THE AUDIBILITY OF COMB-FILTERING DUE TO PERFORATED CINEMA SCREENS

David Elliott ISVR, University of Southampton
Keith Holland ISVR, University of Southampton
Philip Newell Acoustics consultant, Moaña, Spain

1 INTRODUCTION

In cinema audio, the sound needs to be combined to try and create an atmosphere in which the visual and audio images work in conjunction with each other to help engage the viewers. The loudspeakers positioned behind the cinema screen are integral in creating a localized image that matches the positions of the video sources. This means that current cinema screens need to be optimized for both image reflection and sound transmission. Placing the loudspeakers above or below the screen results in a less satisfactory experience with other attendant problems [1]. Because placing the loudspeakers behind a normal, non-perforated, screen would give rise to large amounts of attenuation at high frequencies, perforated screens have been developed to "let the sound through". Currently no material exists that has the qualities required for both perfect reflection of light and perfect transmission of sound. This means that trade-offs need to be made to enable a screen to meet the requirements of both the reflection of light and transmission sound at acceptable level. For years, the quality of the image has dominated cinema specifications, partly due to the fact that early films did not include particularly high quality sound. However, the arrival of digital soundtracks, and as the science of psycho-acoustics has become better understood, the realization that audio quality is greatly important for the overall realism of the cinema experience has led to more research into improving the sound whilst trying to maintain the high definition required for the projection of a high-quality image on to a large screen.

The presence of the screen gives rise to a phenomenon known as 'comb filtering', arising from the addition of delayed versions of an original signal to itself. The severity of the comb-filtering will vary depending on the geometric setup of the system. In particular, varying the distance between the loudspeaker and the cinema screen can change the nature of the comb filtering that the system exhibits. This, in turn, will alter the frequency content of the final signal reaching the listener, and it is this final signal that is essential to the cinema experience.

This paper focuses on characterizing the actual audibility of a comb-filtered signal by means of signal processing. The cinema screen's acoustic properties are used to create a model that simulates the effect of playing a sound through the loudspeaker/screen system. The audibility or otherwise of the comb-filtering is then assessed via a subjective listening test.

In 2013 Newell, Garcia and Holland [2] conducted experiments on the effect of a cinema screen on loudspeaker performance. As a continuation of this research, the experimental data recorded is used here to "fine-tune" the system model to ensure that it behaves in a similar way to an actual physical setup.

2 SYSTEM MODEL

2.1 Comb-Filtering

Consider the loudspeaker/screen system shown in Figure 1. The sound radiated by the loudspeaker is incident upon the screen where some of it is transmitted through, to continue onwards, and some is reflected back towards the loudspeaker. The reflected wave is then re-reflected back towards the screen where it is again partially transmitted and partially reflected. This process continues until the wave has lost enough energy to be considered negligible. Figure 2 shows the effect that these multiple reflections have on the impulse response of an idealised system. In reality, the transmission and reflection properties of the screen are frequency dependent so the modelling of the system is simpler to implement in the frequency domain.

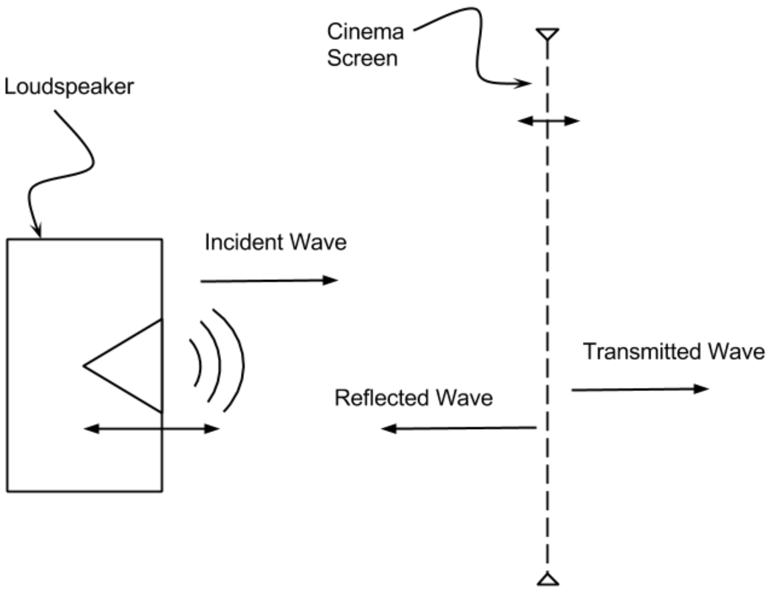

Figure 1 The loudspeaker / screen system

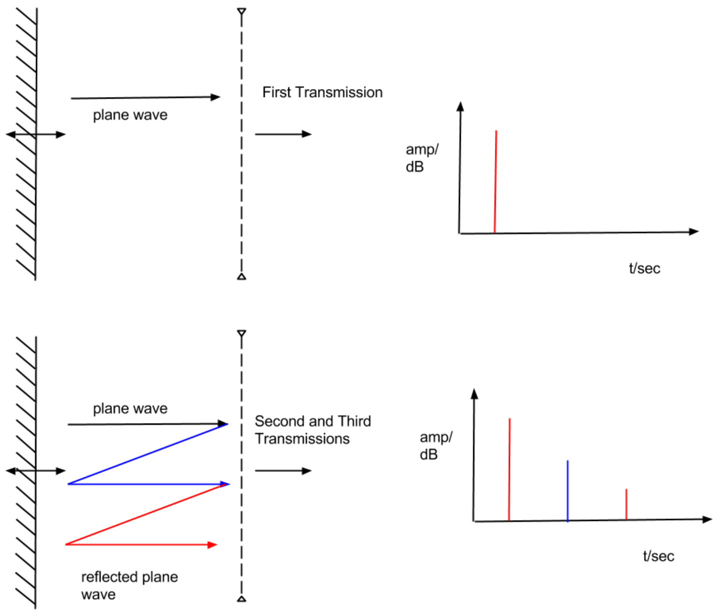

Figure 2 Build up of impulse response due to multiple reflections by screen and loudspeaker

2.2 System Frequency Response

Let $T(f)$ and $R(f)$ be the transmission and reflection coefficients of the screen. Assuming a perfect loudspeaker, the frequency response of the loudspeaker/screen system after only the initial wave has passed through the screen is

$$\hat{H}_1 = \hat{T}(f)e^{-j\omega t_\Delta} \quad , \tag{1}$$

where t_Δ is the delay due to the propagation from the loudspeaker to the screen. After the wave is reflected back to the loudspeaker and then back to the screen, the response becomes

$$\hat{H}_2 = \hat{T}(f)e^{-j\omega t_\Delta}\left(1 + \hat{R}(f)e^{-j\omega 2t_\Delta}\right) \quad , \tag{2}$$

and after N reflections, it becomes

$$\hat{H}(f) = \hat{T}(f)e^{-j\omega t_\Delta}\sum_{r=0}^{N}\hat{R}(f)^r e^{-j\omega 2rt_\Delta} \quad . \tag{3}$$

Accounting for the attenuation of the wave due to spherical spreading, the frequency response at an observer position a fixed distance d from the loudspeaker is then

$$\hat{H}(f) = \frac{\hat{T}(f)}{4\pi d}e^{-j\omega t_d}\sum_{r=0}^{N}\frac{\hat{R}(f)^r}{(1+2r)}e^{-j\omega 2rt_\Delta} \quad . \tag{4}$$

where t_d is the delay due to propagation from the loudspeaker to the observer position.

2.3 Screen Model

Assuming that the tension in the screen, and hence any transverse wave propagation, can be neglected, along with any viscous losses, a perforated screen can be modelled as a limp, porous mass. There are then two mechanisms by which an incident sound wave can be transmitted through the screen; the screen material can be moved by the incident wave as a limp mass with the resultant motion re-radiating the sound, and/or, the sound can pass through the perforations. Following [3], such a system can be considered as two masses in series, giving a resultant mass of

$$m_{res} = \frac{m_{screen}\, m_{perf}}{m_{screen} + m_{perf}} \quad , \tag{5}$$

where m_{screen} is the mass per unit area of the screen and m_{perf} is the effective mass per unit area of the air in the perforations. Given the perforation density D (% area) and the diameter of one perforation d, the mass of the screen is given by

$$m_{screen} = \left(1 - \frac{D}{100}\right)m \quad , \tag{6}$$

where m is the mass per unit area of the screen material, and the effective mass of the air in the perforations is estimated as

$$m_{perf} \approx \rho_0 \left(e + \frac{1.6d}{2} \right) \frac{100}{D} \quad (7)$$

where e is the thickness of the screen. For the screen modelled in this paper, m = 0.43 kgm-2, e = 0.3 mm and D = 1.7%.

2.4 Sound Transmission and Reflection

The net acoustic pressure applied to the screen can be expressed in terms of the complex amplitudes of the incident wave A, the reflected wave B and the transmitted wave C, thus:

$$\hat{p} = \hat{A} + \hat{B} - \hat{C} \quad , \quad (8)$$

and the velocity of the combined limp mass/perforation screen is

$$\hat{u} = \frac{\hat{A} - \hat{B}}{\rho_0 c_o} = \frac{\hat{C}}{\rho_0 c_0} = \frac{\hat{p}}{j\omega m_{res}} \quad . \quad (9)$$

Combining Equations (8) and (9), the transmission through the screen is given by

$$\hat{T}(f) = \frac{\hat{C}}{\hat{A}} = \frac{2\rho_0 c_0}{j\omega m_{res} + 2\rho_0 c_0} \quad , \quad (10)$$

and the reflection from the screen

$$\hat{R}(f) = \frac{j\omega m_{res}}{j\omega m_{res} + 2\rho_0 c_0} \quad .$$

(11)

Figures 3a and 3b show the transmission and reflection factors for the screen modelled in this paper as a function of frequency.

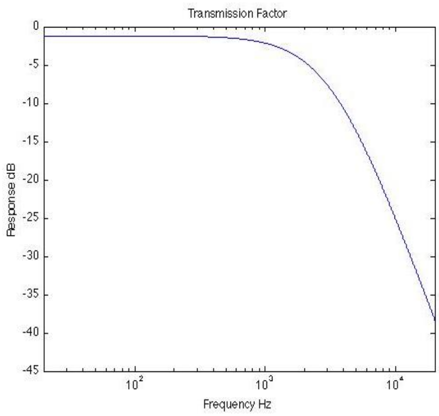

Figure 3a Screen transmission factor

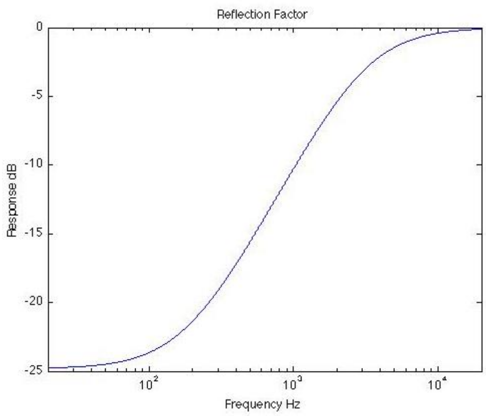

Figure 3b Screen reflection factor

2.5 Screen Loss Equalisation

The screen transmission factor in Equation (10) and shown in Figure 3a acts as a low-pass filter, attenuating the high frequencies. As the subjects taking part in this test were more accustomed to hearing more uniform spectral responses, and in order to aid the A/B/X comparison with the direct, unfiltered sound, the overall screen-loss component of the filtered sounds were compensated for. To this end, the system frequency response used in the subjective tests is as Equation (4) but with the transmission factor removed, thus:

$$\hat{H}_e(f) = \frac{e^{-j\omega t_d}}{4\pi d} \sum_{r=0}^{N} \frac{\hat{R}(f)^r}{(1+2r)} e^{-j\omega 2 r t_\Delta} \ . \tag{11}$$

3 THE SUBJECTIVE TEST

3.1 Aims and Hypothesis

The main goal of the subjective testing was to determine how audible comb filtering actually is when compared to a reference signal that represents a perfectly transmitting cinema screen. This meant that the actual effect that the cinema screen had on the quality of sound could then be quantified, and it could be determined whether or not the problem needed to be addressed. In addition to the investigation into the audibility of comb-filtering in general, the effect of the variation of distance was also explored. This was done to look into whether or not there is a certain distance that yields less audible comb filtering than others, or if there is a trend in the audibility of comb filtering as the distance between screen and loudspeaker increases.

The hypothesis for these tests is that comb filtering is audible and that distance does have an effect on the audibility of comb filtering.

3.2 Generation of Test Signals

The test signal used in the subjective tests was a section of movie soundtrack containing a mix of dialogue, sound effects and music. In order to simulate the effects of the screen comb-filtering, this signal has to be filtered using the frequency response in Equation (11) for a variety of loudspeaker/screen separation distances. To achieve this, the system frequency responses were transformed into the time domain to yield a set of impulse responses which were then convolved with the test signal. It was decided that it was not necessary to attempt the convolution in real time, but instead, the calculations were carried out offline and the results stored for playback during the tests. It should be noted that to keep the impulse responses to manageable sizes, the number of reflections (N in Equation (11)) was truncated to 10, by which time the energy in the reflected impulse is considered to be negligible. In keeping with the objective measurements reported in [2], six loudspeaker/screen distances were simulated: 2cm, 7cm, 15cm, 30cm, 45cm and 60cm. Figures 4a and 4b show the equalised system frequency responses for the 7cm and 15cm spacings respectively.

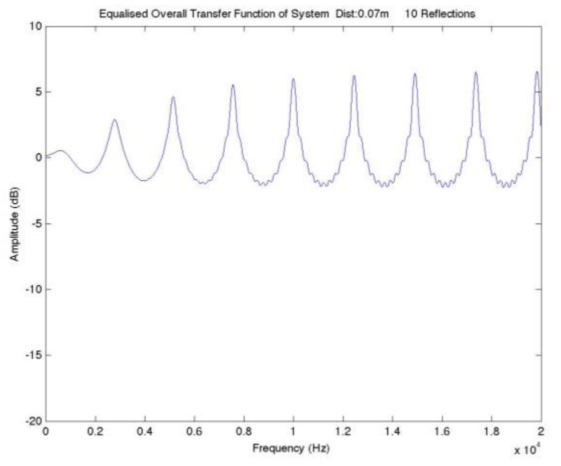

Figure 4a Equalised System Frequency Response with 7cm spacing

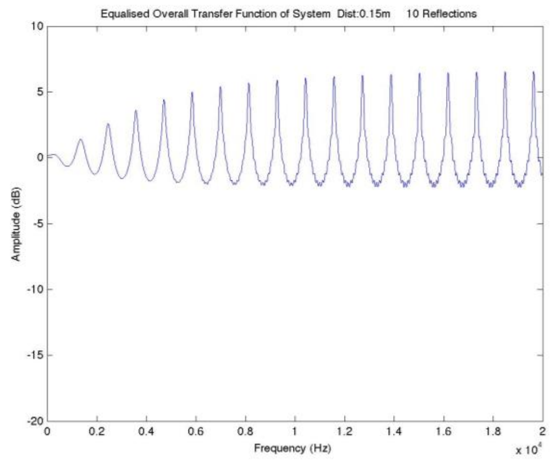

Figure 4b Equalised System Frequency Response with 15cm spacing

3.3 Experimental Setup

The experimental setup was kept as simple as possible to minimize any factors that would affect the sound heard by the subject. For this reason the tests were conducted in the ISVR small anechoic chamber, which is anechoic from 300Hz to 20kHz and has rough dimensions of 5m x 5m x 3m. The convolved samples were played through a single, high quality monitor loudspeaker. Although the chamber has a hard floor surface, the area between the loudspeaker and the subject was covered in sound absorbent material to avoid floor reflections. Consistent with the methods laid out by the ITU Radiocommunication Assembly [4], the loudspeaker was raised to 1.2m from the floor and a chair was placed at a listening distance of 2m from it, on axis. The chair was height adjustable to allow the subjects to make sure their heads were as level with the acoustic centre of the speaker as possible. The maximum sound pressure level for the tests was set to 80dB as this was deemed to be a comfortable listening level and ensured that safety and ethics requirements [5] were fulfilled.

3.4 Test Procedure

To conduct the subjective testing, an ‘ABX Comparator’ was modelled in the Matlab programming environment. This script inputs the database of filtered signals along with a reference signal, which represents the input signal being played through a perfectly transmitting screen (essentially just the input signal). The ABX method consists of two samples, A and B, being played in succession. One sample was randomly chosen as the reference signal and the other was one of the filtered signals. A third sample, X, was then played. This sample was the same as either sample A or B and the subjects’ task was to decide which one out of the two they thought it was. In total there were seven separate comparisons per test, one for each of the filtered samples and one dummy comparison where the reference signal was played twice. As well as A and B being randomized, the order in which the comparisons were played was also randomized via Matlab. The only input required from the researcher was to start each comparison once the subject was ready. The test was repeated once.

Before the test began, each subject was appropriately briefed as to both the process and their rights to end the test at any time. They were then were asked to sign a subject consent form. In turn, each subject was placed in the chair at the listening position and given a test sheet to fill out with their answers. If any subject felt the need for a dummy test, three dummy samples were played and the subject was then asked if they fully understood the process. Once this was established, the test was started. For each comparison the actual identities of each sample were noted on a researcher test sheet. Then, once the test was complete, the subject’s answers were taken, compared with the true answers and the correct choices were noted in Microsoft Excel. Results were stored both separately for the two tests and together as one overall test.

3.5 Confidence

If only one ABX trial is conducted then random guessing would mean that there is a 50% chance of choosing the correct answer each time, so if the participant guessed every answer they would get an average of 50% correct answers. In order for the results to have any degree of confidence, many trials must be performed.

If the number of trials is increased, the chance that the subject is actually able to distinguish between A & B is increased. A 95% confidence level is usually considered statistically significant. This means that if a 95% confidence level is achieved then it can be said that the samples were differentiable.

Reference [6] gives a recommended number of trials for a minimum number of correct answers, as seen in Figure 4.2 below. This is calculated using the equation:

$$N_{\min} = \frac{n}{2} + \sqrt{n} \qquad (12)$$

where n is the number of tests for one comparison. For example, if the number of trials is 10, the minimum number of correct answers is 8, but for 25 trials then only 17 correct answers are required. The test was run twice for 20 subjects.

4 RESULTS OF SUBJECTIVE TESTS

Tables 1 shows the results of the subjective tests conducted in the format that was laid out in Section 3. Tests I and II were conducted with the same twenty participants in the same session.

Test I Sample	No. Correct	Percentage	Test II Sample	No. Correct	Percentage
ref	8	40	ref	12	60
2	11	55	2	9	45
7	14	70	7	13	65
15	13	65	15	11	55
30	14	70	30	12	60
45	10	50	45	11	55
60	9	45	60	8	40

Table 1 Subjective test results

Comparing the results in Table 1 with Equation (12), it can be seen that only the 7cm sample meets the confidence level (14/20) that allows for the statement that the comb filtering is audible compared to the reference signal. However, this is only on the very boundary of the minimum number of correct answers, and if more tests were taken could easily be below. This leads to the general conclusion that comb filtering is, in fact, inaudible at all the distances tested. Further processing of the results, where the results from those subjects who scored particularly low were removed, did very little to increase the confidence that the comb filtering was audible.

5 DISCUSSION AND CONCLUSIONS

From the results obtained during this test it is safe to assume that comb filtering due to a typical loudspeaker/screen setup is not as major a problem as was originally predicted. This is testament to the fact that the only real way to measure audio depreciation is by listening to it. Since all listening tests were conducted under anechoic conditions, the likelihood is that the comb filtering effect will only become *less* audible in non-anechoic conditions. Also, for much of the time, cinema sound does not only emanate from one loudspeaker. There are multiple loudspeakers behind the screen as well as surround sound around the cinema, which can give rise to further sources of unavoidable comb filtering. So, in most cases, the comb filtering due to the screen will become even more negligible in terms of audibility compared to a single channel in an anechoic setting. In addition, the subjective tests used a comparative method to determine audibility of the differences between a comb filtered and non-comb filtered signal. It is far less likely that a subject would have been able to confidently say that there was something amiss with a signal without having instant access to a reference signal to compare it with. So, even if the comb filtering is audible when compared to a reference signal, it is very probable that a person could sit through a film and never notice that any comb filtering effect was degrading the audio.

To conclude, the results of this test suggest that the comb-filtering due to reflection from a typical perforated cinema screen is, at worst, barely audible. It may be detectable if it is specifically being listened for but it is unlikely to be a major factor in determining cinema sound quality.

6 REFERENCES

[1] M. Rettinger, “Sound Transmission through Perforated Screens,” SMPTE Journal, No.91, pp 1171-1174, 1982

[2] P. R. Newell, J. Guijarro Garcia and K. R. Holland, ‘An Investigation into the Acoustic Effect of Cinema Screens on Loudspeaker Performance', Reproduced Sound 2013, Proceedings of the Institute of Acoustics, Vol. 35. Pt. 2. 2013

[3] L. Cremer and H. A. Müller, “Principles and Applications of Room Acoustics”, Essex: Applied Science, 1978

[4] ITU Radiocommuniction Assembly, “Methods for the subjective assessment of small impairments in audio systems including multichannel sound systems”, Recommendation BS.1116-1 (10/97), 1997

[5] Human Experimentation Safety and Ethics Committee, “Guide to Experimentation involving Human Subjects”, ISVR, October 1996.

[6] QSC Audio, “ABX Comparator User Manual”, 2013. [Online]. Available: http://www.ackthud.net/shawnfogg/pics/temp/ABX_manual.pdf [Accessed Feb 2014]