound field [23]

For values predicted above 0.75 s, the measured reverberation times are lower than the predictions. These points relate to the system including 140 ms of delay which is a quarter of the unaltered reverberation time. From this, it could be inferred that the analysis of section 2 is invalid for values of delay which are a significant proportion of the unaltered reverberation time.

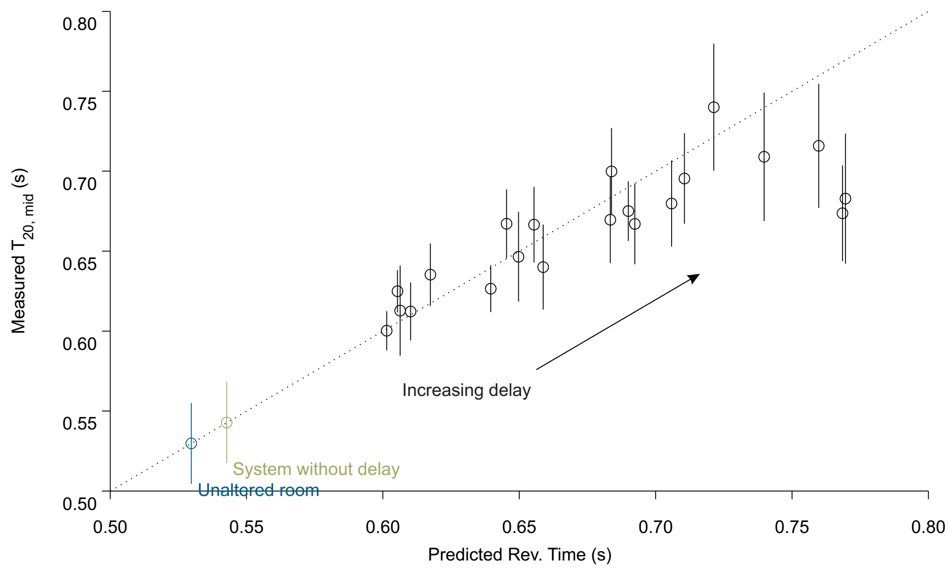

Figure 6: Comparison of the measured reverberation time against prediction. The dotted line indicates unity between measurment and prediction. The reverberation time of the unatlered room and that caused by a system without delay are included for the purposes of comparison.

The results of this experiment show that the analysis is accurate for values of delay which are not too large. More data would be required to ascertain the range of values of delay over which the analysis is accurate. The data from this experiment implies that with delays of less than a fifth of the reverberation time the results will agree with the analytical result. This amount of delay allows significant increases in performance of the system.

5 DISCUSSION

The purpose of this work has been to investigate possible methods which would allow reverberation enhancement to be applied to music practice rooms. The requirements for this type of system are simplicity of setup and operation and a low channel count due to space and budget constraints. The experimental results from the previous section showed that significant increases in reverberation time are possible even with a single channel system. However, this performance may have been achieved using other currently available systems which do not use delay. It is important to consider whether this new system fulfils the requirements of simplicity and low channel count as well as or better than other currently available systems.

The systems mentioned in section 1 which would allow comparable gains in reverberation time with similar channel counts are the LARES and VRAS systems. Both of these use proprietary digital reverberation algorithms. Measures must be taken in both systems to mitigate the change in the stability condition caused by the inclusion of electronic reverberation. For these reasons these system are significantly more complex to implement that the new system presented here.

The possible gain in reverberation time of the VRAS system is limited to avoid double sloped decay curves [24]. With the single channel system that was tested here, which had a mean feedback gain of −16 dB, the maximum recommended reverberation time when using a VRAS system is 0.65 s. As can be seen in Figure 5, this was achieved using a delay of 100 ms without a significant increase in the spatial variation of reverberation time implying that the diffusion of the sound field is not decreased. Similar performance is possible with the new system without the additional complexity required to implement current system such as VRAS. However, further investigation is required to see if the new system causes double sloped decays or other unwanted artefacts such as audible echoes.

6 CONCLUSIONS

This work investigated a new reverberation enhancement system based on a non-in-line configuration including delay in the feedback loop. The simplicity of this system may allow the technology to be used for small scale applications such as music practice rooms. Currently available systems allow large increases in reverberation time, but they often include proprietary digital algorithms to create electronic reverberation and reduce the risk of instability. The new system has been shown to allow similar increases in reverberation time but is significantly simpler to implement.

A mathematical analysis has been presented of the new system. In order to test the accuracy of this analysis, numerical simulations have been implemented and an experiment has been conducted. These have shown that the analysis is accurate. However, the experimental results showed that when the delay is a large fraction of the unaltered room reverberation time, above 25 %, the accuracy of the analysis decreases.

There is scope for further work to investigate the effect of the new system on the early part of the impulse response and, specifically, whether double sloped decay curves are created. However, it has been shown that the new system allows significant increases in reverberation time without a large channel count and whilst retaining simplicity of implementation.

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