NEW ZEALAND ACOUSTICS

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The Acoustical Society
of New Zealand

ACOUSTICS 2024

THE ACOUSTICAL SOCIETY OF NEW ZEALAND CONFERENCE

Acoustics: Reflecting on the past, innovating for the future

Ōtautahi Christchurch | 2-4 September

www.acousticsnz2024.co.nz

The Acoustical Society of New Zealand (ASNZ) will hold the Acoustics 2024 Conference within the heritage-listed Christchurch Town Hall of Ōtautahi Christchurch New Zealand, from the 2–4 September 2024. We invite you to come and reflect on the past in this beautifully restored and refurbished venue, nestled on the banks of the Avon River in the Central City.

Acoustics 2024 will provide engineers, scientists and professionals in all fields of acoustics the chance to exchange views, the latest research and share experiences with colleagues. This location holds significance to our Society, as we continue to enjoy the acoustic design and associated research advances of lateral reflected sound in concert halls, as completed by ASNZ Fellow Sir Harold Marshall in the late 1960's. Be inspired by venue acoustics, integrated technology, iconic architectural features and riverside views. We look forward to guest keynote speakers, sharing a full and interesting programme covering a wide range of topics, along with excellent social functions, and networking opportunities. There will also be a unique opportunity for manufacturers and suppliers to showcase the latest developments in acoustic instrumentation, software, and noise and vibration control products. All of these opportunities are aptly reflected in our conference theme for 2024, *Acoustics: Reflecting on the past, innovating for the future.*

Christchurch is the gateway to the stunning South Island and is easily accessible by international and domestic flights into Christchurch airport, which is only a short 15 min drive to the CBD and venue. Known as the Garden City of New Zealand, Christchurch boasts over 700 parks and gardens, along with 80 kilometres of city walking tracks. As you explore the central city you will discover amazing street art, innovative projects, and state-of-the-art architecture nestled between restored historic buildings. Enjoy the farmers market, go punting down the Avon River, take a tour on the City Tram loop, or have a picnic in Hagley Park. From Christchurch you can explore mountains and the ocean all in one day – and there is something for everyone. Go walking in the Port Hills and take in the breathtaking scenery of the Canterbury Plains and Southern Alps, take a ride up the Gondola, go zip-lining or bike-riding at the Adventure Park, head to the exquisite wine country of North Canterbury, or have a dip in the hot pools at New Brighton pier.

The Acoustical Society of New Zealand Council, and Acoustics 2024 Organising Committee looks forward to welcoming you to Ōtautahi Christchurch next year. We hope that the conference gives you an opportunity to strengthen your existing networks and that you leave with great memories, fresh ideas, and new friendships.

Keep up to date with the latest conference information by visiting: www.acousticsnz2024.co.nz, with Registration and Abstract Submissions opening in early 2024.

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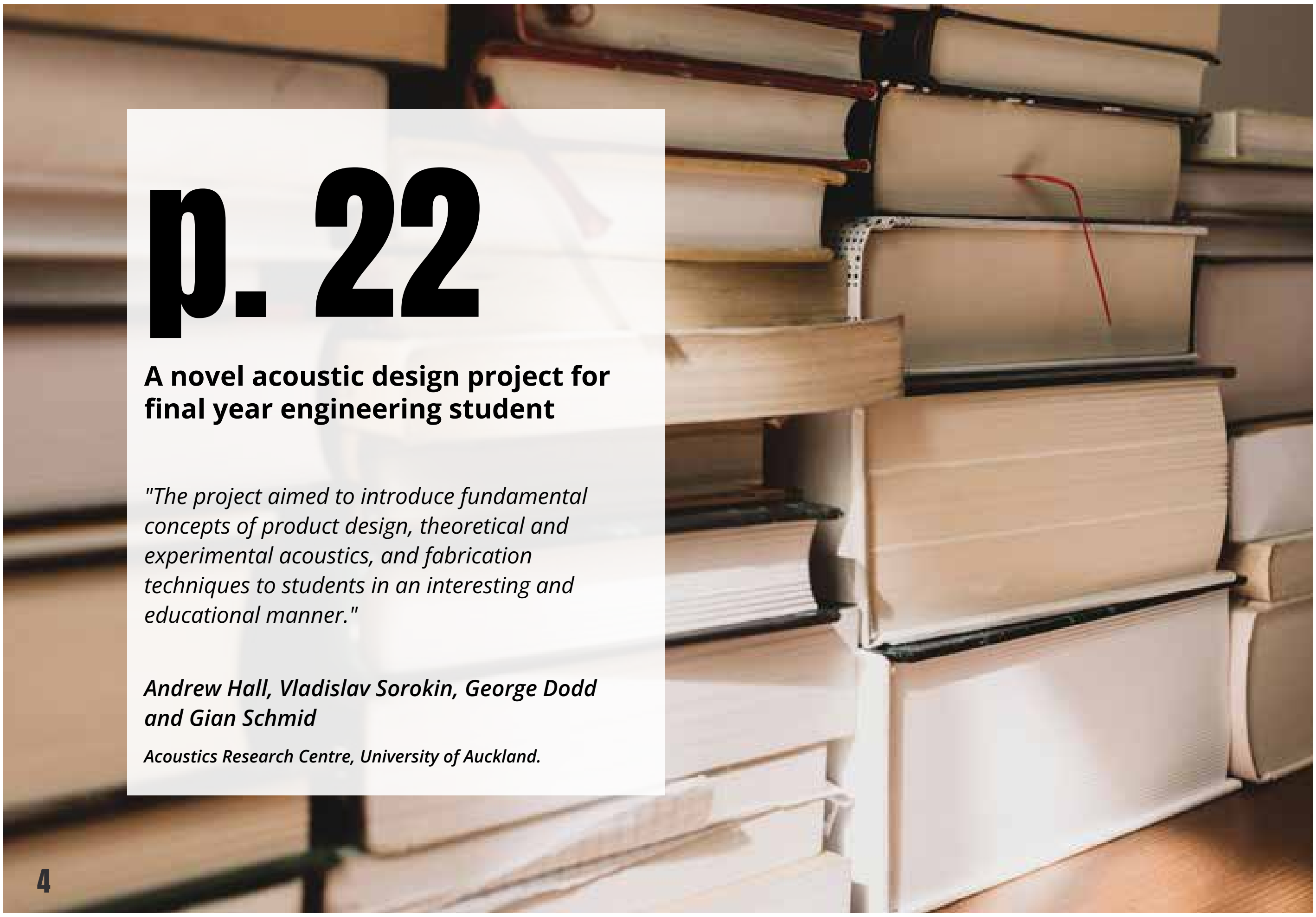

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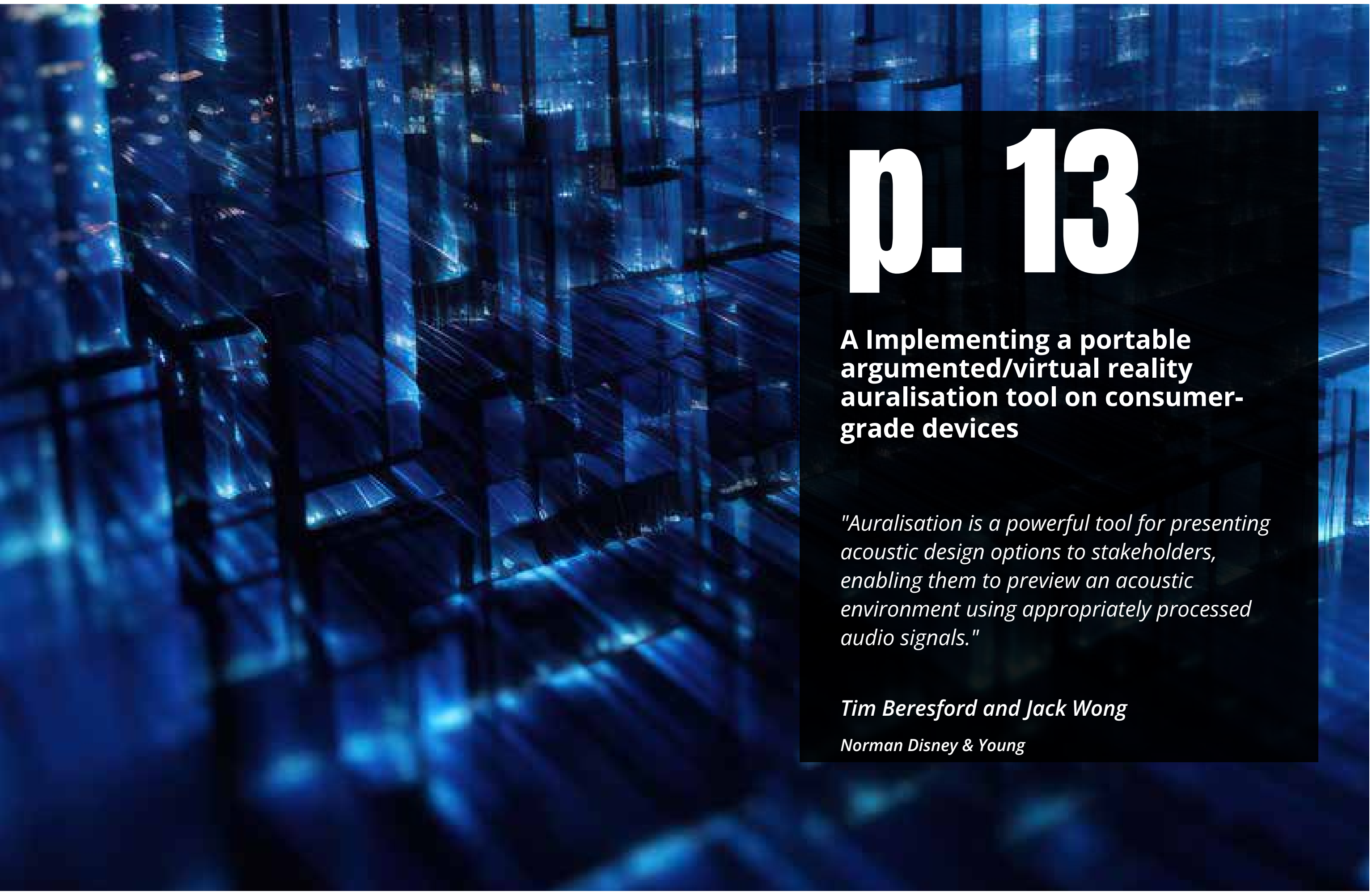
A novel acoustic design project for final year engineering student

"The project aimed to introduce fundamental concepts of product design, theoretical and experimental acoustics, and fabrication techniques to students in an interesting and educational manner."

Andrew Hall, Vladislav Sorokin, George Dodd and Gian Schmid

Acoustics Research Centre, University of Auckland.



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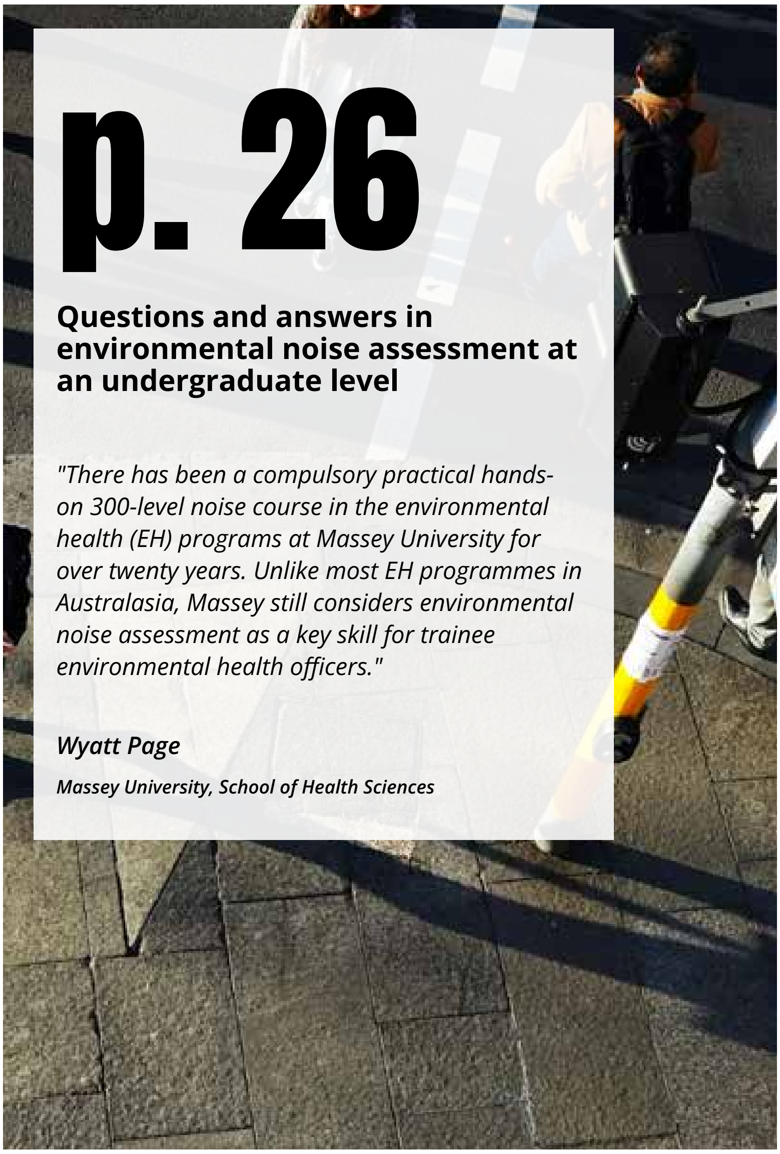
A Implementing a portable argumented/virtual reality auralisation tool on consumer-grade devices

"Auralisation is a powerful tool for presenting acoustic design options to stakeholders, enabling them to preview an acoustic environment using appropriately processed audio signals."

Tim Beresford and Jack Wong

Norman Disney & Young



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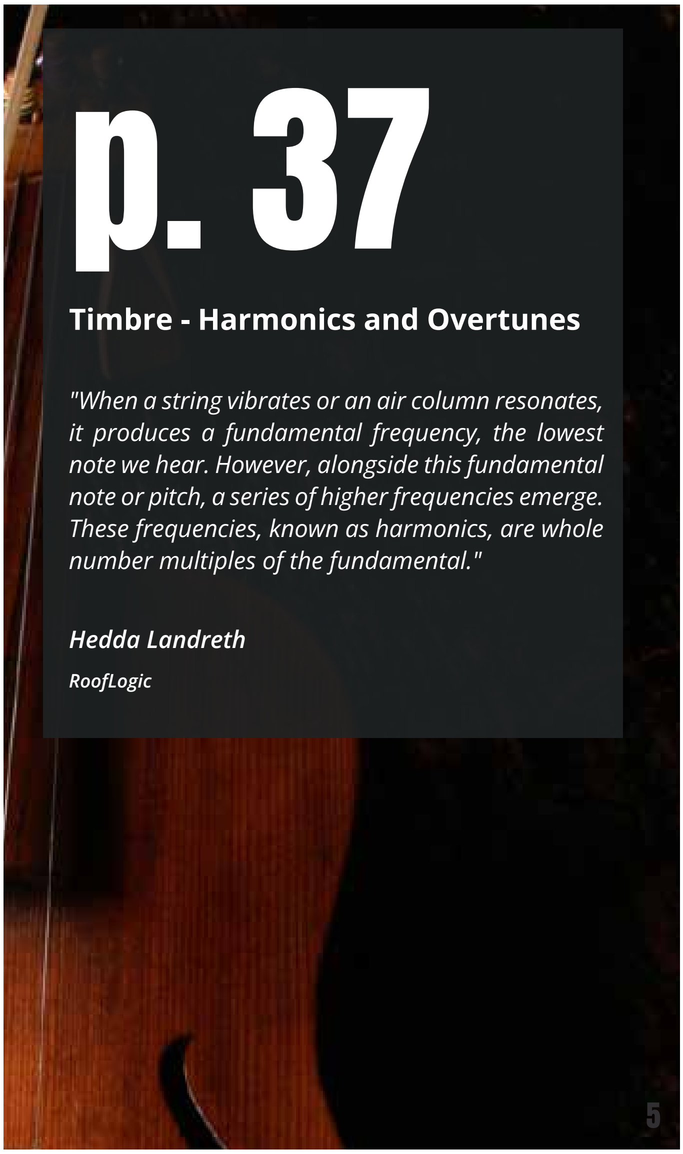
Questions and answers in environmental noise assessment at an undergraduate level

"There has been a compulsory practical hands-on 300-level noise course in the environmental health (EH) programs at Massey University for over twenty years. Unlike most EH programmes in Australasia, Massey still considers environmental noise assessment as a key skill for trainee environmental health officers."

Wyatt Page

Massey University, School of Health Sciences



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Timbre - Harmonics and Overtunes

"When a string vibrates or an air column resonates, it produces a fundamental frequency, the lowest note we hear. However, alongside this fundamental note or pitch, a series of higher frequencies emerge. These frequencies, known as harmonics, are whole number multiples of the fundamental."

Hedda Landreth

RoofLogic

Kia ora koutou,

Again, the end of another years is rapidly closing in, but the weather is picking up nicely. We hope you have enjoyed your local ASNZ hosted end of year event, and any other festivities that you have in store.

To round out the year, SoundPrint held it's third "Find Your Quiet Place" event during October, and, in conjunction with the recent promotion of the SoundPrint app by the ASNZ, a record 30 new venue SoundChecks were added to the database since September. Check out the latest acoustic quality ratings for your local cafe/restaurant published in this issue of the Journal.

I was lucky enough to attend the joint AAS ASA Acoustics 2023 Sydney conference in early December along with fellow Councillor, Christian Vossart. This was a wholesome five-day event covering all types of acoustics (including land-based, and underwater varieties). We also attended the Australian Acoustical Society's Federal Council meeting where plans were developed to strengthen ties between our societies across the ditch.

Along with other ASNZ members, I have been co-authoring a AAAC guideline to help interpret New Zealand Building Code clause G6. Those few sentences in G6 are a veritable Pandora's box, and a lot of good work has been done over the years by the acoustic consulting fraternity to ensure G6 is applied correctly. This guideline looks to capture the views of various consultants working across New Zealand. The guideline is due to be formally published before the end of the year and will be available on the AAAC website: aaac.org.au/Guidelines-&-Downloads

Wishing you all the best for the festive season and we'll see you in the new year.

Ngā mihi,

Tim Beresford

President of the Acoustical Society of New Zealand

Kia ora koutou,

Welcome to the third and final issue of New Zealand Acoustics for 2023. It's that time of year again when Summer joins us and we head off to spend time with our loved ones. Its also the time of year where I thank the team for all the hard work in 2023 on helping create New Zealand Acoustics, specifically Wyatt Page, Hedda Landreth, Edward Dyer and Holly Wright. I also want to take the time to specifically thank all the authors who have contributed this year and of course our advertisers. I ask if you have a paper or piece or are thinking of preparing something and you wish to contribute please get in touch with the team we are always looking for new material.

We have a great selection of papers in this issue, including a paper by our President Tim Beresford and his colleague Jack Wong entitled 'Implementing a portable augmented/virtual reality auralisation tool on consumer-grade devices'. We also have a paper from the Acoustics Research Centre, University of Auckland entitled 'A novel acoustic design project for final year engineering students' and continuing the theme a paper from Massey University Health Sciences from our very own Editor Wyatt Page entitled 'Questions and answers in environmental noise assessment at an undergraduate level'. I recommend you dive into all these papers; they are all well worth the read. We also have our 'bread and butter' in this issue such as the news articles and quiz.

The team at New Zealand Acoustics wish you and your families all a great holiday season and break. Please be safe and we will see you all again in 2024 for Vol 1.

Seasons Greetings,

Lindsay Hannah & Wyatt Page

Principal Editors

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Tim Beresford

Lindsay Hannah

Wyatt Page

NEWS

Acoustic cameras can see sound

Acoustic cameras have an array for microphones that are able to reproduce spatiall information about sound. They even work in slow-motion and echoes look amazing!

https://www.youtube.com/watch?v=QtMTvsi-4Hw

Are electric vehicles too quiet?

At low speeds (below 30 km/h) they are so much quieter that to reduce the risk of accidents for pedestrians the EU has introduced a regula on for electric vehicles, imposing a minimum of 56 decibels at low speeds

https://www.youtube.com/watch?v=2J-oNEZfnZ8&list=WL&index=15

https://www.youtube.com/watch?v=tRc1A-Zf_DE

Could Chat GPT talk to whales?

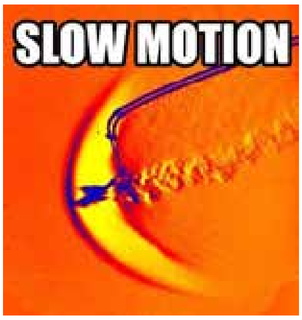
To really understand the structure and evolution of whale language, we first need to understand our own. The evolutionary past of human language is not straightforward. But understanding it's origins might give us more hints about how language is used by our ocean friends.

https://www.youtube.com/watch?v=hph9OeKjg3w&list=WL&index=21

Man goes on hunger strike to protest the noise from a nearby Pickleball court

A Chilliwack, B.C., man is threatening to go on a hunger strike over noise coming from a pickleball court next to his house, which he says the city won't do anything about.

https://www.youtube.com/watch?v=zTGTN6ku_Ho



How does a whip break the sound barrier?

https://www.youtube.com/watch?v=AnaASTBn_K4&list=WL&index=14

The loudest sound in the quietest room

https://www.youtube.com/watch?v=EBClzOM_Lbw&list=WL&index=14

New Western Sydney International Airport flight paths

Proposed flight paths for Western Sydney Airport have finally been made public, with residents learning how noisy aircraft will be over their suburbs.

https://www.youtube.com/watch?v=YExPuSNN6n4&list=WL&index=7

The giant art that keeps planes quiet

https://www.youtube.com/watch?v=7dlLmeage2U&list=WL&index=24

What are those noises after take-off?

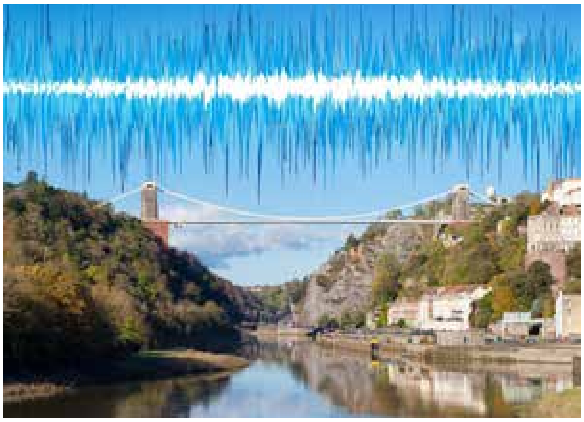
https://www.youtube.com/watch?v=rSwrlzpCkTw&list=WL&index=25

Noise cancelling headphones for your homes

https://www.youtube.com/watch?v=Rz2nNDt2O8o

Have you heard the hum? The source behind the worldwide 'hum'

https://www.youtube.com/watch?v=vul4SYL4QiQ



Why parrots can talk like humans

https://www.youtube.com/watch?v=dBGw7uXc0eo&list=WL&index=70

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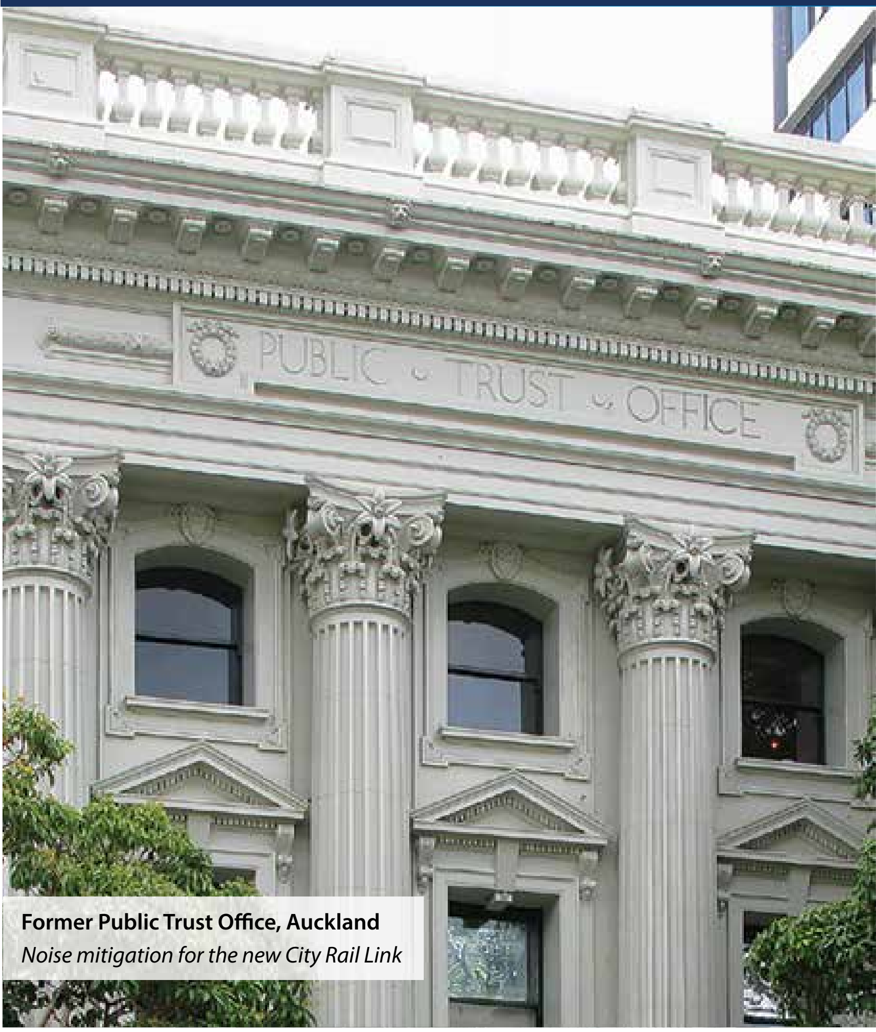
Reduce noise through existing windows & doors



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STUDENT PRIZES

This year, the Acoustical Society of New Zealand awarded two prizes to students studying acoustics at New Zealand higher education institutions. These prizes were awarded for outstanding performance in an undergraduate student project and for completion of several outstanding journal papers on a subject related to acoustics. The undergraduate student prize7 was awarded to Joel Griffin and Tim Peck whilst the other award went to PhD student Robin Go who published four excellent journal papers in 2023 (three in the Journal of Sound and Vibration and one in Applied Acoustics). Below are several paragraphs from these students describing their work and thanking the Society.

Joel Griffin and Tim Peck It was a privilege to receive the Acoustical Society of New Zealand Award for our Part IV research project, undertaken as part of our final year in the Bachelor of Engineering program at the University of Auckland. Our research focused on evaluating the potential of acoustic metasurfaces in enhancing low-frequency sound attenuation within buildings. The idea for the project was put forward by our supervisors Dr Andrew Hall and Dr Vladislav Sorokin, and was inspired by the recent surge of development in acoustic metasurfaces. Recent studies have demonstrated that these metasurfaces can achieve near-perfect sound absorption at frequencies as low as 50Hz while remaining exceptionally thin. Our project aimed to determine if multiple metasurface units could be integrated into a panel to enhance the transmission loss performance of a double leaf partition or serve as a thin low-frequency absorption panel.

We evaluated several concepts from the current literature to identify the most effective metasurface design. Through impedance tube testing of individual metasurface units, we found the most effective metasurface to be a design which features a coiled backing cavity and a narrow aperture. The metasurface works like a microperforated panel, whereby absorption occurs due to viscous dissipation within the aperture at resonance. We then integrated 32 metasurface units into two 15mm thick acrylic panels. These metasurface panels were then separated with a stud to form a sample stud partition, which was tested with a soundbox and laser vibrometry. The tests showed promising results with a peak insertion loss of 16dB at 120Hz with at least 3dB of insertion loss between 102Hz and 192Hz.

Given that the panels are only 15mm thick, these results indicate that acoustic metasurfaces can provide a compact solution to improving the transmission loss of a double leaf partition at the mass-air-mass resonance frequency. The panels may also have an application as a thin, low-frequency absorption panel. Testing was conducted on 1.2 m 2 of metasurface panelling within a reverberation chamber; however, the results were inconclusive due to the small sample size. Further experimentation is required to provide more conclusive results.

The project has helped to instil a keen interest in acoustics for us both, we look forward to using the knowledge and skills learned in future endeavours. We are grateful to the University of Auckland, our supervisors and the ASNZ for supporting our work in this area.

Photograph of the undergraduate student prize winners Joel Griffin and Tim Peck showing their metasurface absorbers.

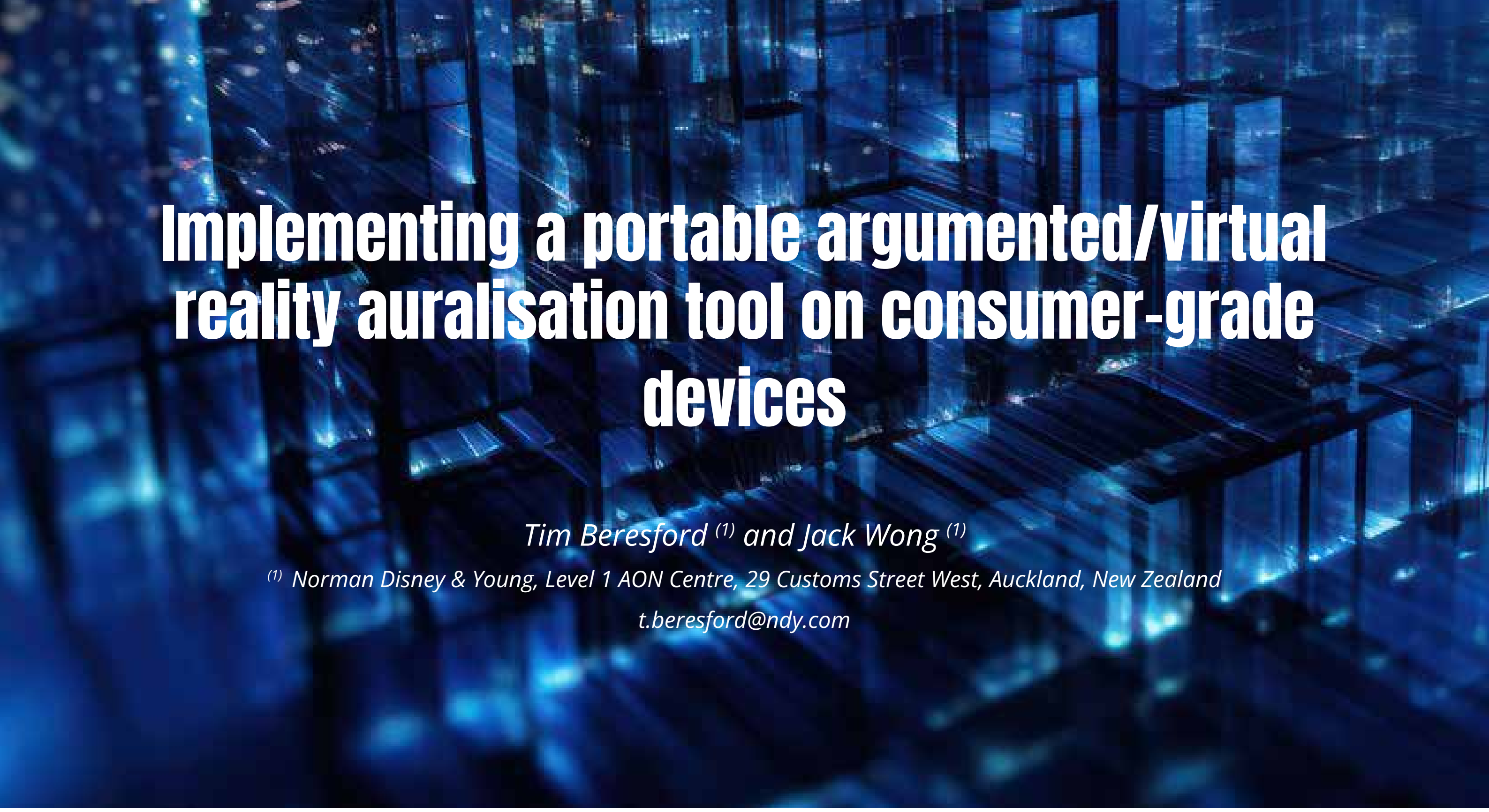
Robin Go

I am very grateful to receive this prize from the Acoustical Society of New Zealand.

I am a PhD student at the University of Auckland studying the noise produced by shrouded unmanned aerial vehicle (UAV) propellers. My interest on UAVs began from doing a summer studentship with Dotterel Technologies, a local company in Auckland, and with my PhD supervisor Michael Kingan, where I helped perform a number of acoustic measurements at the University of Auckland's anechoic chamber of a contra-rotating UAV propeller and varied several parameters such as the propeller diameter to determine the best configuration that would reduce the noise.

I then continued my research on the noise from UAVs by doing a PhD on how an annular shroud or duct affects the noise produced by the propeller. An annular shroud improves the safety and the aerodynamic efficiency of a UAV when hovering. Through this PhD, I had the opportunity to visit several universities around the world and use their anechoic wind tunnels to perform acoustic measurements of a shrouded UAV propeller in an airflow.

During my visit to overseas universities and conferences I learnt that there are emerging technologies for small aircraft to carry passengers or cargo at lower altitudes in urban areas known as 'Urban Air Mobility' where the use of such aircraft will undoubtedly be limited by the noise they produce, and I hope to continue my research down this path after completing my PhD.



Implementing a portable argumented/virtual reality auralisation tool on consumer-grade devices

Tim Beresford [1] and Jack Wong [1]

[1] Norman Disney & Young, Level 1 AON Centre, 29 Customs Street West, Auckland, New Zealand

t.beresford@ndy.com

Abstract

Auralisation is a powerful tool for presenting acoustic design options to stakeholders, enabling them to preview an acoustic environment using appropriately processed audio signals. Current state-of-the-art auralisations require a dedicated listening room with very low reverberation times, very low background noise, and a multi-channel loudspeaker setup. More recently, auralisations have been coupled with virtual reality (VR) visualisations to enhance the user experience, through use of VR headsets. Recent computational power increases in portable consumer-grade devices, such as smart phones and tablets, have meant that rendering a 3D augmented reality (AR) or VR simulation is now a possibility on such devices. AiHear® is an AR/VR auralisation application which utilises the portable device platform in conjunction with an off-the-shelf (calibrated) headphone setup to playback accurate auralisation audio. This paper looks at some of the technical aspects of successfully implementing an AR/VR auralisation application for use on a low-cost consumer-grade platform.

Introduction

Auralisation is a powerful tool for presenting acoustic design options to stakeholders, enabling them to preview an acoustic environment using appropriately processed audio signals. Current state-of-the-art auralisations re-quire a dedicated listening room with very low reverberation times, very low background noise, and a multi-channel loudspeaker setup. More recently, auralisations have been coupled with virtual reality (VR) visualisations to enhance the user experience, through use of VR headsets. However, such listening rooms setups are very expensive and are in no way readily accessible or portable.

Recent computational power increases in portable consumer-grade devices, such as smart phones and tablets, have meant that rendering 3D graphics in an augmented reality (AR) or VR simulation is now a possibility on such devices. Simultaneously rendering binaural audio in the 3D environment is also a possibility, whereby the audio source direction moves/rotates relative to the participant's head position/orientation in the 3D simulated world.

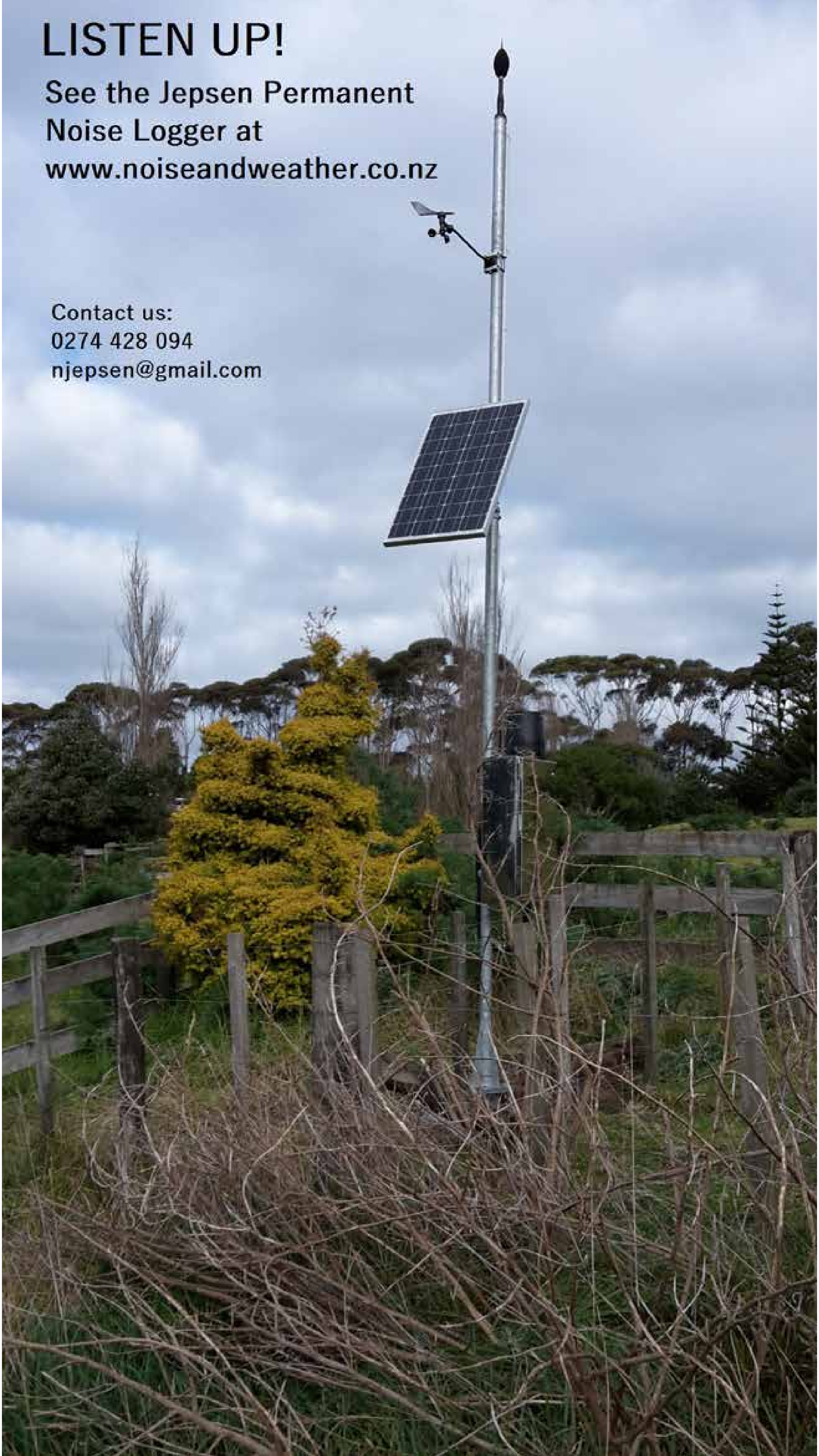
The use of headphones as the audio playback medium has many advantages for a portable AR/VR auralisation system, as discussed later in this paper. Through careful calibration of the AR/VR system playback levels and frequency responses, it is possible to reproduce audio signals with a high degree of spectral accuracy, thus making this platform suitable for engineering-precision auralisations.

Over the past several years, NDY Acoustics has incrementally developed a successful software implementation of AR/VR auralisation, called AiHear®, which utilises portable consumer-grade device technology as its platform. The features of the AiHear® application, along with some of the technical challenges, are discussed in this paper.

VR Versus AR

VR technology has been in common use for several decades now, and many people are familiar with this form of simulation. In virtual reality, the user is fully immersed in a simulated visual environment, whereby the real world is completely replaced by the simulated world.

AR is similar in many ways to VR, in that the user experiences simulated elements which do not exist in reality, however, the simulated elements do not necessarily block out the real world. Instead, the real world is enhanced or augmented with the simulated elements; for example, a simulated chair can be placed within a real room and will remain stationary within the room, even when the user's viewpoint moves/rotates.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The AIHEAR® System

Hardware setup

Figure 1. Typical AiHear® hardware components.

To make the AiHear® auralisation system as portable as possible, and to keep setup costs low, the application utilises only two key hardware components:

- An Apple portable device (iPad Air 3rd generation, iPhone X or better, running iOS 14 or newer*).
- Medium-to-high quality stereo headphones which have been calibrated through the AiHear® system. The current list of calibrated headphones includes Audio-Technica ATH-M40x and JBL Live 660NC**.

*For lower latency and better AR spatial tracking, an Apple device equipped with LiDAR sensors is recommended (e.g., iPad Pro 2nd generation, iPhone 12 Pro).

**The ATH-M40x headphones are wired and require an additional Apple USB-C to 3.5mm adaptor for most current Apple devices. The JBL Live 660NC headphones use Bluetooth to connect wirelessly to the portable device.

Simulation operation

In operation, AiHear® combines pre-recorded audio signals with a 3D simulated model to produce an auralisation which is linked to AR/VR visualisations. Users can choose between AR or VR visuals, depending on their required application.

The environment to be auralised is created in 3D using the portable device running the AiHear® application, as follows:

- The simulation room is created by placing coordinates for each of the room's corners, or by using the simple box-shaped room creation tool.
- The room surfaces are assigned sound absorbing materials (with associated 1/3 or 1/1 octave absorption coefficients). The corresponding room constants and reverberation times are automatically calculated.
- The sound source is placed at the desired location, either inside or outside the simulated room. The source type can be selected as a stationary point source, moving point source or line source.
- 1/3 octave transmission loss filters can be applied to represent different sound insulation properties of the room's walls, for the situation where the sound source is located outside. Composite transmission losses are calculated where a wall consists of multiple materials with varying sound insulation properties.

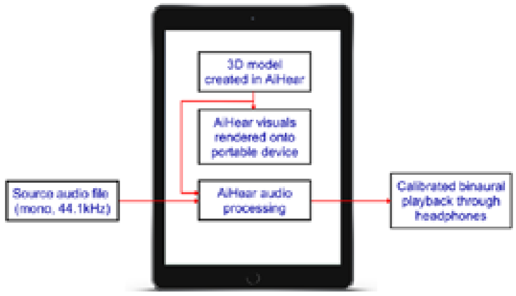

Figure 2. AiHear® functional schematic.

The portable device's screen acts as the lens though which the simulation is viewed. By moving the device around, the user can look around the simulated room which will appear fixed in space relative to the real world. The relative motion/rotation of the portable device to the simulated sound source alters the directional arrival of the audio signals to the user's ears to create a 3D audio environment. For example, a simulated point source moving from left to right across the user's point of view will audibly shift from being louder in the left ear to the right ear to match the visuals. Similarly, if the point source is stationary, but the portable device's point of view is rotated away from the source to the right (clockwise), the audio will become louder in the left and quieter in the right ear.

Figure 3. An example of a simulated line source in AR

AIHEAR® Features

User-defined absorption coefficient and transmission loss data

Users can edit and save their own absorption coefficient and transmission loss data as either 1/3 octave or 1/1 octave band values. Absorption coefficients are saved as "materials" which can also have user-defined visual textures and colours. An additional feature attached to the transmission loss tool is the ability to visualise wall constructions using the in-app wall build-up tool. Users can move closer to the wall cross section to inspect it, whilst hearing the transmission losses the wall construction has to offer.

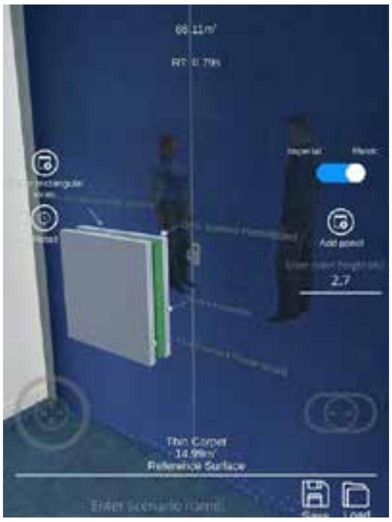

Figure 4. An example of the wall build-up visualisation tool. In this scenario, a conversation is being simulated in the next-door room whilst the sound insulation performance of the seperating wall is demonstrated.

Advanced room geometry setup

In addition to the in-app method of simulated room creation, advanced users can create rooms by writing the corner cartesian coordinates (x, y, z) to a .CSV file which is loaded into AiHear®. There is no limit on the number of corners (or sides) a room can have.

Reverberation simulation

Reverberation is added to the simulated audio playback based on the calculated reverberation time for the room. Early reflections are also modelled to provide better realism to the auralisation.

Saving and loading simulation scenarios

All simulation parameters can be saved as scenarios, which can then be loaded during a presentation, so that different model configurations can be switched between rapidly. This enables direct A/B comparison of different acoustic treatment options.

Multi-user mode

AiHear® allows multiple users to join the same simulation, each with their own individual portable device. One user's device hosts and controls the simulation parameters while other users' devices can join and spectate. Each user views the simulation from their own point of view within the simulated environment and can move closer or further from the sound source and look towards or away from it at their own free will. The theoretical maximum number of spectators is 100, although this has not yet been tested.

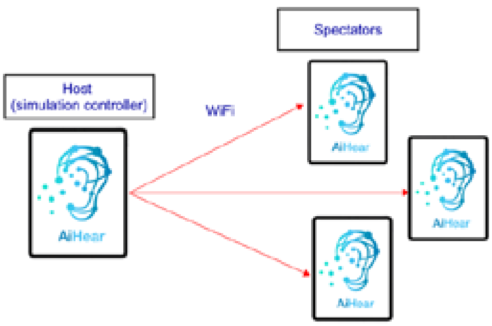

Figure 5. Multi-user mode.

The host user has control over the source audio file, sound source type, visual model, room surface finishes for reverberation control, transmission losses and background noise.

Technical Discussion

This section provides technical discussion on a few key details which apply to the AiHear® method of auralisation.

Headphones for auralisation purposes

The use of headphones as the audio playback medium for auralisations has many advantages over the more typical "listening room" setup utilising multi-channel loudspeakers in an acoustically dead and quiet room. However, implementing headphones for AR/VR auralisations does present some additional technical challenges.

Advantages of headphones	Highly portable
	Low cost
	Good frequency response across the entire audible frequency range
	Playback sound levels are independent of the environment in which the presentation is physically being held
	Sound insulating headphones render the ambient noise levels in the physical presentation environment largely irrelevant. Auralisation of very low sound levels can be accurately presented
	Reverberation within the physical presentation environment is not a limiting factor, i.e., anechoic and free field scenarios can be auralised
Disadvantages of headphones	Head related transfer functions must be applied
	Headphone fit is important for good/repeatable frequency response
	Calibration of the headphones must be completed using an expensive binaural dummy head setup
	The user's head orientation relative to the sound source must be fed into the auralisation engine for correct binaural playback (panning)

Table 1. Key advantages and disadvantages of using headphones for auralisation purposes.


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Modelling of reverberant and direct sound components

Although flexible enough to use in range of applications, AiHear® has been designed primarily as a tool for acoustic engineers. As such, the calculation algorithms within the application are based on the widely used classical room theory for predicting sound pressure levels inside rooms [1]:

$$L_p = L_w + 10\log_{10}\left(\frac{1}{4\pi r^2} + \frac{4}{R}\right) \qquad (1)$$

where L_p is the total sound pressure level at the receiving (user) position, in dB

L_w is the source sound power level, in dB

r is the distance from the sound source to the receiving (user) position, in metres

R is the room constant, related to the mean absorption coefficient for the room.

This equation is implemented in 1/3 octaves in AiHear®.

Equation 1 shows the formula in its most simplified form for an omnidirectional point source located inside a room. Modifications to this formula are applied in AiHear® for different presentation configurations, e.g., line sources, sources outside the receiving room, etc. Further corrections are applied for early reflections which are also included in the AiHear® model.

In equation 1, the direct and reverberant sound components are handled separately and then combined, and AiHear® replicates this approach. Using this widely understood approach enables engineering users to easily match the in-app calculated sound levels and reverberation times to their desktop design calculations.

Air absorption

During development and testing of AiHear®, inclusion of sound absorption due to air (αair) was found to be necessary to improve the realism of the auralisations. This was most important at frequencies from around 4kHz through to 20kHz, where the air absorption becomes rapidly greater with frequency [2].

Realtime 1/3 octave band filters

One key technical challenge of developing AiHear® was creating 1/3 octave band filters which sounded natural (free of audible artifacts), were acoustically accurate with good roll-off characteristics, and which could be implemented in real-time on consumer-grade portable devices across the full audible spectrum (30 bands simultaneously). Through trialling of many different filter configurations (FIR and IIR), an acceptable balance between accuracy and processing speed was struck and implemented in the application.

Pre-recorded audio signals

Users can upload their own audio files to AiHear®, however, the quality of these audio files is of critical im-portance to the accuracy of the auralisation:

- The application only accepts audio files which are mono, 44.1kHz .WAV format
- The spectral balance of the audio files from 20Hz to 20kHz must match the intended sound source spectrum.
- The recordings should be anechoic/free-field or as close to this as possible. Reflections recorded in the source audio file will interfere with and muddy the reverberated sound playback.
- Time varying components of the audio signal must be carefully considered in the context of the auralised scenario. Generally, it is preferable to use audio files which do not vary in level over time. For example, if simulating a single motorbike pass-by as a moving point source in AiHear®, the source audio file should consist of a constant level recording of a motorbike. Using such a file, the movement of the motorbike, as well its received sound level, will be varied accordingly and accurately within AiHear® based on its distance from the receiver location.

Background noise

One highly critical aspect in most auralisations is the background noise which is present during the simulation. This background noise provides audio masking which can either enhance the simulation's accuracy (if properly controlled) or significantly degrade it (if too high or too low). It may be obvious that if the background noise is too high, it will mask important sounds during the auralisation. Similarly, if the background noise is too low, auralised sounds which would normally be masked can appear more prominent than they should be. Therefore, it is important that the background noise is controlled and played back at the correct level.

In the case of auralisations using headphones (assuming the headphones provide a good degree of sound insulation), it is most likely that when the user puts on the headphones, the background noise will drop below where it should be for the purposes of the auralisation. However, having background noise that is too low is preferable to it being too high because it is easier to artificially add in background noise than it is to take it away.

AiHear® has a feature which allows users to load in background audio files and have them played into the auralisation at user-defined sound power levels. The controls for background noise playback are independent of the primary sound source audio controls so the two sources can be overlaid in the auralisation to provide the desired masking effect.

System calibration

Figure 6. AiHear® system calibration being undertaken on a head (and torso)simulator.

The AiHear® application was successively calibrated using a binaural head (and torso) simulator (HATS) for different portable devices and headphone models. The calibration values were saved as 1/3-octave filters within the application, allowing users to select the appropriate filters depending on their specific hardware setup. This calibration process meant the audio playback accuracy of the system could be verified. Calibration was completed with the listener facing directly towards the sound source, i.e., with a neutral-position head transfer function applied.

Final Remarks

Auralisation is a valuable tool for previewing how acoustic environments will sound before they are constructed. It holds a unique place in explaining how sound behaves to non-acoustic experts by enabling them to subjectively experience a hypothetical audio environment with their own ears. This can help stakeholders themselves make informed decisions on sound, without having to interpret complicated reports and the need for an in-depth knowledge of the decibel, reverberation times, background masking, etc.

Coupled with AR/VR visuals, auralisation can be a very immersive experience, and the power of this type of multi-sensory simulation holds a strong place in the future of acoustics.

AiHear® is an innovative front-runner in immersive AR/VR auralisation technology, demonstrating that accurate auralisations can be achieved using low-cost portable consumer-grade devices. We look forward to sharing this technology with all those who has an interest in acoustics, or those who simply have ears.

Acknowledgements

The authors would like to acknowledge the global Tetra Tech team for their ongoing support in the development of AiHear®, and in particular, Arif Zaher, for his tireless enthusiasm for innovation and the development.

We would also like to thank the University of Auckland's Acoustic Testing Service for allowing us to complete the system calibration testing in their laboratory, and Autex Insulation for their interest in, and valuable feed-back on the application.

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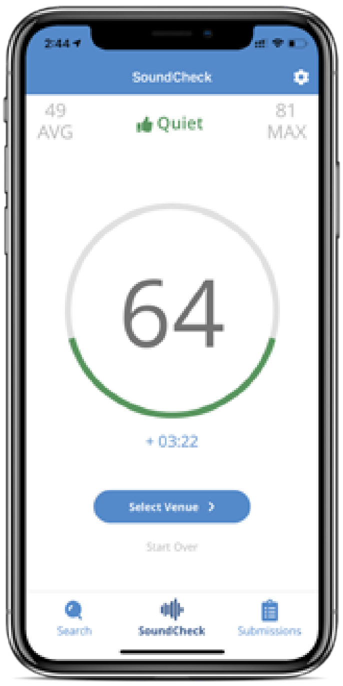

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A novel acoustic design project for final year engineering student

Andrew Hall [1], Vladislav Sorokin [1], George Dodd [1] and Gian Schmid [1]

[1] Acoustics Research Centre, University of Auckland

a.hall@auckland.ac.nz

Abstract

This paper outlines the planning and operation of a design course for mechanical engineering students in the field of acoustics at the University of Auckland. The project aimed to introduce fundamental concepts of product design, theoretical and experimental acoustics, and fabrication techniques to students in an interesting and educational manner. Students were tasked to produce a duct jacket design that would allow free airflow, but which would attenuate sound carried within that airflow. The dimensions of the duct were specified both in maximum height and outer diameter, with a minimum cross-section dimension of free space for airflow. Two fundamental sound mitigation techniques were highlighted, Helmholtz resonators and perforated panels. The volume between the duct and the outer structure provided space to accommodate these mitigation elements, which were to be designed to attenuate a specified sound spectrum. This spectrum comprised two narrow-band components (440Hz and 2kHz), and a wide-band component being a smoothed version of the human voice and centred at 1kHz. The resulting course was run over a 6-week period. Students produced a broad array of implementations which were designed in CAD, modelled analytically and numerically, and fabricated using 3D Printers and laser cutting techniques. The designs were tested using two techniques, a sound pressure level measurement, and measurement in an impedance tube of the tuning of the narrowband suppressors. The resulting designs varied in performance, with the best performing design having a delta dBA reduction of 21.5dB and impressive Helmholtz resonator performance. Initial feedback indicates that student satisfaction with the course was high.

Introduction

The aim of this course was to introduce students to the field of acoustics in an encouraging, exciting, and understandable method. The students were taught the fundamental principles of sound insulation and absorption during the process of designing and fabricating (COVID-19 restrictions permitting) a sound suppressor to operate under certain specified constraints as might be found in real life. Students were also required to work as a team and practise design skills such as acoustic modelling, computer aided design, and manufacturing techniques such as additive manufacturing and laser cutting. The course was run in 2022 for the first time which coincided with the COVID-19 lockdowns. As a result students were not required to manufacturer their end design although they were encouraged to do so.

Project Details

The student brief for the 6 week design course, was to produce a device to fit around a straight duct which would allow free airflow but which would attenuate sound carried with that airflow.

The duct was specified as being linear and having a length of 200mm and a cross-section of arbitrary shape, but with minimum dimension of the cross-section in any direction not less than 75 mm. The duct was to be shrouded within a rigid outer structure also 200mm in length but having a cross-section with maximum dimension in any direction of not more than 250mm.

The volume between the duct and the outer structure provided space to accommodate Helmholtz resonators and a perforated plate absorber which were to be designed to attenuate a specified sound spectrum.

This comprised two narrow-band components one at 440 Hz and the other at 2kHz plus a wide-band component being a smoothed version of the human voice and centred at 1 kHz. A recording of this was provided and a small loudspeaker Bluetooth-connected loudspeaker (AV Note) for reproducing it. The loudspeaker was of a size that would fit inside the duct.

The internal duct was intended to be perforated to permit necks of suitable dimensions for Helmholtz resonators tuned respectively to 440 and 2000 Hz and also suitable size perforations for a perforated plate absorber giving peak absorption at 1 kHz with a bandwidth to matching that of the speech spectrum component.

STUDENTS WERE ASKED TOO:

- Determine the frequency ranges of the provided sound signal to be focused on when designing the suppressor
- Choose which of the two sound suppression techniques would be most appropriate for each of the selected frequency ranges
- Determine dimensions and geometry of Helmholtz resonators, plus the size and number of the perforations for

the wall of the inner tube that will target the corresponding frequency ranges of the provided sound signal

- Create a design that combines the determined dimensions and geometry of Helmholtz resonators and perforations in one device
- Prepare PDF drawings of the device
- Write a report summarizing the design process and your predictions of the device performance.

Students attended a series of lectures and tutorials introducing them to the concepts required to undertake the design of the acoustic device.

STUDENT DESIGN PROCESS

When designing the device and making predictions of its performance, students were allowed to use any of the modelling approaches described below:

- Analytical study via using equations and guidelines provided during the lectures
- Numerical (Comsol) modelling of individual parts of the device
- Physical testing of individual parts of the device (if permitted)

The aim of the modelling was to obtain results that illustrated that the device (or its individual components) would provide suppression in the selected frequency ranges.

STUDENT DEVICES

Figure 1. Examples of student devices

Although students were not required to complete production of the finished device they were strongly encouraged to do so for enjoyment and verification of their design prediction. Seven of the twelve groups submitted their designs for testing, which are shown in Figure 1. The examples show an acoustic jacket around a central open cylinder. This jacket effectively creates an acoustic filter using the Helmholtz resonator and perforated plate techniques.

Experimental Assessment

Two experimental techniques were used to assess the performance of the student devices 1) a measurement of the insertion loss when placed over the loudspeaker, and 2) a sweep frequency response when fitted into an impedance tube. The latter was for checking the tuning and coverage bandwidths.

ANECHOIC CHAMBER MEASUREMENTS

When the finished designs were submitted they were measured in the anechoic chamber of the ARC for their insertion loss compared to that of a plain unperforated tube.

An omnidirectional microphone (UMIK 1) was situated outside of the near field of the tube outlet but well within the range of true anechoic performance of the chamber, Figure 2.

Figure 2. Measurement in the anechoic chamber

The spectrum radiated through the device was measured by taking 32 averages with the REW software and compared with the spectrum measured from the plain tube.

The data obtained were –

1. The suppression of the 440 Hz and 2kHz tones
2. The bandwidth of the suppressed narrow band peaks at whatever frequency they occurred in order to allow for imprecise tuning of the Helmholtz resonators
3. The amount by which the peak of the wideband spectrum component was attenuated.
4. The 3 dB and 10 dB bandwidths of the wideband peak in the response.

Group	L @ 442.7	10 dB Δf	L @ 2014	10 dB Δf	L @ 1000	3 dB Δf	10 dB Df	dB(A)	D dB(A)
Bare Tube	81.3	4	60.6	10	57	572	819	82.9	
Gp 7 cake	L @ 443.6	3 dB Δf	L @ 2017	3 dB Δf	L @ 1218	3 dB Df	10 dB Df		
	46	4.2	44.7	4.4	49.2	209	576	70.9	12
Gp 9	L @ 441		L @ 2005		L @ 1593				
	60.9	2.5	48.8	8.6	41.3	98	491	64.7	18.2
Gp 10	L @ 440.3		L @ 2000.8		L @ 1268				
	42.3	2.3	60.2	2.1	50.6	255	850	73.2	9.7
Gp 2 fully open HR holes									
	L @ 443.5		L @ 2016.2		L @ 1127				
	36.4	2.5	38.1	9.1	39.1	144	820	61.5	21.4
Gp 2 HR holes as fully closed as possible									
	L @ 443.4		L @ 2016.7		L @ 1114				
	36.4	1.5	36.4	8.9	38.1	195	837	61.4	21.5
Gp 4	L @ 443.2		L @ 2015.7		L @ 1433				
	57.3	2.1	59.5	4.7	42.7	236	1004	68.4	14.5
Gp 8	L @ 443.8		L @ 2018.1		L @ 1329				
	59.2	2.1	60.2	4.7	45.4	113	344	68.5	14.4
Gp 6	L @ 443.4		L @ 2015.9		L @ 1330				
	40.9	2.5	56.5	8.9	53.4	152	447	74.5	8.4

Table 1. Results from anechoic testing

5. The overall reduction of the spectrum in terms of the A-weighted Sound Pressure Level.

These were collated and the designs were ranked according to the maximum attenuation achieved in the 440hz, 1 kHz and 2kHz regions and dB(A).

The results (Table 1) for the devices showed a general correlation between the different measures and overall reductions ranging between 8.4 and 21.4 dB(A).

IMPEDANCE TUBE MEASUREMENTS

The impedance tube method was also used to prove the TL performance of the submitted devices under normal incidence excitation. The impedance tube (Figure 3) conformed to the European Standard ISO 10534-2:2001(E). Samples were mounted between two flanges of the impedance tube.

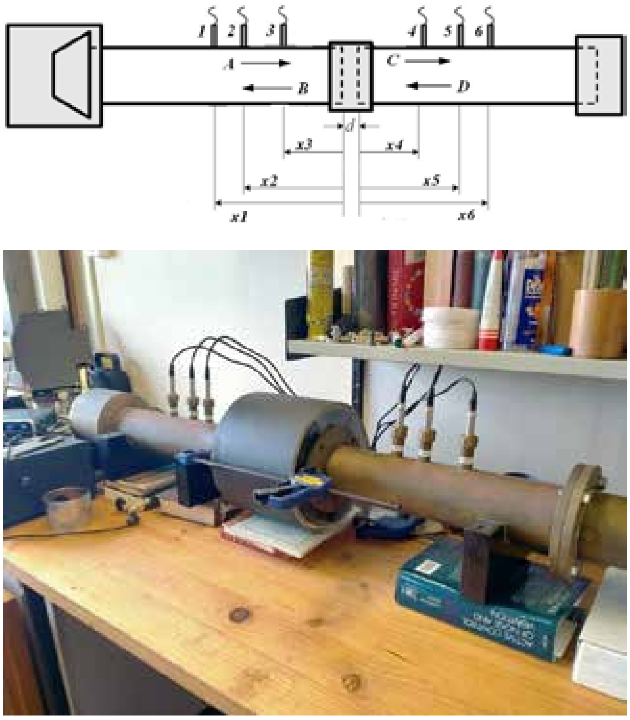


Figure 3. Impedance tube measurements

A loudspeaker generated plane wave sound that propagates down the source side of the tube. Part of the signal is transmitted through the sample which is measured in the receiving tube.

Microphones 1, 2 and 3 are used to find the source side complex wave constants A and B, whilst microphones 4, 5 and 6 are used to find the receiving-side complex wave constants C and D.

The transmission coefficient (t) may be found from:

$$t = AC - BDAA - DD,$$

And the sound transmission loss can be calculated from:

$$TL = 20\,log\left(\frac{1}{|t|}\right)$$

Figure 4 shows the transmission losses achieved.

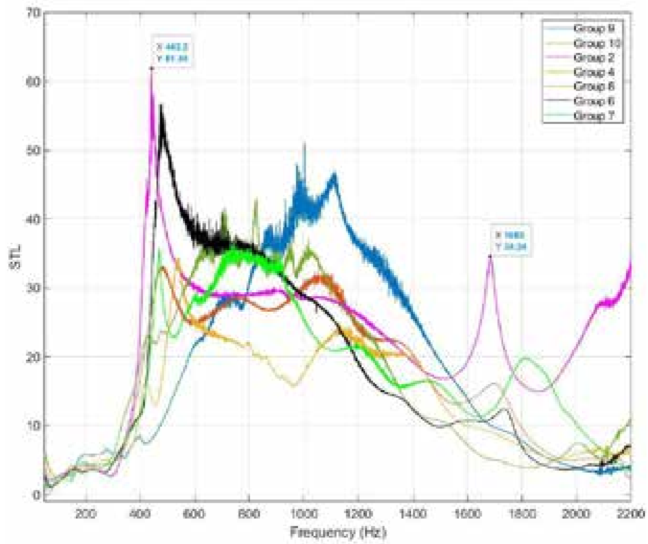


Figure 4. Results of the impedance tube measurements

The Best Device

Based on the conducted measurements using the described techniques, design of Group 2 had the best, with an exceptional performance of Helmholtz resonators and very good performance of perforated plate absorber. They also had a creative "hamburger" style of the device (Figure 5). This design incorporated two sets of adjustable Helmholtz resonators (enabling refinement of the tuning) in addition to well-tuned perforations.

Figure 5. Device of Group 2, the winners

Groups 6 and 7, Figure 6, also produced a design with Helmholtz resonators and a properly working perforated plate absorber.

Group 9, Figure 7, managed to design the best performing perforated plate absorber that showed extremely high transmission loss performance in the 1000Hz broadband region. This, however, was achieved at the expense of not having Helmholtz resonators targeting the 2000 Hz peak. Although their lower frequency (440 Hz) Helmholtz resonators were slightly mistuned, their design still performed second best in terms of subjective sound attenuation, i.e. attenuation as perceived by typical human ear.

Figure 6. Devices of groups 6 and 7.

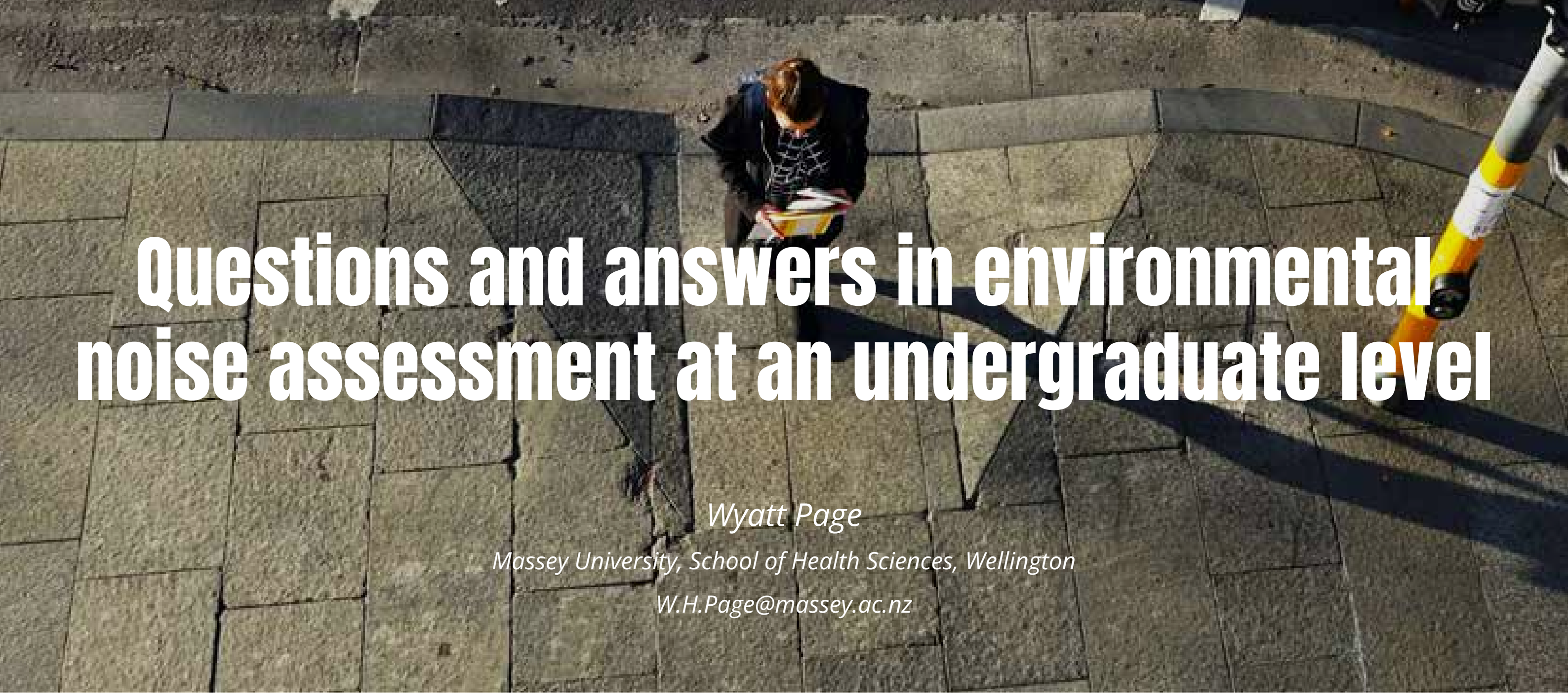
Conclusion

Overall this project was a success and the creativity and variation in design of the devices by the students was commendable given how new the field of acoustics was to the students.

The results prove that it is possible to create a passive suppression system for the outlets of ducting systems and small music devices. We would like to do more research into the possibilities of using this technique for reducing the transmission of noise through trickle ventilation systems around windows.

Acknowledgements

The team would like to acknowledge Frances Fulton for her work during the project helping with tutoring and marking.



Questions and answers in environmental noise assessment at an undergraduate level

Wyatt Page

Massey University, School of Health Sciences, Wellington

W.H.Page@massey.ac.nz

Abstract

There has been a compulsory practical hands-on 300-level noise course in the environmental health (EH) programs at Massey University for over twenty years. Unlike most EH programmes in Australasia, Massey still considers environmental noise assessment as a key skill for trainee environmental health officers. Over the past 13 years that the author has been involved in this course, students have asked many questions, most have been easy to answer, while others have proved more challenging. This paper is a short collection of the more challenging questions and their answers, which should be of interest to noise assessment practitioners.

Introduction

The environmental health (EH) programmes at Massey University have included a compulsory course on noise for over twenty years. The 300-level course, 214.316 Biophysical Effects of Noise and Vibration [1] is practical and hands-on, training environmental health officers (EHOs), and more recently occupational health and safety officers (OHSOs), to carry out noise assessment to professional standards.

Over the past 13 years that the author has been involved in this course, students have asked many questions, most have been easy to answer while others have proved more challenging. This paper is a short compilation of the more challenging questions and their answers. Consistent with the practical nature of the course, sound samples were collected and analysed to assist with answering some of these questions.

Setup

Taking recordings

All the recordings for the experiments in this paper were taken using a Zoom H4n audio recorder. It features up to four tracks simultaneously at up to 96 kHz sampling rate at 24-bit resolution. It has built-in X-Y microphones plus two XLR microphone inputs supporting phantom power. A matched pair of BSWA SM4201 [2] omnidirectional, phantom-powered microphones were used on the XLR inputs. These are precision microphones with Class 1 sound level meter equivalent performance. The inputs were configured to 48 kHz sampling rate at 24-bit resolution, with all filters and limiters turned off to preserve linearity and frequency response. All recordings were saved to the recorder's SD card as uncompressed .wav files (Waveform Audio Format).

Before and after taking recordings, 10 seconds of calibration tone was recorded for each microphone using a standard field calibration at 94 dB and 1 kHz.

Processing recordings

Some time was spent ponding on what software to use to process the sound recordings. My default for custom work usually means MATLAB. I have over 25 years of experience with it, and it provides an incredibly feature-rich technical computing environment. The 'Audio Toolbox' includes a wide range of features, including an implementation of the Sound Level Meter (SLM) object. However, the noise course at Massey is not part of an engineering programme and so students do not have the experience with or access to MATLAB and its associated toolboxes.

Prebuilt applications were also considered. Many sound measurement equipment manufacturers have software available that enables the post-processing of sound recording files. These applications are usually standards-based, but because of their propriety nature and the need to purchase a license, I decided to look at other options.

In the open-source space three are many applications that process audio, but none that I could find that implemented sound level meter functionality, and in particular, the ability to calculate different noise descriptors in a flexible way.

While researching software options for another project, I decided it would be good to learn how to programme in Python [3], the world's most popular programming language, according to the most recent IEEE Spectrum survey [4]. In exploring the libraries and packages available to Python, I had a look at what was available in the acoustics space. Not surprisingly, there is an acoustics library that covers a wide range of areas, from basic decibel quantity manipulations to ambisonics, the Doppler effect, filtering, and so forth. It also includes the implementation of several international standards, in particular:

- IEC 61260 2014 - Performance requirements for band-pass filters

- IEC 61672 2013 - Performance specifications for sound measuring instruments
- ISO 1683 2015 - Specifies reference values used in acoustics
- ISO 1996-1:2003 and ISO 1996-2:2007 - Description, measurement, and assessment of environmental noise
- ISO 9613-1:1993 - Calculation of the absorption of sound by the atmosphere
- ISO/TR 25417 2007 - Definitions of basic quantities and terms

The implementation of IEC 61672 2013 [5] provides all the functions needed to process sound recordings just like a modern sound level meter. It leverages the functionality of the Python signal processing library and is available for anybody to use.

Anaconda Navigator [6] was chosen as the tool to install and manage a self-contained isolated Python environment that did not need Administrator rights. This tool also allows the easy installation and maintenance of packages (libraries) without modifying the system's Python installation. Finally, the Jupyter notebook [7], a web-based, interactive computing notebook environment for Python, was used to interactively develop the processing code. It allows you to edit and run readable documents while describing the data analysis as you go along. The whole environment is flexible, easy to use, and well-suited to the technical abilities of many of our students taking the noise course.

Questions and Answers

Microphone inclination

Sound level meters use omnidirectional microphones, meaning that ideally, they measure sound equally well about a hemisphere, in the direction that the microphone is pointed. In practice, real microphones start to become more directional above about 3 kHz. Figure 1 shows the free-field correction curves for a modern ½ inch microphone capsule, a B&K Type 4176 [8], designed for Class 1 SLMs. These corrections represent the increase of sound pressure caused by the diffraction of the sound waves around the microphone. At 3 kHz, there is about a 1 dB difference between the on-axis (0o) response and the response for angles above 60o. This difference increases significantly with frequency.

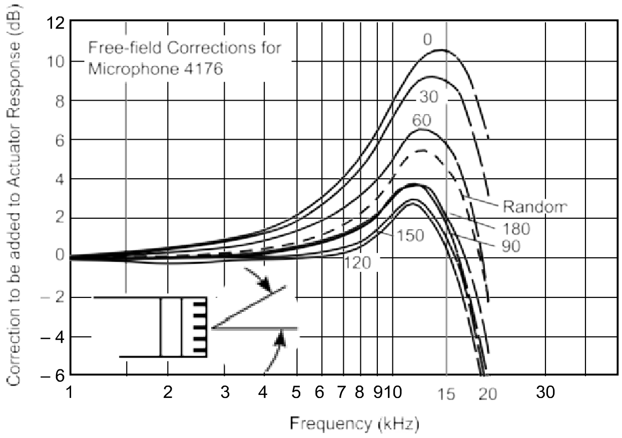


Figure 1. Free-field correction curves for various angles of incidence for a B&K Type 4176 capsule

When I learned to take environmental noise measurements, the wisdom and practice passed on to me was that the microphone of the SLM should be angled upwards at about 30 degrees. A student new to the noise course asked why? The simple answer is that the nearest reflecting surface is usually the ground (1.2-1.5 m away when following NZS 6801:2005 [9]) and to get a fair measurement of the sound pressure, pointing the SLM upwards helps reduce ground reflection and gets a better estimate of the sound level. However, there is no mention of this in NZS 6801:2005 and reports from professionals often include pictures with the SLM placement and setup, typically showing the microphone parallel to the ground. So, how much of an effect does inclining the microphone upwards have?

Setup

- A matched pair of Class 1 microphones on a custom mounting plate on a standard tripod, one horizontal and the other at 30 degrees upwards (see Figure 1).
- The tripod height was set so that the centre of the microphones is at 1.35 m, the mid-point of the preferred range (1.2-1.5 m) in NZS 6801.
- Microphones are recorded simultaneously on the Zoom H4n audio recorder. Channel 1 (left) for the 30o upwards microphone and channel 2 (right) for the 0o (horizontal) one.
- Recordings were taken of road traffic noise on Adelaide Road (Wellington) outside McAlister Park.
- Ten seconds of calibration tone were recorded before and after taking the recordings.

Figure 2. Microphones setup on a custom mounting plate on a tripod with windshields in place

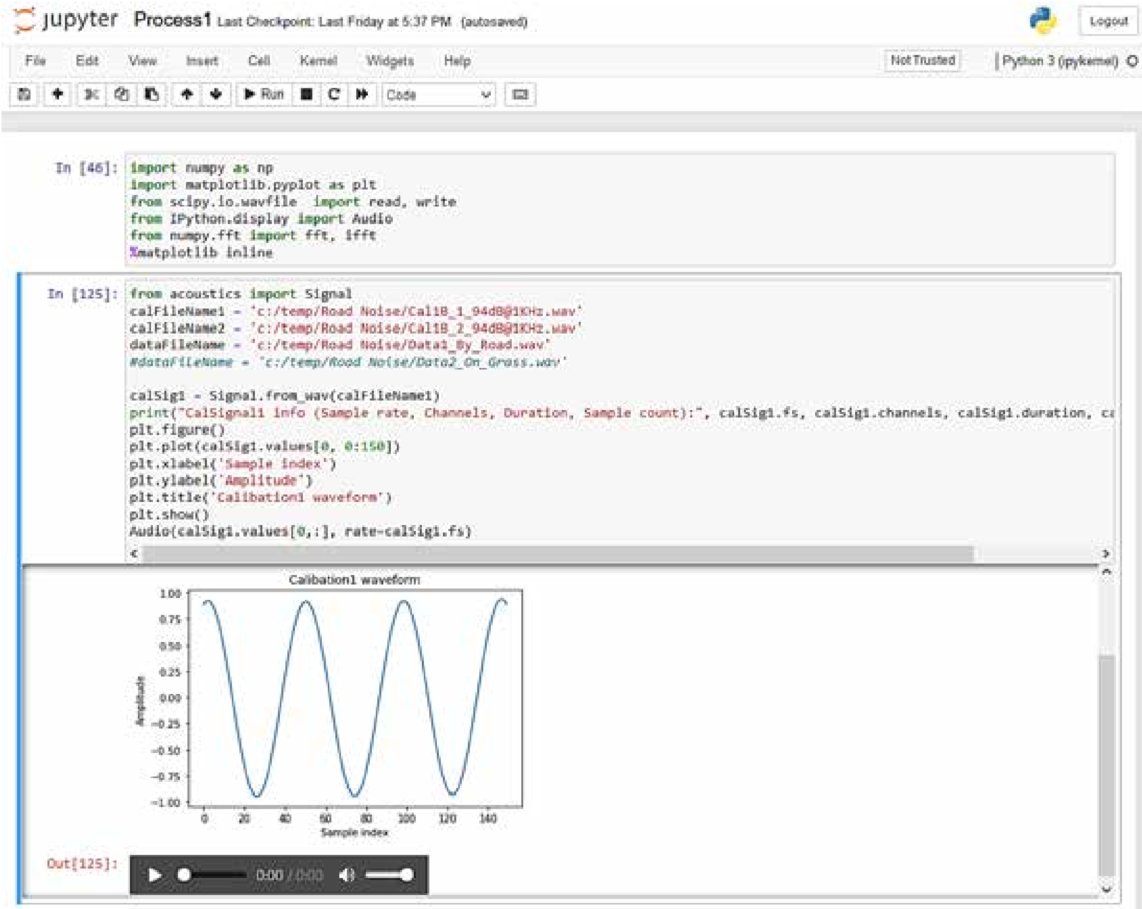


Figure 3. Snapshot of the top of the Jupyet notebook and the Python code to process the recordings

Road traffic noise was chosen because it is often used as the practice noise source when environmental health students carry out practical fieldwork. When the traffic flow is relatively continuous, the sound propagation can be modelled as a cylindrical source. Two main components make up traffic noise: tyre noise and engine noise. In a 50 kph zone, for most vehicles (unless accelerating), tyre noise dominates, occurring low down at the road-tyre interface, whereas engine noise is generally higher up. Considering this, one would expect that a horizontally inclined microphone will measure a slightly higher sound pressure than one inclined upwards as there will be a higher contribution from the ground reflection.

Results processing

Python scripts were developed to process the audio recordings to calculate the standard environmental noise descriptors, L_{Aeq} and L_{AFmax} at 1-second intervals.

Figure 3 shows the top of the Jupyter notebook showing the start of the Python code to process the recordings. Part of the channel 1 signal (the 1 kHz calibration tone) is shown and below it is the audio control that allows the signal to be played. The steps in the notebook code are:

1. Read the before (taking recordings) calibration wav files
2. Calculate calibration factors for each microphone
3. Check calibration factors by them applying to the calibration recordings
4. Read the recordings wav file(s)
5. Applying A-frequency weighing to the recordings using a zero-phase filter
6. Apply the calibration factors
7. Calculate the equivalent time-average levels (LAeq,1s)
8. Calculate the maximum F-time weighted levels (LAFmax) every second
9. Calculate the difference signal between the two channels for the two noise descriptors. Display the results and calculate summary statics: min, mean, median, and max
10. Read the after (taking recordings) calibration wav files, calculate calibration factors, and compare them to the before values.

Results

Table 1 shows the results of processing the data. Two slightly different locations were used for the tripod placement. The first was on the hard footpath close to the road, while the second was further back (about 3.5 m) on the grassed area of a park. In both cases, the tripod was well away from any other surfaces.

The first four rows of table 1 relate to the microphone calibration based on the 'before' recording of the calibrator tone. As the microphones are a matched pair, the calibration factors are very similar, with only a 0.15 dB difference at 1 kHz. When the calibration factors are applied to the tone calibrator recordings, both levels are 94 dB to three decimal places, confirming the calibration.

The last row of table 1 shows the calibration factors calculated from the 'after' recording of the calibrator tone. The values have changed only very slightly, and the difference has reduced to 0.09 dB, a 0.06 dB change from the before values.

Looking at the mean and median difference for the two descriptors at both measurement locations, they are all negative, indicating that the sound pressure is slightly higher for the horizontal microphone.

Source	Quantity	Value (dB)
Before – tone	Cal. Factor: channel 1 (30°)	-3.666
	Cal. factor – channel 2 (0°)	-3.515
Before – tone	Cal. level – channel 1 (30°)	93.998
	Cal. level – channel 2 (0°)	94.001
Traffic noise from footpath	$L_{Aeq,1sec}$ difference: Mean	-0.23
	Median	-0.26
	L_{AFmax} difference: Mean	-0.23
	Median	-0.24
Traffic noise from grassed area	$L_{Aeq,1sec}$ difference: Mean	-0.20
	Median	-0.18
	L_{AFmax} difference: Mean	-0.19
	Median	-0.19
After – tone	Cal. factor – channel 1 (30°)	-3.681
	Cal. factor – channel 2 (0°)	-3.590

Table 1. Microphone inclination results

Measuring from the hard surface of the footpath, the mean and median differences for $L_{Aeq,1sec}$ are -0.23 and -0.26 dB respectively. For L_{AFmax} the mean is the same, but the median is slightly lower. From the grassed area, the difference decreases very slightly (by 0.03 to 0.08 dB) for both descriptors. The reduction is less than expected given that the grassed area was soft and damp underfoot.

Overall, the effect of inclining the microphone upwards results in about 0.2 dB reduction in the sound pressure level for both noise descriptors for the traffic noise when measured from the hard surface of the footpath.

Further analysis

To try and better understand the reason for the difference in measured sound pressure level, additional analysis was carried out in Python to look at the third-octave spectrum of the signals. Figure 4 shows the (calibrated) third-octave spectrum of the horizontal (0°) microphone signal for the recording taken on the footpath. Both Z and A (frequency) weighted spectrums are shown for comparison. The Z-weighted spectrum below 1 kHz shows a small drop to 400 Hz before a steady increase reaching a maximum in the 50 and 63 Hz bands. Above 1 kHz, the spectrum drops at about 8 dB per octave to the 3.15 kHz band and then at about 12 dB per octave to the 16 kHz band at which point the noise floor of the measurement setup would have been reached. As expected, A-weighting has the largest effect on frequencies below 1 kHz with the spectrum decreasing at about 8 dB per octave.

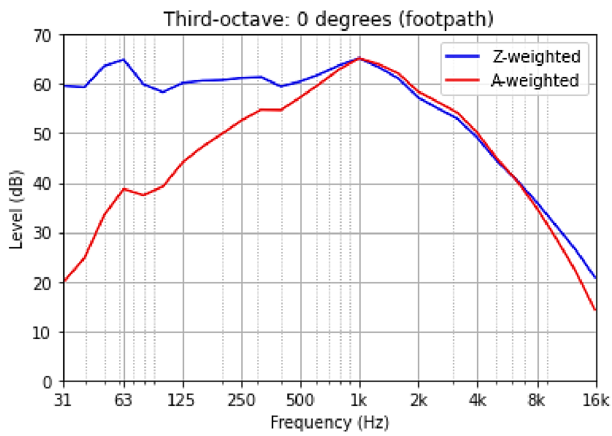


Figure 4. Third-octave spectrum of the 0° inclined microphone for trafftic noise taken from the footpath

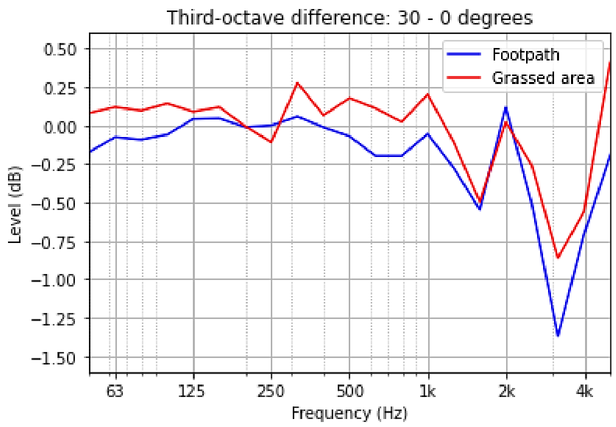


Figure 5. Third-octave spectrum difference between microphones for the traffic noise

The first thing to notice is that there is a lot of variation in the spectral difference between the microphone signals across the frequency bands. It is much more complex than I was expecting. The difference for the footpath location is consistently negative, as expected, but less negative below 1 kHz than predicted based on the results of table 1. For the grass location, the difference is close to zero up to 1 kHz (except at 250 Hz), significantly less than predicted based on the results of table 1. Interestingly, the spectrum difference markedly decreases (-0.75 dB) at the 1.6 kHz band before returning close to zero at 2 kHz, then rapidly decreasing (-1 to -1.5 dB) to 3.125 kHz before swinging back to close to zero at 4 kHz. This indicates that ground reflection in these upper bands is highly frequency dependent. The mean values below the 1 kHz band for both measurement locations are less than the mean values in Table 1. The reason for this is the substantial decrease at 1.6 kHz and 3.125 kHz which overall significantly contributes to the A-weighted noise descriptor values. Below 1 kHz for the grassed area location (with the soft damp ground) the difference between the two microphones is close to zero, which is much more in line with expectations. The exception is 250 and 316 Hz bands, where there is a distinct down then up change.

As the last analysis, I looked at my observational notes to identify any periods of low or no nearby traffic. I then opened the original audio files in Audacity [10] (an open-source, multi-platform, audio editor and recorder application) and extracted the very quiet sections into a single file. No quiet sections were identified from the grassed area recording, but 15 seconds in total were identified from the footpath recording. Listening to the wav file, there was still distant traffic sound but none of it was near the microphones. The previously developed Python scripts were used to process the edited recording.

Source	Quantity	Value (dB)
Traffic noise from footpath	$L_{Aeq,1sec}$ difference:	
	Mean	-0.10
	Median	-0.04
	L_{AFmax} difference:	
	Mean	-0.10
	Median	-0.03

Table 2. Microphone inclination results for quiet sections

Table 2 shows the difference values of the two descriptors from the footpath location based on the 'quiet' sections of the recording. For both descriptors, the mean and median differences are substantially reduced compared to the value from the whole recording (see Table 1) but are still negative. This is consistent with the expectation the local ground reflection effects are less significant when the traffic noise is well distant from the microphones.

Further experiments

The peer reviewers of the draft version of this paper had a range of suggestions for further experiments, these included:

1. Experiments at different tripod heights between 1.2 to 1.5m.
2. Controlled experiments with a known wide-band source in an anechoic chamber at different microphone inclinations.
3. Microphones were used without an SLM body attached. What effect might this have had on the measurements?

The first idea is a natural follow-on from the microphone inclination experiment and will be explored in the next section. The other ideas are also good and may be explored in the future but are outside the scope of this paper.

Tripod height

Section 6.1.2 of NZS 6801:2008 states that whenever practical, measurements should *"...carried out at least 3.5 m from any reflecting surface other than the ground, and 1.2 to 1.5 m above the immediate ground level".*

So, what effect is there on the measurements, if taken at 1.5 m compared to 1.2 m off the ground? On face value, the lower height will have a higher contribution from ground reflection, whereas the higher height is likely to have a better direct path (line of sight) signal.

To test whether this is the case, traffic noise data were collected at the same site as previously. A custom mounting system was used with microphones horizontal. The channel 1 microphone was at 1.5 m and channel 2 at 1.2 m from the ground (see Figure 6).

Figure 6. Microphone setup with the top one at 1.5m and the lower at 1.2m off the ground

Source	Quantity		Value (dB)
Traffic noise from footpath	$L_{Aeq,1\ sec}$ difference:	Mean	+0.64
		Median	+0.60
	L_{AFmax} difference:	Mean	+0.74
		Median	+0.72
Traffic noise from grassed area	$L_{Aeq,1\ sec}$ difference:	Mean	+0.57
		Median	+0.57
	L_{AFmax} difference:	Mean	+0.56
		Median	+0.57

Table 3. Microphone heights results, 1.5m - 1.2m height

Table 3 shows the mean and median difference between the two microphones (1.5 m – 1.2 m height) for the two descriptors, at each location. The first thing to notice is that all values are positive. Thus, the sound pressure picked up by the more elevated microphone is overall higher than for the lower one. This implies that for this case, better line-of-site (greater direct sound) is more significant than the higher (ground) reflected sound contribution likely to be experienced by the lower microphone. This effect persists from the grassed area but is reduced by 0.07 to 0.12 dB compared to the footpath location.

Third-octave spectrum differences were calculated between the microphones at both locations and the results are shown in Figure 7.

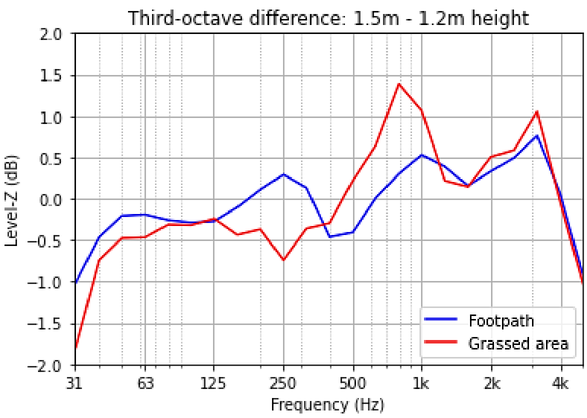


Figure 7. Third-octave spectrum difference between 1.5 and 1.2m microphones for the traffic noise

Much like we saw for the inclination experiment, the spectrum difference at both locations is highly frequency-dependent. Below 500 Hz, the difference is slightly negative (except at 250 Hz for the footpath location), implying a higher sound pressure is being received by the lower microphone. But in the frequency bands from 630 Hz to 4 kHz, the difference is positive, implying a higher sound pressure is being received by the upper microphone. As this part of the spectrum is relatively unaffected by A-weighting, it has a more significant contribution to the two noise descriptors, which is why they are all positive in table 3.

Summary – Inclination and height

Based on the results of the microphone inclination experiment at a mid-range tripod height (1.35 m), if the microphone is inclined upwards, measured values should be less sensitive to tripod height. However, based on the tripod height experiment, using the highest allowable height, more direct-path sound is collected at this height, resulting in an increase of about 0.6 dB for both descriptors compared to the lowest allowable height.

One reviewer of the draft paper asked, “Isn’t the point of making the measurement at 1.2-1.5 m above ground level to get a representative measurement that includes the ground effect?”. Yes, the aim is to collect representative measurements and this will include the contribution from ground reflections as it is usually the nearest surface. However, the experiments show that the tripod height effects on the measured sound pressure level are more significant than a slight inclination of the microphone upwards.

Finally, an additional advantage of using the higher height of 1.5 m is that it is approximately at adult ear height, so is more likely to be representative of the sound pressure experienced at the ear.

Measuring Lmax

Noise measurement standards use a mixture of conventional (exponentially time-weighted, frequency-weighted) descriptors (metrics, or measurement quantities) and integrating (-averaging, frequency-weighted) descriptors. It takes some time for students to get their heads around what each of these different noise descriptors measure and their purpose.

The standard NZS6801:2008 defines Lmax as the maximum A-frequency weighted, F-time-weighted sound pressure level, L_{AFmax}. It goes on to say that for the purpose of the standard, if Lmax is derived from measured short-LEQ values of 100-125 milliseconds duration, it shall be taken as equivalent to Lmax derived from F-time weighted measurements. In the standard, a short-LEQ value is $L_{Aeq(t)}$ for t ≤ 1 second. So, putting this together implies that:

$$\text{Lmax} = L_{AFmax}(t) = \max((L_{Aeq,\ 125\ ms}),\ t) \quad (1)$$

So, a question I have often been asked by students is, are these truly equivalent?

The key difference in terms of the mathematical description of a conventional noise descriptor using time-weighing and one using a time-average equivalent level, is that the first uses exponential integration with a time-constant, while the latter uses simple linear integration over an integration period.

In IEC 61672-1:2013, it says that A-weighted and F-time-weighted sound level LAF(t) at observation time t can be represented by

$$L_{AF(t)} = 10 \log\left(\frac{1}{\tau_F}\int_{-\infty}^{t} e^{-(t-\xi)/\tau_F}\, p_A^2(\xi)\, d\xi \,/ {p_0}^2\right) \quad (2)$$

Where:

τ_F is the exponential time constant in seconds for F-time weighting;

ξ is a dummy variable of time integration from some time in the past, as indicated by -∞ for the lower limit of the integral, to the time of the observation;

$p_A(\xi)$ is the A-weighted instantaneous sound pressure;

p_0 is the reference pressure of 20 μPa.

Similarly, the standard defines the time-averaged or equivalent continuous A-weighted sound level at observation time t, as:

$$L_{Aeq,t} = 10 \log\left(\frac{1}{T}\int_{t-T}^{t} p_A^2(\xi)\, d\xi \,/ {p_0}^2\right) \quad (3)$$

Where:

T is the averaging time interval (integration time);

ξ is a dummy variable of time integration over the averaging time interval ending at the time of observation t.

Looking at equations (2) and (3), the main difference is the use of the exponential weighting term, $e^{-(t-\xi)/\tau_F}$ (which is always less than 1) and that the integral for $L_{AF(t)}$ may start more than τ_F before t. Given that τ_F = 0.125 seconds, then when T is the same or similar in value, the effect of the exponential term should be small. So how small, for a real-world signal?

Setup

The measurement setup was the same as for the microphone inclination assessment experiment. The only difference was that the sound source was that of roofers installing a new corrugated iron roof. This was chosen as it was happening next door while I was trying to work from home and because it contained significant impulsive sounds, primarily from the stapling of the roofing underlay to the roof structure.

Figure 7 shows a snapshot of a section of the raw audio recording. The impulsive nature of the stapler sound is evident from the discrete pressure bursts each time a staple is driven. As for the previous inclination experiment, channel 1 (left) was the microphone inclined upwards at 30° while channel 2 (right) was for the 0° horizontal inclination.

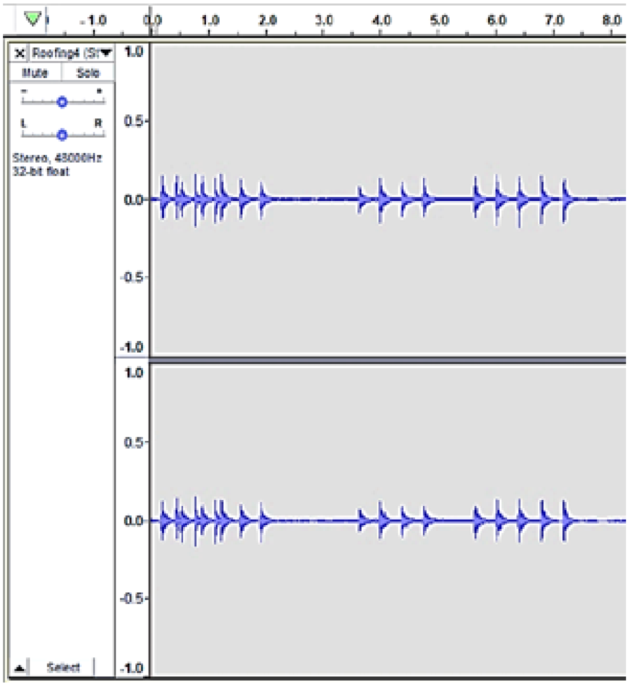


Figure 7. A section of the audio recording showing the impulsive stapling bursts

Results

Python scripts were developed to process the audio recordings to calculate LAFMAX and the maximum $L_{Aeq(0.125\ sec)}$ at 1-second intervals and then produce the difference statistics value between the two descriptors. Before and after calibration checks were also performed.

The results in table 4 show that as expected, for the impulsive stapling sound, the difference between the two ways of calculating Lmax, is very small, averaging about 0.036 dB with a maximum of 0.055 dB. $L_{AFmax(1\ sec)}$ was always higher than max($L_{Aeq,0.0125\ sec}$, 1 sec) and ever so slightly higher for the recording where the microphone was inclined upwards at 30°. This makes sense, as the stapling sound occurred about 3.5 metres off the ground and so was better captured by the upward inclined microphone.

Source	Quantity	Value (dB)
Roofing stapling noise – 30° incline	$L_{AFmax(1\ sec)}$ - max($L_{Aeq,0.125\ sec}$,1 sec) stats: Min Mean Median Max	 0.003 0.037 0.035 0.055
Roofing stapling noise – 0° incline	$L_{AFmax(1\ sec)}$ - max($L_{Aeq,0.125\ sec}$,1 sec) stats: Min Mean Median Max	 0.002 0.036 0.034 0.054

Table 4. Roofing noise Lmax descriptor difference

Pause and Back-erase

The underlying guidance provided in NZS 6801:2008 is that nominally a 15-minute sampling scheme is used, with the provision that a substantially longer period is often required for the measurement to be representative of the sound under investigation ('target sound'). One of the reasons given in section C6.3.3 is *"…pauses to exclude extraneous sound not under investigation. Examples include passing traffic, or aircraft, bird calls, and dogs barking."*. This is reiterated in section 8.5 Fluctuating Sound - "… *(excluding pauses, or periods of data exclusion)*, may be appropriate".

The companion base standard, NZS 6802:2008 [11], continues this narrative. In section C6.2.2, it states: *"The simple method allows use of coding of sound samples for subsequent processing, as well as use of back-erasure, data exclude, and pausing during measurements."*. In section B3.2.4, concerning the measurement of the residual sound, it states *"Direct measurement may require use of back-erase, pause and data exclude functions…"* to ensure extraneous short-term transient noise is not included.

At face value, the reason one might use back-erasure and pausing (a feature provided on most name-brand sound level meters), *"to exclude data that contaminates the measurement with extraneous sounds"*, seems reasonable. Back-erase is commonly implemented in one of two ways:

1. Manually pressing the back-erase button, erases the last 5 seconds (or some other small, time value) of the measurement data.
2. Manually pressing back-erase adds a timestamp 'exclude marker' to the measurement file and this maker stays on until it is manually turned off, but the measurement file is continuous.

Pausing feature implementations are more straightforward, the measurement is manually paused and then manually un-paused (continued), resulting in a discontinuous time measurement file.

So, what, I hear you say? Well, one of my students, who was doing the noise course and had just completed the compulsory course, 214.216 Environmental and Public Health Law [12], proposed the following courtroom conversation:

Defence: Mr Officer, can you confirm that your measurements of the sound under investigation were sufficient and representative?

Officer: Yes, I can.

Defence: I understand that the sound level meter that you used has a pause and back-erase feature. Did you make use of this feature?

Officer: Yes, I did, I used it to exclude measurements that I considered to be contaminated by extraneous sounds, like passing traffic and dogs barking.

Defence: How were you able to confirm that the measurements you excluded did not affect the measurement of the sound under investigation?

Officer: I can't, the measurements were not recorded.

Hopefully, you can see where this is going. If the officer is not careful, they will dig a hole that is going to be hard to get out of.

In the noise course, I recommend that they do not use the pause feature or back-erase (the last 5 seconds or so) feature. Instead, they record continuously and use their observational notes to note any extraneous noise and the approximate time of occurrence. After the recording is complete, they can then use the sound level meter software to see if the exclusion of the segment with the extraneous noise has a significant effect on the measured descriptors.

The implementation of 'back-erase' by 'exclude-marker' goes hand-in-hand with observational notes and ensures the integrity of the measurements.

Conclusion

After teaching a noise course for many years, one would have thought that all variations of pertinent questions would have been asked and satisfactorily answered. It is clear from the short collection of topics covered in this paper, that this is not the case. I have more questions that at face value seem simple to answer but will need to be explored in the future to provide a more satisfactory answer.

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Timbre - Harmonics and Overtones

Hedda Landreth

RoofLogic

When a string vibrates or an air column resonates, it produces a fundamental frequency, the lowest note we hear. However, alongside this fundamental note or pitch, a series of higher frequencies emerge. These frequencies, known as harmonics, are whole number multiples of the fundamental. In simpler terms, they are two times, three times, four times, and so on, the frequency of the fundamental.

The term "overtone" is a broad and inclusive term used to describe any resonant frequency that is higher than the fundamental frequency. This includes harmonics, but also inharmonic partials, which are often too faint to be audibly perceived.

Overtones cannot be heard independently, as the fundamental frequency always remains dominant. Therefore, they do not alter the pitch of the sound, however, they enrich the character and texture of the sound. Musical instruments each have a unique balance of overtones, imbuing every instrument with its own sonic signature.

The timbre of an instrument is determined by the spectral composition of the sound and overtones that it emits. It is what allows us to differentiate between different musical instruments or voices producing the same note at the same volume. In addition to the clear distinctions among various instruments, even those belonging to the same family can produce notably different sounds due to variations in timbre.; a cheap violin will have a very different sonic signature from a Stradivarius and of course the person playing the instrument also has a significant role in shaping the timbre.

Certain instruments emphasise overtones, amplifying their presence, while others project the fundamental more prominently. In the woodwind family, the clarinet accentuates the sound of harmonics, while the oboe emphasises the fundamental tone (which may explain why it is the instrument that sounds the A (440Hz or there abouts) for the rest of an orchestra to tune to). The size and characteristics of an instrument influence the balance of overtones it produces, in general, lower-pitched instruments tend to have more audible harmonics. For example, a cello produces more harmonics than a violin, even when they play the same pitch. Percussion instruments, such as crash cymbals, shimmer with countless overtones that are so prominent that they start to defy the very notion of pitch.

Timbre encompasses the subtle nuances, tonal variations, and harmonic richness that make a sound unique and recognisable. Timbre is what gives a gentle acoustic guitar strum a warm and intimate quality, or a piercing electric guitar solo its edgy and aggressive nature. By combining and layering different timbres, musicians have the ability to create intricate sonic landscapes. The interplay of timbres shapes the overall texture and mood of a musical piece, allowing for the exploration of emotions and the creation of musical narratives.

Moreover, timbre helps to convey specific emotions and evoke atmospheres. A sombre melody played on a solo cello elicits a different emotional response compared to the same melody played on a bright piano, saxophone or even a violin. The subtlety offered by timbre allows composers and performers to craft a sonic language that resonates deeply with the listener. Just as an artist uses different brushes to create various textures and visual effects in a painting, composers utilise timbre to add depth, colour, and expressiveness to their compositions.

"Nasal"	Referring to a loud fundamental pitch with minimal overtones.
"Rich" or "thick"	Describing a sound filled with multiple overtones.
"Muddy"	Characterising a sound where overtones overpower the fundamental pitch.
"Distorted"	Often used for compressed sound waves with amplified middle frequencies and reduced high and low frequencies
"Breathy"	Describing sounds where audible unpitched airflow is present.
"Vibrato"	Describing the fluctuation of frequencies, resulting in subtle pitch variations.

SUPER QUIZ

1. Sounds produced by the vibration of the vocal folds are said to be ....and thus are characterized as having a(n) ....

2. Sibilants (Sibilant sounds) always have their energy centred in the range........................

3. The changing pitch pattern in the spoken voice that provides expression, is called........................

4. True or False? Vowel formants change with pitch inflections?

5. Normal speech, a 1m distance, has a sound level of around ....
   A. 30-35 dB B. 40-50 dB
   C. 55-65 dB D. 75-85 dB

6. The vowel is characterized by energy peaks called .... that correspond to a specific configuration of the vocal tract.

7. **True or False?** The 'ossicles' is part of a human ear anatomy

8. The bulk of information content in speech lies in the frequency range
   A. 50 Hz to 1kHz B. 250Hz to 5kHz
   C. 300 Hz to 3 kHz

9. What is an **audiometer** used for?

10. **True of False?** 'ANCON' is a UK civil aircraft noise computer model?

11. What is **'CRTN'** an acronym for?

12. Describe to a non-expert what the 'direct sound field' is?

13. Who is the **Doppler Effect** named after?

14. What is the recorded speed of sound at 20°C at sea level?

15. What is the recorded speed of sound in space?

16. What is **'JND'** with respect to psycho-acoustics?

17. **True of False?** 'L_{Apeak} is the maximum A-weighted sound pressure level occurring withn a specified time period such as 15 minutes?

18. What is the Lombard Effect?

19. **True of False?** 'Mel' is a unit of pitch?

20. What is STITEL?

21. What is a semi-anechoic chamber or room?

22. Define the terms *R* and *R'* and where you may come across these terms in acoustics?

23. What is noise immission?

24. **True or False:** Z-weighting is zero frequency weighting?

25. What is 'specific noise source' defined as?

26. What do the following symbols mean? *P*, SWL, L_{WA}

27. What is *'structural acoustics'*?

28. What is the following equation related to?

$$\boxed{\nabla^2 u - \frac{1}{c_0^2}\frac{\partial^2 u}{\partial t^2} = 0}$$

29. The portrait below is of who and what are they known for?

30. 'Noise' is derived from a Latin word that means?

UPCOMING EVENTS

Acoustics 23, Sydney

4 December - 8 December 2023

International Convention Centre Sydney (ICC Sydney) 14 Darling Drive, Sydney, NSW, Australia

25th International Congress on Acoustics (ICA 2025)

18 May - 23 May 2025

New Orleans Marriott 555 Canal Street, New Orleans, LA, United States

26th International Congress on Acoustics (ICA 2028)

11 September - 14 September 2028

Pestana Casino Park Hotel Rua Imperatriz D. Amélia, Funchal, Portugal

XVIII Argentine Congress of Acoustics

6 December - 7 December 2023

Universidad Nacional de Quilmes Roque Saénz Peña 352, Bernal, Buenos Aires, Argentina

Forum Acusticum Euronoise 2025

23 June - 26 June 2025

FYCMA Ortega y Gasset, 201, Málaga, Spain

53th International Congress and Exposition on Noise Control Engineering (INTER-NOISE 2024)

25 August - 29 August 2024

Cité des Congrès de Nantes 5 rue de Valmy, Nantes, France

54th International Congress and Exposition on Noise Control Engineering (INTER-NOISE 2025)

4 September - 27 September 2025

WTC Events Center Av. das Nações Unidas, 12551 - Brooklin Novo, São Paulo, SP, Brazil

Note: Dates and information is subject to change. We encourage you to go directly to the source web site of each event to ensure you have the latest and most up to date information.

SOUNDPRINT

The ASNZ has teamed up with **SoundPrint** to provide this curated list of acoustic ratings for food and beverage venues across Aotearoa (replacing the previous **CRAI** ratings). This data is collated from submissions made by users of the SoundPrint app, which rates venues based on the ambient noise levels present at the time of review and a subjective impression of how easy it was to hold a conversation. SoundPrint ratings follow a decibel scale, and these correspond with our awarded star ratings as follows:

Quiet		moderate	loud	very loud
70 dBA or below + subjectively "great" for conversations	70 dBA or below	70 - 75 dBA	75 - 80 dBA	80 dBA or above
★★★★★	★★★★	★★★	★★	★

The list below contains submissions from the past 3 years only. The numbers in parentheses are the total reviews over this period.

AUCKLAND		
Bellota, Auckland	★★	(1)
Birkenhead Brewing Company, Birkenhead	★	(1)
Brickhouse Espresso Bar, Auckland	★★★	(1)
Brothers Beer, Auckland	★★★	(1)
Chamate, Auckland	★★★	(2)
Copia, Remuera	★★★★★	(1)
Corner Bar, Auckland	★★★★	(1)
Dear Jervois, Herne Bay	★★★	(1)
Dizengoff, Ponsonby	★	(1)
Fabric Cafe Bistro, Hobsonville	★★★★★	(1)
Ginger, Remuera	★★★★	(1)
Kind Cafe & Eatery, Auckland	★★★★	(1)
Lieutenant, Auckland	★★	(1)
Little Bird Unbakery, Ponsonby	★★★★★	(1)
Little Creatures Hobsonville, Hobsonville	★★★	(1)
Little Culprit, Auckland	★★★★	(1)
Masala Indian Restaurant, Pukekohe	★★★	(1)
Pasta & Cuore, Auckland	★★★★★	(1)
Poni Room, Auckland	★★★	(1)
Seoul Night, Auckland	★★★	(1)
Siso Bar And Eatery, Auckland	★	(1)
St Pierre's Sushi & Seafood, Auckland	★★★★★	(1)
Sumthin Dumplin, Auckland	★★★★	(1)
The Brewers Co-operative, Auckland	★★★★★	(1)
The Chamberlain, Auckland	★★	(1)
The Dark Horse, Auckland	★	(1)
Tok Tok, Hobsonville	★★	(1)
Toto Cucina, Auckland	★★	(1)
BAY OF PLENTY		
Ohope Charter Club, Ohope Beach	★★	(1)
CANTERBURY		
Black And White Coffee cartel, Christchurch	★★★★	(1)
Coffee Culture, Papanui	★★★★	(1)
Coffee Culture, Christchurch	★★★★★	(1)
Columbus Coffee, Papanui	★★★	(1)
Doubles, Christchurch	★★★	(1)
Kohan Japanese Cuisine, Lake Tekapo	★★★★★	(1)
Kum Pun Thai Restaurant, Christchurch	★★★★	(1)
Little Poms, Christchurch	★★★	(1)
Mac's South Bar & Café, Christchurch	★★★★★	(1)
Meshino, Saint Albans	★★	(2)
Misceo Cafe & Bar, Ilam	★	(1)
Poppies Cafe, Twizel	★★★	(1)
Strange Bandit, Burnside	★★★★	(2)
Strawberry Fare, Christchurch	★★★★	(1)

SoundPrint can be downloaded from the Apple App Store and Google Play Store. More information and reviews are available at the website: https://www.soundprint.co/locations/nz

Terrace Tavern, Christchurch	★★★	(1)
Two Thumb Brewing Co Ltd, Christchurch	★★	(1)
Volstead Trading Company, Christchurch	★★★★★	(1)
HAWKE'S BAY		
Hunger Monger, Napier	★★★	(1)
Mister D, Napier	★★	(1)
NELSON		
Columbus Coffee, Nelson	★★★★★	(1)
Sprig & Fern Hardy St, Nelson	★★★	(1)
Sprig & Fern Tavern, Nelson	★★★	(1)
The Free House, Nelson	★★★★	(1)
OTAGO		
1876 Bar & Restaurant, Queenstown	★★	(1)
Farelli's Trattoria, Queenstown	★	(1)
Margo's queenstown, Queenstown	★★★★	(1)
My Thai Lounge, Queenstown	★★★★	(1)
The World Bar, Queenstown	★★	(1)
Wolf Coffee Roasters, Arrowtown	★★★★★	(1)
WAIKATO		
The Vine Eatery, Taupo	★★	(1)
WELLINGTON		
Boulcott Street Bistro, Wellington Central	★★	(1)
Caffe L'affare, Te Aro	★★	(2)
Charley Noble, Wellington	★	(2)
Crab Shack, Wellington Waterfront	★★	(1)
Crumpet, Wellington	★★	(1)
D4, Wellington	★	(1)
Dillinger's, Wellington	★★★★	(1)
Dirty Burger, Wellington	★★	(1)
Dragon Fly, Te Aro	★★	(1)
Flamingo Joe's, Pipitea	★★	(1)
Foxglove, Wellington Central	★★★	(1)

Hashigo Zake, Wellington	★★★★★	(2)
Ivy: Underground, Wellington	★	(1)
Liberty restaurant, Wellington	★★	(1)
Logan Brown Restaurant & Bar, Wellington	★★★	(1)
Mexico, Lower Hutt	★★★★★	(1)
Neo Cafe & Eatery, Wellington	★★	(2)
Panhead Tory, Te Aro	★★★★★	(1)
Preservatorium, Wellington	★	(1)
Rosie's Red-Hot Cantina & Taco Joint, Wellington	★	(1)
Scopa Caffé Cucina, Wellington	★★	(1)
Seashore Cabaret, Petone	★★	(1)
St Johns Bar, Te Aro	★★	(1)
Te Papa Cafe, Wellington	★★★	(1)
Viva Mexico, Wellington	★★	(1)

Acu-Vib Electronics
ACOUSTICS AND VIBRATIONS

NATA
Accredited Laboratory No. 9262

Vibration Calibration at Acu-Vib

The Acu-Vib **calibration laboratory** has **NATA accredited and traceable vibration calibration systems** that cover a wide range of frequencies, vibration levels and load capabilities:

For low frequency transducers such as **geophone based systems or high sensitivity accelerometers**, the long stroke air bearing table ranges from 0.5 Hz to 250 Hz with mass capability up to 5 kG and velocity levels up to 200 mms^{-1}.

For **higher frequencies or high acceleration levels**, the high force shaker system ranges from 8 Hz to 5 kHz for mass loading up to 10 kG limited by force up to 1000N or acceleration levels up to 1000 ms^{-2}.

The test frequencies and levels can be chosen within these ranges to **suit the customer needs**.

Acu-Vib Electronics can issue Nata endorsed certificates for the following equipment:

- Ground & Building Vibration Loggers
- Geophones
- Accelerometers (Triaxle / Single Axis)
- Whole Body Vibration Analysers & Seat Pads
- Hand Arm Vibration Analysers & Sensors
- Sound Level Meters
- Acoustics and Vibration Calibrators
- Noise Dosimeters (Including Instrinsically Safe)
- A wide range of Vibration Transducers and Microphones

FAST FREIGHT to and from New Zealand (DHL)

enquiries@acu-vib.com.au acu-vib.com.au Castle Hill, NSW 2154 Australia +61 (02) 9680 8133

Acu-Vib Electronics
CALIBRATIONS SALES RENTALS REPAIRS

SVANTEK
sound and vibration

NATA
NATIONAL ASSOCIATION OF TESTING AUTHORITIES, AUSTRALIA
Registered Laboratory No. 9262
Acoustic and Vibration Measurements

SUPER QUIZ ANSWERS

1. A1-1 Voiced (The vibration of the vocal folds produces a pitch and therefore the voice is said to be voiced.) and A1-2 Pitch (The periodic vibration of the vocal folds produces a pitch.)
2. 5- 10 kHz (Sibilants are high frequency noise bands).
3. Inflection (Inflection refers to changes in vocal pitch).
4. False (Pitch changes are the result of the vocal folds vibrating at different rates. Therefore they are largely independent of the resonant patterns of the vocal tract which are the result of tongue position).
5. C (Normal speech is in the sound pressure level range of 55-65 dB L_{pA}).
6. Formants (Formants are the energy peaks in the spectrum of a vowel).
7. True (The term "ossicle" literally means "tiny bone". Though the term may refer to any small bone throughout the body, it typically refers to the malleus, incus, and stapes (hammer, anvil, and stirrup) of the middle ear).
8. B. The bulk of information content in speech intelligibility lies between the range of 250 Hz and 5 kHz, although audibility lies between the range of 20 Hz and 20 kHz.
9. An audiometer is an instrument used to measured hearing sensitivity.
10. True. ANCON is a UK civil aircraft noise computer model developed by UK CAA that calculates contours from data describing aircraft movements, routes, noise generation and sound propagation.
11. 'CRTN' is an acronym for Calculation of Road Traffic Noise.
12. The the 'direct sound field' the area in which the sound is perceived directly from the source without being reflected.
13. The Doppler Effect is named after the physicist Christian Doppler.
14. The speed of sound at 20°C and sea level is said to be 343 m/s.
15. What is the speed of sound in space is 0 m/s as sound cannot travel in a vacuum. Sound is a wave, which means it spreads through the vibration of particles in a medium, such as water or air. Since a vacuum is empty space, there is no medium for the sound to travel through. But sound energy can be transferred over very short distance in space under the right conditions, see: Physicists demonstrate how sound can be transmitted through vacuum.
16. 'JND' is a concept in psycho-acoustic measurement, being 'just noticeable difference' between two (acoustic) stimuli which is just noticeable in some defined condition.
17. False? 'L_{Amax} (not L_{Apeak}) is the maximum A-weighted sound pressure level occurring within a specified time period such as 15 minutes. It is usually an F-time weighted RMS value (L_{AFmax}), but can also be derived from short Leq values, eg. $L_{Aeq,\ 125\ ms.}$
18. The Lombard Effect is when a speaker raises their level voice in response to an increase in background noise, in turn increases the background noise level (positive feedback).
19. True? 'Mel' is a unit of pitch being the pitch of any sound judged by listeners to be n times that of a 1 mel tone in n mels.
20. STITEL is version of STI (speech transmission index) for telecommunications systems.
21. A semi-anechoic chamber or room is room with anechoic walls and ceiling but with a sound reflecting floor.
22. The terms R is the sound reduction index and the term R' the apparent sound reduction index. The terms are used in building acoustics. R is he weighted sound reduction index for a partition or single component only while R' a field measurement which attempts to measure the sound reduction index of a material on a real completed construction (e.g. a wall between two offices spaces).
23. Noise immission is the amount of sound exposure or sound received at a particular receiver location. Sound level meters measure sound immission.
24. True. Z-weighting is zero frequency weighting defined in standards such ISO standard 61672-1. It was introduced to relaced "flat" or "Linear: frequency weighting often adopted by some manufacturers.
25. The 'specific noise source' is the noise under investigation, the 'target sound' in NZS6801, for assessing the likelihood of complaints?
26. *P* is often used to indicate sound power (W), whereas SWL and L_{WA} are often used as abbreviation for A-weighted sound power level (the sound power on a dB scale relative to 1 pW).
27. 'Structural acoustics' is the study of the mechanical waves in structures and how they interact with and radiate into adjacent media. The field of structural acoustics is often referred to as vibro-acoustics in some parts of the world such as Europe and Asia. People working in the field of structural acoustics are known as structural acousticians.
28. The scalar wave equation, where ∇ is the *nabla* operator, and $\nabla^2 = \nabla \cdot \nabla$ is the (spatial) Laplacian operator.
29. The portrait is of French scientist Jean-Baptiste le Rond d'Alembert who discovered the wave equation in one space dimension.
30. 'Noise' is derived from the Latin word 'nausea', meaning measickness. It also may have come from the Latin word 'noxia' a variant of 'noxa', meaning to harm, damage, hurt or injure.