TOWARDS A GENERALIZED THEORY OF LOW-FREQUENCY SOUND SOURCE LOCALIZATION

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1 INTRODUCTION

Sound source localization has been the subject of numerous research projects, dating back over one-hundred years with Rayleigh's work in the 1800s[1]. The century-plus of investigations into how humans determine a sound's origin has resulted in a fairly robust understanding of the mechanisms involved, especially for localization in the horizontal plane. Three-dimensional localization is perhaps less-well understood, however studies of head-related transfer functions[2] (HRTF) have given a fair deal of insight.

A new hypothesis is proposed that is informed and validated through a series of straightforward and repeatable experiments. The experimental findings are compared both to previously published subjectively-based results and to a series of informal listening tests performed recently at the University of Derby using two rooms of disparate size.

Low-frequency sound source localization, however, seems to generate considerable amount of disagreement between audio/acoustics researchers, with some arguing that below a certain frequency humans cannot localize a source[3-8] with others insisting that in certain cases localization is possible, even down to the lowest audible of frequencies[9-14]. Nearly all previous work in this area depends on subjective evaluations to formulate theorems for low-frequency localization. This, of course, opens the argument of data reliability, a critical factor that may go some way to explain the reported ambiguities with regard to low-frequency localization.

The resulting proposal stipulates that low-frequency source localization is highly dependent on room dimensions, source/listener location and absorptive properties. In some cases, a source can be accurately localized down to the lowest audible of frequencies, while in other situations it cannot. This is relevant as the standard procedure in live sound reinforcement, cinema sound and home-theater surround sound is to have a single mono channel for the low-frequency content, based on the assumption that human's cannot determine direction in this band. This work takes the first steps towards showing that this may not be a universally valid simplification and that certain sound reproduction systems may actually benefit from directional low-frequency content.

2 SOUND SOURCE LOCALIZATION

The foundation of our current understanding of sound localization was laid by John Strutt (a.k.a. Lord Rayleigh) in 1876[1]. Rayleigh's duplex theorem describes the most important aural cues for localization; however the theorem fails to describe localization for all scenarios. Due to this, additional research projects have been carried out, some being directed specifically at multi-channel sound reproduction (the Hass effect) with others looking at the physical structure of our ears (HRTF) as well as the mostly subconscious mechanics employed during everyday listening (head rotation). This section briefly reviews sound localization and highlights the current understanding of low-frequency localization, emphasizing disagreement between previously published research findings.

2.1 Rayleigh's duplex theorem

Sound source localization in the horizontal plane is generally described with Rayleigh's duplex theorem. The horizontal plane covers left-to-right and front-to-back, as opposed to the median plane (front-to-back, up-to-down) and the frontal plane (left-to-right, up-to-down).

The two localization mechanisms described by the theorem are interaural time delay (ITD) and interaural level difference (ILD). ITD operates on the fact that aside from sound located directly to the front or rear of a listener, the propagation distance from sound source to each ear is different, hence a time of arrival delay between ears. This delay is often referred to as the binaural delay and is calculated using Eq. 2.1 where r is the head radius (s), φ is the angle off center of the source (rad) and c is the speed of sound in air (m/s).

$$ITD = \frac{r(\varphi + \sin\varphi)}{c} \tag{2.1}$$

Binaural delay is at its maximum when a source is located directly to either side of the head, and equals roughly 0.65 ms (depending on head size). It is believed that the human hearing mechanism operates not directly on binaural delay, but on binaural phase difference which is caused by the ITD (Eq. 2.2).

$$\Phi_{ITD} = 2\pi r(\varphi + \sin\varphi) \tag{2.2}$$

This idea suggests that there is an upper frequency limit for accurate localization with ITD since beyond a certain frequency the binaural delay corresponds to more than one-half wavelength (or 180°), resulting in ambiguity of phase difference. This limit is calculated with Eq. 2.3:

$$f_{\max} = c[2r(\varphi + \sin\varphi)]^{-1} \tag{2.3}$$

Assuming a head radius of 9 cm, the maximum frequency for ITD accuracy for a centered source is 1.91 kHz, while a source directly to either side of the head gives a maximum frequency of 741 Hz. It is clear that the accuracy of ITD is dependent on the direction of the source, which is interesting as that is precisely what ITD is used to determine!

Placing aside the ambiguity arising from source location and frequency, it is clear that ITD excels at localization over the lower portion of the audible frequency band (generally below 1 kHz). It is necessary, therefore, that a second localization mechanism exists which specializes in the higher frequency range. This is the second component of duplex theorem called interaural level difference (ILD) or interaural intensity difference (IID). It is referred to as ILD for the remainder of this paper.

The best way to think of ILD is to imagine a spotlight located in the horizontal plane pointed towards the center of the head. When the spotlight is directly in front or behind the head, both ears are illuminated equally (i.e. they receive the same sound level). As the spotlight rotates about the head it loses a direct path to one of the ears, thus causing that ear to be in the shadow of the head. In acoustical terms, this means that the shadowed ear receives lower sound levels than the other ear. The difference between received levels at each ear is called ILD and can be as much as 20 dB when the source is directly to either side of the head.

As with ITD, there is a frequency limit for ILD. This is due to the fact that as wavelength increases (frequency decreases) the head causes progressively less of a shadowing effect since its relative size to wavelength is insignificant[1]. Generally, this limit is taken as the frequency at which the head diameter is less than one-third of the wavelength. The ILD limit is calculated using Eq. 2.4:

$$f_{\min}\big|_{\varphi=\pi/2} = \frac{1}{3}\frac{c}{2r} \tag{2.4}$$

Equation 2.4 gives the minimum ILD frequency for a source directly to either side of the head which if the head radius is assumed to be 9 cm, the minimum frequency for accurate ILD localization is 635 Hz. This limit can be calculated for other source locations, but is beyond the scope of this brief review of the duplex theorem. Nevertheless, it is clear that ILD operates best at higher frequencies and picks up roughly where ITD loses accuracy. Therefore, Rayleigh's duplex theorem proffers a good general description of how humans localize sound in the horizontal plane.

There is a significant hole in the localization explanation provided by duplex theorem, which is that there is no way to accurately differentiate from front-to-back and also no clear mechanism to handle vertical localization. This problem is called the cone of confusion as it creates a cone of uncertainty for each ear, where the apex of the cone is located directly at the ear and the cone moves outward from the head.

It is vital to understand that the cone of confusion is not an actual issue with human hearing (as it is often misinterpreted as), but is a shortcoming of duplex theorem. In the real world, there are additional mechanisms used to pinpoint a sound's direction, giving a much more robust system than that operating solely with ITD and ILD. One such mechanism is discussed in the following section.

2.2 Head-related transfer functions (HRTF)

A considerable amount of past and current research has focused on the effects the physical structure of the human head and ears has on localization. This work led to head-related transfer functions whereby the transfer function of the ear is largely dependent of the angle of incidence. The directionally-sensitive transfer functions are largely an effect of reflections off the pinna in the outer ear which results in comb-filtering of the received signal. It is thought that the human hearing mechanism trains itself early in life to detect the minute differences in response to aide in localization[2], it is in effect a form of pattern recognition that becomes more evident when a 3-dimensional map of HRTFs is observed.

Since the human ear is asymmetrical, HRTFs resolve (especially when higher frequency components are present within a signal) the cone of confusion allowing for more accurate front-to-back and vertical localization, with head rotation providing further evidence. Since the propagation delay between the direct sound and the pinna reflections is very short, comb-filtering is likely to occur only at higher-frequencies, so still may lead to some ambiguity in the low-frequency range. This issue is explored more in Section 2.4.

2.3 The precedence effect

An additional localization phenomenon that applies to multichannel sound reproduction is the precedence effect (*a.k.a.* the Hass effect)[15]. The effect is applicable to two sounds arriving at a listening location at slightly different times and is used to create virtual sound sources, away from the physical location of the loudspeakers.

The default case in this instance is where there is no delay between the two sources. Assuming the sources are in a left-right stereo configuration, there will be a phantom source directly in the center of the soundscape. As small amounts of delay are applied to one source (up to 1.0 ms) the virtual source will begin to move towards the source without any delay. To this point, the effect is mimicking ITD from Rayleigh's duplex theorem. Beyond this small delay is where the precedence effect begins to take hold.

When one source is delayed by 1 - 30 ms, the sound will be localized to the un-delayed source. The two sounds are still perceived as only a single event, despite the delay between them. This is valid even if the delayed sound is up to 10 dB louder than the other[15]; Level differences greater than that can lead to two separately perceived events. This is the core of the precedence effect; the earlier arriving sound is dominant in localization where the later sound is largely masked by the first[15]. The effect fades when, in addition to large level differences, one source is delayed by more than 30 ms. In these cases two separate events will be perceived and localized individually.

2.4 Current understanding of low-frequency localization

At present, there is no consensus on whether humans can detect low-frequency directionality, and perhaps more importantly, if it even matters if detection is possible. In this research, fourteen previously published works were reviewed in order to ascertain the current state of affairs on this subject. Of the fourteen works, six conclude low-frequency localization is possible and/or important, six argue it is impossible and/or unimportant and two give mixed opinions on the matter.

Among the research supporting that low-frequency localization is not possible and/or important is the work of Borenius[3] where subwoofer crossover frequency and delay settings were examined to determine at what frequency/delay the location of the subwoofer can be determined in a small listening room (7.0 m x 5.2 m x 2.7 m). Borenius found that there is very little directional information below 200 Hz and none at all below 100 Hz. This suggests that a mono subwoofer configuration is ample for multichannel home theater sound systems. The insight was made, however, that speech signals are highly sensitive to delay (especially if the subwoofer signal arrives first), which at the very least supports the importance of proper time alignment, even if directionality is not an issue.

Another thorough piece of research supporting the “not localizable” group was carried out by Welti[4], who has undertaken a considerable amount of work on small-room low-frequency acoustics and reproduction over the past decade. Welti’s tests compared mono and stereo subwoofer configurations in a small listening room (6.7 m x 7.3 m x 2.7 m). His conclusions were that the only noticeable difference was a comparison between a mono subwoofer at the front wall midpoint and a stereo subwoofer configuration with sources directly to the left and right of the listener. Comparisons between mono and stereo two-subwoofer systems with front corner placement gave no indication a difference could be heard. Welti did note however that these tests in no way prove there are no merits to multichannel subwoofer reproduction, thus leaving the door open to the possibility that in some cases it may make a difference.

The work of Kelloniemi *et al*.[5] also concludes that direction cannot be judged in the low-frequency band. Their tests were again performed in a single listening room (6.35 m x 5.58 m x 2.71 m) where they were examining subwoofer crossover frequencies. The work found that crossover frequencies above 120 Hz reveal the subwoofer’s location, but this is somewhat dependent on the placement of the main left and right loudspeakers. This is in line with the findings of Borenius[3], which gives confirmation to both experiments as they were carried out in rooms of similar dimensions and were both testing crossover points.

Zacharov *et al*.[6] examined subwoofer positioning in the context of film soundtracks in home theatre environments. Their conclusion was that as long as the subwoofer crossover is around 85 Hz, one subwoofer is sufficient and listeners cannot hear the difference between subwoofer locations (front center, rear center and directly to the left of the listener were tested). Interestingly, all three subwoofer locations were at wall midpoints in a single room (5.03 m x 6.03 m x 2.63 m), which gives similar modal excitations. This is in slight disagreement with Borenius in that delay does not seem to be a huge issue in this work, however the test signals were very different in nature and the subwoofer crossover frequency was 85 Hz rather than 100 – 200 Hz. It is unclear whether the insensitivity to position supports Welti’s work[4] or not.

Similar conclusions stemmed from comparable experiments to those of Zacharov *et al.*[6] in the work of Kugler and Theile[7]. In their work, a full-range stereo system was compared to a left, right and subwoofer system with varying crossover points and subwoofer locations in one listening room (6.68 m x 6.62 m x 2.71 m). Their conclusion was that as long as the crossover frequency is below 100 Hz, one subwoofer is enough and does not contribute to directional information; very much in line with the work of Zacharov[6], Kelloniemi[5] and Borenius[3].

One piece of work that takes subjective results similar to those presented already in this section and provides a form of objective explanation was carried out by Benjamin[8]. These experiments used tone bursts in two similarly sized, but acoustically different, listening rooms (6.75 m x 4.64 m and 5.48 m x 4.59 m). Benjamin finds, like Borenius[3], that there is little directional information received by listeners below 200 Hz. The reasoning for this is that in this range room reflections change the phase of the received signals (since they arrive well within one wavelength of each other) causing issues with ITD cues. He does state, though, that listeners may still have a strong sense of localization; it just may be incorrect. Interestingly, in another piece of work focusing on localization in Ambisonic systems[9], Benjamin *et al*. state that below 400 Hz velocity vectors provide localization cues, which is not quite in line with the reasoning in his other paper[8].

One piece of work that does support the ability to localize low-frequencies was carried out by Braasch *et al*.[10] and is largely an extension of Kugler and Theile's work[7] comparing full-range and left, right plus subwoofer systems. In this work, subwoofers were placed at each of the five standard surround sound locations in two small listening rooms of unspecified dimensions. Their listening test results lead to very different conclusions from those of the "not localizable" group. Braasch *et al*. suggest that low-frequency signals are often localizable and, therefore, multiple subwoofers should be used. Their ideal recommended configuration is to have a subwoofer to the left and to the right of the listening area since this will give the best left-to-right ITD cues, while ignoring any front-to-back information, which was found to be less localizable.

Subkey *et al*.[11] confirm the work of Braash *et al*.[10] in a piece of work targeted at directional cues for auditory imaging in auditoria. Tests were carried out in a single listening room (4.5 m x 3.6 m x 2.5 m) with subwoofers in each room corner. These blind listening tests judged whether listeners could determine which source was emitting the sound. The conclusion was that left-to-right discrimination is very strong in the subwoofer band, even as far down as 25 Hz. Front-to-back directional sensitivity, on the other hand, was much reduced, possibly due to the weak ITD clues.

Auditory imagery is also the focus of Martens' work[12] whereby a mono and a stereo two-subwoofer system are compared in an anechoic chamber with the subwoofers located directly to the left and to the right of the listener. Results indicated that the stereo subwoofer configuration enhanced the auditory imagery, primarily by providing better distance perception. This is in line with informal tests carried out by one of the authors of this work at an outdoor music festival during summer 2012. The subwoofer system was wired to allow for toggling between a mono and stereo feed. Switching from mono to stereo during the performances gave exactly the impression described by Martens, where there was more depth to the auditory image.

Finally, Griesinger has performed a considerable volume of research concerning localization, primarily targeted at concert halls, but also applying to home listening[13,14]. In one experiment, a low-frequency stereo image (two subwoofer system) is directly compared to a mono subwoofer placed at the spot the virtual source from the stereo system should rest. As in Borenius' work[3] Griesinger found that filtered speech signals were much easier to localize. His results show that left, right and center positions can be localized well, and critically makes the point that room size is highly important in localization (which no other work has explicitly addressed), going on to suggest in a free-field low-frequency localization should give very accurate results.

Griesinger continues in a later piece of work[14] inspecting how accurately humans can locate a low-frequency source in a listening room (dimensions unspecified). He concludes that localization is possible (with 20 degree accuracy) down to the 63 Hz octave band. Areas containing one or more room-mode node (zero-pressure point) gave more difficulty which is likely due to large phase differences which do not correspond to accurate ITD cues. Avoiding these points, though, the results support the conjecture that low-frequency localization is possible in closed spaces.

An interesting idea revealed by comparison of the research supporting the case that low-frequency localization is not possible and/or important is that five of the six cases only test a single listening room and make no mention of the potential effects of room size. Additionally, nearly all of these experiments (and many of the pro-localization works, for that matter) seem to only use a centrally-located listening position. A central position suffers from a number of nodal points, which Griesinger indicated cause poor localization[14]. These two missing pieces of the low-frequency localization story (room dimensions and listening location) provided the key inspiration for this work which is fully explained in the following section.

3 NEW HYPOTHESIS FOR LOW-FREQUENCY LOCALIZATION

In order to work towards a generalized theory of low-frequency sound source localization it is necessary to go beyond the one-room, one listening location subjective evaluations characteristic of most previously published work[3-14]. In this research, four key variables are identified: room dimensions, listener location, frequency and subwoofer location. The primary aim is to relate these variables directly to localization with a small set of equations. The equations must be derived using objectively-obtained data (rather than subjective evaluation data) so that experiments are repeatable, allowing for verification and validation. Once the proposed set of equations is formed, an initial set of listening tests can be carried out in an attempt to provide subjective verification.

3.1 Simulation results

The most practical approach to formulating this set of equations is to begin in the virtual world where it is possible to simulate any number of configurations. A proprietary finite-difference time-domain (FDTD) simulation toolbox[16] served as this virtual environment. To ensure time-efficiency (and considering the preliminary nature of this work), simulations were arranged in two dimensions and consisted of nine listening locations (Fig. 3.1). Two single subwoofer locations were tested in each virtual space, a corner and a front wall midpoint placement. Room size variation was based on length-to-width ratios where 6:5, 5:4 and 5:3 were tested. The physical dimensions were calculated based on a set of fixed room lengths (5 m, 10 m, 15 m, 20 m, 25 m, 30 m, 40 m, 50 m, 60 m, 70 m, 80 m, 90 m, 100 m).

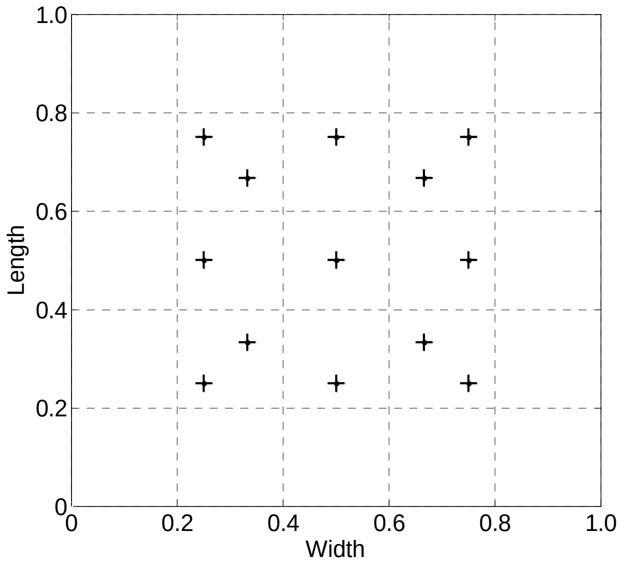

Figure 3.1 Listening location configuration (width and length are normalized)

Virtual recordings were made at each listening location by spacing two omnidirectional microphones at 20 cm to approximate the ears (as suggested by Griesinger[14]). From these closely spaced recordings, velocity vectors were calculated since they have been shown in previous work[9] to be crucial in low-frequency localization. The velocity vector direction for each listening location was compared to the true direction vector pointing towards the source and the differences were calculated over time (an example vector error plot is given in Fig. 3.2).

After calculating directional error for each simulation (which were run at 30, 60, 90 and 120 Hz) two types of plots were generated. First, a room size vs. time below 15° directional error plot shows how room size relates to how much accurate directional information a listener receives (the plot contains a trace for each listening location, as shown in an example in Fig. 3.3). Second, frequency vs. time below 15° directional error plots were generated for each dimension ratio tested (with one plot each for 20 m, 40 m and 60 m room lengths). An example of this is given in Fig. 3.4.

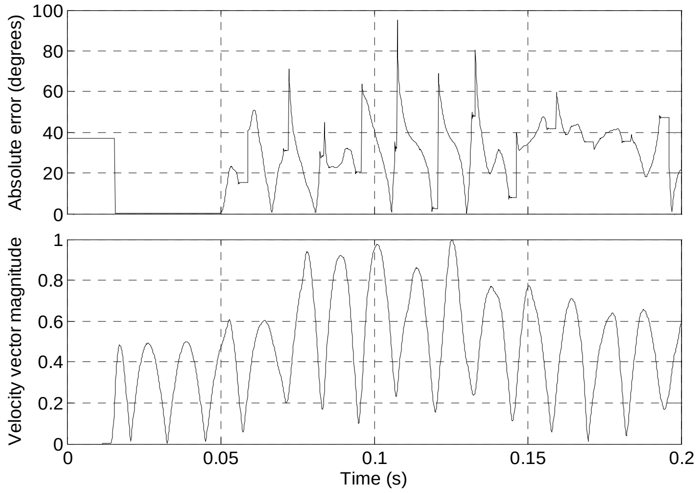

Figure 3.2 Example vector error and velocity vector magnitude plots for a 12 m x 10 m room with a corner subwoofer (40 Hz test tone) and a listening location at (4 m, 3 m)

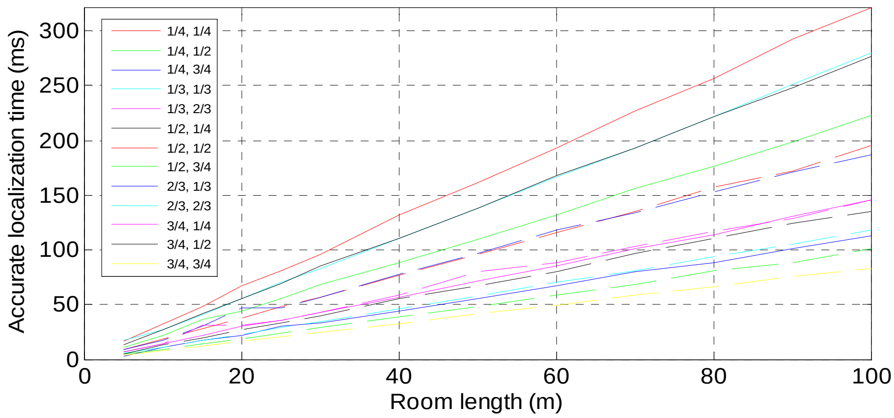

Figure 3.3 Room size vs. time below 15° directional error (corner sub, 5:4 length-to-width ratio)

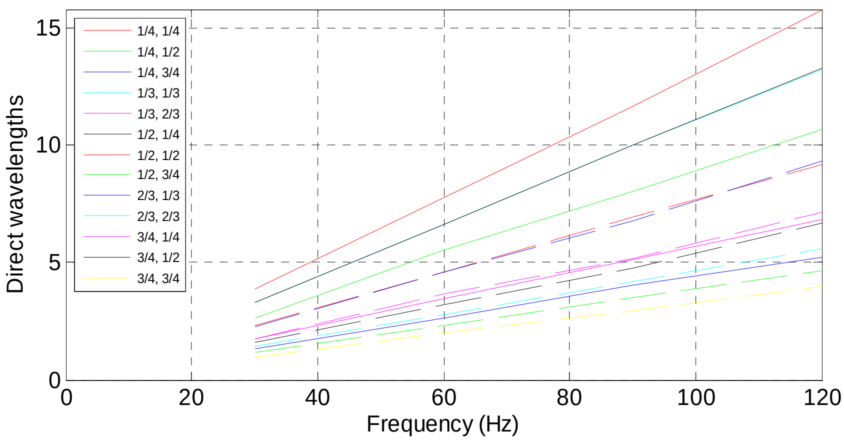

Figure 3.4 Frequency vs. direct wavelengths plot (corner sub, 40 m x 32 m room)

All data stemming from these plots was reduced to a set of two equations: one for room corner subwoofers (Eq. 3.1) and one for wall midpoint subwoofers (Eq. 3.2). These equations are used to determine the number of uncorrupted wavelengths received at a specified frequency (f) before there is more than 15% localization error. Each equation takes into account the room length (l), length-to-width ratio (r) and a normalized distance from listener to source, d (0 – 1). The variable coefficients in the equations were determined with best-fit lines of the data under examination.

$$\lambda_{corner} = 1000\, fl\left[7.22 - 5.42d - 2.68r + 2.02dr\right] \tag{3.1}$$

$$\lambda_{midpoint} = 1000\, fl\left[2.83 - 1.55d - 1.09r + 4.76dr\right] \tag{3.2}$$

Equations 3.1 and 3.2 generate plots for predicting the period of time the wavefront is uncorrupted by boundary reflections for a specified set of room dimensions and subwoofer location (covering all possible listening locations). As an example, plots are generated for a 5 m x 4 m room with a corner (Fig. 3.5a) and wall midpoint (Fig. 3.5b) subwoofer and the same for a 60 m x 50 m room (Fig. 3.6).

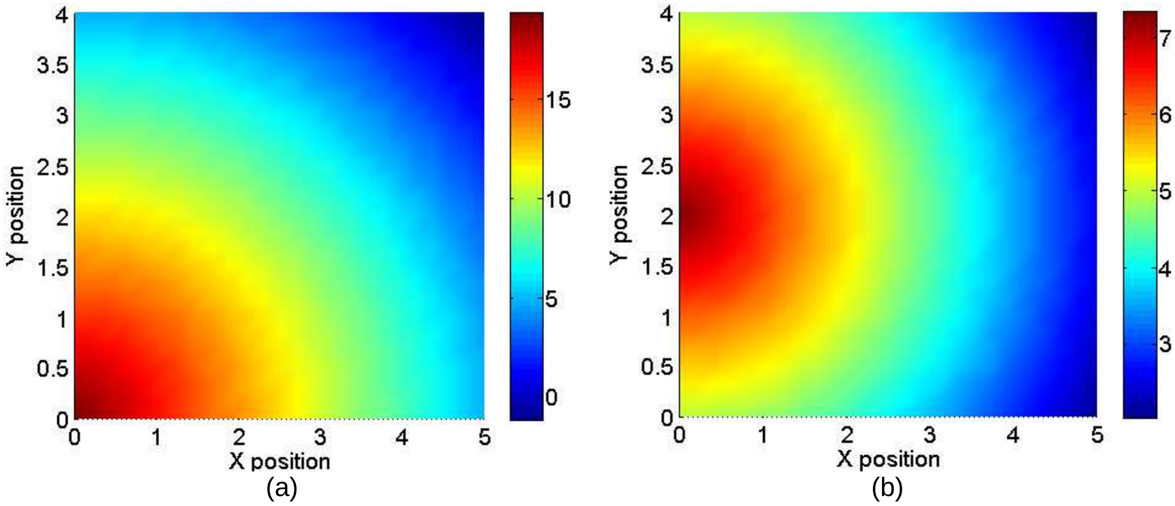

Figure 3.5 Uncorrupted localization time (ms) in a 5 m x 4 m room with a subwoofer at (a) a room corner and (b) at a wall midpoint

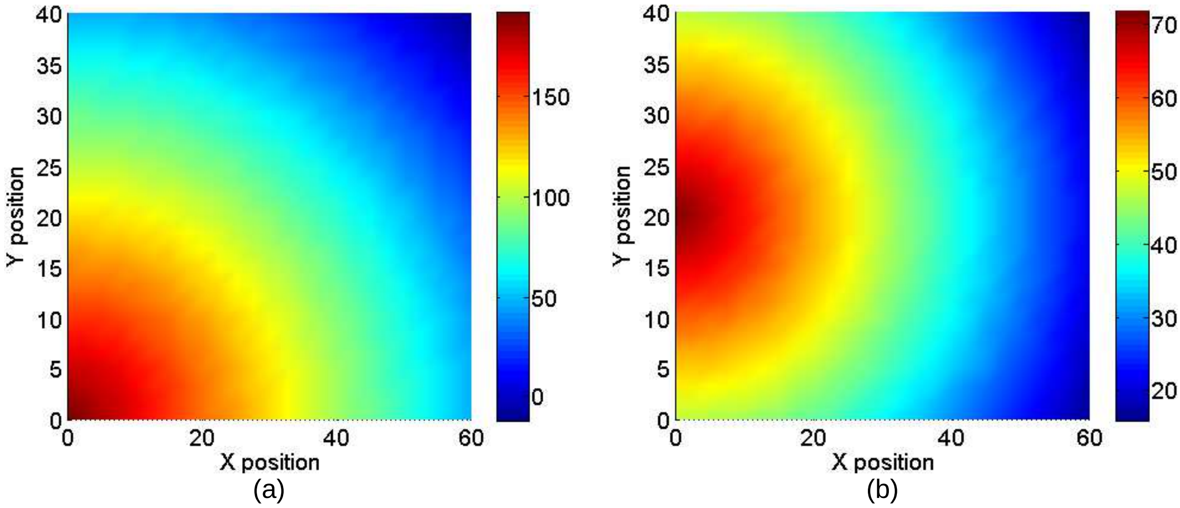

Figure 3.6 Uncorrupted localization time (ms) in a 60 m x 40 m room with a subwoofer at (a) a room corner and (b) at a wall midpoint

What the example plots indicate intuitively is that listeners closer to a subwoofer (and further from side and rear room walls) benefit from having additional time (and a higher direct-to-reverberant sound ratio) to localize a source before reflections corrupt the ITD phase information. Room size and dimension ratios appear to only affect the scale of the uncorrupted localization time, while corner subwoofers achieve enhanced localization compared with wall midpoint subwoofers.

The key point here is that as room size increases listeners benefit from a longer duration of uncorrupted ITD phase information, allowing for precise localization (although this is strongly dependent on listener and subwoofer location as well a signal characteristics). This extra time is believed to be critical especially at the onset of a signal where human ability for low-frequency directional discrimination is likely to be at a maximum compared for example to a pseudo steady-state signal where reflections will induce continual impairment.

3.2 Subjective evaluation

Informal subjective evaluations were carried out in order to provide a degree of confirmation of the simulated results in Section 3.1. Tests were carried out in two rooms at the University of Derby's Markeaton Street campus: a large lecture hall (16.23 m x 9.83 m x 6.50 m) and a multichannel sound lab (6.3 m x 6.9 m x 3.0 m). An omnidirectional subwoofer was placed in the corner of each room and driven by a series of 50-cycle tone bursts at 30, 60, 90, 120 and 150 Hz.

The test subjects were blindfolded and brought into each of the darkened rooms, one at a time and placed on a swivel chair. Between each frequency test they were rotated in the chair to blur their orientation, but were always aligned to face the forward direction. As each tone burst was played the subjects were asked to point in the direction they perceived the sound to originate. These tests were carried out in two listening locations per room. In the large lecture hall the relative positions were approximately (1/3, 3/4) and (1/4, 1/4) and in the multichannel room they were (3/4, 3/4) and (1/4, 1/4). Test results are presented in Table 3.1.

Frequency (Hz)	Perceived directional absolute error (degrees)															
	Large lecture hall								Multichannel sound room							
	S_1		S_2		S_3		S_4		S_1		S_2		S_3		S_4	
	L_1	L_2	L_1	L_2	L_1	L_2	L_1	L_2	L_1	L_2	L_1	L_2	L_1	L_2	L_1	L_2
150	225	0	180	0	235	45	235	45	235	0	135	0	235	45	180	45
120	45	0	135	0	0	0	45	45	45	0	45	0	0	0	45	45
90	90	0	225	0	235	0	135	45	45	0	0	0	0	45	90	45
60	0	0	0	0	180	0	45	45	45	45	0	45	0	45	90	45
30	45	45	225	135	0	0	135	135	45	45	180	235	235	0	135	135

Table 3.1 Subjective evaluation results ($S_{1\text{-}4}$ = test subjects, $L_{1\text{-}2}$ = listening locations)

The key point illuminated by the subjective evaluation results is that listening location makes a significant difference in low-frequency localization. In both rooms, the location furthest from the subwoofer exhibited poor localization, with specifically poor performance across all frequencies in the larger room. The location closer to the subwoofer, on the other hand, gave consistently good localization over most frequencies. In addition to benefiting from a longer duration of uncorrupted directional information, the closer listening location has a higher direct-to-reverberant sound intensity ratio, also allowing for improved localization. The closer location did give poor localization at 30 Hz in the small room, but this is as predicted by the simulations since the time for accurate localization gives only three-quarters (at best) of an uncorrupted wavelength at 30 Hz, which is insufficient for accurate localization.

Although these subjective tests were informal in nature, they give reasonable indication that the equations derived from the simulation data (Eqs. 3.1 & 3.2) are accurate, shown best by the improved localization performance with closer proximity to the subwoofer. Additionally, there is indication that room size does indeed affect the lowest possible localizable frequency. In this case, the larger room allowed for good source identification at all frequencies tested when nearer to the subwoofer while the smaller room gave difficulties at 30 Hz. As most subwoofers are calibrated to reproduce around 20 – 200 Hz, these results indicate that the conjecture that humans cannot tell direction at low-frequencies is not accurate and, in fact, localization is possible and quite good in many cases in this range.

4 CONCLUSIONS & FUTURE WORK

This research provides a first step towards a generalized understanding of low-frequency sound source localization. It by no means provides a definitive answer to this long-standing question, but presents some objectively-obtained data to confirm or disprove many of the conflicting hypotheses regarding this subject.

First, the work confirmed what most researchers have previously suggested: below around 200 Hz, low-frequency localization in small rooms tends becomes impaired. This frequency limit, however, is not one-size-fits-all (as Greisinger suggested[13]), but depends largely on room dimensions. Larger rooms allow for localization of lower frequencies, but smaller rooms prevent any noticeable directionality in the subwoofer band.

Furthermore, both simulation and subjective results in this work highlight the importance of listener location. Listeners closer to a subwoofer benefited from a longer duration of uncorrupted ITD cues while listeners located further away exhibited directional confusion. This is a point that has not been addressed in the literature, although it appears to be a crucial aspect in understanding this subject. The bulk of the reviewed work only examines centrally-located listening points, which characteristically gives poor localization due to phase issues[8,14].

While the equations stemming from the simulation results (Eqs. 3.1 & 3.2) give calculations for the number of uncorrupted wavelengths for localization (factoring in frequency, room dimensions and listener location), they do not indicate the minimum number of wavelengths necessary for accurate localization (or room absorptive properties, which must be examined at some point).

Analyzing the findings of previous work, however, a picture emerges of this wavelength requirement. Borenius' work[3] indicated that minimal localization is possible below 200 Hz in a 7.0 m x 5.2 m room. Using the derived equations in this work, the configuration should give 2.4 uncorrupted wavelengths at 200 Hz (with 1.2 wavelengths at 100 Hz, where it is indicated that no directional information is noticeable). Similarly, Kelloniemi *et al*.[5] showed that directional information is not necessary below 120 Hz in a 6.25 m x 5.58 m room. This corresponds to 1.5 uncorrupted wavelengths. Zacharov *et al*.[6] proposed that localization is effective down to 85 Hz in a 5.03 m x 6.03 m room, which gives one uncorrupted wavelength. Lastly, Kugler and Theile[7] state that one subwoofer is ample for frequencies below 100 Hz in a 6.68 m x 6.62 m room, corresponding to 1.4 uncorrupted wavelengths.

Three of the four works support the minimum number of uncorrupted wavelengths for accurate localization is around 1.4. This may be a good initial suggestion for determining what frequencies can be localized in a room. If a listener receives more than 1.4 uncorrupted wavelengths, the source is localizable at and above that frequency. It must be noted, though, that the wavelength calculation for each of these results was based on the assumption that the listening location in the tests was in the room center. The data in this research has indicated that localization is sensitive to listener location; therefore the 1.4 wavelength suggestion must be investigated in future work.

Overall, it appears that certain low-frequency signals can be localizable under the right conditions. Large venues (such as cinemas, theaters, clubs, auditoriums and outdoor sites) should not contribute significantly to impairment of low-frequency directional data and may actually benefit from the employment of multichannel subwoofers. It has yet to be determined, however, if directional low-frequencies impacts the overall listening experience when high-frequency localization cues are also present, although from an artistic perspective the reproduction of signals that are predominantly low-frequency may well be enhanced by improved directionality. This is a key question for future work as are methodologies for enhancing low-frequency directionality in smaller listening spaces. Nevertheless, it seems the conjecture that humans cannot localize low-frequencies is inaccurate and, in reality, the question concerning localization demands the response: "it depends".

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