SOUND PRESSURE LEVELS IN CLOSE PROXIMITY TO SOUND-REINFORCEMENT LOUDSPEAKERS

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1 INTRODUCTION

Over-exposure to loud sound causes irreversible hearing injury[1]. Galvanized by the World Health Organization's "Make Listening Safe" initiative[2], there is growing momentum behind efforts to develop a global regulatory framework to protect the hearing of audience members in entertainment venues, whilst recognizing and appreciating the profound personal, cultural, and economic benefits of live music.

A small number of countries, most notably in European nations, already have regulations or guidelines in place to protect the hearing of audience members at venues and events[3]. Some of these include requirements restricting audience access to the area directly in front of the loudspeakers, where sound levels are generally at their highest. For instance, guidelines from the UK Health and Safety Executive[4] state that, wherever possible, patrons should not be allowed within three metres of any loudspeaker, and that under no circumstances should the separation be less than one metre. Similarly, a local restriction imposed by the City of Ghent in the Flanders region of Belgium requires that audience members should be kept at least one metre away from any loudspeaker[3]. Austrian regulations[5] require patron access to the area around loudspeakers to be restricted if the sound level exceeds 100 dB L_{Aeq} in that area.

Restrictions on access to the area directly in front of loudspeakers are intuitively well motivated. However, there is little published research addressing how sound levels vary in this area. Research on this topic could help to inform the development of the most suitable recommendations for inclusion in safe-listening guidelines.

The change in sound pressure level with increasing distance from a sound source is well understood in the hypothetical case of a perfect point source radiating into the free field. In that case, the inverse square law dictates that the sound pressure level will decrease by 6 dB per doubling of distance from the source (the "6-dB rule"). Figure 1 shows how, under the 6-dB rule, an increase in sound pressure level of 20 dB is predicted when moving from 3 m to 30 cm away from a sound source.

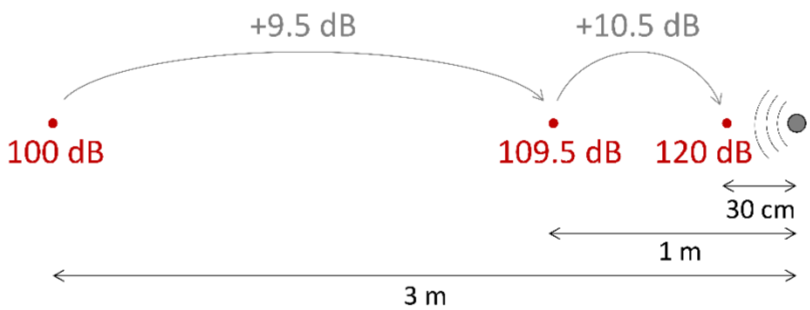

Figure 1 *Illustration of rapidly increasing sound level as one approaches a sound source. These values are for the hypothetical case of a point source in the free field; the change in sound level as one approaches a real loudspeaker will differ from the values shown here.*

However, real loudspeakers, especially those used in sound-reinforcement applications, are far from true point sources (although compact sound-reinforcement loudspeakers are sometimes referred to as "point-source" loudspeakers owing to the configuration in which they are typically used). Indeed, line-array systems, which have become the popular choice in professional applications, are specifically designed not to act like an acoustic point source, but rather, within limits, like a line source, for which the sound pressure level falls off at a rate of 3 dB per doubling of distance[6]**.** It is important to add that, in real situations, the degree to which a line array actually behaves like a line source is heavily dependent on frequency and distance from the source[6].

Further complexities arise from variation in the acoustic environment into which the loudspeaker radiates, the physical presence of a human body in the sound field, and the nature of music as a complex, broadband signal. We must also contend with the fact that, close to the loudspeaker, in the acoustic "near-field", the sound field is a complex mixture of circulating "evanescent" waves and propagating spherical waves, which causes unpredictability in measurement and modelling of the sound pressure level in this area[7]**.**

The aim of the present study was to establish empirically how the sound pressure level arriving at the ears of a human listener varies with distance directly in front of loudspeaker systems typical of those commonly used in live-music venues.

2 METHODS

2.1 Overview

Sound levels were measured at a range of distances (between 30 cm and 3 m) from two different loudspeaker systems. The measurements were made in Metronome[8], a purpose-built 400-capacity live-music venue in Nottingham, UK. The venue is 17 m long by 11 m wide and has a ceiling height of 4.3 m. The venue features a distributed mixture of acoustically absorptive and diffusive treatments across the walls and ceiling and has a mid-frequency RT60 reverberation time of around 0.5 seconds. All testing was carried out in March 2021, when the venue was unoccupied.

2.2 Measurement equipment

2.2.1 Dummy-head recordings

Relative sound pressure levels arriving at the ears of a dummy head (Soundman OKM) were measured using in-ear binaural microphones (Soundman OKM II Classic/Studio A3). The microphone signals were amplified and digitized using a Focusrite 2i2 USB audio interface and subsequently recorded as uncompressed, stereo, 48 kHz, 24-bit WAV files using Reaper digital audio workstation software running on an Apple Macbook Pro laptop. Once initially set, care was taken to ensure that no changes were made to the gain structure of the recording chain, to ensure that levels could be directly compared across recordings. The dummy head was mounted on a clothed mannequin to simulate the presence of a human body in the sound field.

2.2.2 Reference measurements

Reference measurements were made using a Brüel & Kjær Type 2250 hand-held sound level meter with a Type 4189 free-field ½" microphone. Calibration of the sound level meter was performed using a Brüel & Kjær field calibrator immediately before and after the full set of measurements. The drift from start to end of the measurements was below 0.1 dB.

2.3 Loudspeaker systems under test

Photographs of the dummy head and mannequin positioned in front of the two loudspeaker systems are provided in Figure 2.

2.3.1 "Point-source" public address loudspeaker

The first loudspeaker system that we tested was a compact, self-powered unit (Electro Voice ELX112P) representative of the type of loudspeaker commonly used for sound reinforcement in smaller venues. The specifications of the loudspeaker were as follows:

Frequency range:	50 Hz – 20 kHz
Amplification:	1000 W Class D
Low-frequency transducer:	12" woofer
High-frequency transducer:	1.5" titanium diaphragm compression driver
Dimensions (H x W x D):	607 mm x 362 mm x 340 mm

The loudspeaker was mounted at head height on a tripod placed on the floor in front of the stage.

2.3.2 Line-array system

The second loudspeaker system was a compact line-array system (L-Acoustics KIVA II 6-box array) powered by external amplification (L-Acoustics LA4X). Line-array systems are a common solution for professional sound reinforcement in mid-to-large venues. The specifications of the line-array system were as follows:

Frequency range:	70 Hz – 20 kHz
Amplification:	1000 W Class D (per box)
Low-frequency transducer:	2 × 6.5″ neodymium cone drivers (per box)

High-frequency transducer: 1 × 1.75″ neodymium compression driver (per box)
Dimensions (H x W x D): 202 mm x 525 mm x 357 mm (per box)

The line array was suspended from the ceiling to one side of the stage, in the position that it would ordinarily be installed during operation of the venue. However, during testing, the array was lowered so that its vertical centre was just above head height and rotated inwards so that it pointed towards the centre of the rear wall. The venue's permanently installed subwoofer system was deactivated during testing.

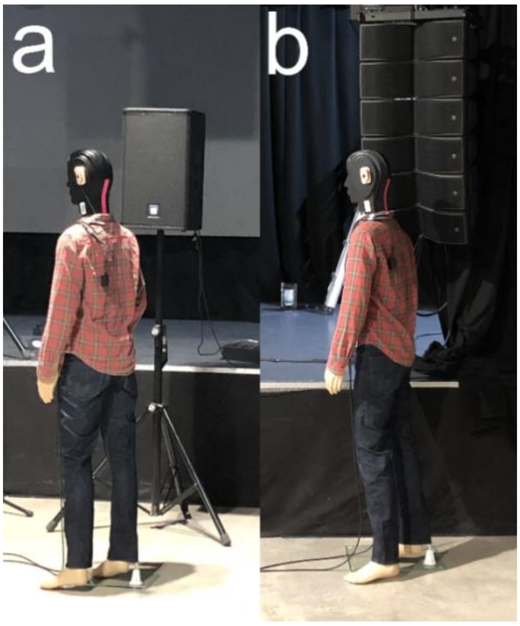

Figure 2 Photographs of the dummy head and mannequin in front of (a) the point-source loudspeaker and (b) the line-array system.

2.4 Test stimulus

A steady noise signal was generated using MATLAB (MathWorks, Natick, Massachusetts, United States) having a frequency spectrum matched to the long-term average spectrum of popular pre-recorded music[9]. This signal was generated by starting out with a frequency-domain signal with the desired magnitude spectrum, adding random phase information, and then converting the signal to the time domain using the inverse fast Fourier transform. The resulting broadband signal was bandpass filtered between 80–16,000 Hz to remove energy at very low and very high frequencies. The power spectral density of the test signal is plotted in Figure 3.

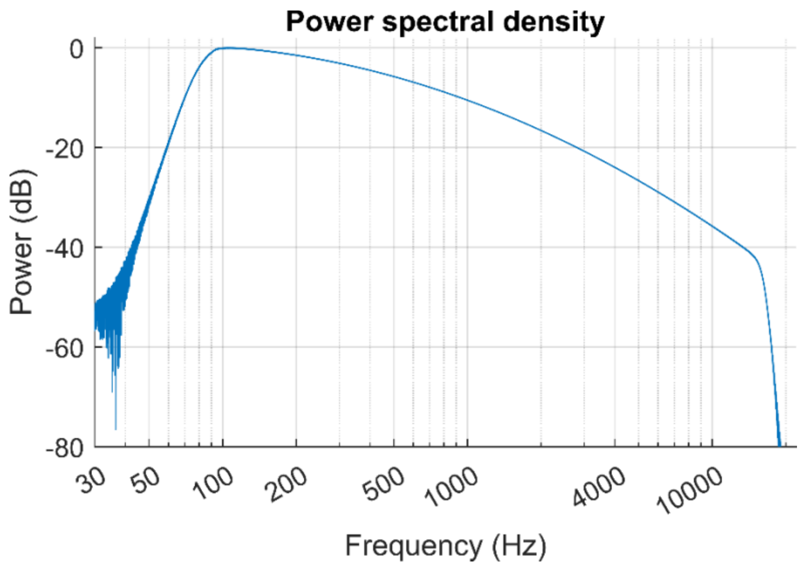

Figure 3 *Power spectral density of the test signal used to excite the loudspeaker systems under test. The test signal was designed to have a power spectrum matched to the long-term average spectrum of popular pre-recorded music.*

During the measurements, the test signal was played through the loudspeaker system under test at a level of approximately 90 dB L_{Aeq} at one metre. The level was set reasonably high to simulate realistic operating conditions and to ensure that all measurements were well above the ambient noise floor.

Because our primary measurement system used microphones that were not designed specifically for measurement purposes and could potentially have been overloaded by high sound levels, we included within the test stimulus a check for linearity of the measurement chain. The test stimulus was digitally edited so that its level was attenuated by 6 dB for the first 15 seconds (half of the total stimulus duration). We checked the recordings to confirm that the measured level increased by 6 dB at the midpoint of the stimulus (see Section 2.6.1), which would indicate that the microphones and recording chain were responding linearly at the sound levels used during testing.

2.5 Measurement procedure

For each loudspeaker system under test, measurements were made at distances of 30 cm, 1 m, 2 m, and 3 m (measured from the loudspeaker grille to the centre of the dummy head). Multiple measurements were made at each distance to assess the sensitivity of the measurements to small shifts in microphone location. For the point-source loudspeaker, we made four measurements at each distance, two with the ears of the dummy head in vertical alignment with the woofer and two with the ears in line with the high-frequency driver. For the line-array system, we made three measurements at each distance, with the dummy head shifted laterally by a few centimetres between each measurement. In all cases, the dummy head was orientated at an azimuthal angle of 60 degrees to the axis of the loudspeaker, this angle having been chosen to produce a maximal level difference between the exposed (right) and shielded (left) ears. Each measurement lasted 30 s in total (i.e. a single reproduction of the test stimulus, including the 6 dB step up in level occurring at 15 s).

The following additional measurements were made using the reference sound level meter, each with a measurement duration of 30 s: i) a measurement of the ambient noise level in the venue, using the moving-microphone technique[10]; ii) for each loudspeaker system under test, a measurement of the free-field on-axis sound pressure level at a distance of 1 m; iii) for each loudspeaker system under test, a measurement of the reverberant field in the rear half of the venue during continuous playback of the test stimulus, using the moving-microphone technique[10].

2.6 Analysis

Labelled audio recordings were exported from Reaper for analysis using MATLAB (MathWorks, Natick, Massachusetts, United States).

2.6.1 Procedure for confirming linearity of the measurement system

An automated thresholding procedure was applied to segment each audio recording into 15-s measurement windows. After digital application of the A-weighting function, for each window, we computed the time-averaged (L_{Aeq}) level at the left and right ears. We then calculated in each case the change in level from the first to the second measurement window, so that this could be compared with the known value (6 dB) that was embedded within the test stimulus.

2.6.2 Estimation of direct sound level

To better characterize the behaviour of the loudspeaker systems under test, while reducing the influence of the specific venue in which testing was undertaken, we estimated the direct sound level from the loudspeakers by adjusting for the reverberant field level. The adjustment was made based on the assumption of energetic addition of the direct and reverberant sound components. In practice, because measurement distances were short (≤3 m) and the venue had a relatively low reverberation time, estimates of the direct sound level were close to the original measured values. The adjustment for the reverberant field was at most 2 dBA and typically less than 0.5 dBA.

3 RESULTS

3.1 Ambient noise level

At the time of testing, the ambient noise level in the venue was 28 dB L_{Aeq}. This was deemed sufficiently low to allow testing to proceed with negligible risk of interference from background sound.

3.2 Confirmation of measurement linearity

Figure 4 shows an example of the measured shift in sound level corresponding to the step-change increase that was embedded within the test stimulus. In this example, the measured increase (5.96 dBA) closely matched the expected value of 6 dB.

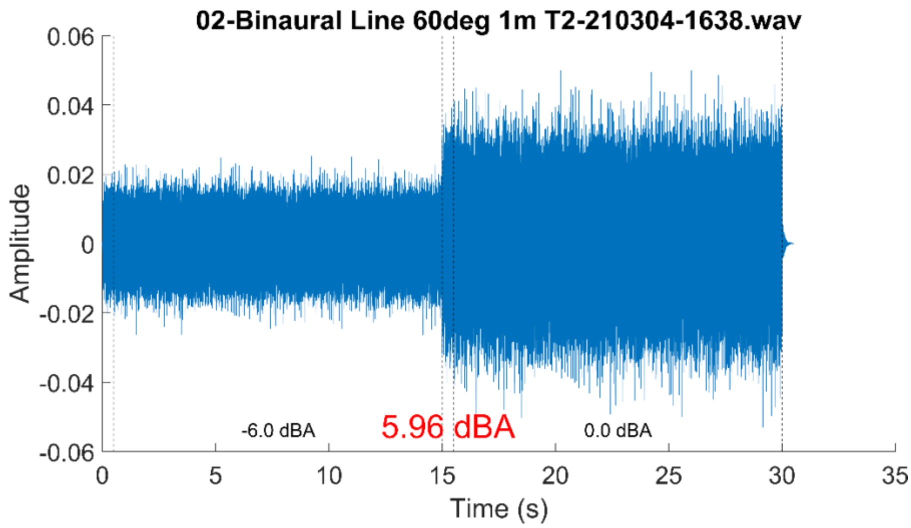

Figure 4 *An example of the procedure used to confirm linearity of the measurement system. In this example, the increase in level computed from the recorded audio signal (5.96 dBA) closely matched the expected value of 6 dB that was embedded within the test stimulus.*

Figure 5 plots the discrepancy between the measured increase and the expected increase (assuming linearity of the measurement system) for all recordings. After considering that at least part of the discrepancy for the point-source loudspeaker seems to have come from a compressive non-linearity in the response of the loudspeaker itself (baseline discrepancy of around -0.15 dBA irrespective of measurement distance), non-linear effects in the measurement chain were small (<0.1 dBA) for absolute sound pressure levels up to 100 dBA. Above 100 dBA, the discrepancy became noticeably larger, indicating increasing non-linearity in the measurement chain. This was likely due to the in-ear microphones becoming overloaded at the highest sound levels. To ensure the validity of our measurements, we took our final readings from the first 15 s of each recording (corresponding to the period in which the stimulation level was attenuated by 6 dB), meaning that all our measurements were made at an absolute sound pressure level below 100 dBA.

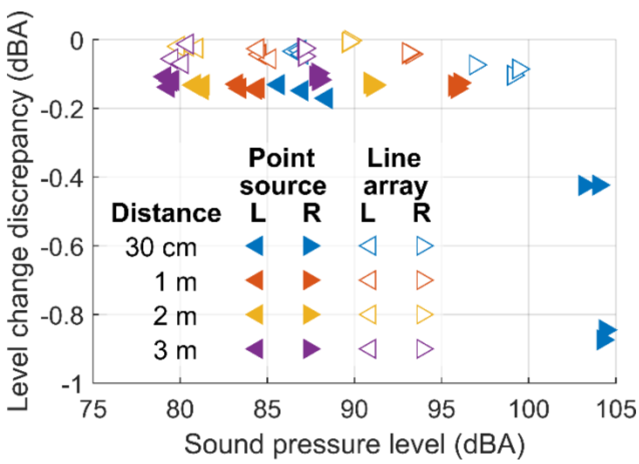

Figure 5 *Measurement chain linearity test results. The discrepancy between the measured increase midway through the test stimulus and the expected value (6 dB) is plotted against absolute sound pressure level at the measurement position. Non-linearity in the measurement chain became pronounced only at absolute sound pressure levels above 100 dBA.*

3.3 Primary measurement results

The primary measurement results are plotted in Figure 6 and reported numerically in Table 1. To aid interpretation, Figure 6 also shows quadratic fits to the datapoints (solid red and black lines for the right and left ears, respectively), as well as predictions under a simple free-field model based on the 6-dB rule (dashed lines).

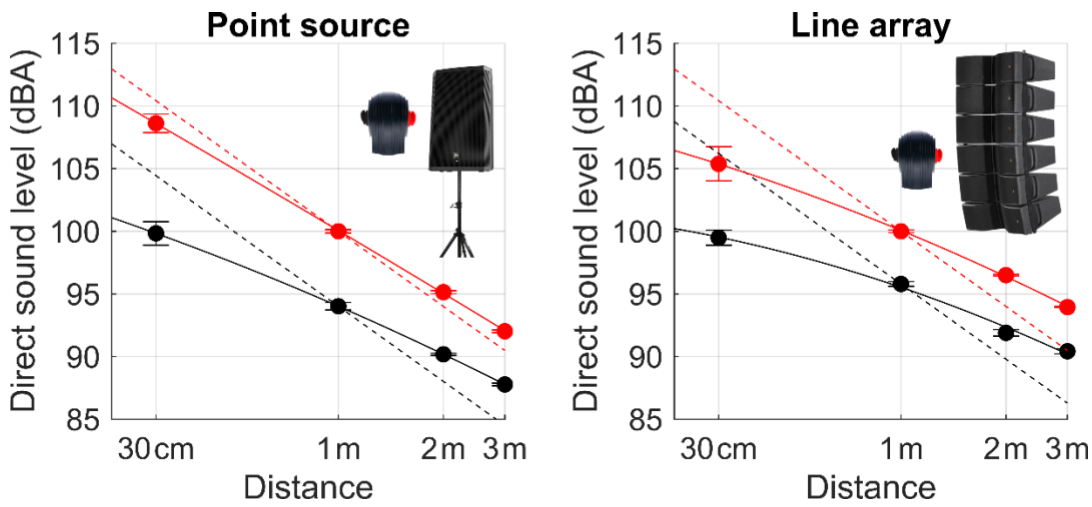

Figure 6 Primary measurement results for the point-source loudspeaker (left) and the line-array system (right). Data for the exposed (right) ear of the dummy head are plotted in red, while data for the shielded (left) ear are in black. The dashed lines show what would happen under a simple free-field model based on a 6-dB change in level for every doubling/halving of distance. All data were normalized to an absolute sound pressure level of 100 dBA at the right ear at 1 m for comparison.

Distance	30 cm	1 m	2 m	3 m
Point source				
Left ear	99.8 (0.94)	94.0 (0.30)	90.2 (0.09)	87.8 (0.10)
Right ear	108.6 (0.74)	100.0 (0.13)	95.1 (0.12)	92.0 (0.11)
Line array				
Left ear	99.5 (0.60)	95.8 (0.19)	91.9 (0.26)	90.4 (0.21)
Right ear	105.4 (1.36)	100.0 (0.11)	96.5 (0.07)	93.9 (0.05)

Table 1 Primary measurement results. Mean (standard deviation) direct sound level, $L_{Aeq,\,15s}$ (dB), for each loudspeaker system at each distance. All data were normalized to an absolute sound pressure level of 100 dBA at the right ear at 1 m for comparison.

4 DISCUSSION

4.1 Sound pressure levels fall off at a rate lower than 6 dB per doubling of distance close to real loudspeakers

The slopes of the solid lines in Figure 6 (quadratic fits to the measurement results) are shallower than those of the dashed lines (hypothetical free-field model based on a 6-dB change per doubling/halving of distance). This implies that the (A-weighted) sound pressure level falls off at a rate lower than 6 dB per doubling of distance in the region close to real loudspeakers.

Compared to the hypothetical reduction of 20 dBA when moving from 30 cm to 3 m away from a sound source, for the two loudspeaker systems tested here, the measured reduction was between 12.1 dBA and 16.6 dBA for the point-source loudspeaker (left and right ears, respectively) and between 9.1 dBA and 11.4 dBA for the line-array system. This corresponds to an effective rate of reduction of up to 5.0 dBA per doubling of distance for the point-source loudspeaker and up to 3.4 dBA per doubling of distance for the line-array system. Thus, in this close-proximity region, and when considering the A-weighted sound pressure level in response to a test stimulus with a spectrum representative of musical material, the “point-source” loudspeaker showed an effective rate of reduction slightly shallower than that for a true point source (5.0 dBA vs. 6 dBA), while the line-array system showed an effective rate of reduction slightly steeper than the theoretical minimum achievable by a line source (3.4 dBA vs. 3 dBA).

4.2 Exposed versus shielded ear

Naturally, the ear that pointed towards the loudspeaker experienced higher sound pressure levels than the ear that was partially shielded by the head. The difference between the ears was more pronounced at locations closer to the loudspeakers. For the point-source loudspeaker, the interaural level difference was 4.2 dBA at 3 m, rising to 8.8 dBA at 30 cm. For the line-array system, the interaural level difference was 3.5 dBA at 3 m, rising to 5.9 dBA at 30 cm. Since we estimated direct sound levels by adjusting for the reverberant field, this distance-dependent effect does not arise due to an equalizing effect of room reverberation at the more distant measurement positions. Instead, larger interaural differences at positions within 1 m of the loudspeakers are attributable to curvature of the wavefront in the near-field region and the way in which this interacts with the head, torso, and pinnae[11].

4.3 Measurement uncertainty is high immediately in front of loudspeakers

We found measurements of sound pressure level to be more variable immediately in front of the loudspeaker (at 30 cm) than at further distances. At 30 cm, the standard deviation across repeated measurements (with small lateral/vertical shifts of microphone position) was on the order of 1.0 dBA, while at distances of 2–3 m it was in most cases an order of magnitude smaller at around 0.1 dBA. In terms of the absolute range between our minimum and maximum measurements made under notionally the same conditions, at 30 cm the average range was 1.8 dBA, while at 2–3 m it was only 0.3 dBA. We attribute the increase in measurement uncertainty at 30 cm to the complexity of the sound field in the near-field region, which makes the measurements sensitive to small shifts in microphone position.

4.4 Implications for safe-listening guidelines

4.4.1 Restrictions on access to the area immediately in front of loudspeakers

In the relatively small number of countries where regulations or guidelines currently exist to protect the hearing of audience members in entertainment venues, it is common to recommend (or require) that audience members should not be allowed to approach within a certain distance (e.g. one metre) of the loudspeakers. The results of the present study confirm that such restrictions are well motivated: although the increase in sound levels as one approaches a real loudspeaker is smaller than predicted by a simple free-field model based on the 6-dB rule, we nonetheless found that sound levels were as much as 8.6 dBA higher at 30 cm than at 1 m. This equates to a reduction in the "safe-listening" time by a factor of 7, e.g. one hour at 1 m distance being equivalent in terms of overall sound exposure to just 8 minutes at 30 cm.

Perhaps counter-intuitively, sound levels can be expected to increase more rapidly when approaching a relatively small ("point-source") loudspeaker than when approaching a larger loudspeaker system with dimensions comparable to the human body. This is relevant, because it is in smaller venues in which compact loudspeakers are most likely to be used, and it is also in these venues that audience members are most likely to be able to approach the loudspeakers unhindered. This analysis does not, however, take account of the fact that larger loudspeaker systems will generally be capable of higher output and will typically be used with more powerful amplification to provide coverage to larger venues. In short, the area immediately in front of the loudspeakers, and especially within one metre of them, is likely to be hazardous at any live-music venue or event, regardless of the specific type of loudspeaker system in use.

One might think that "common sense will prevail," and that no one would choose to stand immediately in front of the loudspeakers. However, it is important to consider other factors that might lead people to occupy these positions, perhaps against their better judgement, such as venue over-crowding or poor sight lines to the stage. Venues should seek to resolve such issues by other means wherever practicable.

Requiring that audience members are kept a minimum distance from the loudspeakers may be easier said than done in many cases. Especially in smaller venues, such restrictions might mean a reduction in audience capacity, and the use of physical barriers might introduce new risks unless the area in front of the stage is continuously monitored by security staff. In many cases, elevating the loudspeakers above head height may be a preferable solution, compared to guarding the area in front of the loudspeakers, this approach often having the added benefit of improving the uniformity of sound distribution across the audience area[12].

4.4.2 Procedures for measuring sound levels in venues

Our results confirmed that measurement uncertainty increases when sound levels are measured very close (<1 m) to loudspeakers. This should be taken into consideration when devising protocols for sound-level monitoring or enforcement in venues, as measurements made close to the loudspeakers may suffer from poor reproducibility.

In most cases, sound levels are likely to be measured at a more distant location, such as at the front-of-house mixing desk. However, the need for measurements close to the loudspeakers could conceivably arise, for example, in jurisdictions that require an upper sound level limit to be met at the most exposed (i.e. loudest) location in the audience area.

Where measurements of the sound level immediately in front of the loudspeakers are called for, but there is concern over measurement reproducibility, one solution could be to measure the sound level slightly further from the loudspeakers and apply an agreed correction to estimate the level immediately in front of the loudspeakers. Based on our measurements for a typical "point-source" public address loudspeaker, we propose that a reasonable worst-case assumption for the size of the necessary correction for measurements made at 2 m distance would be 15 dBA.

4.5 Limitations

The primary limitation of the present study is that only two types of loudspeaker system have been tested and all testing has taken place within a single venue. Caution is therefore needed when generalizing the present findings to different loudspeaker systems operating in different venues. Nonetheless, we believe that our results are, broadly, generalizable, in so far as the two loudspeaker systems we have tested are representative of the sorts of system commonly found in small to mid-sized venues, and the venue in which we undertook our testing was a purpose-built performance space with neutral room acoustics.

A procedural limitation is that we have assessed only broadband (A-weighted) sound levels in response to a specific test stimulus. That test stimulus was designed to have a frequency spectrum matched to the long-term average spectrum of popular pre-recorded music, and so, again, we expect our results to be broadly generalizable. However, we have not addressed here the extent to which A-weighted sound levels at different distances from the loudspeakers might be influenced by varying spectral content across different songs, sets, or genres.

A final limitation, related to our focus on broadband A-weighted sound levels, is that we have not assessed low-frequency sound levels near to subwoofer loudspeakers. While most regulations and guidelines designed to protect audience members' hearing focus primarily, if not exclusively, on A-weighted criteria, concern has been raised about the risk of exposure to high levels of low-frequency sound directly in front of ground-based subwoofers[13].

5 CONCLUSIONS

We used a dummy head with in-ear microphones to assess variation in the (A-weighted) sound pressure levels arriving at the ears of a simulated human listener standing at different distances from two types of sound-reinforcement loudspeaker. Our main conclusions are that:

1) Sound pressure levels close to real loudspeakers fall off at a rate between 3 dBA and 6 dBA per doubling of distance, with a value closer to 3 dBA expected for line-array systems and a value closer to 6 dBA expected for smaller "point-source" loudspeakers.
2) Under realistic conditions, sound levels may be up to around 17 dBA higher at 30 cm from the loudspeakers than at 3 m, corresponding to a 46-fold reduction in the "safe-listening" time at the closer location.
3) Sound-level measurements made very close (<1 m) to loudspeakers show relatively poor reproducibility under small shifts of microphone position; measurements further away (≥2 m) from the loudspeakers are recommended, with a correction applied if necessary.

6 REFERENCES

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