CONSIDERATIONS FOR THE GENERATION AND MEASUREMENT OF LOW FREQUENCY EFFECTS IN CINEMA ROOMS

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Little has been published about the repercussions of different source locations and measuring positions for the Low-Frequency Effects (LFE) loudspeakers in cinemas and dubbing theatres. The aim of this study is to determine the effects of the number and position of the loudspeakers on the uniformity of the response over the listening area, and to assess the effect of the measuring of those responses by the choices made regarding the positioning of the microphones within typically used arrays.

INTRODUCTION

Some organisations issuing recommendations for the installation of LFE loudspeakers have long suggested that they should be installed asymmetrically, below the screen. The reason given for this is to avoid the symmetrical driving of the low-frequency modal responses of the rooms, the sources typically being centred about 20% of the distance from one side-wall and 33% of the distance from the opposite side wall. However, other organisations, and many designers, have recommended the mounting of the LFE loudspeakers in tighter-packed clusters. Furthermore, it has long been suggested that a single microphone position is inadequate for measuring the response of an LFE channel because the long wavelengths involved make spacial variation in the measurements inevitable. Typically, 4, 5, 8 and 10-microphone arrays have been used, but as rooms differ so much in size, shape and acoustic properties, no universal instructions exist about precisely how to place the microphones in such arrays.

This paper examines how the choice of low-frequency source positions, and the microphone positions in a 5-microphone array, can affect the measured average responses over the designated listening areas. It also discusses how the positioning of the loudspeakers can affects the evenness of coverage. The results of the spacially averaged responses are compared with the responses at individual microphone positions, in order to assess how representative the averages

1 MEASUREMENT SET UP AND TEST PROCEDURE

Figure 1 shows the distribution of the two LFE loudspeakers in a Dolby certified, European dubbing theatre. At the bottom of the figure can be seen the two, asymmetrically placed loudspeakers. The 21 dots represent the positions of the measuring microphones used for this investigation. Some of the room construction details are shown in Figure 2 and 3. The room dimensions of the isolation shell were approximately 11 m x 7 m x 5.5 m (length x width x height).

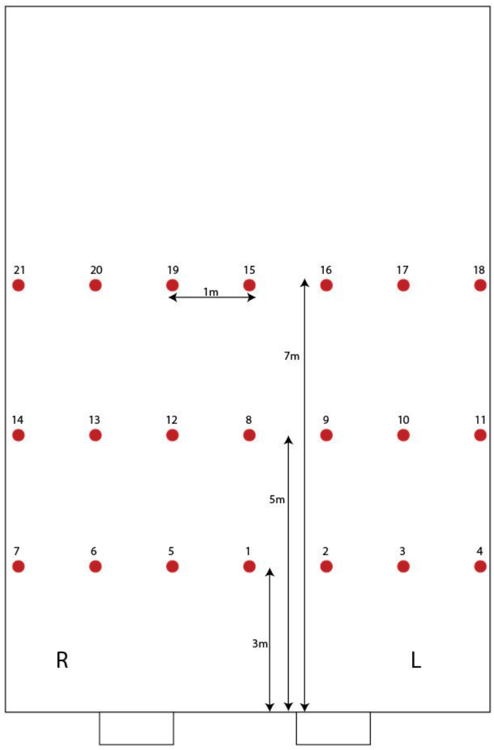

Figure 1: Locations of the two LFE loudspeakers and the array of 21 microphones

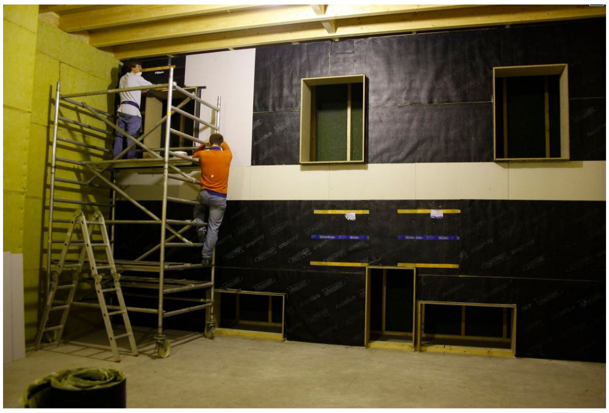

Figure 2: Loudspeaker layout and construction. The front wall is heavy and rigid

Figure 3: The rear of the room prior to testing. Wall and ceiling surfaces are absorbent down to low frequencies

The room had a very rigid front wall in which the loudspeakers were mounted, as shown in Figure 2, plus a rigid, concrete-based floor. The wall opposite the loudspeakers was, essentially, a one-metre deep absorber, effective down to very low frequencies. The other two walls and the ceiling were moderately absorbent down to very low frequencies. RT60 (decay time) and waterfall plots of the room at the mixing position, when driven from the centre-front loudspeaker, are shown below in Figures 4 and 5. The room under test is typical of a modern generation of low decay-time dubbing theatres. Many new cinemas also exhibit low decay times.

Pink noise was used as a stimulus signal for the measurements, which were made using a dual-channel FFT system. The pink noise was injected directly into the input connector of the LFE-channel amplifier in order to bypass any system-processor modification of the test signal. Switching between single and dual loudspeaker measurements was done by means of connecting or disconnecting the connectors on the back of the amplifiers. All measurements were taken with an Earthworks model M30 teat microphone, at a height of 80 cm from the floor. Measurements were taken and analysed with two different software analysers, using the same hardware.

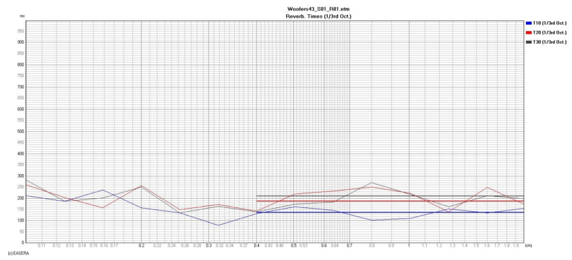

Figure 4: Measured RT60 times of the room under test at the central 7m position when driven from the front wall

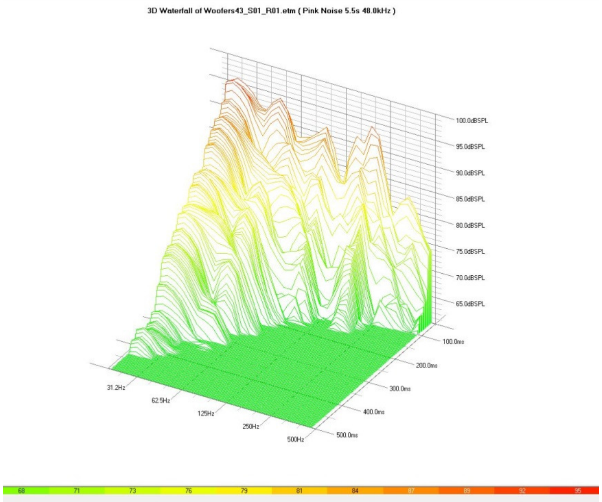

Figure 5: Waterfall plot of the room under test.

As the absolute level was not relevant to the tests being carried out, no precise level calibration was undertaken except to ensure that there was good resolution without any clipping in the system. As an operator was required to be in the room during the tests, a playback level of approximately 78 dBA on the non band-limited signal was chosen in order to comply with regulatory safety requirements (written in dBA), measured using a calibrated Audiotools IOS based SLM and iAudiointerface hardware. The unfiltered response of the loudspeaker extended beyond 1 kHz. Background noise within the room was below NC20, thus allowing adequate measurement resolution without the need for higher levels to be used. The levels shown in the analysis plots included in this paper do not represent the actual measured levels, and should be discounted.

2 MEASUREMENT RESULTS

The plots presented below, in Figures 6.1—6.21, represent the measurements taken at the positions 1—21 shown in Figure 1, respectively. Each measurement graph contains three results. In all cases, the top plot (in red) is the measurement taken with both loudspeakers running. The measurement below this on the key to the right of the graph, numbered sequentially higher, is the left loudspeaker only. The measurement at the bottom of the key to the left of the graph, numbered sequentially the highest, is the right loudspeaker only. Left and right channels are identified “from the house”, such that if the listeners were looking at the front wall from the listening area, the left loudspeaker would be on their left.

To further clarify, Woofers 0 is both loudspeakers, Woofers 1 is the left loudspeaker and Woofers 2 is the right loudspeaker. The numbering sequence follows this format, sequentially, on all figures from 6.1—6.21.

Row 1 measurements are taken at 3 m from the screen, row 2 at 5 m, and row 3 at 7 m.

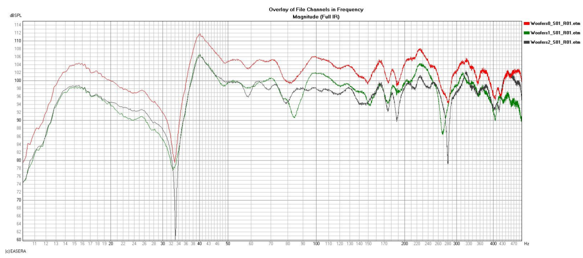

Figure 6.1: Position 1, row 1

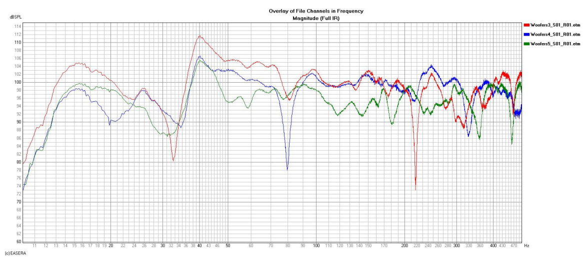

Figure 6.2: Position 2 row 1

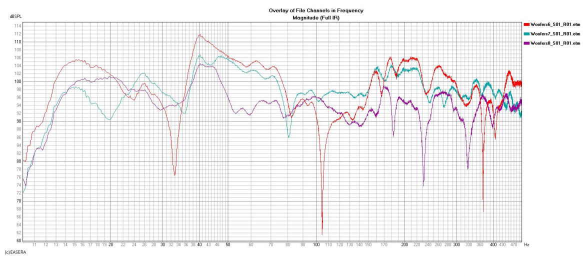

Figure 6.3: Position 3 row 1

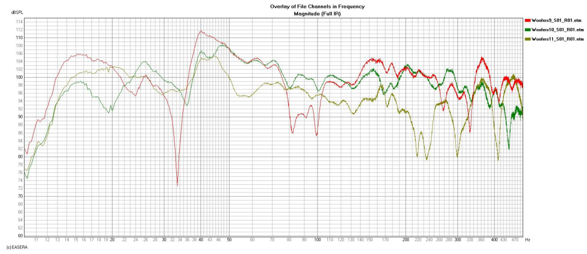

Figure 6.4: Position 4 row 1

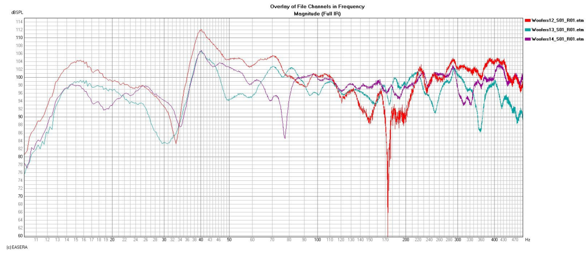

Figure 6.5: Position 5 row 1

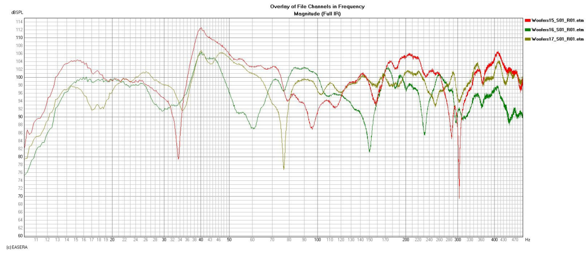

Figure 6.6: Position 6 row 1

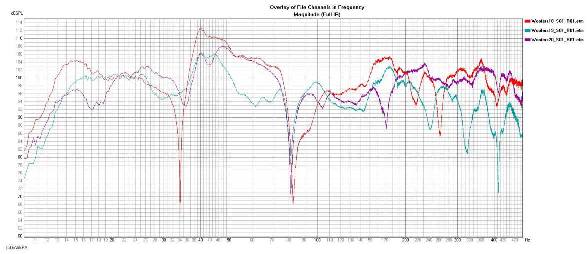

Figure 6.7: Position 7 row 1

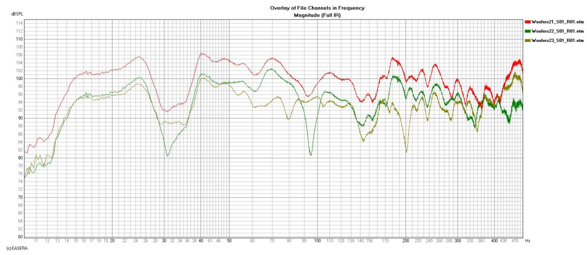

Figure 6.8: Position 8 row 2

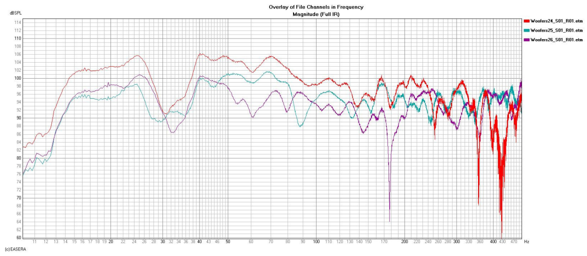

Figure 6.9: Position 9 row 2

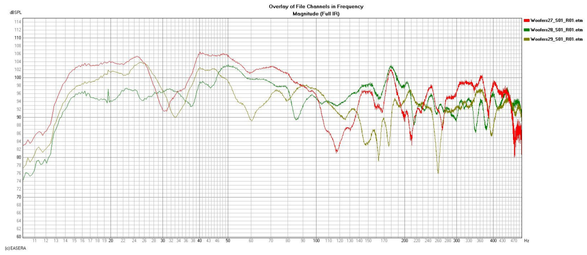

Figure 6.10: Position 10 row 2

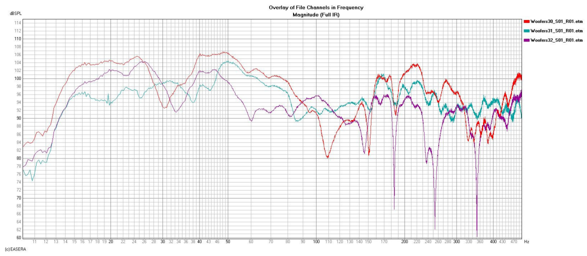

Figure 6.11: Position 11 row 2

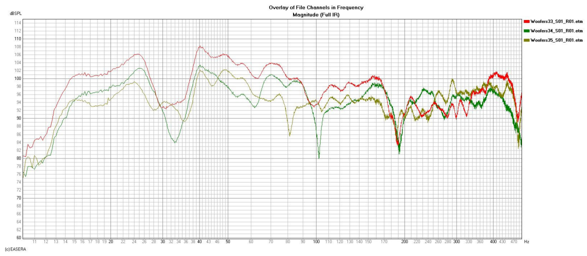

Figure 6.12: Position 12 row 2

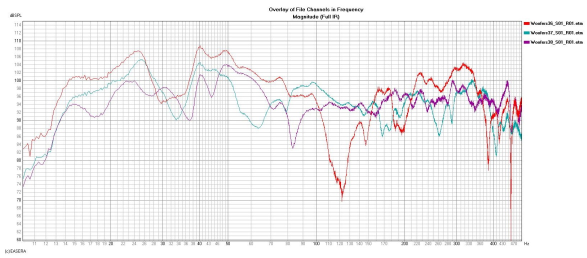

Figure 6.13: Position 13 row 2

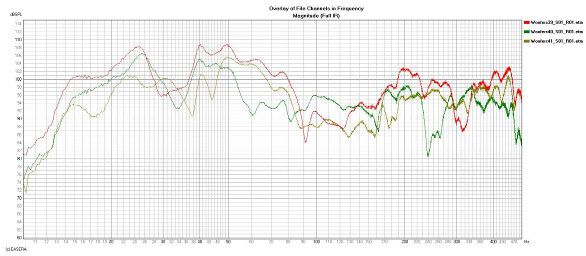

Figure 6.14: Position 14 row 2

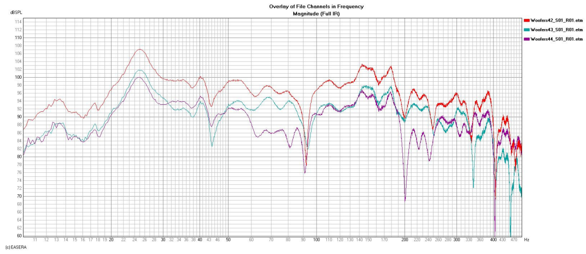

Figure 6.15: Position 15 row 3

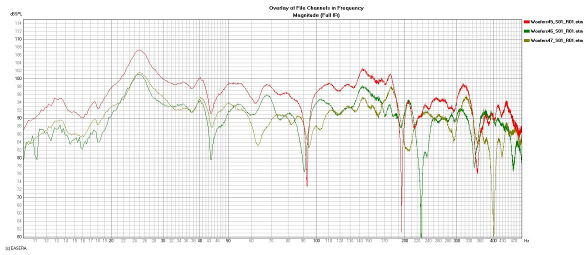

Figure 6.16: Position 16 row 3

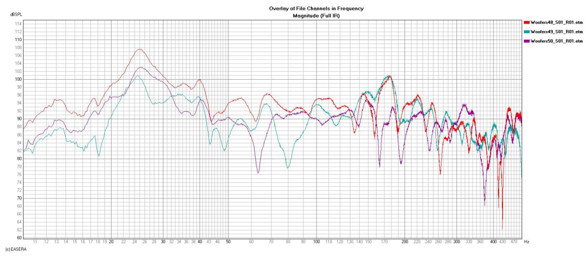

Figure 6.17: Position 17 row 3

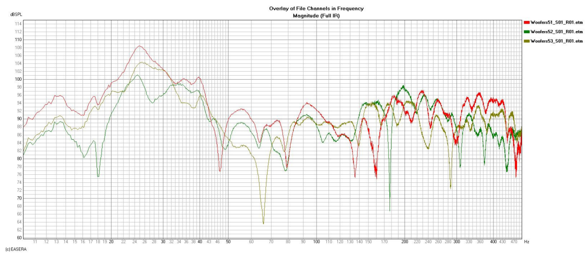

Figure 6.18: Position 18 row 3

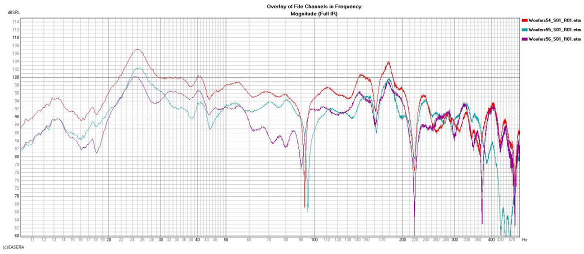

Figure 6.19: Position 19 row 3

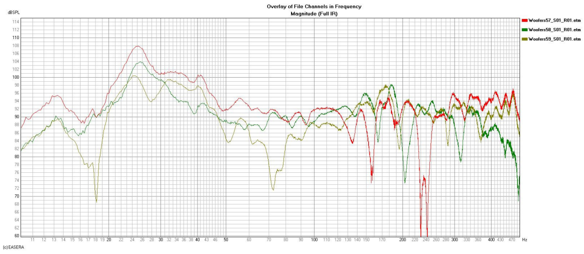

Figure 6.20: Position 20 row 3

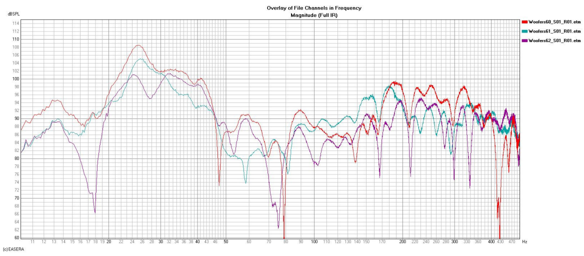

Figure 6.21: Position 21 row 3

2.1 Spectrographs

Further to the data collected at the 21 static points, three measurements were taken using the SMAART Live 5.3 spectrograph function and are shown in Figures 7.1—7.3. These measurements were taken at the same height from the floor as all previous measurements. The spectrograph was used to collect a measurement of the response for each and both loudspeakers in a continuous line, travelling from left to right of the room along measurement row 1.

The purpose of this measurement was to see the fine detail at a great number of points across the room, both relative to the interaction of the two loudspeakers and the interaction of each loudspeaker with the room itself. The left hand side of each spectrograph represents the left hand side of the room. The vertical scale is the frequency on a logarithmic scale, and the colour represents sound level: black/blue being quiet and red/white being loud. Data pre and post microphone travel has been blacked out from the spectrograph to avoid confusion.

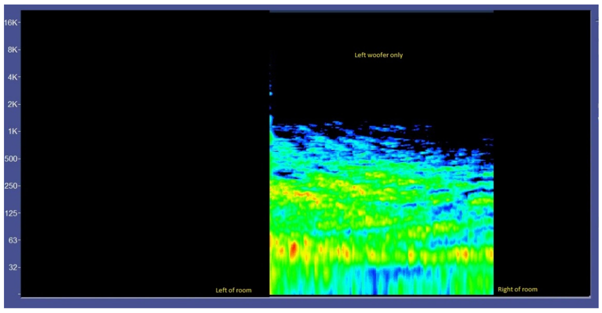

Figure 7.1: Spectrograph of left woofer across row 1

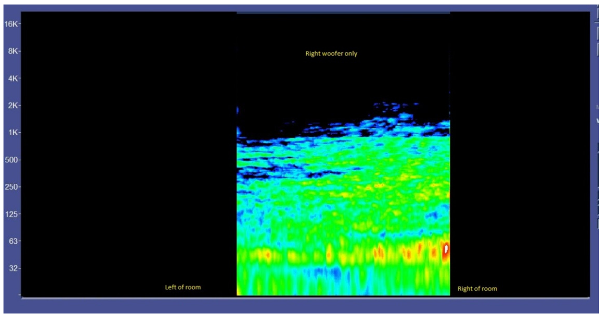

Figure 7.2: Spectrograph of right woofer across row 1

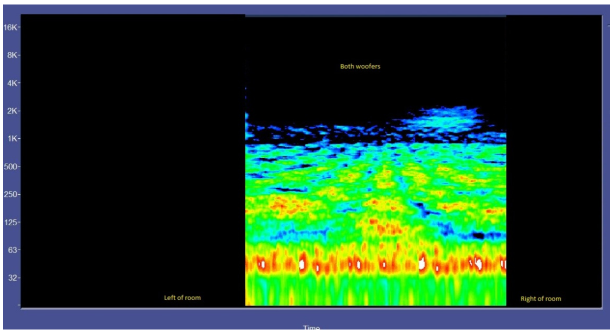

Figure 7.3: Spectrograph of both woofers across row 1

3 DISCUSSION OF RESULTS

The task of capturing the data yielded 65 sets. Given the constraints of the physical size and layout limitations of this paper, it may well be difficult to see global trends because it is difficult to display the plots in a meaningful arrangement with any level of detail. However, the authors were able to print and lay out the data in a more easy to see format. Many sets of plots were printed and laid out in a representative arrangement for overall, visual evaluation. It is recommended that anyone wishing to closer inspect the data should do likewise.

3.1 Multiple source evaluation

In order to make a comparison between single or separated LFE sources, an overlay was created of the seven results from each row of measurements. This was to demonstrate the similarity or difference in response, laterally across the room, as the angle of incidence to the listener from each LFE source and the proximity to the side walls was changed.

It is worth noting that the effect of the side walls in the room under test is attenuated in comparison to a conventional room due to the side walls being fitted with very effective wideband absorber systems, shown in Figure 8. Measurements were taken closer to these side walls than would normally be considered valid in rooms with more rigid walls.

Figure 8: Detail of acoustic treatment of the walls. Behind the waveguides are multiple-membrane absorbers

Figures 9—17 show the results of overlaying the seven plots of, first, the dual LFE loudspeaker pair, and then each single LFE loudspeaker, across each row.

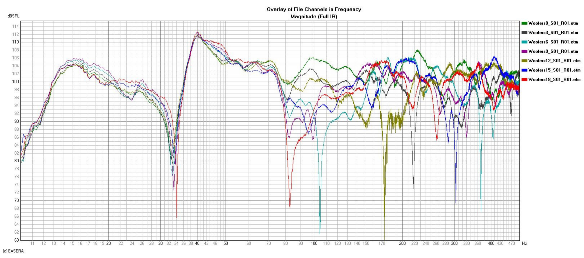

Figure 9: Row 1, both LFE loudspeakers

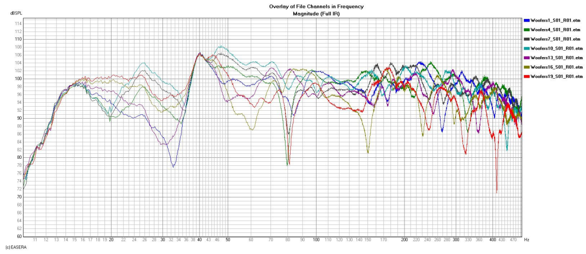

Figure 10: Row 1 left LFE loudspeaker only

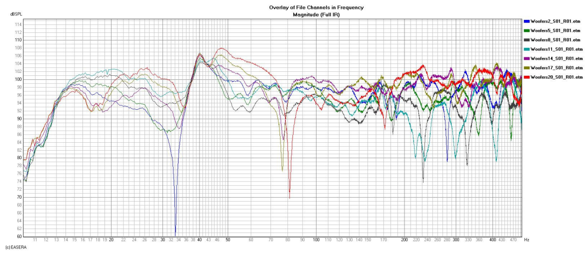

Figure 11: Row 1 right LFE loudspeaker only

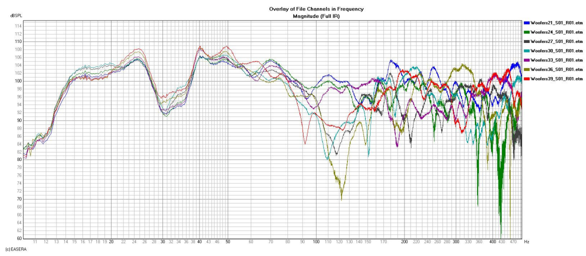

Figure 12: Row 2 both LFE loudspeakers

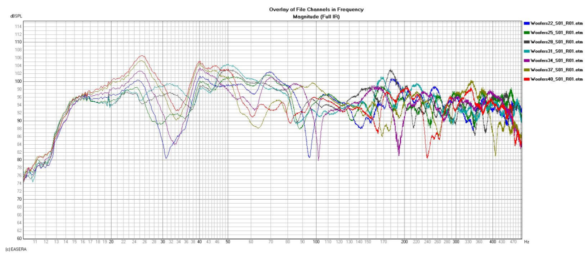

Figure 13: Row 2 left LFE loudspeaker only

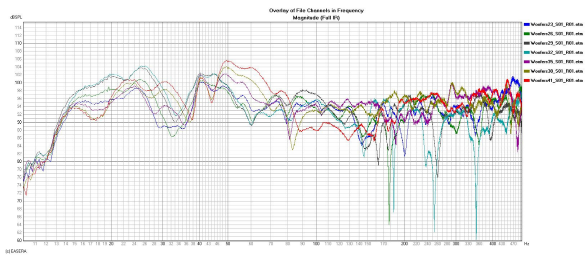

Figure 14: Row 2 right LFE loudspeaker only

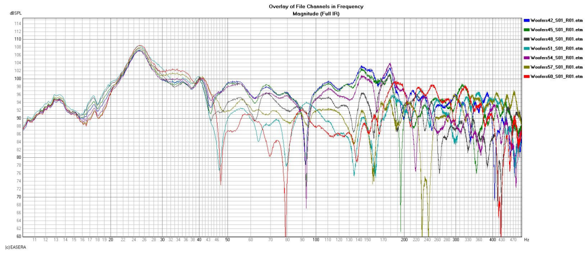

Figure 15: Row 3 both LFE loudspeakers

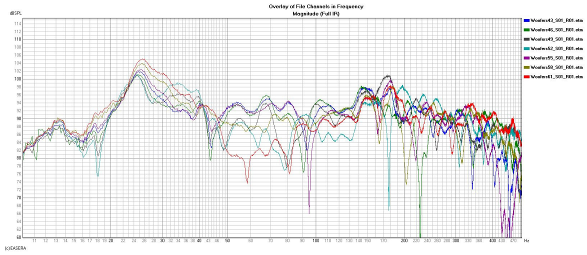

Figure 16: Row 3 left LFE loudspeaker only

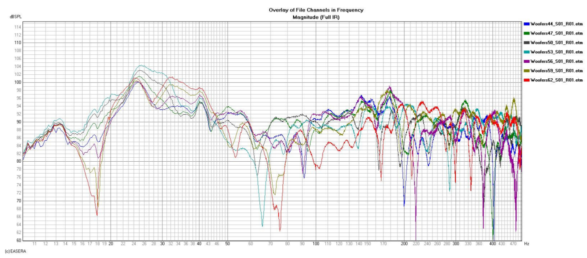

Figure 17: Row 3 right LFE loudspeaker only

From the results shown in Figures 9—17 it can be seen that with two LFE sources running there is an even and consistent similarity of responses from position to position in the very low frequency region, below 40 Hz; far more so than with an individual source. There is therefore some evidence to support the theory that the dual source arrangement does even out some of the room anomalies that affect the point sources. However, there is a very clear negative effect to this concept which can be seen in all of the comparisons. In each case, above a noticeable transition frequency, the correlation significantly decreases. In the case of the room measured for this study, this transition frequency is in the middle of the useful band of the LFE channel. Above this frequency it can be seen that the dual source produces a far less consistent coverage over the measuring area than the single sources. In the upper LFE region, between 80 Hz and 120 Hz, there appears to be a greater deviation between positions than from the single source. There exists up to 10 dB of deviation between the seven row-measurement positions with the single sources, yet there is somewhere between 15 dB and 20 dB deviation with the dual source. Furthermore, it can be seen that there is a greater difference between all sources from row to row as the measurements progress back down the room. In general, the dual LFE system seems to be generating significantly more variation between measurement positions than the single LFE sources in all but the lowest section of the LFE frequency range.

Analysis of the individual measurement plots 6.1—6.21 shows a further effect of the dual LFE concept. As there exists such a sudden and significant departure from the uniform coverage above a certain frequency, it can be seen that there is a considerably differing balance of bass to sub-bass from measurement position to measurement position; far more so than with either single source. There is a distinctive tilt to the frequency responses with the dual LFE system the further off axis that the measurements are taken, whereas the single LFE systems demonstrate a generally more flat trend.

Closer inspection of the raw data from Figures 6.1—6.21 tends to indicate that the dual LFE system produces the best results in a region down the centre line of the room, corresponding to positions 1, 2, 5, 8, 9, 12, 15, 16, 17, 19 and 20. This observation is consistent with a long recognised phenomenon that live-event systems engineers have known about for many years, and have frequently referred to as ‘Bass Alley’. This is observed where wide, laterally-spaced, low frequency loudspeakers, or groups of loudspeakers, concentrate the majority of their sound pressure down the centre of the audience area whilst exhibiting considerable fall off in level outside that region. The data from outside this zone, in the case being studied here, tends to show rather uneven responses above the very lowest frequencies, and indicates that the lateral separation of the low-frequency loudspeakers gives rise to horizontal beaming.

The higher spacial resolution of the spectrograph plots in Figures 7.1—7.3 generally support the results seen in the higher amplitude resolution shown in Figures 6.1—6.21. It is evident that there is a clearly defined interaction in Figure 7.3 which follows a distinct frequency pattern relative to position, which is very clearly not present in Figures 7.1 and 7.2. There is also a far greater amplitude range between the highest and lowest sound levels at various frequencies in Figure 7.3, indicating the greater inconsistency from position to position. It was not possible within the scope of this study to adjust the position of the sources, but theory would suggest that if the sources were further apart, in a bigger room, the frequency at which the response uniformity of the separated sources begins to degrade would be lower.

In addition to the 21 sets of amplitude-response plots, there were generated a similar number of time-response plots. Presentation of all of the plots would make for an excessively long paper, but two of them are included which representatively demonstrate the effect that the two LFE sources have on the arrival times at different positions in the room. Figures 18 and 19 show the single and dual arrivals at listening position 18, from the single and dual sources. Position 18 was chosen for this example as it is mid-way between the most on-axis and the most off axis measurement positions. The result of the less synchronous arrival was a distinct ‘hollow’ sound when listening to pink noise.

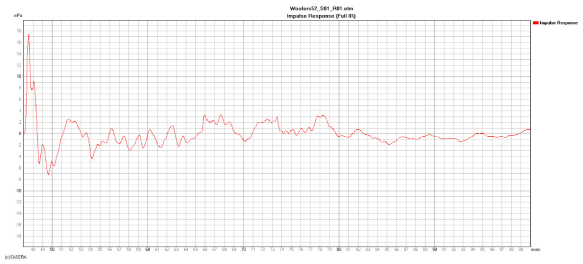

Figure 18. Impulse Response: Left LFE source only at position 18 row 3

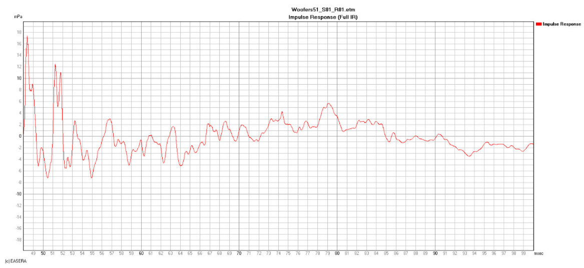

Figure 19. Impulse Response: Both LFE sources at position 18 row 3

3.2 Multiple test microphone positions and averaging of the results

One aspect of the range of measurements presented here is that it is obvious that, whether looking at one LFE source or two, there is a significant inconsistency from one measurement position to another. There are visible underlying trends when moving either laterally or longitudinally around the room, but it would be difficult to identify from a brief observation as to what was typical of the room as a whole. Considering the spectrograph data as a representation of a single line, akin to a string drawn across the room from left to right, 3 m back from the source and 80 cm from the floor, it would suggest that that there is an infinite variation of responses within the room as a whole. As this room is largely non-reverberant, it is probable that the results presented here are in many ways a better-case scenario. A more reflective and/or reverberant space would no doubt further complicate the sound field, and hence disassociate still more the in-room readings from the source responses. Despite the high degree of acoustic control and the very low level of resonant energy, the modal response of the room is still extremely complicated.

It is currently widely accepted that in order to gain an overall assessment of the response in the room as a whole it is good practice to take measurements at a number of locations, and average them to obtain a more representative idea of the sounds perceived from seat to seat. The concept suggests that averaging multiple positions will smooth out many individual anomalies, and indeed this concept would appear logical. There are many theories about how to go about the calibration or periodic response verification of the low-frequency loudspeaker systems in theatres, and there are also various opinions about how many microphones to use. Nevertheless, the quantity of microphones which can be used in practice is frequently dictated by the number of inputs on the typical equipment that is within the means of the average field engineer, or by the abilities of the software and hardware at the time of system development. Four and five-microphone arrays are currently the ones most commonly used, although 8-microphone arrays are becoming more popular for the calibration of professional rooms.

There are also differences in how the inputs are summed or multiplexed and averaged. Some of the systems use a hardware summing amplifier that simply sums the microphone signals electronically. Other systems sum the signals digitally, within the analysis algorithm, first analysing the impulse response then time-aligning the signals before performing the processing, in order to avoid cancelations between different locations. Where sources are multiplexed in a simple analogue multiplexer, or where the signals are just summed within single channel analysis software, prior to processing, there can be little validity in the analysis results as the differing path length distances from the source to each microphone being summed will result in time-shifting of the component signals under test. This time-shift will introduce anomalies into the measurement that are nothing other than a function of the geometrical arrangement of the microphone array. The displayed result will be modified by this function, in effect showing a result which is the average modified by the capture array layout. There are some commercial analysis systems in use that simply electronically sum multiple analogue microphone signals into a single channel analyser without any consideration for the time domain and the problems this causes. These systems do not provide a valid result. In order to provide a correct average of the component signals it is necessary to use a dedicated multi-channel analysis engine capable of compensating for the time differences of the captured data before displaying a sum of the individual microphone signals, or which performs a function that ignores the time domain altogether such as power summing.

For the purposes of this paper, various methods of averaging were first applied to the raw data. One set of averages was taken without time alignment, whilst others were taken with time alignment. Further sets were taken using various different methods of averaging including Magnitude [linear], Energy [squared] and Complex Vector. Despite the different methods rendering different results, the one thing that was consistent across all methods was the amount of disagreement between the post averaging data. The results published here are those of the time aligned Complex Vector process, but it would have been equally descriptive of the overall situation had any of the alternative methods been used. The differences between the results of the different arrays were also of a similar order.

The data was analysed off-line, from the sample size of 21 positions. Processing was carried out using AFMG EASERA software, windowing was not used due to the nature of the room, length of test signal and required data. The microphone position matrix chosen in each case was deemed to cover an area representative of that which the majority of field engineers or technicians would choose to use if measuring such a room. The microphone positions were one metre apart laterally, and two metres apart longitudinally. Row 1 was chosen to be in a position which would represent what could reasonably be the first row of the audience area in a small cinema.

It was decided that, for averaging purposes, five microphones would be adequate to represent what typically happens in the field whilst allowing a sufficient number of data points for the measurement process required for this study. Four sets of five measurement positions were chosen by the operator, based on positions that would be reasonably representative of what professional field engineers would typically choose. Each set of data from each position was loaded into the analysis software for processing, and an average was generated. All source plots were time aligned prior to averaging, as would be done by most systems.

Figures 20—28 show the overlay of the raw data and the result of the averaging. Averages were chosen from the positions as follows:

Average 1: Mic 20, 19, 15, 16 and 17
(constrained to three-quarters distance down the room)

Average 2: Mic 20, 12, 15, 9 and 17
(variation of Average 1)

Average 3: Mic 21, 12, 6, 8, and 11
(general, typical array)

Average 4: Mic 20, 6, 8, 17, and 3
(wide array covering whole measurement area)

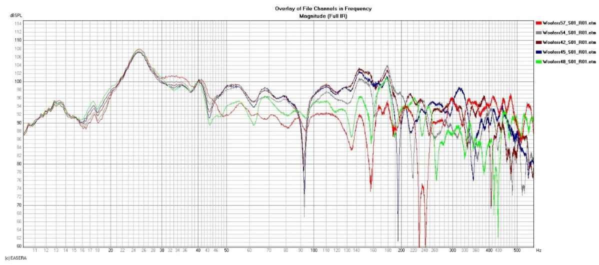

Figure 20: Components for Average 1

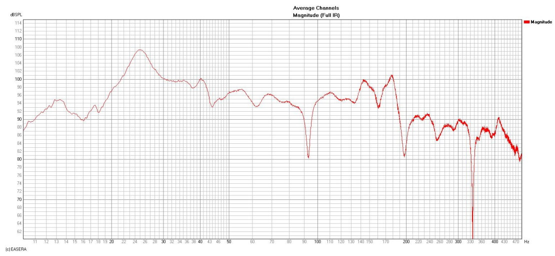

Figure 21: Average 1

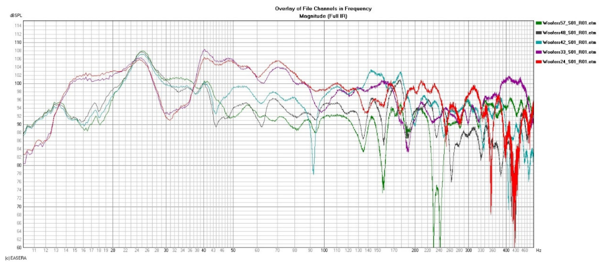

Figure 22: Components for Average 2

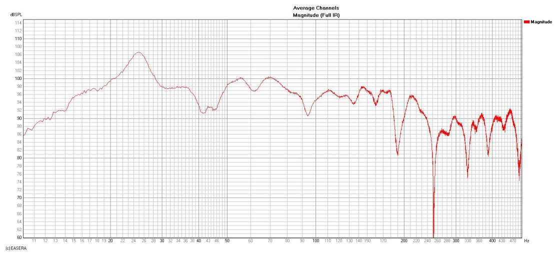

Figure 23: Average 2

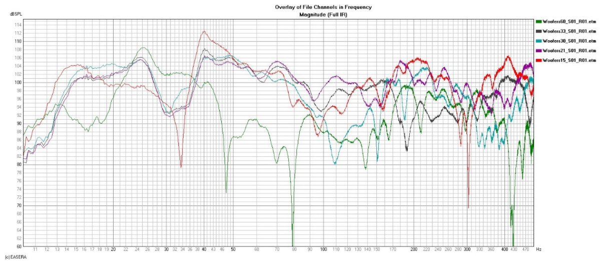

Figure 24: Components for Average 3

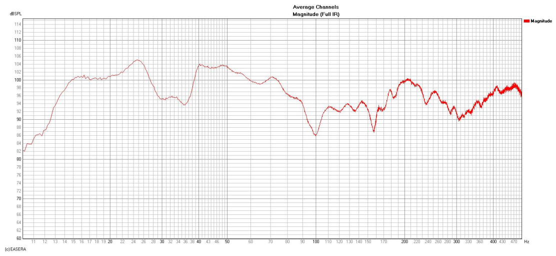

Figure 25: Average 3

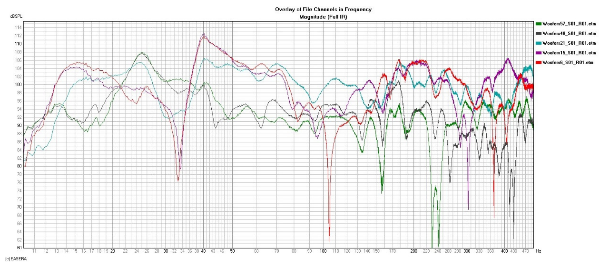

Figure 26: Components for Average 4

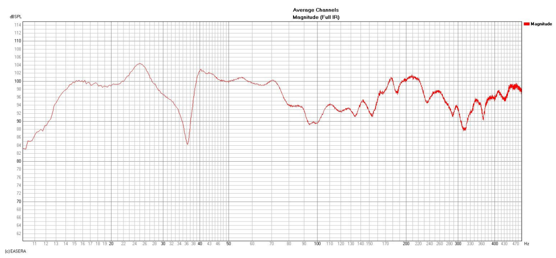

Figure 27: Average 4

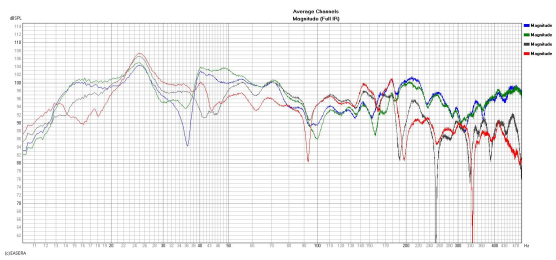

Figure 28: Overlay of Averages 1, 2, 3, and 4

From the results presented here it can be seen that there is considerable variation in all areas, which was perhaps to be expected following the observations discussed earlier regarding the complexity of the modal response of the room. It can also be seen that the raw input data is, in each case, very different from position to position. An inspection of Figures 20, 22, 24, and 26 shows that the source data from each set of measurement points indeed exhibits a high degree of variability. For this reason, simply choosing capture positions by eye, as is frequently done, will almost surely result in a highly variable set of input data.

Figures 6.1—6.21 show that there is significant dissimilarity between the responses at all but a few adjacent positions. Averaging such widely different data points only results in averaging what are effectively randomly-chosen responses from an extremely complex sound field. As Figure 28 clearly shows, moving one or two sample-points results in a significant disagreement between average plots, in some cases up to 10 dB, yet nothing, of course, has changed in any individual seat position. Average 2 has three of the same data points as Average 1, and the two different points are only moved to the next-adjacent microphone positions yet a significantly different measured result is apparent. It is obvious from Figure 28 that, depending on which data points were chosen in the first place, there is a high level of variation between all of the averaged results yet all the microphone positions that were chosen would have been valid choices for a typical field engineer. No point chosen was in any extreme position. This raises the question as to which of the four averages is most valid as each one would be a reasonable choice. Averages 1 and 2 were close packed measurement points, Averages 3 and 4 covered more of the room, but both methods represent current practice in cinema loudspeaker system measurement.

The principle observation from these results is that by varying the position of the test microphones, alone, we can get a greater degree of difference in the test results than if we had left the microphones in place and changed to a different loudspeaker system.

It can be seen that the difference between the four average plots is less than the difference between the sets of five individual microphone plots from which the averages were taken. However, there is no evidence to suggest that any of these plots are in any way more correct than any other plot taken in this process. If the averaging process were to produce the greater degree of representation of the actual perceived sound in the room, as is so often claimed, it would be reasonable to expect to see a greater degree of agreement between the four resulting average plots, and certainly to expect to see a pattern that represents the room across all four averages. This is surely a prerequisite for the concept of the calibration and equalisation of the sources from averaged measurements. The more the frequency resolution shown in the results, the more the positional variations become obvious.

At the time when an earlier version of this paper was being prepared [3], a paper was published by Dolby describing some rather similar measurements being carried out, albeit for a different path of investigation. [4] It is interesting to note that the degree of positional variation in the responses was very similar to those presented here, and so helps to corroborate the validity of this measurement process.

3.3 The effects of extreme smoothing on the consistency of the averages

It has been suggested in some influential quarters that the use of whole-octave-band smoothing would give more reliable and consistent results. It was decided to apply this concept to the results shown in Section 3.2 of this paper, to see if this suggestion has merit.

The results shown in Figures 29—37 depict the same sets of source data represented as one-octave-smoothed plots, along with the result of the averaging of the time-aligned raw data. The results of the four sets of five position-averaged plots were then overlaid to show the level of agreement.

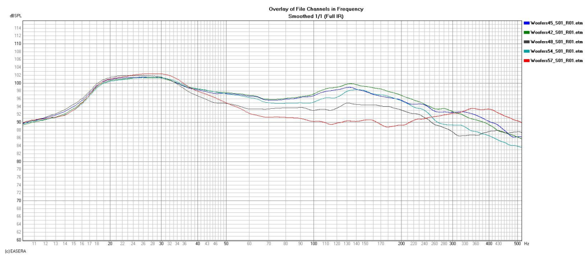

Figure 29: Components of average 1 – Whole octave smoothed

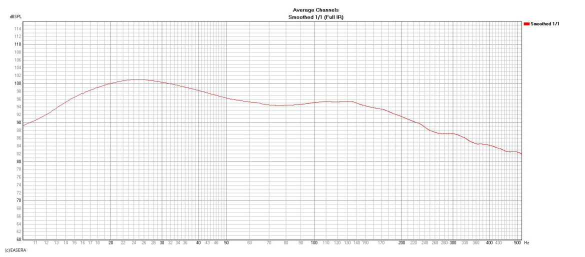

Figure 30: Average 1 – Whole octave smoothed

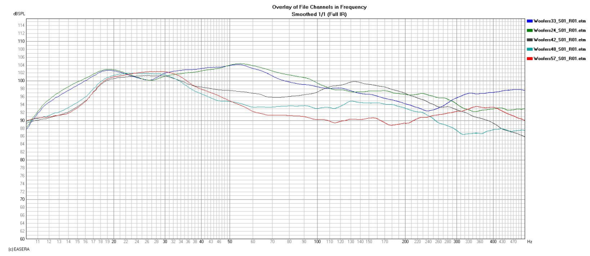

Figure 31: Components of average 2 – Whole octave smoothed

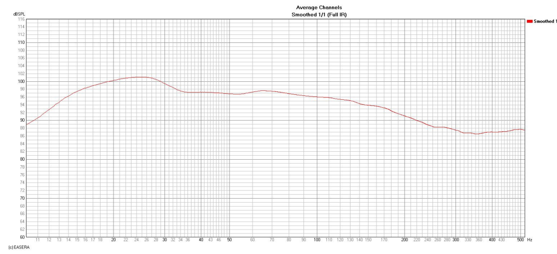

Figure 32: Average 2 – Whole octave smoothed

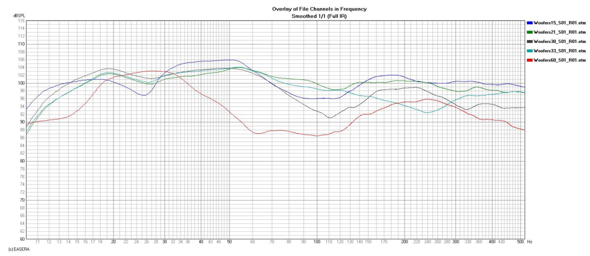

Figure 33: Components of average 3 – Whole octave smoothed

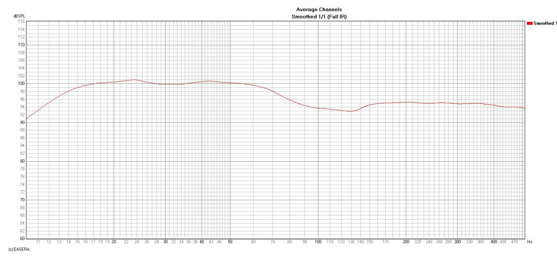

Figure 34: Average 3 – Whole octave smoothed

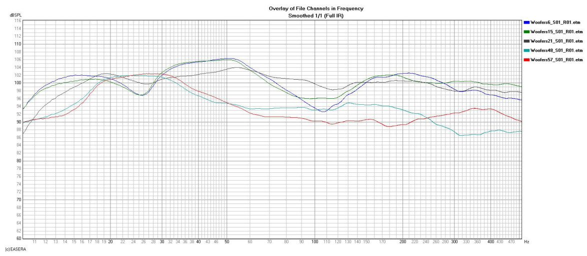

Figure 35: Components of average 4 – Whole octave smoothed

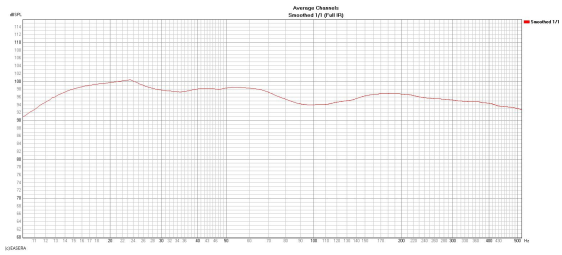

Figure 36: Average 4 – Whole octave smoothed

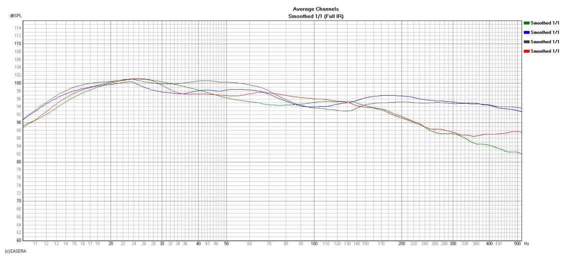

Figure 37: Overlay of averages – Whole octave smoothed

While it can be seen in Figure 37 that there is certainly a closer agreement between the four sets of averages, there still exists a considerable level of disagreement, especially in the higher ranges. It can be seen from the low-frequency, truncated measurement that the measurements are certainly not within only one or two of decibels of disagreement. The differences are significant.

Further to this, it is evident that the majority of useful analytical data has been smoothed away by the severe amount of applied smoothing. Figure 38 shows the degree to which much useful data is lost under extreme smoothing, and shows that in certain instances (around the 20 Hz and 125 Hz bands) the smoothing has introduced errors which could lead an operator to make different adjustments. Figure 38 shows the effects of a more appropriate amount of smoothing (one-sixth octave) and how that follows the raw data in a far more representative manner.

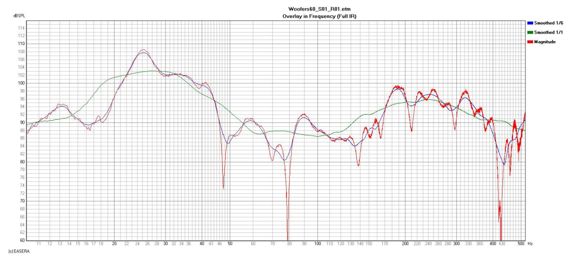

Figure 38: Overlay of the effects of smoothing on an example audio signal: red: unsmoothed; green: octave smoothed; blue: 1/6th octave smoothed

In light of all the above results, the question arises as to whether there is any deep-in-the-room measurement system which can be used to standardise the responses, or whether the source responses should be the preferred characteristics for standardisation.

4 CONCLUSIONS

4.1 Separated-source LFE loudspeaker systems

It has been shown that the use of separated LFE loudspeaker positions, horizontally displaced across the front wall of a theatre, does not produce a more even, overall distribution of sound from seat to seat (at least in rooms of relatively low decay time). With the exception of good similarity in the response of the lower portion of the frequency band, below 50 Hz, it is evident from this study that dual-source, horizontally-displaced systems can produce a significantly less-even coverage. It has been shown that the use of a single source produces a more even spectral coverage over the wider 20 Hz to 120 Hz band, with fewer irregularities in both the time and amplitude domains.

Figure 39 shows the results of a simulation of the summed low-frequency response of the output of two loudspeakers, 5m and 6m from the microphone, under free-field conditions, which highlights the point being made. The first interference notch is when the half wavelength equals one metre, which in this case is at about 170 Hz. As the difference in the path lengths between the sources and each listener increases, the frequency of the cancellation dip will lower. The cancellation frequency therefore differs for different positions in the room, and hence cannot be corrected, irrespective of what may be shown for any given set of microphone-position averages. The simulation is in general agreement with the measurements presented in this paper.

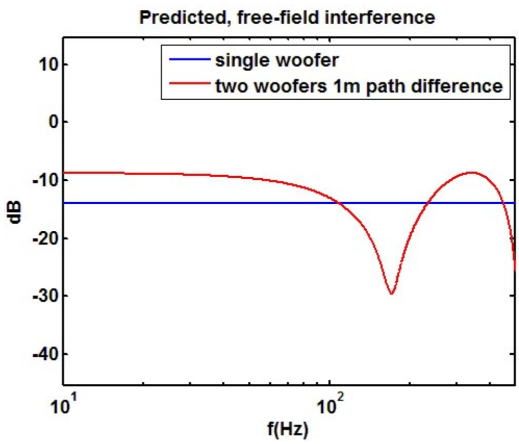

Figure 39: Predicted free field interference of two sources at an off-axis listening position

4.2 Multiple microphone analysis

1. The results of this experiment show that the variation from position to position within a room is so uneven that the arbitrary choice of any five microphone positions, as opposed to the careful choice of one single microphone position, is not necessarily a way to ensure a more accurate or repetitive measurement of the low-frequency response. Irrespective of any average which may be measured, nothing can be done in a calibration process to change the relative degree of variability from one seat to another.

2. It has been shown that moving a few microphones within a typical array by no more than two metres can result in significantly different average measurements. These can vary by up to 10 dB in different places in the spectrum. Given the high degree of variability from one position to another, in what is a highly complex modal response within a room, the differences in the averages should come as no surprise.

3. If, as shown in this paper, there is such variability in the results from multi-microphone arrays with only moderately different microphone positions, it would seem improbable that even a simple measurement using this method could be accurately replicated on subsequent maintenance visits without very careful attention being paid to the precise positioning of the microphones prior to any new measurements. Given the time constraints usually applying during commercial theatre analysis, it is doubtful that many maintenance technicians could even hope to dedicate the required level of care and attention to the positioning of each microphone in a multi-microphone array on routine maintenance visits. In fact, it has been shown in other literature that system alignment engineers, when left to their own devices, tend not to position the microphones in similar manners to each other, even when given specific instructions on how to do so. [5] This gives rise to the tendency for the technicians to suspect during maintenance visits that something has changed in the performance of the system when, in fact, only the microphone positioning may have changed. For routine maintenance checks, the careful positioning of a single microphone in a previously, carefully-recorded location, for comparison to a previous response plot, may be a more practical option.

4. Given the difference between the four sets of position-averaged results shown in this study, it is difficult to know which one is the most ‘correct’. None of them represent the true response at any given listening position. There is such a great variation in the response from the source signal in different places in the room that there is little prospect of the selection of any five positions corresponding to the average responses taken at any five other positions.

5. The variability in the results of these measurements highlights the danger of using such techniques for the purpose of the equalisation of low-frequency loudspeaker systems in theatres. Previous work has shown that the typical sorts of equalization currently applied to loudspeakers in cinema rooms rarely match the true frequencies of the response irregularities. Therefore, it is equally rare that such adjustments actually ‘correct’ the response in any precise sense of the term. [6] The results shown in this paper indicate how widely those measured irregularities can vary, even when using arrays of five microphones.

6. It is evident that there is no direct 1:1 relationship between any of the averaged response plots and the individual components from which that average was calculated. It would be hard to understand how an attempt to use the average plot, which cannot be verified back to any physical reference point, to apply corrective equalisation to a system could result in an equalisation curve that relates to any, real, listener position in the room. No individual listener is subject to the overall-area response represented by the average response, so it cannot be said with any certainty that the average response relates to any specific listener experience. The raw data of the captured responses, themselves, show that each listener is experiencing a different, measured, frequency balance which cannot possibly be related back to the resulting equalisation derived from any of the averages that have been derived. Evidence has shown that application of any corrective equalisation which is the result of a measurement taken at a single point in the far field, when applied to any other specific listening position in the room, can result in a greater deviation from target in the other location [6]. It is hard to understand how any applied equalisation resulting from averages that are shown not to exist at any

verifiable listening point in the room can be any more valid than any other method. Nothing is being done which could reduce the degree of seat-to-seat response differences.

7. It should be noted that, entirely independent of the work carried out within the scope of this paper, the manufacturers of one of the most comprehensive and leading multi-channel live data analysis systems do not recommend the use of averaged results as a sole system tuning reference [7]. Despite this, many in the industry continue to advocate the method as an optimal sole solution, and indeed go so far as to incorporate it into automated equalisation systems.

8. Although the addition of whole-octave-band smoothing does render closer agreement between the four sets of averages, the results still disagree by a significant amount, and not just one or two decibels. What is more, there is very little resolution in such measurements, and to some extent the octave-smoothed data begins to show areas where the smoothed result begins to disagree with the raw data. In general, significant deviation begins to be evident between the reality of the raw data and the displayed trace when the system analysis is carried out with resolution of less than one-third octave. That having been established, it is recognised that some of the suggestions which have been made by some consultants to limit the resolution to whole octaves have been intended to preclude the tendency for some less-skilled technicians to chase the centre lines of the response targets and apply excessive equalisation, in the belief that the closer the average response fits the target on the analyser, the better the sound must be. Unfortunately, this belief is still wide spread.

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