Play that funky music: Making 3D acoustic measurements of instruments under performance conditions.

Conference or Workshop Item

PLAY THAT FUNKY MUSIC: MAKING 3D ACOUSTIC MEASUREMENTS OF INSTRUMENTS UNDER PERFORMANCE CONDITIONS.

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1. INTRODUCTION

This research involves the design and implementation of a high spatial resolution acoustic radiation measurement system for the purpose of collecting data about the acoustic radiation in 3 dimensions from musical instruments while they are being played by musicians (examples[1]). It is also possible for the system to be used to capture 3 dimensional data about other sound sources.

Each musician may, when they perform upon their instrument, attempt to play consistently with the same dynamics and tone, the same timing and *feeling*, but will ultimately fail[2]. This is not a failing in the musician themselves, but rather a function of the fact that musicians are, like all people, biological beings, and therefore cannot perform exactly the same movement with the consistency required for scientific data gathering when the purpose of the movement is to actuate another object (the instrument in this case). The problem has been attempted to be solved by taking the musician out of the measurement process, and by mechanically actuating the instrument in order to make repeatable and consistent measurements[3,4,5], but this rather defeats the purpose, as the instrument will not play (in performance) without the musician present.

The very presence of the musician will change the acoustic behaviour of the musical instrument as the musician will absorb some of the acoustic energy radiated from the instrument[6], especially at high frequencies, and to a lesser extent, the mid-range and lower frequencies within the audible spectrum. (This is related to the physical wavelength of the sound pressure wave – where it is smaller than the dimensions of the human body of the musician it will tend to be absorbed, but when longer it will tend to diffract around the body and will not be absorbed much, if at all).

So, it has been proposed to make the acoustic radiation measurement system compatible with an instrument actually being played. This will involve a grid of sensors which surround the musician, taking measurements of data at spatially averaged points, but with higher spatial resolution than in any of the previously surveyed literature[7]. This will allow the interpolation of that data in order to predict the frequency-amplitude response and the phase of those frequency components in <u>any</u> given direction around the instrument (or other sound source) which has been measured.

In order to make high resolution measurements of the acoustic behaviour of musical instruments under performance conditions, a measurement system has been designed which consists of a number of interconnected parts. These include the sensors arranged in array on a 3 dimensional grid, signal conditioning circuits to bias the sensors, a set of analogue to digital convertors to send the signals to the data capture devices, and an array of hard disk data recorders to capture the data being generated by the sensors. When the data is captured, it can be then organised, processed, analysed and displayed in various ways.

2. DESIGN ASPECTS

Measurement System Design
Designing this system has presented a number of challenges including how to arrange the sensor array for the best spatial coverage of the spherical surface area around the sound source, how to interface the sensors with the analogue to digital convertors, how to synchronise the analogue to digital convertors together so that the data is synchronised on all the data streams feeding the hard disk data recorders, and how to synchronise a number of hard disk data recorders together so that the data from each can be used together.

The measurement system could be used within a fully anechoic environment (without reflections from the floor) or within a semi-anechoic environment (with reflections from the floor), giving us both purely theoretical results (anechoic), along with results which would more fully reflect the behaviour in the real world (semi-anechoic).

Sensor Array
The sensor array arrangement will be in a 3 dimensional grid pattern as on a geodesic dome as this presents the most even data point distribution on the surface of a sphere[8]. There will be 120 points where the sensors are placed, and the data at these points will be captured. When processed, this data can be interpolated to predict the behaviour of the sound source/musical instrument in any direction around the performer. The number of data points has been chosen to reflect both the requirement for high spatial density of data points[1], and is a multiple of blocks of 24 channels, which is the maximum number of simultaneous data channels that each hard disk data recorder can capture simultaneously.

Each sensor is an omnidirectional pressure sensor, selected from a batch to have the closest tolerances, expected to be within 1-2 dB at all relevant frequencies. The sensors require biasing and interfacing to the analogue to digital convertors[9].

Sensor Biasing
The interface circuit between the sensor and the analogue to digital convertor requires 2 stages.
The circuit must supply power to the sensor (known as *biasing* the sensor), and also ensure that the sensor output is compatible with the analogue to digital convertors which feed the data recorders. (These circuits can also be referred to as 'signal conditioning circuits' or 'sensor biasing circuits'). The analogue to digital convertors require a differential input signal, and so the sensor biasing circuit must have a differential signal output, or at the very least an impedance-balanced quasi-differential circuit which will have the Common Mode Rejection Ratio (CMRR) of a differential circuit, but with only one terminal carrying the signal. The other differential input terminal will be earthed via the same impedance as the signal terminal, but will not carry an opposing polarity signal. This will preserve the CMRR characteristics of a differential circuit, but will result in 6dB less gain as the signal to which a 'difference' is made will be effectively 0Volts.

An electronic circuit has been identified[10] which will interface the sensors to the analogue to digital convertors, using surface mount components which will fit inside the outer shell of the cable connector used to connect the sensor to the analogue to digital convertor unit. Currently a supplier for getting a large number of PCBs made utilising this circuit is under investigation.

Convertor Synchronisation
The synchronising clock signal is generated by a master clocking unit (Applied Research and Technology Syncgen). It is running as an analogue square wave at 48kHz. This is transmitted as a single ended positive going 5V peak square wave signal on 75ohm coaxial cable according to Annex B of AES Standard AES11-2009[11].

Synchronising Data Recorders
The data recorders used are model Alesis HD24, which have been successfully interfaced with their synchroniser/master control unit (model Alesis BRC). The master control unit was selected on the

basis that it can synchronise the HD recorders from the clock signal for the analogue to digital convertors, derived from the master clocking unit. The testing thus far has shown that the synchronisation controls work without any major problems, but the synchronised data capture is not yet fully tested.

Sensors
Various sensor types have been identified at this point which may be suitable for this project[9]. It is necessary to test and evaluate samples in order to ascertain which is the most suitable both in terms of sensor-to-sensor consistency (as stated earlier, it is hoped that a tolerance of approx. 1-2dB will be achievable), and also the overall quality of the individual sensors. Although one application of the measurement system could be based on the differences between measurements taken from each sensor location, the linearity in terms of the frequency-amplitude performance is quite important. Any overall deviation from a linear frequency response will negatively affect the quality of the data gathered.

3. DATA PROCESSING

Sensor Array Design and Construction
After any further data capture synchronisation problems are resolved, the sensor array grid can then be designed and constructed. The sensor array will be placed on a quasi-spherical surface at approx. 2 metres from the sound source at the centre of the sphere. A similar working distance has been used for these measurements elsewhere[12].

The current design for the array grid is for it to be fabricated as an aluminium grid based geodesic frame, with the sensor locations at the node points of the frame. The actual aluminium elements will need to be designed in a CAD application, and will need to be erected and dismantled for transport to various measurement locations, for example the Acoustics Laboratory / anechoic chamber at the Open University site in Milton Keynes. The structural members will need to be thin or perforated in order to present the least amount of reflection surface to the pressure wave radiating from the sound source at the centre of the spherical grid, and the node points will also need to have minimal orthogonal surface area while retaining strength to support the structure.

Data Capture
For the measurements to me made, the system will be transported to the anechoic chamber in the Acoustics Laboratory at the Open University in Milton Keynes, where the measurements will be made. The Open University has approved the use of their facilities for this research work.

The data gathered will be of high spatial resolution 3 dimensional acoustic behaviour of musical instruments. It is expected that a number of instruments will be measured, including some for which no data currently exists[7].

Data Analysis
Analysis of the data can then take place, including investigation into the display of the 3 dimensional sound radiation data, to determine how best to analyse the data. The data can be analysed using a spherical harmonic decomposition[13] allowing the encoding of the sound radiation pattern in a more efficient way.

It will also be possible to verify that a high spatial resolution (with more points on the sensor array grid than previously used) will result in less high frequency errors when interpolating the data to predict the acoustic behaviour (sound output quality) in any random direction from the sound source/musical instrument. This will require some interpolation of the data to be performed and predictions generated of the frequency amplitude response at various different directions corresponding to between-measured data points[8].

4. INITIAL TEST DATA

Initial testing of a small number of sensor positions has taken place, using 7 microphone sensors in the horizontal plane, in the following configuration as illustrated in Fig.1.

Sensor position	**Angle**
Position 1	-90^o
Position 2	-60^o
Position 3	-30^o
Position 4	0^o
Position 5	$+30^o$
Position 6	$+60^o$
Position 7	$+90^o$

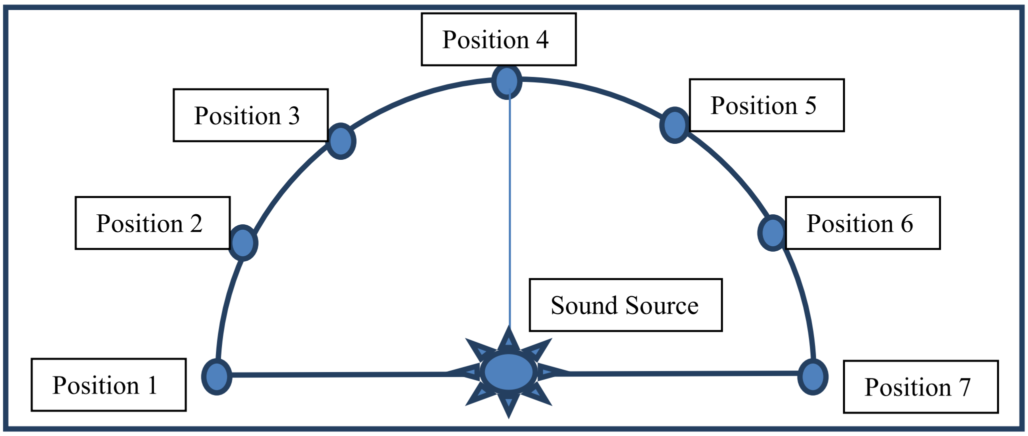

Fig. 1. Test Sensor Positions

The sound source used is a Lowden model O-35 steel string acoustic guitar, playing a single open ‘A’ string (fundamental frequency of 110Hz). Each microphone sensor was placed 1m radially from the centre of the sound source.

The resulting data has been interpolated (in order to test the data processing methodology), and an approximation for the frequency-amplitude curve for radiation at intermediate horizontal position has been generated. The graphs show frequency in Hz along the horizontal axis, and signal amplitude in dBFS (decibels Full Scale relative to the 24bit digital recording medium) along the vertical axis. The frequency-amplitude curves for each microphone sensor position are shown here (Fig.’s 2-8):

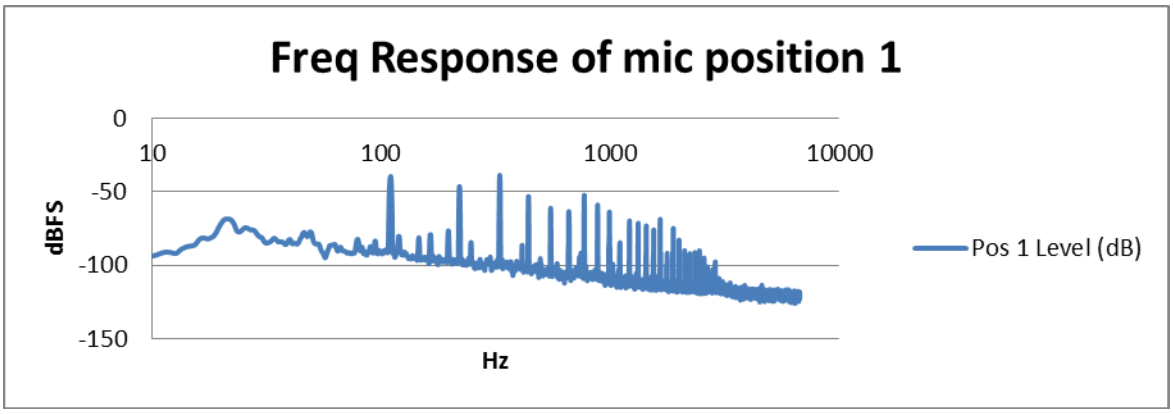

Fig. 2. Frequency Response of mic 1.

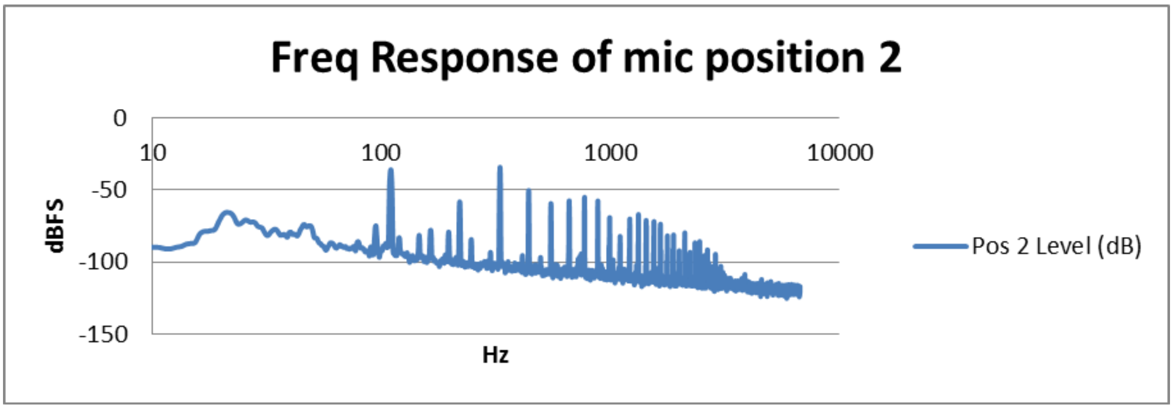

Fig. 3. Frequency Response of mic 2.

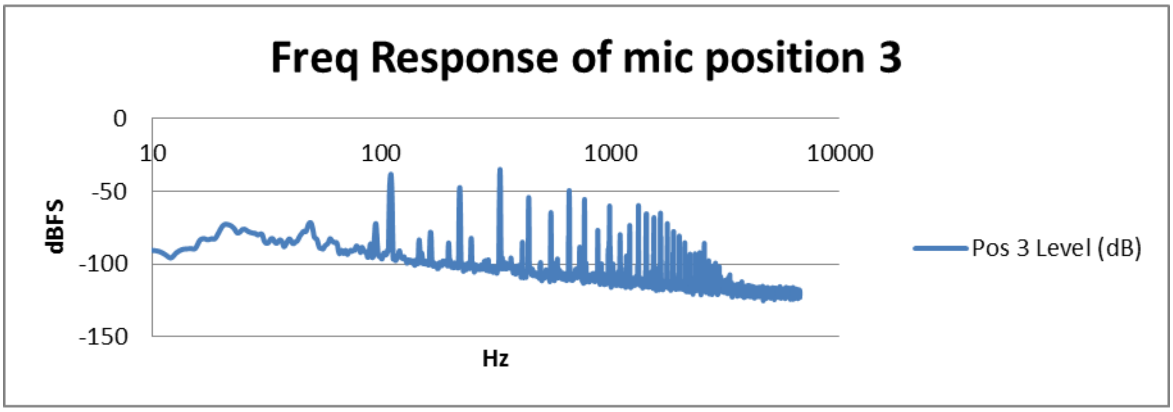

Fig. 4. Frequency Response of mic 3.

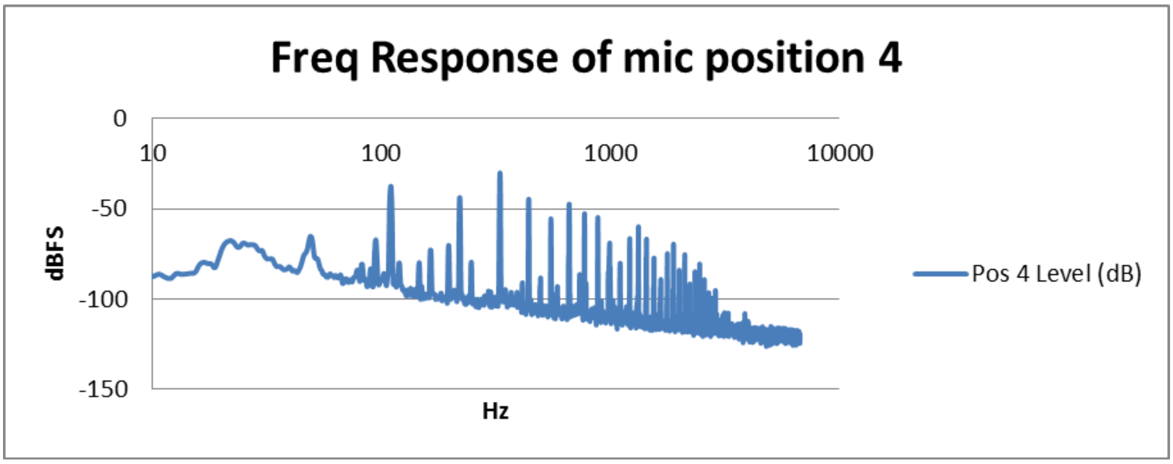

Fig. 5. Frequency Response of mic 4.

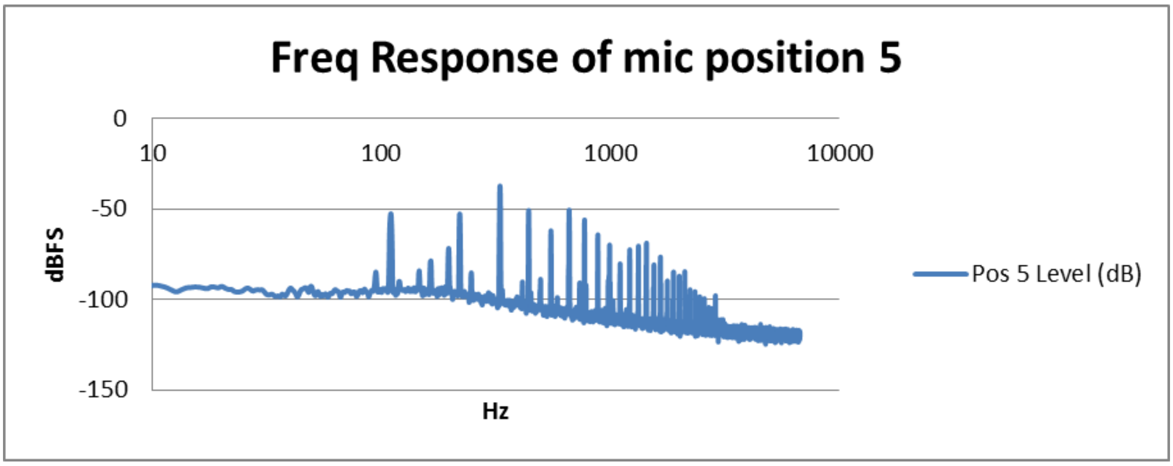

Fig. 6. Frequency Response of mic 5.

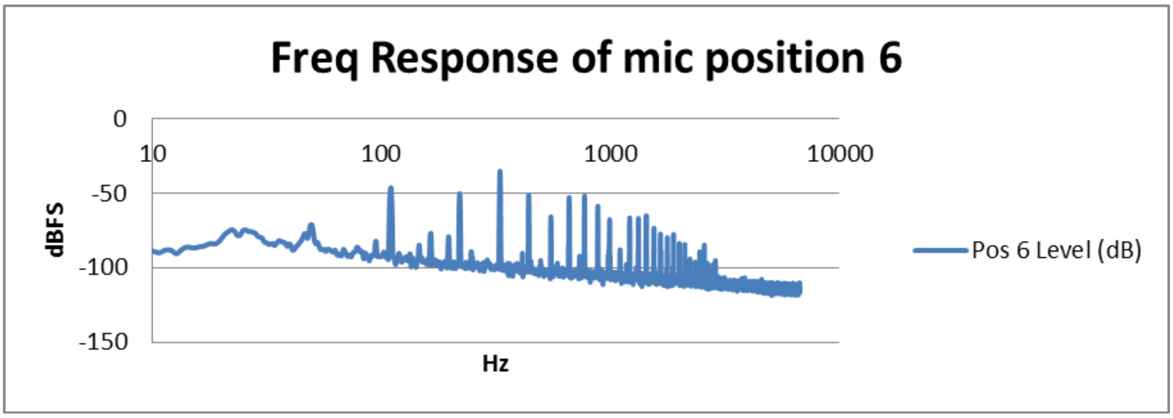

Fig. 7. Frequency Response of mic 6.

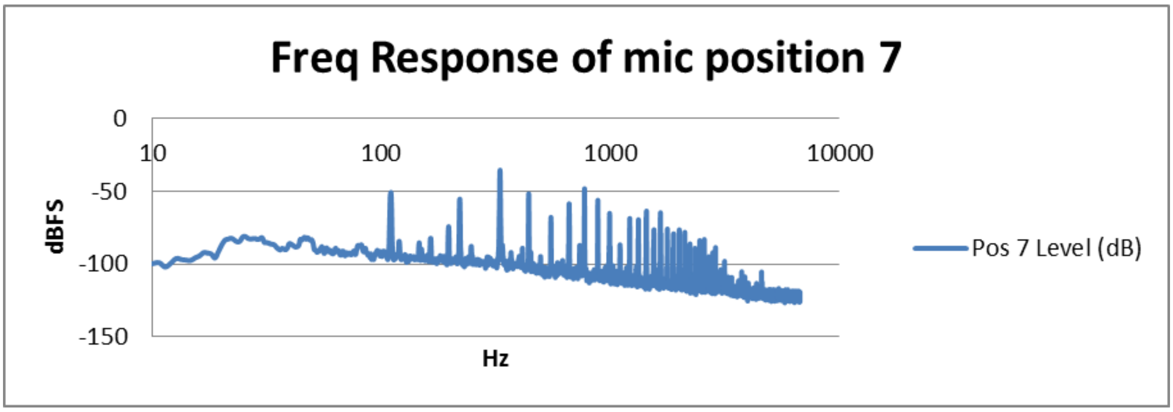

Fig. 8. Frequency Response of mic 7.

The interpolation in the first instance (as a proof of concept within the project) used was a linear interpolation between the amplitude values at each frequency, weighted towards the relative distance of each sensor node position. The following graphs (Fig.'s 9 and 10) show some of the data interpolated for a point mid-way between two of the sensors, at microphone position 4 and position 5:

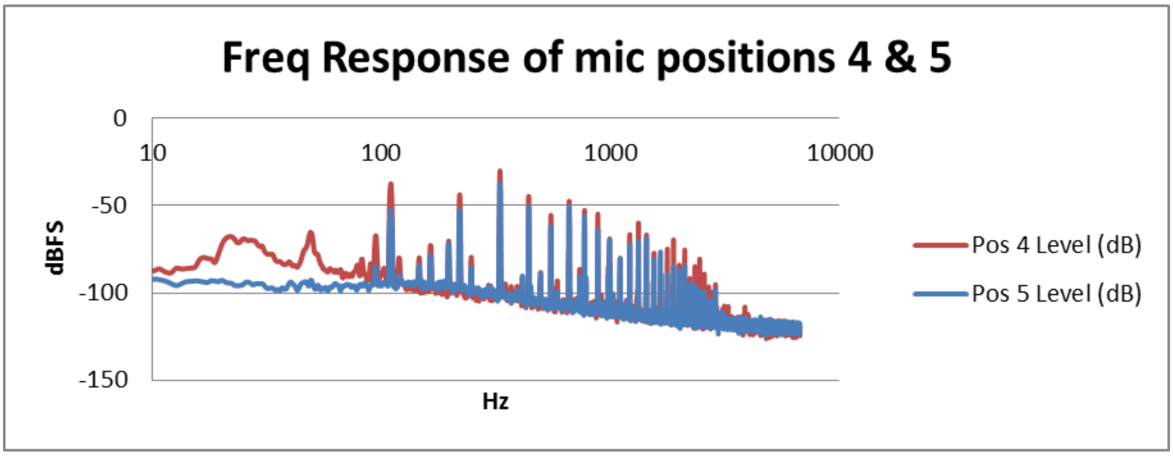

Fig. 9. Frequency Response of mics 4 and 5 shown

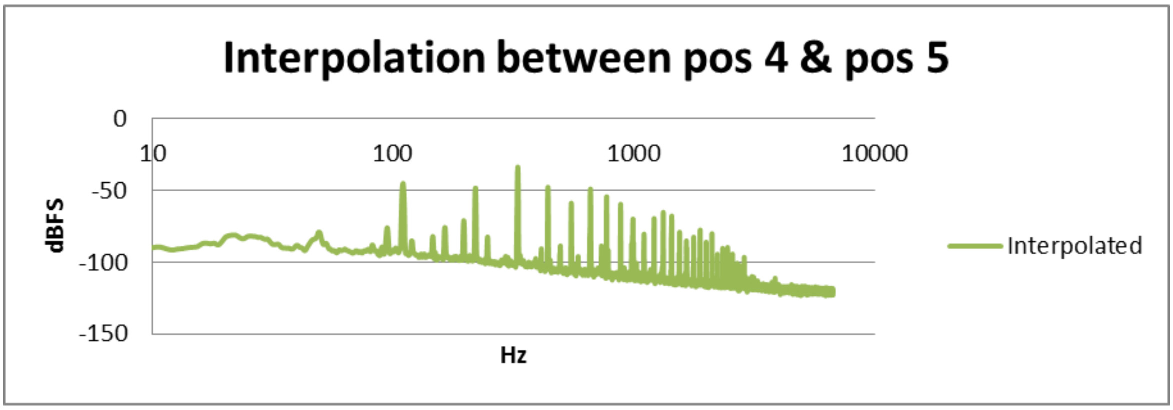

Fig. 10. Interpolated data between positions 4

This demonstrates that an interpolation can be done in order to derive the frequency amplitude response at any radiation angle from the sound source, up to the point where the wavelength is of the order of the spacing between the microphone sensors (and the radiation angle may be narrow enough that the energy falls between 2 sensor positions). In the example case shown here, the microphone spacing is ~0.52 metres, and so the relevant wavelength of sound at normal temperature and pressure is ~650Hz. In the example shown here, the first 5 harmonics of the sound source are below this limit, and so the interpolation is valid for these harmonic components.

5. DATA SETS

It is expected that the capture of data for sound sources/musical instruments for which no data currently exists will be one of the contributions to the body of knowledge in the subject area of musical acoustics[7,14]. The data that is measured for instruments for which there is currently a data set will be used to verify the existing data sets[7], and also to improve on the spatial resolution of the currently available data sets[7,8,15], as this research uses almost twice the resolution of existing or previous measurement systems[1,7,8,12,15]. In addition to this, it is proposed to verify that using high spatial resolution data results in less high frequency errors in the data interpolation when predicting the acoustic behaviour of a musical instrument in any random direction, which has previously been suggested in the relevant acoustic measurement literature[8].

6. CONCLUSION

It is possible to utilise the measurement system for further work beyond the scope and duration of this research project. This may include the gathering of a large data set of the acoustic radiation characteristics (frequency-amplitude and phase response in all directions) for musical instruments for which no existing data set is currently available. It may be possible to use the 3 dimensional data to develop a spherical harmonic based mathematical model of a measured sound source which could be used for synthesising sound sources[12].

If the measurement system were to be used in a performance environment (for example a convert hall) the measurements would reflect the influence of that particular acoustic environment upon the instrument (and musician) being measured. This data may be useful in the future for research into the differences between musical performance in an acoustically sympathetic environment and musical performance in an acoustically dampened space (for example a heavily damped recording studio environment). This could be useful in informing the musician of the differences they (and their audience) could expect when playing in each of these types of spaces.

7. REFERENCES

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