Spatial analysis of acoustic support on auditorium stages: modelling and measurement using high-order transducers

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ABSTRACT

Acoustic support on auditorium stages is conventionally quantified using the stage support parameters (e.g., ISO3382-1). These parameters are derived from room impulse responses measured between an omnidirectional source and receiver. For early stage support (ST_{Early}), the source and receiver are horizontally separated by 1 m, and the degree of support is represented as an energy ratio between early reflections (20-100 ms) and the direct sound (0-10 ms). The project described by this paper introduces two modifications: firstly, the source and receiver are co-located (or at least, concentric); and secondly, the source and receiver can both be resolved into higher order spherical harmonics (enabling detailed spatial analysis). This concept has been realised within room acoustical modelling software, as well as in a prototype transducer array for room acoustics measurement. This paper describes the methods used to achieve modelling and measurement in this way, provides examples of such work, and outlines the potential practical benefits of spatial analysis of the acoustic response around a point on stage.

INTRODUCTION

An auditorium stage environment provides acoustic support to musicians and human speakers in a number of ways. Acoustic early reflections make a performer's own sound seem louder, without which it would seem that the sound was disappearing into space. The ability to play in an ensemble (playing in time, in tune, and in balance – i.e. to perform with effective interaction) is also enhanced through acoustic support from reflections. Performers may also find some satisfaction from being able to hear their own sound reverberate in the auditorium (Gade 1981, 1989a, 1989b).

Although methods for measuring acoustic support on stages have existed for many years and are detailed in a widely applied international standard (International Organization for Standardization, 2009), there remains some question over their adequacy (Dammerud et al. 2010). Current measurement techniques are mainly based on energy ratios, between a nominated evaluation range and the first 10 ms of a room impulse response, measured between omnidirectional transducers separated by a distance of 1 m. The most commonly used of these parameters is ST_{Early} (also known as ST1), for which the evaluation range is between 20 ms and 100 ms, to represent the strength of the early energy (compared to the energy of the direct sound and first-order floor reflection). One clear advantage of the current techniques is that the measurement is simple and inexpensive to implement. However, such measurements do have some vulnerabilities in representing the quantity and quality of stage support, and such vulnerabilities can be avoided with more detailed measurement methods. This paper explores the potential of more detailed methods.

Typically, when we (the authors) measure conventional stage support parameters, we position the omnidirectional loudspeaker at what we might define as the measurement location, and then make four measurements, with the microphone at 90 degree intervals around the loudspeaker (at a distance of 1 m) (*c.f.* Ternström *et al.*, 2005; Cabrera *et al.*, 2010). Stage support values are calculated for each of the four measurements, and values are exponentially averaged to yield the final single value for the measurement location. Several measurements at various locations would normally be used to characterise the stage as a whole, and it is usually possible to see a pattern of variation in stage support over the stage area. However, it is common to see a 2 dB range of ST1 values around the loudspeaker in a single measurement location using this approach, and fluctuations can be as much as 5 dB. Reasons for such fluctuations do not necessarily include the less-than-ideally omnidirectional sound source, because a source that has near-perfect horizontal omnidirectionality can be used (*e.g.*, Brüel & Kjær 4295). Instead the fluctuations mainly come from the size of the area spanned by the 2 m diameter circle over which the measurements are made. Such fluctuations would be reduced, to some extent, if the source and receiver were closer together, but the closer together they are, the more difficult it is to obtain a good signal-to-noise ratio in the measurement (because the direct sound from the source increases in amplitude so a greater dynamic range is needed to accommodate both the direct sound and room reflections). Furthermore, close measurements may be more prone to shadowing effects due to the size of the source (mainly in the high frequency range), as well as near-field radiation effects (mainly in the low frequency range).

In this project we simplify the theoretical concept of stage support measurement by using a co-located source-reciever pair. This allows us to characterise the acoustics around a single point, rather than between a pair of points. Hence the measurement location is better-defined, and can be characterised without the degree of uncertainty seen in conventional measurements. While this is conceptually simpler than conventional measurements, its practical implementation is considerably more complex (including the challenge of managing signal-to-noise ratio). Indeed, no source or receiver can function as a point, and so literal physical implementation is impossible.

We are implementing the concept in two ways. Firstly through a physical transducer array, consisting of concentric spheres (a sphere of loudspeakers and a sphere of microphones). The second implementation is in computer simulations, in which the source and receiver can be points that are very close (or actually co-located, depending on the limits of the modelling software). In computer modelling, attaining adequate signal-to-noise ratio does not pose any problem, but instead some limitations arise in the approximations made by the modelling algorithm.

The second concept developed in this work is the spatial analysis of the room acoustics around the measurement point. Ueno and Tachibana (2003) developed a system for measuring the spatial environment of stages using six orthogonal cardioid microphones near an omnidirectional sound source (for the purpose of laboratory auralization of stage environments). Cabrera *et al.* (2010) derived directional stage support parameters using a similar approach. In the present work, we take this type of spatial analysis further, by characterising both the source and receiver via a set of arbitrary directivities. These directivities can be expressed as beams (which are formed to approximately evenly sample all directions around the measurement point), or as spherical harmonics (which are the spatial-frequency representation of directional characteristics).

IMPLEMENTATION

Transducer array

A prototype concentric microphone and loudspeaker array has been constructed based on a dodecahedron. Each of the twelve faces has an independent loudspeaker driver occupying most of its surface. A small microphone is mounted on each of the docecahedron vertices (*i.e.*, there are 20 microphones). At the time of writing, this device has been designed and constructed, but physical testing and measurements remain to be done.

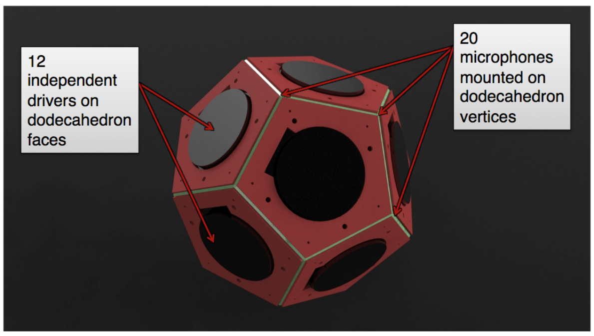

Figure 1. Computer-rendered model of the transducer array.

Each driver is in a sealed enclosure (laser-cut from 4 mm plywood), and these enclosures tessellate to form the dodecahedron. The driver model is Aurasound NS3-193-8A: a 70 mm nominal diameter driver capable of relatively large excursion (10 mm peak to peak). The small neodymium magnet structure of this driver is very useful in reducing the required size of each loudspeaker enclosure (thereby bringing the drivers closer together than would be possible with a conventional ferrite magnet). However, by bringing the loudspeakers close together, the available volume of the enclosures is reduced to less than that which would normally be chosen for the driver, which raises the low cut-off frequency of the loudspeaker's transfer function. The volume of each enclosure (subtracting displacement from the driver) is 0.219 L. The drivers (considered individually) provide useful radiation from 200 Hz and above (their low frequency response is limited by the low compliance of the small sealed enclosure and the radiation impedance implicit in their small radiating area). The theoretically calculated transfer function from a single driver mounted in its enclosure is shown in Figure 2.

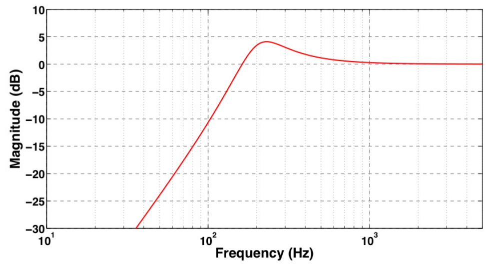

Figure 2. Calculated magnitude of the transfer function of each loudspeaker driver, based on Thiele-Small parameters.

The microphone model is Knowles Acoustics FG-23329-C05. The microphone array was designed based on the principles of higher order Ambisonics (HOA) capture. HOA is based on the decomposition of the sound field around a point into its spherical harmonic components. The ability of a microphone to capture a soundfield to be encoded into HOA signals depends on the spatial arrangement of its transducers. The limits of a microphone array to encode HOA signals up to a desired order is limited by transducer positioning errors and transducer noise at lower frequencies and spatial aliasing at higher frequencies. At lower frequencies the range of the array can be increased by using more transducers (oversampling). The distance between transducers has to be reduced to correctly extend the higher frequency limit.

The actual spacing of the microphones was limited by the requirement to house the loudspeaker array. As mentioned previously, the loudspeaker array was made as compact as practical, given the size of each driver and the walls separating each loudspeaker enclosure. As a result, adjacent vertices (*i.e.*, the microphone positions) are 79.5 mm apart. By calculating encoding filters for the microphone array, we can predict its performance by comparing the frequency average error in single frequencies between the expected spherical harmonic expansion coefficients and the obtained results. A theoretical estimate of the signal-to-noise ratio is introduced to the response calculations to model the error at low frequencies (it should be noted that the placement error is not included in this calculation). The inverse of the error estimate is then obtained and results are expressed in decibels to show the signal-to-noise ratio. Figure 3 shows this for the microphone array at HOA orders up to 3.

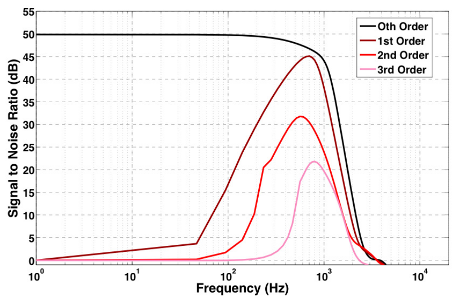

Figure 3. The microphone array's signal-to-noise ratio for spherical harmonic orders 0 to 3 as a function of frequency.

The useable frequency range for 3rd order encoding is approximately 500 Hz – 1300 Hz.

The device is suspended using cables, so as to avoid any substantial support structure that would interfere with the radiation and reception directional sensitivity.

Our initial aim for this prototype device is to make co-located source-receiver measurements (with spatial analysis) in the 1 kHz octave band, which is well within the frequency limits of the loudspeaker and microphone arrays, and is also within the frequency range most relevant for acoustic stage support.

The synthesis of source directivity from a compact loudspeaker array has been the subject of several recent studies (*e.g.*, Pasqual, Arruda & Herzog 2010; Pasqual, Herzog, Arruda 2010; Zotter & Pasqual 2011), and there are various ways by which a measurement can be made. One approach is to measure from one loudspeaker at a time – requiring this procedure to be done twelve times. This can be time-consuming, especially considering measures that may be taken to optimise signal-to-noise ratio. Using this approach, generalised directivity patterns (such as the spherical harmonic series) are synthesised in the analysis stage *post hoc*. Another approach is to make a smaller set of measurements by synthesising spherical harmonics. Hence, a first-order measurement requires four iterations, while a second-order measurement requires nine. Alternatively, other sets of directivity patterns could be synthesised.

Optimising the signal-to-noise ratio in a measurement is more difficult than for conventional measurements with spatially separate transducers. The direct sound (from loudspeaker to microphone) is not relevant to the analysis, and so is truncated from any impulse response. However, the direct sound has high amplitude, relative to the amplitude of room reflections. One technique is to use a long duration swept sinusoid (or repeated shorter swept sinusoids) such that the dynamic range after deconvolution becomes very large. Another technique is to use repeated short duration (e.g., 3 ms) excitation signals, so that the direct sound is not present when the room-reflections return to the device. We are currently still examing the most effective approach for this.

Computational room acoustics modelling

A computational simulation of co-located source-receiver measurement has been implemented using two room acoustics modelling programs: MCRoomSim (Wabnitz *et al.* 2010) and ODEON (Rindel 2000; www.odeon.dk). Examples in this paper were produced using the first of these, which models the acoustic response of a rectangular room using a hybrid ray-tracing and image-source approach, incorporating scattering. MCRoomSim supports high order sources and receivers, and a source-receiver pair can be co-located. ODEON is a well known and versatile room acoustics modelling program that can model arbitrary room shapes. While it does not directly support high order sources and receivers, these can be indirectly incorporated into a model. A high order source is achieved by succesively modelling a source with the directivity for each phase sign of each degree and order of the desired spherical harmonic series at the source location. Hence, seven such sources are required for first-order spherical harmonics (including the zeroth order component). A high order receiver is achieved in ODEON by analysing the directional of arrival of the rays (as was done previously by Favrot and Buchholz (2010)).

For this paper, we model a simple rectangular room in MCRoomSim with the dimensions 30 m x 15 m x 9 m (height). The absorption coefficient and scattering coefficient of all surfaces at all frequencies is 0.5. In order to assess somewhat stage-like conditions, the selected measurement points are 6 m from the end of the room.

Parameters

In this paper we consider two parameters that could be used to characterise stage acoustics. The first is an adapted version of early stage support, ST_{Early}. In its standard form, ST_{Early} is measured between an omnidirectional source-receiver pair horizontally separated by a 1 m interval (between acoustic centres). Octave band values are calculated as per the following equation:

$$ST_{Early} = 10\log_{10} \frac{\int_{0.02s}^{0.1s} p^2}{\int_{0s}^{0.01s} p^2} \tag{1}$$

Values are measured in the 250 Hz – 2 kHz octave bands, and the single-number value is the arithmetic average of these band values (International Organization for Standardization, 2009). ST_{Early} values tend to be negative: surveys of concert halls by Gade (1989b) and Beranek (2003) yield a median value of -14.6 dB, with a range of 8 dB.

In calculating the modified ST_{Early} values, we use the same value for the denominator (*i.e.*, the integrated squared sound pressure over the first 10 ms of the impulse response, at a 1 m distance from the source). In order to acquire this value, the source needs to be calibrated – using an external microphone at a distance 1 m from the acoustic centre of the source. If amplifier gain is kept constant, and floor is flat and acoustically reflective, then the value of the denominator should be quite constant. This approach allows for direct comparison between values of our modified ST_{Early} and the standard approach to measuring ST_{Early}. Recently Wenmaekers *et al.* (2012) addressed this problem similarly.

In our adaptation of ST_{Early}, this value for the denominator is used not only for the omnidirectional ST_{Early} measurement, but also for directional measurements. Rather than squared pressure, the numerator is generalised to integrate the squared amplitude of the received sound (which, strictly speaking, represents a combination of pressure and pressure gradient). On average, a more directional sensitivity pattern should have a lower ST_{Early} value than the omnidirectional value at that position, but when the sensitivity is directed towards an area of strong early reflection, the ST_{Early} value should be greater than the omnidirectional version.

For this paper we consider the simple case where one of the transducers is omnidirectional, and the other has cardioid directivity (which is directed, successively, to many evenly spaced angles around the measurement point). This is a case where reciprocity applies, so the result of the measurement will be the same regardless of which transducer (loudspeaker or microphone) is the directional one. Note that the diffuse field sensitivity of a cardioid pattern is 4.77 dB less than that of an omnidirectional pattern, and so 4.77 dB is added to the value of ST_{Early} to compensate for this (as done previously by Cabrera *et al.* (2010)). While cardioid ST_{Early} may vary considerably with direction, the spatially averaged value (integrated over the sphere) should be close to the omnidirectional value after this 4.77 dB compensation has been made.

The evaluation range in the numerator for ST_{Early} also needs a small adjustment when source and receiver are co-located, because the direct sound arrives at the microphone at the same instant as it leaves the loudspeaker (instead of 2.9 ms later). Hence the integration period in the numerator should be 22.9 ms – 102.9 ms for maximum compatibility with the standard measurement. Further adjustments might be made based on the actual electro-acoustic latency of a physical measurement system and the speed of sound. This might appear to be a trivial adjustment, but it can affect the values significantly (by several decibels) when there is reflected sound in the vicinity of 20 ms.

The second parameter that we consider is a measure of sound field diffusivity at the measurement point. There are many ways by which diffusivity (or isotropy) can be characterised, but most approaches are concerned with the distribution of energy over the sphere of possible arrival directions at the measurement point (Gover *et al.* 2005; Bassett 2012). There are also some approaches that consider only the temporal features of a single channel impulse response (e.g., Defrance & Polack 2008), but these are unnecessarily indirect if spatial microphony is available. However, if the phase (or time) relationship of the sound arriving from all directions within the evaluation period is neglected, then a measurement is unable to distinguish a random soundfield from a converging spherical wavefront focussed onto the measurement point (to take an extreme case). Since focusing can be an important issue in real room acoustics, our approach to quantifying diffusivity can be thought of as spatio-temporal diffusivity (rather than purely spatial diffusivity). The sound field arriving at the measurement point is spatially analysed in terms of a spherical harmonic series (retaining temporal information). The covariance matrix of this series is calculated, and the diffusivity metric is based on the variance of the eigenvalues of this covariance matrix (Jin and Epain 2012). For the purpose of this paper, we use the same time period as for ST_{Early}, and the somewhat speculative intention of this diffusivity calculation is to provide some information on the quality of the early stage support.

MODELLED CASE STUDY

Effect of source-receiver separation on ST_{Early}

As was noted in the introduction, in standard measurements of ST_{Early} on a real stage, the particular position of the microphone on the 1 m radius circle around the acoustic centre of the source often has a remarkable effect on the derived value. For this reason, it is prudent to make four (or probably more) measurements around the loudspeaker if the aim is to characterise the stage support as it varies across the stage area. Considering this, it seems likely that ST_{Early} values would also vary considerably if the source-receiver distance is varied (including if they are collocated) – even if the integration time period is shifted appropriately.

Using the modelled room in MCRoomSim, ST_{Early} values were calculated on a grid of microphone positions (with omnidirectional source and receiver) around and at the source, on the horizontal plane. The grid spacing is 0.2 m, extending 1.2 m from the source in each dimension. The results of this example simulation are shown in Figure 4, which has the ST_{Early} value for the co-located transducers at its centre, and also shows a circle 1 m in radius. The interesting result here is not the actual values, but the large variation in values that occurs around the circle, and across the whole plane. Although this is the result of computer simulation, experience with real measurements is not at odds with this degree of variation. The measurement point is position A (as described in the next section) on the midline of the room (hence the symmetry in Figure 4).

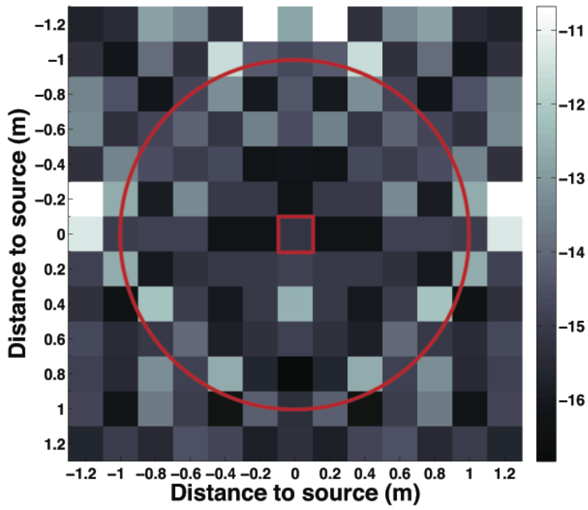

Figure 4. ST_{Early} calculated for a fixed omnidirectional loudspeaker (located at the centre of the grid) and 169 omnidirectional receiver positions in the vicinity of the source, including the co-located case. The circle represents a distance 1 m from the source.

Directional ST_{Early}, and the effect of position

Now we examine the consequences of directionality in ST_{Early} (as described earlier). Using the computer-modelled room, four co-located ST_{Early} calculations were made with one transducer having cardioid directivity, while the other is omnidirectional. The lobe of the cardioid pattern is aimed horizontally, and successive calculations are made for 72 angles (at 5 degree azimuth increments). This process is conducted at four locations in the modelled room, all at a height of 1.5 m (to represent the standing mouth/ear height of a singer) and 6 m from the back of the stage. Positions A-D are 0, 2, 4 and 6 m, respectively, from the midline of the 15 m wide room. Calculated values are shown in Figure 5.

As might be expected, ST_{early} increases as the measurement position moves closer to a side wall. Note that position D is closer than 3.45 m to the wall, so the first order side-wall reflection is not included in the energy used to calculate ST_{early} – and yet the value still is the greatest of the four positions. There is a strong bias in directional stage support towards the 'upstage' direction – i.e., to the closest of the end walls. In general terms, results are similar to the separated transducer measurements that were made in a real theatre, reported by Cabrera et al. (2010) (1 m separation, 1st order cardioid microphone and omnidirectional loudspeaker). Also, the modelled directional ST_{early} values appear to be compatible with their omnidirectional counterparts.

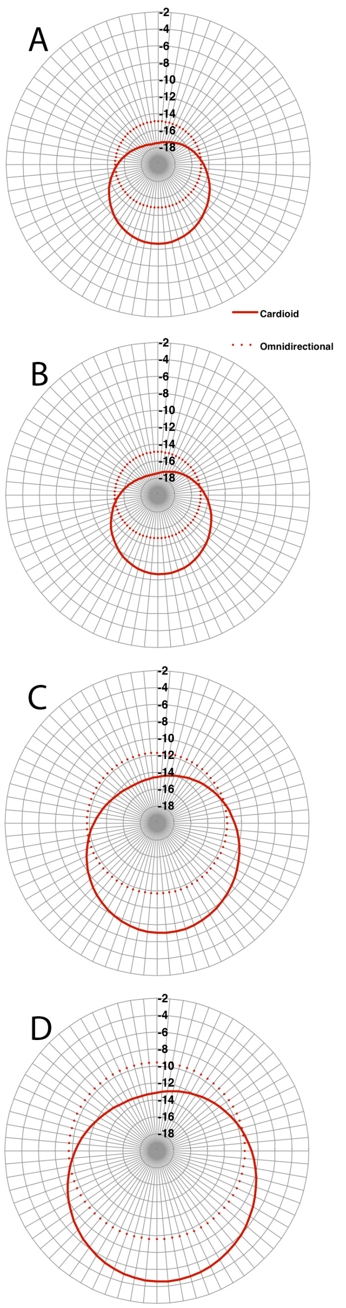

Figure 5. ST_{Early} calculated between co-located omnidirectional and cardioid transducers, with the directional lobe of the cardioid transducer rotated around a horizontal circle. For reference, omnidirectional ST_{Early} is shown by the dashed circles. Each chart represents values for a position (A is halfway across the stage, and D is 1.5 m from a side wall, with B and C at intermediate positions).

Diffusivity

The diffusivity of the sound returned to the measurement point during the ST_{early} integration period was calculated for positions A and B. For an omnidirectional source, values are 0.4 for position A, and 0.45 for position B (on a scale from 0-1). This is interesting because these two positions have approximately the same value of ST_{early}, but the diffusivity differs because position A is exactly halfway along the width of the room, and so there is spatial symmetry in the sound field – whereas position B does not have this characteristic.

Figure 6 shows the diffusivity calculated for a directional (cardioid) source, which varies depending on the elevation and azimuth angle to which it is directed. The reduced diffusivity of position A (compared to B) is clearly seen again. Furthermore, greatest diffusivity occurs when the sound source is directed horizontally. A small reduction in diffusivity is observed if the source is directed towards the upstage wall (the end wall closest to the positions) – the upstage direction is on the equator in the righthand side of the figure.

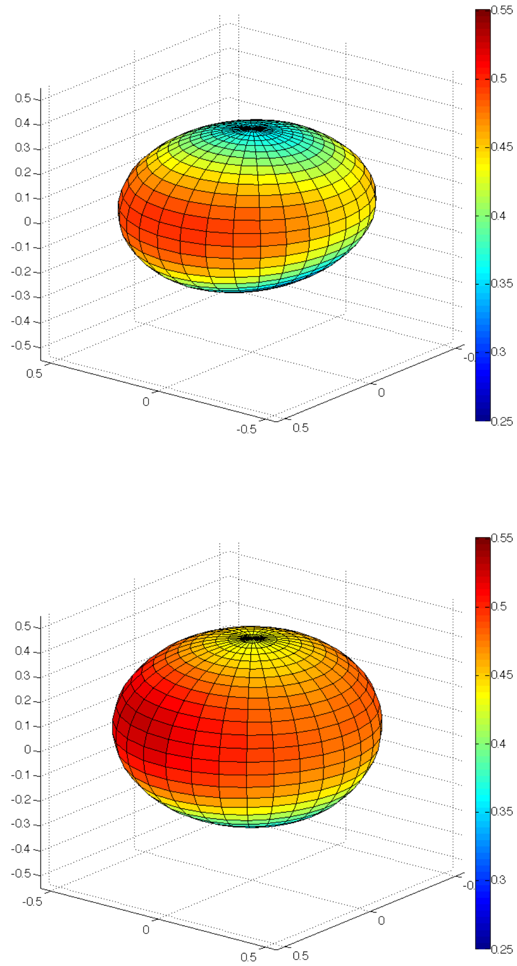

Figure 6. Spherical plot of diffusivity of the sound returned from a cardioid sound source (within the ST_{early} integration period), showing how values vary with the direction at which the source points. The upper chart is for position A and the lower chart is for position B. The vertical axis extends in the room's vertical direction; the lower axis on the left is in the direction of the room length; and the lower axis on the right is in the direction of the room width.

DISCUSSION

The approaches taken to stage acoustics characterization that have been outlined in this paper still require further development and testing for real-world deployment, but they show some promise. The relevance of these and similar approaches to the experience of performers on stages needs to be explored, refined, and validated, via subjective tests. Nevertheless, the advantages of the proposed approach include reduced uncertainty from transducer positioning, and greater detail available for analysis (which can be used to address some of the vulnerabilities of the standard approach).

The physical transducer array described in this paper is clearly rather limited in function. While its working frequency range is within the range relevant to stage acoustics, it would be desirable to have a device that covered the full range currently used (250 Hz – 2 kHz octave bands), and beyond. This can be achieved through a more elaborate design, almost certainly with more microphone channels.

The simple approach to computational room acoustics modelling taken in this paper was merely for illustrative purposes. Modelling of more realistic stage environments would require the greater flexibility provided, for example, by ODEON – for which we have also developed implementation techniques. It is likely that a greater range of calculated values would occur on some of the more realistic stage environments that would be modelled in ODEON.

For the analyses in this paper, we used the 20-100 ms early energy integration period from the standard form of ST_{Early} (with appropriate time-shift to compensate for co-location). However this rectangular integration window is one of the vulnerabilities of ST_{Early} – because a reflection can be near one of the integration limits, and whether or not it is in the window can significantly affect the calculated value. This vulnerability is especially acute at the start of the integration period, where the energy of an individual reflection can often be relatively strong. Perhaps a more robust approach is to use a window without sudden transitions, and there are many possibilities (which are beyond the scope of the present discussion).

The denominator (i.e., the reference energy) for stage support is another area that deserves re-examination. In the present work, we used the same denominator as for standard meassurements, but this would be problematic if the spectral resolution of the measurement were to be increased, because of the comb filter between direct sound and floor reflection (at the 1 m radius microphone position). The tuning of this comb filter is particular to the spatial relationship between the direct sound and floor reflection, and this relationship does not apply to subsequent energy received by the microphone. For this reason, it may make sense to use the free field sound emited from the loudspeaker (without floor reflection) as the reference in more detailed measurements.

There are other approaches that can be taken to the quanitification of reflected energy. Strength factor, G, which is often used to represent the energy in the audience area from the stage, can be adapted to stage support. Brunskog *et al.* (2009) proposed a modified version of strength factor, known as room gain (G_{RG}). Their approach, which is focussed on autophonic speech, uses a head and torso simulator to measure the increase in energy transmitted from mouth to ears due to the room environment. Later, Pelegrín-García (2011) proposed a modification of their technique to calculate a variant of stage support, known as voice support (without the direct sound in the numerator). However, these approaches are only suitable for quantifying the autophonic voice support (or gain) in acoustic environments because of their dependence on a head and torso simulator. Ranjbari (2012) proposed a form of on-stage strength factor, essentially the same as conventional strength factor, except that the transducers are on stage: G_{self} is measured at 1 m (like conventional stage support). The important distinction between strength factor and stage support is that the former includes the direct sound in the numerator, while the latter does not – meaning that the minimum value for G is 0 dB (or 10 dB if the reference position is at the conventional 10 m), whereas there is no minimum value for stage support. When a source-receiver pair are in close proximity (such as in a head and torso simulator), the direct sound may be strong, making a strength factor measurement insensitive (because if the reflected sound energy is relatively weak, the result will be near 0 dB). For co-located measurements, the direct sound has very high amplitude (conceptually, the amplitude is infinite if source and receiver are co-located points), and so the direct sound must be excluded from the numerator.

If a stage is being characterised as a whole, there may be not particular benefit in co-locating source and receiver because the acoustic response of the room at particular locations is not of interest. However, if the acoustic response at particular locations is of interest, the co-located source-receiver removes the degrees of freedom and consequent uncertainty associated with separated transducers (which, using the current method, yields substantial variation in the derived value, depending on the loudspeaker's position on the 1 m radius circle). This has the potential to provide a more reliable map of variation in acoustic quality across a stage area, and also can be an important step in auralization.

CONCLUSIONS

Whether the benefits of the techniques examined in this paper justify their costs (logistical, computational and financial) remains to be seen, but is seems very likely that high spatial resolution transducers (especially spherical microphone arrays) will be part of the normal toolkit used for room acoustics measurements in coming years. Some of the benefits demonstrated in this paper include: being able to identify the directions (and hence surfaces) from which acoustic support is provided (quantified by conventional values of stage support); accounting for various instrument directivities (and rotations thereof) thereby enumerating the sensitivity of support to directivity; removing the uncertainty associated with a 2 m diameter measurement circle; and introducing spatiotemporal diffusivity and associated statistics (which should contribute to the understanding of the *quality* of stage support), especially for solo performers.

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