UNIVERSITY OF EXETER "THE FORUM" - USE OF AURALISATION TECHNIQUES TO INFLUENCE THE DESIGN, A CASE STUDY

S Di Rosario Buro Happold Ltd, London, UK
A Meachin Buro Happold Ltd, Bath, UK

1 INTRODUCTION

A new development at the University of Exeter named "The Forum" was designed by Wilkinson Eyre Architects with Buro Happold providing multi-disciplinary engineering services including acoustic consultancy. The project includes a large courtyard (the "Main Street") covered by a mixed timber, glass and ETFE roof structure as well as new-build classrooms, laboratories, auditorium and a refurbished library.

This paper describes in detail only the Forum Main Street as this was the main subject of the auralisation presentation to the client in July 2009. Additional auralisations were performed that simulated concert noise break-in to acoustic sensitive spaces (laboratories, classrooms and auditorium) but they are only briefly described here.

Auralisation was used to simulate various sources of noise (rain, occupants, PA system) and music performers in the Main Street. A particular concern was break-in noise from the nearby concert venue.

A multi-source auralisation presentation was given to the client and design team. Attendees gave a positive response to this, considering it effective in portraying a complete simulated aural environment and helpful in forming a clear understanding aspects of the acoustic design. The auralisation empowered the client to take important decisions on acoustics criteria in order to balance quality and cost.

This paper describes in detail how various technical issues were tackled to ensure accurate simulations. Decisions made as a result of the auralisation are outlined.

The paper also aims to open a discussion on the use of auralisation techniques in the design of complex environments.

2 SITE DESCRIPTION

The Exeter Forum includes construction of a variety of spaces including cafés, teaching spaces, break-out spaces, conference / reading rooms, career and education centre, retail shops, bookshop, bank, 400-seat auditorium and refurbishment of the existing library. The covered Main Street is at the centre of and interconnects all these spaces; its volume is approximately 13,000m^3. The development connects several existing buildings i.e. the Great Hall (a music venue), Devonshire House (an administration building), the shopping mall and the main library. The development is adjacent to the university concert hall (the Great Hall – used mainly for pop and rock concerts) and is surrounded by the quiet environment of the campus.

Figure 2—1 shows a roof plan of the site labelled with descriptions of the main acoustics issues.

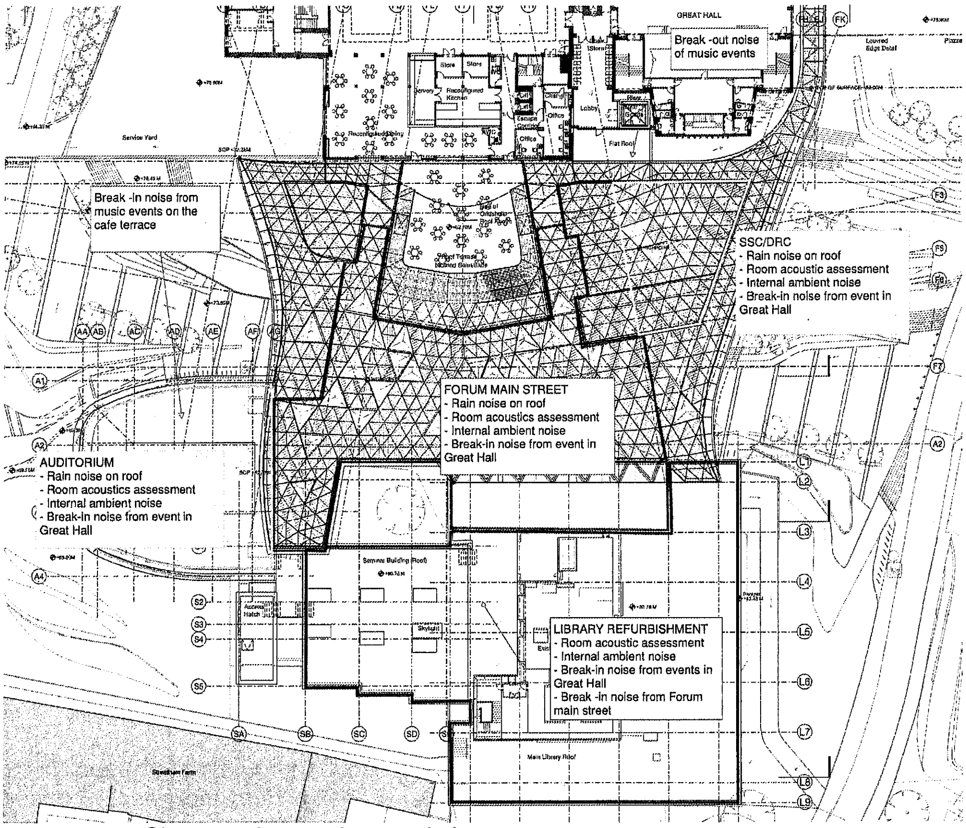

Figure 2—1 Site overview and acoustic issues

The main acoustic issues affecting the Main Street and addressed by the auralisation are:

- Rain noise from the roof
- Break-in noise from music events in the Great Hall
- Noise from the café
- Noise from social events in the Forum Main Street
- Subjective quality of musical performance in the street
- PA system speech intelligibility

3 ACOUSTIC CRITERIA

At an early stage of the project development, a number of provisional acoustic criteria were set. Some of the most important are given in the table below. One of the objectives of the auralisation exercise was to challenge these where not easily achievable.

Location	Maximum concert noise / dB LAeq	Maximum heavy rain noise / dB LAeq	Maximum T_{mf} /s	Minimum PA intelligibility / STI*
North seating terraces (outdoor)	50			
Street	45	65	2.0	0.50
Silent Study	35			
Exploration Lab	35	55		
Auditorium	35	55		
SSC Disability Resource Centre	40	60		
Library entrance		60		
Library open plan		60		

STI rating is mean value minus one standard deviation as per BS EN 60849:1998 "Sound systems for Emergency purposes"[15]

4 SOUND FILE CALIBRATION

All the audio files used in the auralisation simulation were recorded using a Core Sound Tetramic (A-format microphone)[3] or have been taken from professional field recording library or websites in mono format.

There is no standard method to control or document sound levels represented by Ambisonic systems, so we have developed the following system of recording this information:

- Ambisonic microphone sensitivity information is obtained from the microphone supplier, based on tests in anechoic conditions
- Pistonphone calibrators are used to confirm the sensitivity of omni-directional microphones
- Electro-acoustically-generated calibration tones (for Ambisonic microphones) or pistonphone reference tones (for omni-directional microphones) are recorded at the start of each recording session
- A 'recording record sheet' is kept for every recording, noting calibration levels, equipment used, environmental details and any other necessary information
- Sound files are stored electronically, and file names indicate the sound level represented by full scale (FSD) signals, for example, *recording_FSD108dB.wav*
- A record is kept of any sound file manipulation carried out, and output files are named to indicate sound levels represented by full scales
- Assumptions must be made to estimate the sound levels represented where sound files are used for which no level calibration information is available. The assumptions used are recorded and files are named to indicate full-scale level representations

These measures systematically track what sound levels files represent, and allow Auralisation to be calibrated to absolute sound levels with the maximum possible accuracy.

5 ACOUSTIC PREDICTIONS AND SOUND FILE PREPARATION

An electro-acoustic computer model using CATT-Acoustic was used to calculate room acoustic conditions. Figure 2—2 shows an image of this model including receiver positions. Sources were included to represent the various sound sources. First-order Ambisonic B-format impulse responses were calculated for each source-receiver combination. These impulse responses were convolved with appropriate sound files to generate Ambisonic sound files for each source-receiver combination, with calibration maintained throughout the process.

Two options for acoustic absorbers in the Main Street referred to as option 1 and option 2. Option 1 included a larger area of absorbent materials.

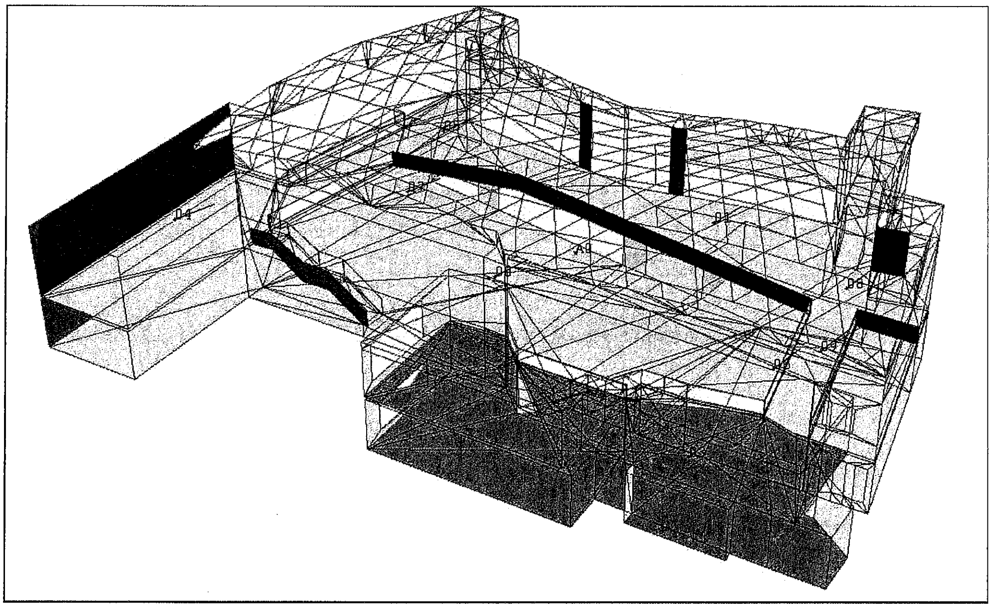

Figure 2—2 Forum Main Street CATT-Acoustic model

5.1 Rain noise

5.1.1 Rain noise level calculation

Laboratory procedures for measuring the rain noise performance of different constructions are defined in BS EN ISO 140-18:2006[1]. This standard defines 'heavy' rainfall conditions to be used for laboratory measurements. The BRE paper "Building Bulletin 93 - Information on rain noise from roof glazing, polycarbonate roofing and ETFE roofing"[6] indicates that 'heavy' rainfall has a return period of roughly every 50 years in the UK. The standard also defines 'intense' rainfall, which is much lighter and more representative of common weather conditions, having an approximate return rate of two years. The BRE paper states that sound levels from intense rainfall are approximately 20dBA lower than heavy rain. For this auralisation we demonstrated both heavy and intense rain.

The BRE paper[6] includes laboratory measurement data for rain noise on several roofing materials, and the data for ETFE (without rain noise suppressor) and double-glazed panels were used as the basis for calculations. Noise produced by rain impacting on the opaque panels of the roof (timber panels with sound absorption on the front) was assumed negligible compared to the noise generated on the clear areas.

Six hemi-spherical sources distributed throughout the roof area are used in the CATT-Acoustic model, each having a sound pressure level adjusted according to the area of roof represented.

5.1.2 Rain noise audio file preparation

To generate the rain noise anechoic sound file, we have followed these steps:

- Use recording of rain outdoors on hard surface
- Apply frequency response filtering to make spectrum of sound match measured spectrum of roof material as reported in the BRE paper[6]
- Calibrate FSD of file so that intensity L_I matches L_I reported in BRE paper[6]
- For each source used in the CATT-Acoustic model, convert from FSD_I to $FSD_{P,1m}$ according to:

$$FSD_{P,1m} = FSD_I + 10\log\left(\frac{Q}{4\pi}\right) + 10\log A = FSD_I - 8 + 10\log A \quad (1)$$

Q=2 for hemi-spherical sources used; A in this equation is area of the rain noise source.

The audio file used for the simulation of rain noise was taken from the on-line repository of professional field recordings FREESOUND[5]; all files are released under the creative commons sampling plus 1.0 license (http://creativecommons.org/licenses/sampling/1.0/legalcode).

The file was recorded using an omni-directional measurement microphone in a large parking area under a solid concrete roof but open at the sides; the recorded rain noise represents the impact noise of the rain on the parking floor.

The recording has been modified and equalized to match lab data for "heavy rain" conditions on an ETFE roof.

Figure 3—3 shows the frequency spectra published in the BRE paper[6] for ETFE and glass elements and the frequency spectrum of the recorded file before any audio processing. Figure 3—4 shows equalization applied to the sound sample to match BRE document[6] values. Equalization was performed using Audacity[16] audio processing software.

Note that an alternative method of preparing the sound file was tried then abandoned. This was based on a recording of rain on roof window from below. The sound was then layered many times with random delay in an attempt to create the sound of rain on a larger area of glass. Subjectively, this was unsuccessful as the sound took on a very unrealistic metallic and rhythmic quality.

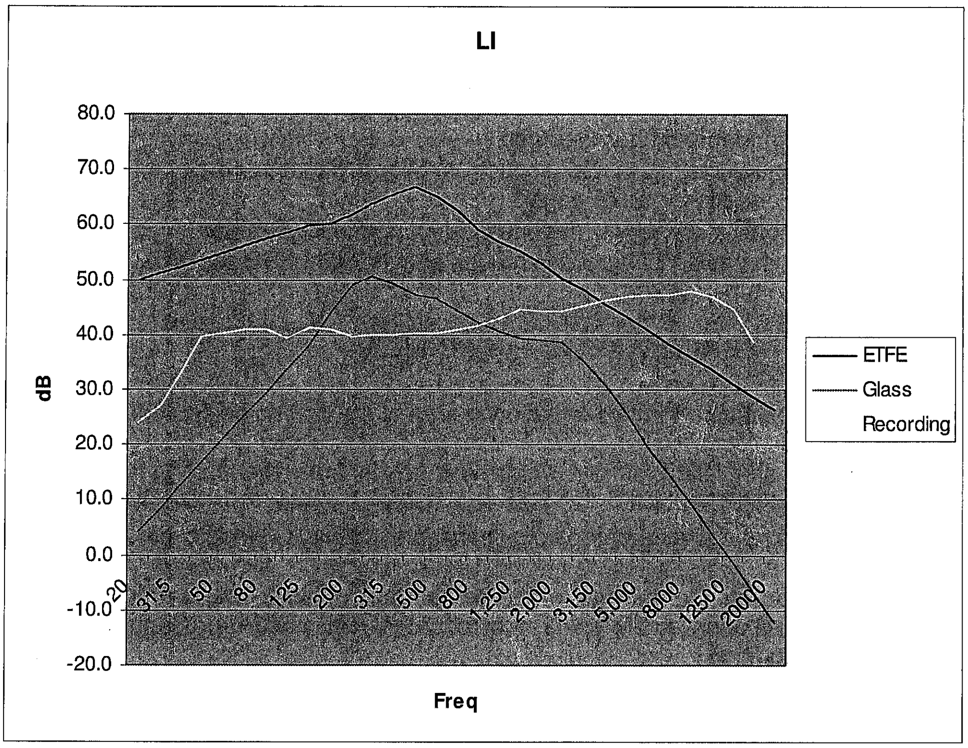

Figure 3—3 Frequency spectra for rain noise recording and BRE measurements for ETFE and glass

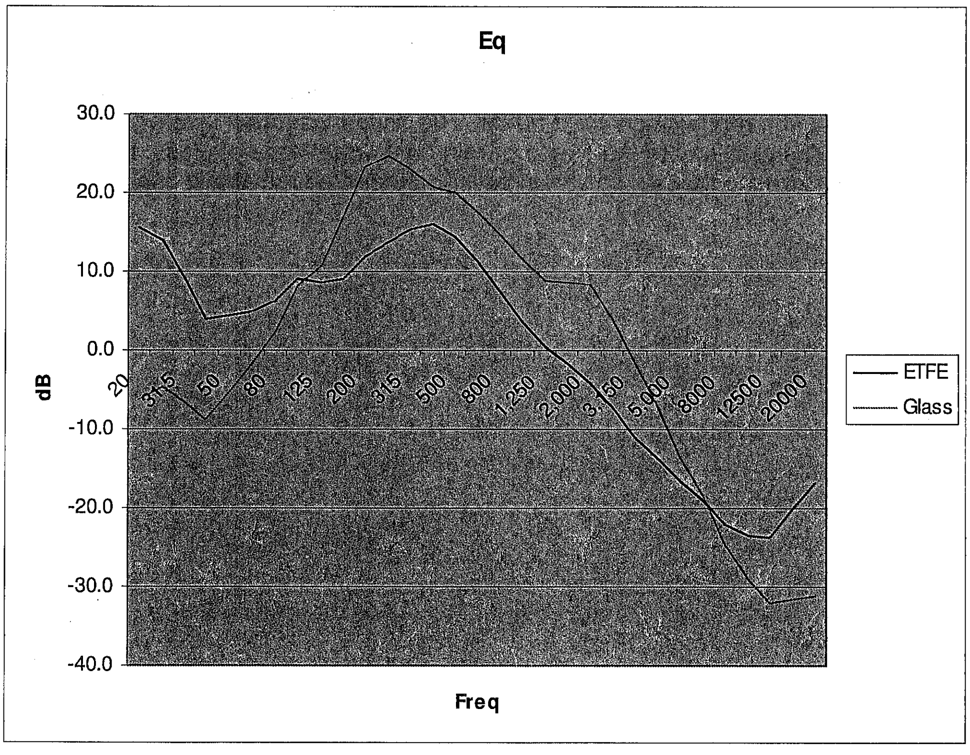

Figure 3—4 Equalization filter applied to the rain noise sound sample (absolute gain not shown)

5.2 Great Hall concert noise

Sound level measurements were made of concerts held in the Great Hall, outside the hall façade facing the new development. The effects of distance attenuation, barriers and building envelope sound insulation were calculated to determine the sound power levels of sound breaking into the Main Street. These sound levels were then assigned to hemi-spherical noise sources at the main noise break-in locations.

Recordings were made using a SLM with omni-directional microphone simultaneously with the acquisition of sound level data; this allowed us to obtain calibrated monophonic recordings. The FSD values of the files were calculated from this information. The recording used for the simulation represents the concert held by the rock band "The Enemy" on 2nd of April 2009.

The file was then subjected to additional processing based on:

- Calculated level outside various parts of the façade based on spherical propagation theory and barrier effects
- Filter to simulate sound insulation of the façade
- Included in the room acoustics computer model for convolution processing

5.3 Café noise

5.3.1 Cafe source file recording

The source file used for this simulation was recorded by Buro Happold acoustic team members in Paris in 2008 in a small café as shown in Figure 3—5.

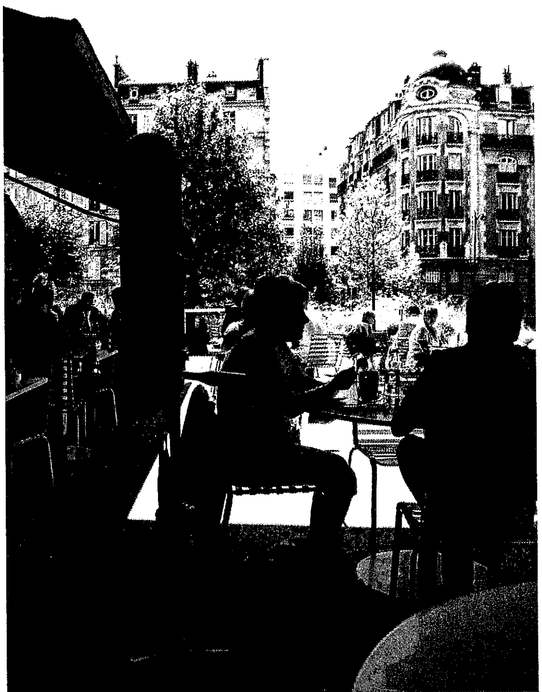

Figure 3—5 Café noise recording, showing microphone wind-shield on tripod

The extent and occupancy of the recorded café was much smaller than the café in the Forum Main Street so additional sound processing was performed on the file to calibrate and obtain a more accurate subjective perception of the noise produced. The recording was also layered several times in order to create the sound illusion of a bigger crowd. The procedure used is explained in section 5.3.3.

5.3.2 Calibrating the café recording

The capsule used for the café recording was cardioid (one capsule of a Core Sound Tetramic[3]). This type of capsule cannot be straightforwardly calibrated using a pistonphone, so the procedure below was used.

The effective FSD of the microphone was calculated from:

$$FSD_P = -20\log\left(\frac{20\mu Pa}{1Pa}\right) + L_E + C_C - 20\log\left(\frac{V_S}{1V/Pa}\right) - 20\log D + C_D - G_N \tag{2}$$

Where FSD_P = Sound pressure level represented by Full Scale Deflection
L_E = electrical calibration RMS level in dBV
C_C = Correction for calibration crest factor = 3dB for a sine calibration signal
V_S = Voltage sensitivity of microphone
D = Peak value of calibration tone in file, as a proportion of FSD (full scale deflection)
C_D = Correction for microphone directivity (see below)
G_N = Normalisation gain applied to file

The C_D correction factor is to account for the lower sensitivity of a directional capsule to random incidence than an omni-directional microphone. According to Sound System Engineering by Don Davis[7], an omni-directional microphone has an RE (random efficiency) of 1 and Cardioid has RE of 1/3. The correction term is given by:

$$C_D = 10\log\left(\frac{RE_{omni}}{RE_{cardioid}}\right) 10\log\left(\frac{1}{\frac{1}{3}}\right) = 4.8dB \tag{3}$$

5.3.3 Effective distance, area and sound power level of café recording

The café recording was made within the café rather than of the whole café. Therefore we needed to calculate what the effective area represented by the recording is.

Consider the café as an even distribution of sources in a plane with intensity I. The sources included are between r_{min} and r_{max} distance from the receiver (i.e. a circular disc of sources with a round hole in the middle).

The power of sources between r and r+dr distant is:

$$W = 2\pi Irdr \tag{4}$$

And the pressure at the receiver due to these sources is:

$$P = \frac{P_0}{I_0}\frac{2\pi Ir}{4\pi r^2}dr = \frac{P_0}{I_0}\frac{I}{2r}dr \tag{5}$$

Where P_0 =reference pressure (20 μPa)
I_0 = reference intensity (10^{-12} W/m^2)

Total pressure at the receiver is then:

$$P_{tot} = \frac{P_0}{I_0}\frac{I}{2}\int_{r_{min}}^{r_{max}}\frac{1}{r}dr = \frac{P_0}{I_0}\frac{I}{2}\left[\ln|r|\right]_{r_{min}}^{r_{max}} \qquad (6)$$

To calculate the 'mean effective distance' of sources from the receiver (cancelling out constants):

$$r_{eff} = \frac{\int_{r_{min}}^{r_{max}} rP\,dr}{\int_{r_{min}}^{r_{max}} P\,dr} = \frac{\int_{r_{min}}^{r_{max}} 1\,dr}{\int_{r_{min}}^{r_{max}} \frac{1}{r}dr} = \frac{\left[r\right]_{r_{min}}^{r_{max}}}{\left[\ln|r|\right]_{r_{min}}^{r_{max}}} = \frac{r_{max} - r_{min}}{\ln\left(\frac{r_{max}}{r_{min}}\right)} \qquad (7)$$

The sound power level of the represented area is:

$$L_W = L_P + 11 + 20\log r_{eff} \qquad (8)$$

Sound power level represented by full-scale deflection is:

$$FSD_W = FSD_P + 11 + 20\log r_{eff} \qquad (9)$$

If a recording is used as an omni-directional anechoic source in CATT-Acoustic, its level needs to be given in terms of sound pressure level at 1m, given by:

$$FSD_{P,1m} = FSD_P + 20\log r_{eff} \qquad (10)$$

For the above, the values of r_{min} and r_{max} for the recording must be estimated from the distances to the nearest and farthest sound sources.

The sound sample is then layered a number of times (with random delay) given by A_{actual}/A_{eff} (A_{actual} is the area of the actual sound source) to give it the required degree of complexity (related to the number of audible sources). The resulting sound file is then convolved with the impulse response for an onmi-directional source in the café location generated using the CATT-Acoustics model. Determining the correct number of layers to create an accurate depiction of source complexity is not straightforward, particularly if a directional microphone is used. In this case we made the subjective judgement that the recording represented an area of 8m^2 and layered the sound file eight times to produce a signal complexity of the actual café area of 63m^2.

5.4 Large event in the street

Source audio file used for the simulation of a large function on the balcony at first floor has been downloaded from the on-line repository of professional field recordings FREESOUND[5].

The file was recording using an omni-directional measurement microphone on a large open air terrace in the middle of the people; the recorded noise a good representation of a party in semi-anechoic conditions.

In order to evaluate the FSD of the file, the noise level of 200 people talking has been calculated based on the levels for normal voice levels published by ANSI[14].

The sound file is then convolved with the impulse response for an onmi-directional source in the event location generated using the CATT-Acoustic model.

Note that with more detailed information on recording conditions, the same layering process described in section 5.3.3 could be used.

5.5 Performer in Main Street

The source audio file for the performer was taken from the Bang & Olufsen "Music for Archimedes"[13] library choosing the anechoic cello recording of "Gavotte" by Martini.

In order to evaluate the FSD of the file, the measured emission level of a Stradivarius was reduced by 10 dB, as suggested in "Acustica Musicale e Architettonica" by Cingolani, Spagnolo[8].

The directivity function of the instrument to be used in CATT-Acoustic was obtained from the repository of musical instrument directivity CLF files of the Aalto University in Finland[9].

The anechoic sound file is convolved with the impulse response for a source with this directivity in a typical performer location generated using the CATT-Acoustic model.

5.6 PA announcement

Auralisation of sound reproduced by loudspeakers is a relatively well-established procedure so presented no serious new technical challenges.

The source audio file used for simulation of the PA system announcement was received by email from a PA system manufacturer and represents a typical emergency message.

The loudspeakers in the PA system design were:

- 1 qty Community Entasys column loudspeaker (http://communitypro.com/index.php/product-list/102-entasys)
- 4 qty Penton MUS 40/TC column loudspeaker (http://www.pentonuk.co.uk/products/high-performance-column-loudspeakers.asp)

Impulse responses were generated relating signals fed to these loudspeakers to the sounds heard at receiver positions using CATT-Acoustic. The source file was then processed by convolution with the impulse responses calculated and levels calibrated according to the level measured in the software at receiver positions.

6 AURALISATION

6.1 Reproduction system

The auralisation simulation was performed in the existing "quiet room" in the University of Exeter library in July 2009 and was presented to the main architect and the client team by Alistair Meachin and Serafino Di Rosario of Buro Happold.

The reproduction system used for the simulation was a horizontal only first-order Ambisonic loudspeaker rig comprised of:

- 4 qty Genelec 8030A powered bi-amplified loudspeaker in a square configuration (4 m x 4m) at approximately 1.4 m height
- Dell Laptop running Audiomulch[12] software patch[11] for Ambisonic decoding and playback
- Edirol UA-101 USB soundcard

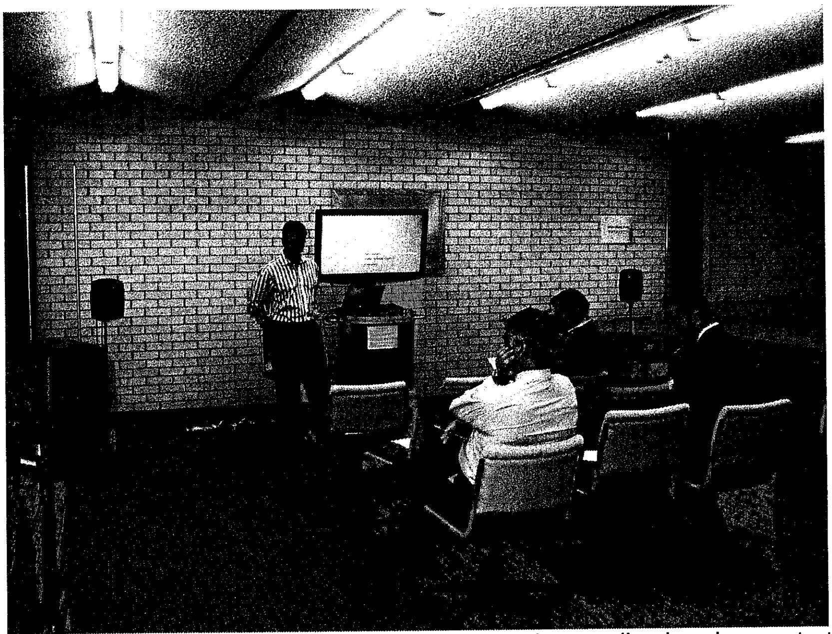

Figure 3—6 Client representatives listening to the auralisation demonstration

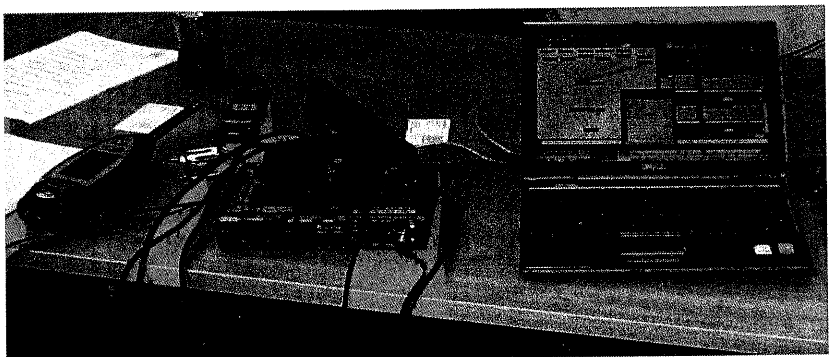

Figure 3—7 Auralisation source equipment

The room used for the simulation had approximately the following characteristics:

- Large volume = 380 m^3
- Carpet floor
- Plastered and brick walls (no specific materials for acoustic absorption, T_{mf} = 1.4 s)

The room despite being called "silent" had a significant ambient noise level ($L_{Aeq,\,5\,min}$ = 43 dB) due to the ventilation system; this noise effectively masked some of the quieter sounds in the reproduction including lower sound pressure level concert break-in.

6.2 Reproduction calibration

The reproduction system was calibrated according to the procedures described in one of the authors' previous papers[2].

In summary, system equalization was performed in two stages:

1. Equalize the sound pressure response of individual loudspeakers over the usable frequency range
2. Apply a single filter to all loudspeakers to compensate for the low-frequency response increase when using the loudspeakers in an Ambisonic arrangement (due to more that one loudspeaker being active for a given source direction).

Sound level calibration was also performed by reproducing a band limited pink-noise signal (over the range 250Hz to 2kHz) at a known level and checking the value in the sweet spot of the system using a sound level meter.

An additional check was then performed by measuring the reproduction level of one of the auralisation files.

6.3 Presentation method

Before the demonstration, the audience was briefed on the purpose, limitations and scope of the auralisation. Limitations mentioned included:

- Selection of source files was based on our interpretation of typical events. Real sounds will vary
- Transmission paths from the Great Hall (concert venue adjacent to site) to the Forum are very complex and difficult to predict accurately, some aspects of the calculations were based on estimated factors and our experience
- Performer reproduction is based on a single example of instrument, genre, location and performance standard. Of course different performances, instruments and locations will sound different
- Concert noise is based on a typical concert event, but noise can vary according to the type of music and to the sound engineer's decisions
- Accuracy of reproduction level is based on assumptions as per our "Auralisation level calibration" paper[2]
- The listening room ambient noise and reverberation masked some effects

Sounds were then played to demonstrate the various noise sources, and in various combinations of simultaneous playback to illustrate masking and other effects of multiple sound sources. Listeners were free to request any of the simulated receiver positions and any combination of sound sources.

During playback, visualisations of views from the simulated receiver positions were displayed on a screen in front of the audience.

For many of the sounds we informed listeners of both the original design criteria and the predicted performance, stating in which cases targets had been reached.

7 RESULTS

The demonstration enabled attendees to discuss and express their subjective impressions of the predicted soundscapes. Note that this would have been harder using some alternative reproduction methods such as headphones, as these can inhibit conversation. As a result of hearing the simulations, attendees were happy to give clear opinions and decisions on the illustrated aspects of the acoustic design, whether considered inadequate, acceptable or too onerous.

A summary of outcomes of the auralisation are shown in the table below.

Location	Original limit criteria	Predicted value	Comments received during auralisation
Noise from concert in Great Hall			
North seating terraces (outdoor)	50 dB L_{Aeq}	65 dB L_{Aeq}	Acceptable Supposing that people there will be aware of the concerts in the Great Hall
Street	45 dB L_{Aeq}	46 dB L_{Aeq}	Acceptable
Silent Study	35 dB L_{Aeq}	31 dB L_{Aeq}	Acceptable
Exploration Lab	35 dB L_{Aeq}	36 dB L_{Aeq}	Acceptable
Auditorium	35 dB L_{Aeq}	25 dB L_{Aeq}	Acceptable
SSC Disability Resource Centre	40 dB L_{Aeq}	36 dB L_{Aeq}	Acceptable Low-frequency noise is dominant and was found quite disturbing during auralisation; acceptable because SSC will be closed during concerts
Rain noise (intense rain)			
Street	45 dB L_{Aeq}	52 dB L_{Aeq}	Acceptable
Library entrance	40 dB L_{Aeq}	52 dB L_{Aeq}	Acceptable
Library open plan	40 dB L_{Aeq}	36 dB L_{Aeq}	Acceptable
Auditorium*	35 dB L_{Aeq}	35 dB L_{Aeq}	Acceptable
Exploration lab	35 dB L_{Aeq}	26 dB L_{Aeq}	Acceptable
SSC DRC	40 dB L_{Aeq}	37 dB L_{Aeq}	Acceptable
Reverberation time			
Main Street (option 1)	1.8s T_{mf}		Acceptable
Main Street (option 2)	2.1s T_{mf}		Not Acceptable

It was discussed during the auralisation that intrusive noise from concerts in the Great Hall will be more acceptable due to the occasional and social nature of events and would not be considered as an annoyance despite high sound levels that did not comply with the original ambient noise targets.

8 CONCLUSIONS

This paper describes the case study of the acoustic design of the University of Exeter Forum, where auralisation techniques have been used to influence the design.

Key conclusions of the study are:

- Presentation of multi-source auralisation was considered by clients and architects to be effective in portraying a complete simulated aural environment and helped them to clearly understand many aspects of the acoustic design and increase their confidence in it
- Auralisation demonstrations can influence the design of buildings especially when a decision is required by architects or clients. Auralisations help design team members, who are not professional acousticians, to clearly understand acoustic requirements and their effect on design so that related costs can be justified
- Auralisation demonstrations set a scene in which informed discussions of the acoustic conditions can be held
- It is important to make clear the limitations of auralisation simulations
- The method of quantifying sound file 'complexity', related to the number of included sources, would usefully be the subject of further study to inform how sounds should be layered in auralisations.

9 REFERENCES

1 EN ISO 140-18:2006, "Acoustics – Measurements of sound insulation in buildings and of building elements – Part18: Laboratory measurement of sound generated by rainfall on building elements", ICS 91.120.20

2 S Di Rosario, A Meachin, "Auralisation Level Calibration", Proceedings of the Institute of Acoustics, Vol.30 Pt.6 2008

3 Tetramic, http://www.core-sound.com/TetraMic/1.php

4 VVmic for tetramic, http://vvaudio.com/products/vvmic

5 The freesound project, http://www.freesound.org/

6 BRE, Test report: Measurement of rain noise on roof glazing, polycarbonate roofing and ETFE roofing, test report number 220312, http://www.bre.co.uk/pdf/BRE_Report_220312.pdf

7 Don Davis, E. Patronis jr, Sound System Engineering, Focal press, 2006, ISBN 13:978-0-240-80830-7

8 S. Cingolani, R. Spagnolo, Acustica musicale e Architettonica, Utet Libreria, 2005.

9 http://auralization.tkk.fi/node/44

10 Lee, R., A.J. Heller and E.M. Benjamin, "Is my decoder Ambisonic?", presented at the 125th AES convention, San Francisco, USA, 2-5 October 2008;

11 Wigware Ambisonics plug-ins, http://www.brucewiggins.co.uk/?page_id=78

12 Audiomulch, http://www.audiomulch.com/

13 Bang & Olufsen (1992) Music for Archimedes. CD B&O 101

14 ANSI S3.5-1997, Table 3 and 4

15 BS EN 60849:1998, "Sound systems for emergency purposes", ICS 13.320

16 Audacity, http://audacity.sourceforge.net/