Audio and Acoustical Response Analysis Environment (AARAE): a tool to support education and research in acoustics

Densil CABRERA[1]; Daniel JIMENEZ[2]; William L. MARTENS[3]

[1] The University of Sydney, Australia

[2] The University of Sydney, Australia

[3] The University of Sydney, Australia

ABSTRACT

AARAE is a MATLAB-hosted environment for measurement and analysis of room acoustics and audio systems, designed to support education and research in audio and acoustics. While it already provides a wide range of test signals, and measurement, processing, and analysis methods, one of its main features is easy extensibility. Students new to signal processing (and MATLAB) can use it from its graphical user interface (without any need to write code), whereas more advanced students can develop implementations of pre-existing or novel measurement, processing and analysis algorithms using templates that allow quick integration of functions into the graphical user interface. AARAE was introduced into acoustics teaching at the University of Sydney in 2014, and it is used in both introductory and advanced units. While most of the code for the current version of AARAE was written by the authors, coursework students have contributed to functions in areas such as speech intelligibility, distortion, reverberation parameters and background noise rating. This paper illustrates how AARAE is used in teaching, highlighting some recent student projects in AARAE. It also illustrates how AARAE has been used to support research and research training.

Keywords: Measurement Techniques, Signal Processing, Architectural Acoustics, Audio Systems, Education, Software. I-INCE Classification of Subjects Number(s): 72, 74

1. INTRODUCTION

Measurement in architectural and audio acoustics increasingly employs signal processing, and so computer software is almost always used to support measurement and analysis. In research and education, open source software has advantages: the software can be validated not only by performance testing, but also by examining the code (and bugs quickly fixed); code can be modified and extended by users to support particular projects (and research often draws on or proposes new analysis methods); and when applied in teaching, the availability of functioning programs with open source code allows students to simultaneously develop a top-down and bottom-up understanding. This paper describes a project called AARAE that is intended to support research and teaching concerned with audio and acoustic measurement, all implemented within the MATLAB environment.

Several other somewhat similar projects exist. Cabrera et al. (1) developed PsySound3 using MATLAB to make accessible a wide range of audio signal analysis algorithms, especially ones from psychoacoustics. While one possibility for AARAE might have been to develop it as an extension to PsySound3 (PsySound4?), there were several reasons why this was not done, one of which was the relatively complex computational approach in PsySound3, which makes it quite difficult for a novice to add a new analyser to it. Furthermore, while PsySound3 was designed for sound analysis, most of the data that it deals with is non-audio data – i.e. data from the results of analysis – AARAE has a much stronger focus on processing and displaying audio data.

Developed within the Institute for Technical Acoustics (RWTH University, Aachen), the

[1] densil.cabrera@sydney.edu.au
[2] djim9123@uni.sydney.edu.au
[3] william.martens@sydney.edu.au

ITA-Toolbox (2) is the project that compares most closely with AARAE. The ITA-Toolbox also uses MATLAB, and a graphical user interface, for audio and acoustic system measurement and analysis. The ITA-Toolbox is a more mature project than AARAE, differing in emphasis but with considerable overlap. Both projects are intended to support teaching and research in their home institutions, but freely available to all.

Although the purpose of this paper is essentially to outline the scope of the AARAE project, a general description of a software project is not well suited to a conference paper (it is better suited to a software guide or manual). Instead this paper gives a brief overview of certain features of the AARAE project, and then presents examples of analyses currently available within AARAE.

2. OVERVIEW

The code for AARAE can be divided into framework and content. The framework is designed to provide a user interface, a practical workflow and to facilitate the addition of new content. The framework supports many aspects of measurement and analysis, including: the acquisition of audio data (by loading, recording or generating), calibration of the system or of particular signals, display of audio and non-audio data and saving data in various formats. Audio data in AARAE is held in a structure format that includes mandatory fields (e.g., for the audio data and sampling rate), other common fields (e.g., for specifying calibration offset, channel names, band frequencies, etc.) and other properties of the audio (which might be particular to the type of audio). The audio waveform within the audio structure can be multidimensional, with the first dimension for time (samples), the second for channels, the third for bands, and higher dimensions not pre-defined. Additional audio fields may also be included in an audio structure – most commonly, 'audio2' is used to store an inverse filter that can be used to create an impulse response by convolution with the processed 'audio' field (e.g., if the content of the audio has been played and re-recorded through an audio or acoustic system). Once in the AARAE workspace, audio structures can be selected, displayed, played and re-recorded, edited, processed, analysed, deleted, and exported in various ways through simple operations in the graphical user interface (GUI) – which is illustrated in Figure 1.

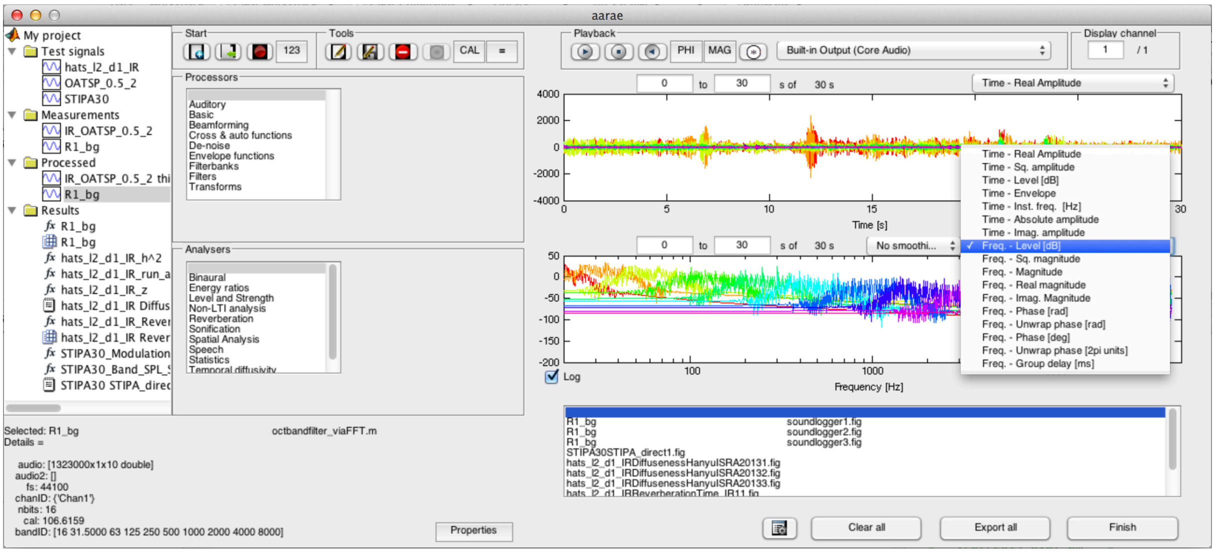

Figure 1 – Screenshot of AARAE's main graphical user interface, showing the pop-up chart options.

As shown in Figure 1, the AARAE main GUI includes two visual display charts, which automatically update to show the selected audio in the chosen format (either time domain or frequency domain). These charts produce independent editable figures by clicking on them. The '=' button provides subplots so that the content of multiple audio structures can be compared, or to simultaneously display multiple channels and bands of a single audio structure or to compare non-audio data that was output by an analyser. Above these charts are playback controls, allowing normal ('>'), time-reversed ('<'), random-phase ('PHI'), flat magnitude spectrum ('MAG'), and convolved ('*') playback, some of the principles of which are discussed by Cabrera and Ferguson (3). Time reversed playback may be helpful in hearing the time structure of an impulse response (because backward masking has a smaller effect than forward masking). Random phase playback removes any variation in time envelope, thereby allowing the listener to focus entirely on the distribution of sound across the spectrum. Flat magnitude spectrum playback does the opposite of that, removing the spectrum envelope from the reproduced sound. In the last case, the audio is convolved with a reference sound-file, which by default is a short recording of anechoic speech, and this can be helpful in appreciating the effect of an impulse response. Further visualization and sonification tools are in the Analysers section of AARAE. Thus AARAE aims to promote understanding of audio content by providing multiple methods of displaying audio data.

As an example of a workflow in AARAE, a user might first connect the measurement transducers and audio interface to their computer, and conduct system calibration – which can compensate for system latency, inverse filter the system response (if desired), and determine the gain calibration offset for each input channel (typically done using a microphone calibrator). Then a test signal could be generated using one of AARAE's generators (e.g., one that generates a swept sinusoid). If desired, the signal can be repeated, and these repetitions can be at constant or stepped gain. Repeated signals could be useful for the analysis of time variance or non-linear distortion, or simply for increasing signal-to-noise ratio using linear time-invariant assumptions (e.g., by excluding any atypical recordings of the signal from a synchronous average). A silent cycle can also be generated – which is treated as if there were a signal present – and this is used for determining the effective signal-to-noise ratio using the same inverse filtering process as is used for the signal (on the assumption that the noise is steady state). This signal (or sequence of signals) is played and recorded through the system under test, and its inverse filter can be applied to the recording to derive the impulse response. If multiple sweeps were generated, then the impulse responses can be synchronously averaged or stacked in dimension 4 (or dimension 2 if the recording is single-channel) for further processing. Often other recordings would also be made, such as of background noise, other test signals, and probably other impulse responses. AARAE offers a wide range of processors including basic editing functions, filters, filterbanks, envelope functions, cross and auto functions, and so on, some of which might be applied at this stage in the workflow. Finally the user may run some analysers to determine the values of acoustic parameters (e.g. reverberation time, speech transmission index, inter-aural cross-correlation, etc.), and these analysers generate tables and charts, and also return data to the AARAE workspace for further plotting and comparison. AARAE allows multiple tables within a figure to be copied to clipboard, for pasting directly into a spreadsheet. The entire session can be exported to a directory – including audio data, calibration data, charts, tables and other non-audio data. Afterwards, the data can be loaded directly into MATLAB for further analysis, or can be reloaded into the AARAE environment.

3. CONTENT

The content of AARAE is divided into four types of functions: calculators, generators, processors and analysers. The AARAE framework accepts new content simply by adding function files (m files) to the appropriate directories (within the 'Calculators', 'Generators', 'Processors', or 'Analysers' directories). Templates are provided so that users can develop and add new content quickly, especially if they have pre-existing code that they wish to port to AARAE. Generators are primarily used to generate test signals, such as swept sinusoids, noise, impulses, tones, the STIPA signal, etc. Processors have audio input and output, and include filters, filterbanks, envelope operations, transforms, and many basic editing functions. Analysers have audio input, but output charts and tables (as MATLAB figures) and data is returned to AARAE for further interactive plotting. Calculators are the miscellaneous category of AARAE content, examples of which include an image-source room acoustics model and a calculator of the dissipation of sound in air. Calculators are intended to support measurement and analysis by providing the expected theoretical values or responses for comparison with their measured counterparts. The following subsections of this paper illustrate and discuss selected parts of AARAE's content, focusing mainly on analysers.

3.1 Sound level and loudness

One of the most basic and important questions in acoustic measurement is 'how much sound is there?' and, as any acoustician knows, there are many ways of answering this question. Hence AARAE offers several ways to quantify the amount of sound, and these are highly likely to expand in the future. Issues to consider in quantifying the amount of sound include considering spectral distribution and weighting, temporal distribution and integration, and the units used, and other possible processing (e.g., loudness analysis or deriving single-number ratings). An important aim of AARAE is to provide many approaches to analysis, so that intelligent decisions need to be made by the person doing the analysis, rather than relying on pre-packaged analysis templates that remove decision making from the process. Figure 2 gives examples of various ways of analyzing the strength of sound in a single audio recording (a recording of solo operatic singing).

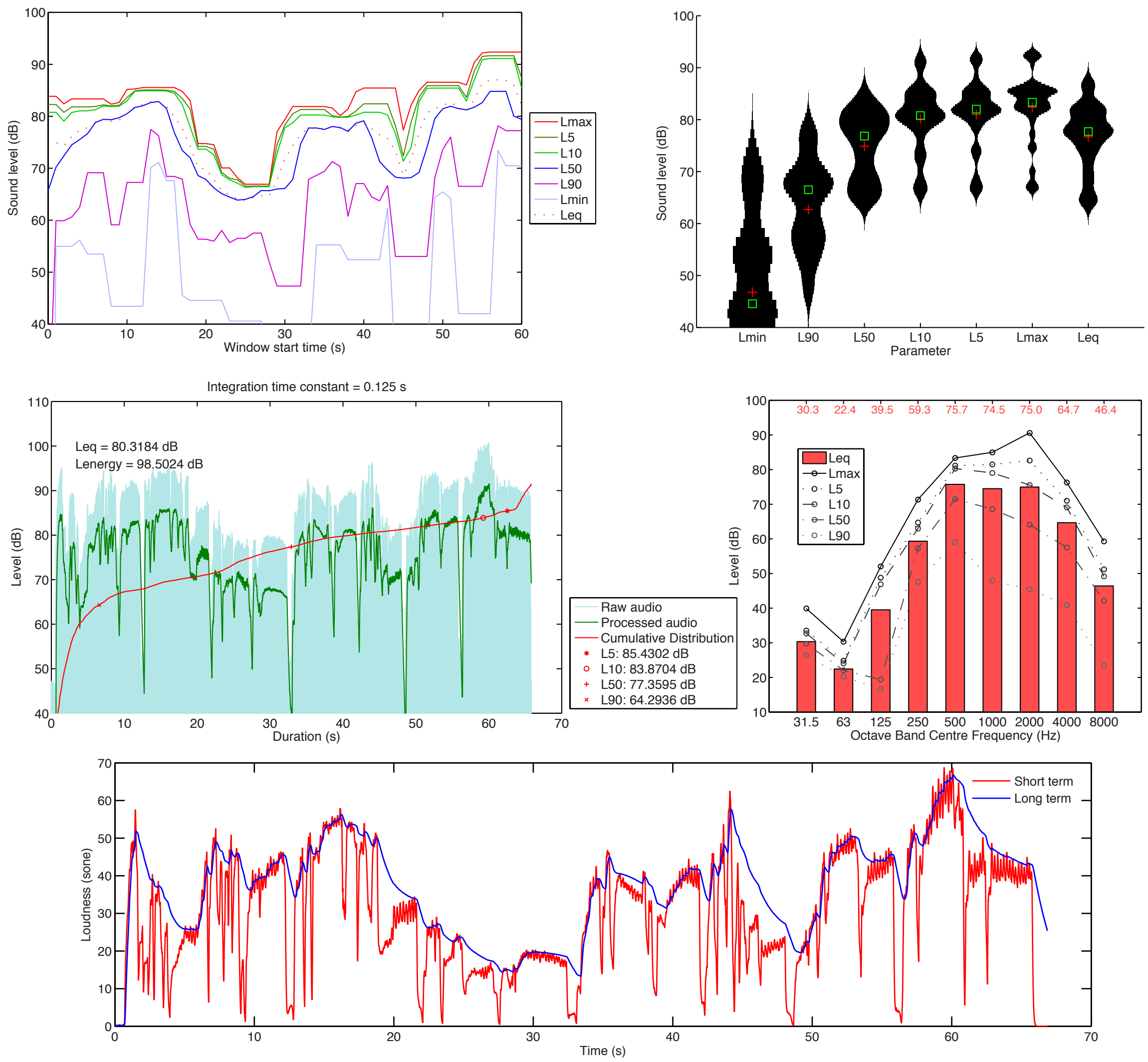

Figure 2 – Various ways of representing the strength of sound of a recording (in this case, a recording of an unaccompanied singer). The top left chart shows time-varying sound pressure level statistics, using a 5 s moving window. The top right chart shows distributions of the level statistics from the top left chart. The middle left chart shows the original instantaneous levels in light blue, the A-weighted fast integrated level in green, and the cumulative distribution (with percentiles) in red. The middle right chart shows octave band Leq (in red) and percentiles for each band. The bottom chart shows the time-varying loudness following Glasberg and Moore (4).

In 2014, Nicholas Lynar and Guy Hopkins (both masters students) implemented ANSI S12.2-2008 Noise Criterion (NC) and Room Noise Criterion (RNC) methods of rating background noise in rooms in an AARAE analyser (5). While many of the comparable noise rating systems are well suited to spreadsheet implementation following octave band analysis, RNC requires the analysis of low frequency fluctuation in the noise, which is not practical to do with a spreadsheet, but is well suited to a signal processing environment like MATLAB.

3.2 Reverberation parameters

While reverberation time is almost certainly the most used and best understood of the room acoustical parameters, measuring it well can involve several subtle issues. In a round robin study of reverberation time analysis using a single measured impulse response, Katz (6) found substantial divergences between results. One of the most common issues in correctly deriving reverberation time from impulse responses is the presence of background noise. While impulse response measurement techniques are usually designed to substantially reduce background noise, significant noise is quite common in the extremes of the frequency range. Approaches to dealing with background noise include finding the optimum truncation point for reverse integration (7,8) non-linear curve-fitting based on a model that includes exponential decay combined with steady background noise (9), subtracting the background noise (squared amplitude) from the squared impulse response prior to reverse integration (10) and extrapolation of the impulse response decay envelope beyond the noise floor (8,11). The effectiveness of approaches such as these have been examined in detail by Guski and Vörlander (11) and Venturi et al. (12), showing that highly accurate results can be obtained when the right analysis approach for the impulse response is taken. Hence AARAE provides substantial flexibility in reverberation time analysis. Rather than simply providing a generic analysis algorithm, AARAE allows the user to choose the analysis method.

As an exploratory tool, AARAE includes an analyser that calculates reverberation time over a wide range of end truncation points and evaluation ranges, so that the stability of the reverberation time calculation can be investigated (and perhaps the best way to characterize the reverberation can be determined). Figure 3 shows an example of a result of an impulse response measurement (63 Hz octave band only) that is affected by noise. The steep rise in the reverberation time value at long truncation points and large evaluation ranges is due to the noise artifact, but some stability in the values is apparent for moderate evaluation ranges (e.g., T20 does not appear to be noise-affected). In this example, very short evaluation ranges yield unstable results (not due to noise, but due to the particular form of the decay function). The zero reverberation time values occur when the attempt to perform a linear regression fails (and so represent unavailable data).

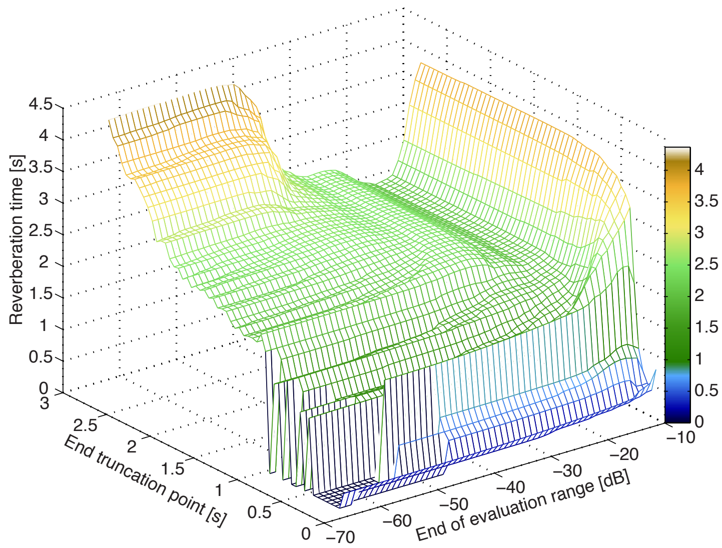

Figure 3 – A plot generated by one of AARAE's reverberation time analysers, which shows how the calculated reverberation time is affected by the impulse response end truncation (from 50 ms to the end of the recorded impulse response) and evaluation range (with ending values from -10 dB to -65 dB, equivalent to T5 to T60). This result comes from an auditorium measurement in the 63 Hz octave band.

A masters student, Grant Cuthbert, wrote the initial code for the conventional reverberation time analyser of AARAE and the code was later extended by the authors. This analyser allows the calculation of reverberation time with various approaches to noise floor treatment, including no treatment, and implementations of Chu's (10), Lundeby's (8) and Xiang's (9) approaches. These robust noise floor treatments have been helpful in large non-specialist class coursework teaching in which students have recorded handclaps for reverberation analysis. While handclaps are generally unlikely to provide a high signal-to-noise ratio, a reasonable approximation of reverberation time is usually achievable in at least some of the octave bands using Xiang's non-linear curve-fitting method or Lundeby's automatic truncation method. This low-tech approach to measuring reverberation time avoids the need to discuss some of the signal processing concepts that are needed to understand more conventional swept sinusoid measurements, minimizes the use of unfamiliar technology, and also encourages students to attend to the sonic experience of sound decaying in a room. An example of a hand clap reverberation time analysis is shown in Figure 4, and while none of the approaches succeed in the 125 Hz octave band, the non-linear curve fitting and automatic truncation methods give useful working approximations in the noise-affected 250 Hz and 500 Hz bands.

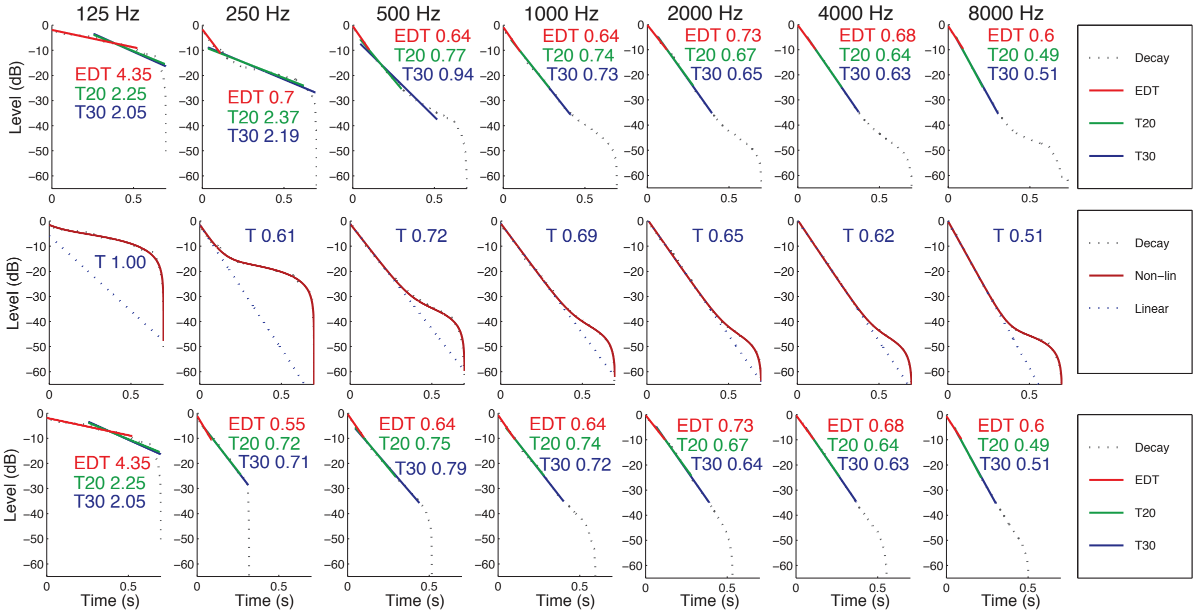

Figure 4 – Example of reverberation time analysis of a handclap recorded in a room by a group of students. Charts show the Schroeder reverse integration curve and the regression lines that are used to calculate reverberation time in the 125 Hz to 8 kHz octave bands. The top row of charts shows the result of linear regressions with no special treatment; the middle row shows the result on non-linear regression; and the bottom row shows the result of linear regression following automatic truncation. Values shown are early decay time (EDT), reverberation time (T20 and T30) for the linear regression methods, and reverberation time without any specific evaluation range (T) for the non-linear regression method.

AARAE also supports the measurement of reverberation time using the interrupted noise method, by averaging the squared decays of a sequence of statistically independent noise bursts. While this method is often not included in room acoustics measurement software (because impulse response-based methods are more efficient and can achieve greater signal to noise ratios), its inclusion in AARAE is helpful in education – as interrupted noise measurements give a more immediate experience of reverberation than indirect methods that require inverse filtering, and also because the relationship between impulse response and interrupted noise methods needs to be explained in acoustics teaching so that the purpose of reverse integration is understood.

Currently AARAE supports the measurement of impulse responses using a variety of methods such as exponential sweep (13), optimized Aoshima time-stretched pulse (14), maximum length sequence (15), Golay complementary sequences (16), transfer function methods, direct measurement with a delta function (or pulse train) and hybrid approaches.

Other approaches to reverberation analysis that are supported by AARAE include the loudness-based reverberance parameters proposed by Lee et al. (17), and a port of a blind reverberation time estimation method by Löllmann and Jeub (based on 18). The first of these is an example of how AARAE can be used to support research – the analyser is an implementation of an algorithm proposed in a journal publication, giving other researchers and coursework students easy access to the outcome of the research project. This type of approach facilitates the incorporation of research into teaching. The blind reverberation estimation analyser is an example of how quite complex pre-existing code can be ported to the AARAE environment, using a ‘calling function’, i.e. a small function that interfaces between AARAE and the pre-existing code.

3.3 Speech intelligibility parameters

Speech intelligibility analysis has been a focus of attention in the first year of AARAE’s development, especially methods to calculate the speech transmission index (STI) (19). The focus on this topic has been the result of other research projects that required STI and speech intelligibility index (SII) analysis.

Cabrera et al. (20) examined several implementations of the indirect method of measuring the STI (i.e., calculating the STI from an impulse response, and adjusting for octave band signal and noise sound pressure levels), finding some vulnerabilities in each of the surveyed software implementations. These included issues relating to the octave band filterbank design, the frequency resolution at which the modulation transfer function was calculated, and the adjustment for auditory masking and threshold, as well as other unidentified issues that led to inaccurate results. As a result of that work the AARAE implementation is particularly robust, and in cases where results differ between software AARAE is most likely to be giving the correct result.

One of the more difficult aspects of making indirect STI measurements is specifying the speech level. IEC60268-16 provides some guidance on this in Annex J, with an algorithm that excludes the gaps in the speech in the calculation of overall speech level (19). This is implemented as an analyser in AARAE (see Figure 5), allowing realistic octave band levels of speech to be used.

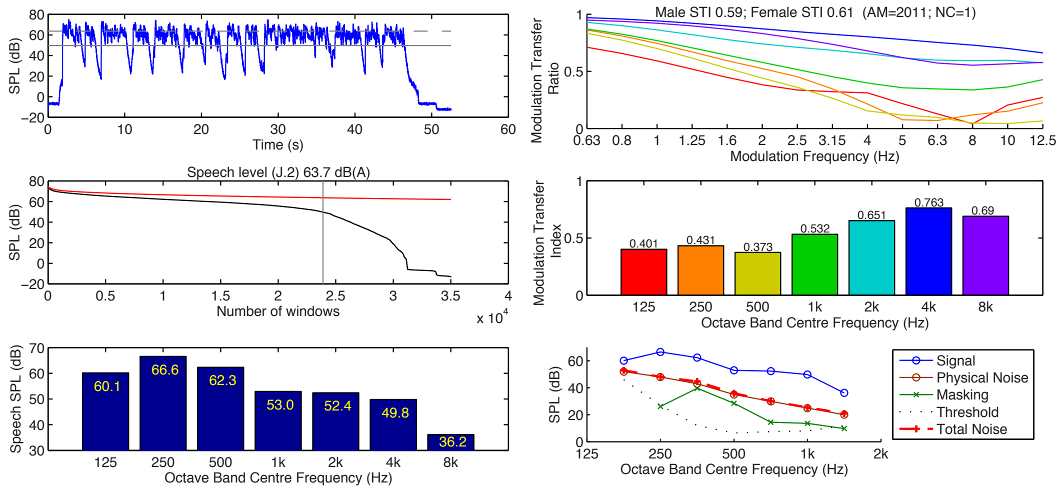

Figure 5 – An analysis leading to the calculation of STI using recorded speech (left charts) to calculate speech level, combined with the analysis of a measured impulse response (right charts). The speech level is determined so as to exclude the gaps in the speech by sorting the intensity values of a large number of 15 ms windows, and finding the point at which the speech intensity is 14 dB below its cumulative sum (middle left chart, indicated by the vertical line). The solid horizontal line in the top left chart shows the corresponding threshold for selection, with octave band sound pressure levels of selected data in the bottom left chart. The STI value can be interpreted in terms of the modulation transfer function (top right chart), the modulation transfer index (middle right chart) and the various sources of noise (bottom right chart).

AARAE includes a STIPA (speech transmission index public address) and full STI generators and analysers, allowing direct measurement using modulated noise (the full STI measurement is made from a sequence of seven STIPA-like signals over 3 to 5 minutes). Clearly it is much more effort to make STIPA (and direct STI) measurements in this way than with purpose-designed hardware that generates a test signal at a calibrated level and automates the calculation, but there are some advantages in hosting STIPA in AARAE. Measuring STIPA from AARAE allows comparison with the indirect method, and it gives more control over the measurement and analysis than a dedicated hardware system (e.g., in customizing the measurement signal, using a particular loudspeaker, or in analysing the results). Although the process of loudspeaker calibration might be considered onerous, it could be seen as a benefit from an educational perspective. Once calibrated, the sound pressure level of the STIPA signal could also be used to indicate speech level in indirect measurements.

AARAE includes an analyser that calculates SII using the modulation transfer function method, which was included for comparison with STI. As part of a research project conducted by Adam Opsata (a masters student), an analyser has been included in AARAE that estimates STI values from recordings of speech (21). More details are given in another paper at this conference (22).

3.4 Harmonic distortion

Currently AARAE supports two methods of harmonic distortion analysis: direct measurement using stepped tones (systematically testing tones of various frequencies and gains), and measurement using exponentially swept sinusoids. The latter was implemented in 2014 by three masters students, Adam Opsata, Nathan Ashmore and Adrian Clarke, based on previous studies by Farina (13) and Abel and Berners (23). The measurement method involves a sequence of sweeps (usually quite long duration, such as 60 s), stepped gain (e.g. 6 dB steps), preceded by a silent cycle. The silent cycle allows distortion to be distinguished from noise in the analysis of the impulse responses. A high audio sampling rate is desirable, because harmonics are at integer multiple frequencies of the fundamental. Figure 6 gives an example of the results of analysis using 48 kHz sampling rate (i.e. not a high sampling rate), showing an apparent drop-off in total harmonic distortion at 8 kHz which is actually a measurement artifact due to the lack of data from the third harmonic above that frequency (evident in the individual harmonic distortion chart) – 8 kHz is the Nyquist frequency divided by 3.

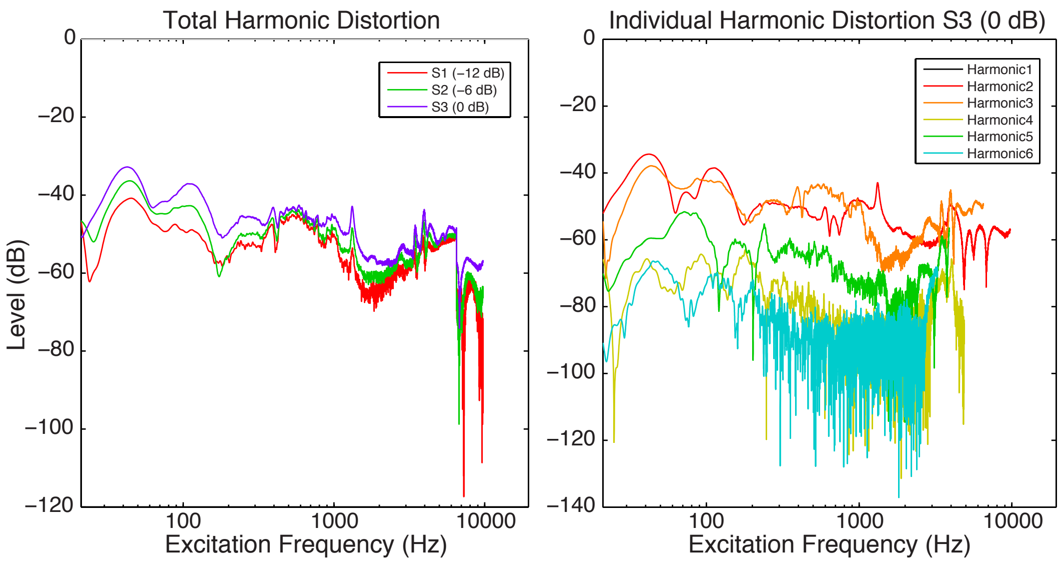

Figure 6 – Plots generated by one of AARAE's harmonic distortion analysers, from measurements made on a small low quality loudspeaker. The harmonic distortion spectra shown are relative to the linear response spectrum. On the left, total harmonic distortion is shown for three measurements (60 s duration exponential sweeps, with 3 dB gain steps). The right chart shows the individual harmonic distortion (harmonic 1 being the linear response, which has a value of 0 dB by definition) for the greatest gain sweep.

3.5 Spatial analysis

The use of spherical microphone arrays in architectural acoustics measurement has been growing over the past decade, and AARAE supports such measurements (e.g., using the 32-channel Eigenmike, or similar device). The HOAToolbox by Nicolas Epain is used for much of the spatial microphone signal processing in AARAE, with AARAE analysers adapted from code by PhD student

Luis Miranda. Figure 7 gives an example of analysis using an impulse response from an omnidirectional loudspeaker to a spherical microphone array on the stage of an auditorium (the transducers were 1 m apart, as is normal for stage acoustics measurements). The spatial distribution of the sound field at the microphone can be seen, with a prominent back wall reflection in the early part of the impulse response (20-100 ms, top left) and a more diffuse sound field in the late part of the impulse response (100 ms onwards, bottom left). Sound field diffusivity can be enumerated in various ways, and using Gover et al.'s method (23) the early sound field's diffusivity is 0.63, while the late diffusivity is 0.94. The top right chart shows the evolution of diffusivity during the course of the impulse response (using Gover's method), showing an initial rapid increase in spatial diffusivity, plateauing at a high level. However it is also interesting to consider diffusivity from a purely temporal (rather than spatial) perspective – the analysis of which is also supported by AARAE. One approach, by Abel and Huang (24), considers the degree to which the waveform in short time windows has a Gaussian distribution (which can be evaluated using methods related to standard deviation or kurtosis). Results for this, using only the 0th order spherical harmonic (i.e. omnidirectional) are shown on the bottom right chart, indicating a 'mixing time' of about 50 ms.

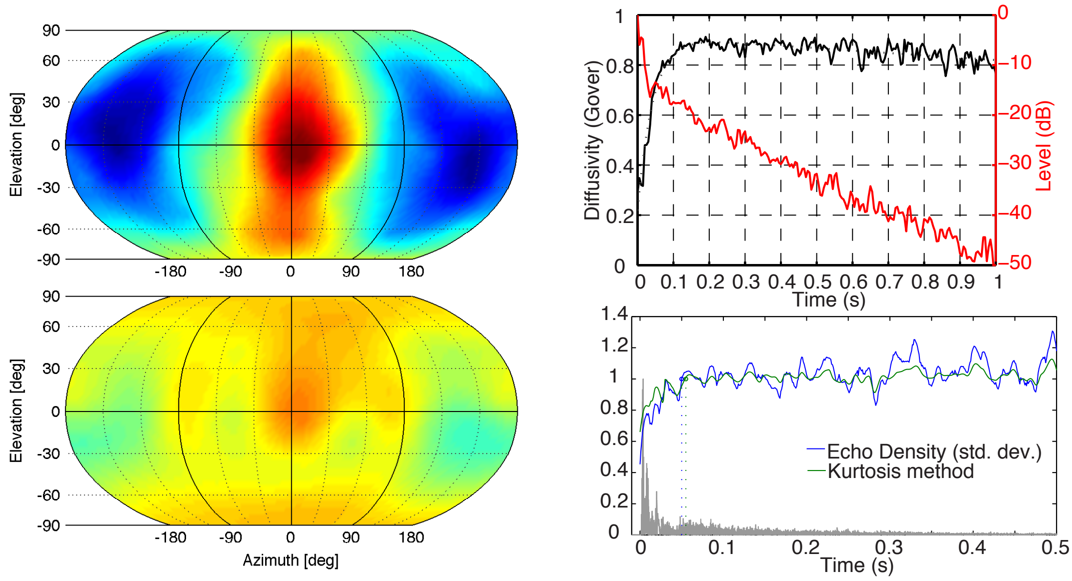

Figure 7 – Plots describing the diffusivity of an on-stage auditorium impulse response, recorded using a 64-channel spherical microphone array (Visisonics). On the left, the spatial distribution of energy is shown (on a 7 dB scale from blue to red), the upper chart showing the 20-100 ms period, and the lower showing energy distribution from 100 ms onwards. The upper right chart shows the evolution of spatial diffusivity (along with the level decay), and the lower right chart shows the evolution of temporal diffusivity.

4. CONCLUSION

This paper gives a brief outline and overview by example of the scope and capacity of the AARAE project. This project is likely to develop considerably over the next few years as it is further integrated into teaching and research, and as more students do projects using and extending the code. It will most naturally find application in systematic analysis projects that require the integration of established measurement methods that can be efficiently managed within the MATLAB environment. It is hoped that others (outside the University of Sydney) also find the project of use, and it is freely available for download from http://aarae.org.

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