Questions and answers in environmental noise assessment at an undergraduate level

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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 (0°) response and the response for angles above 60°. This difference increases significantly with frequency.

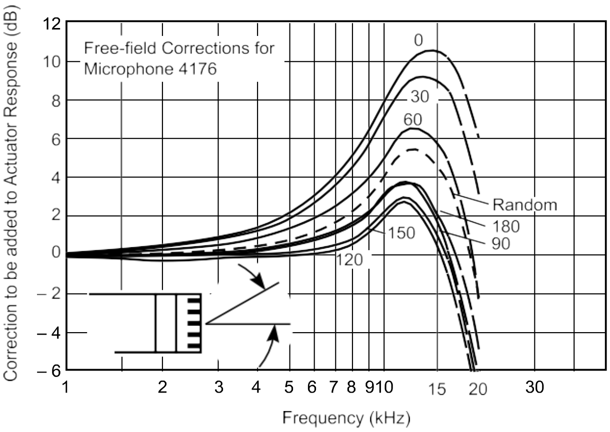

Figure 1. Free-field correction curves for various angles of incidence for a B&K Type 4176 capsule

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 30° upwards microphone and channel 2 (right) for the 0° (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

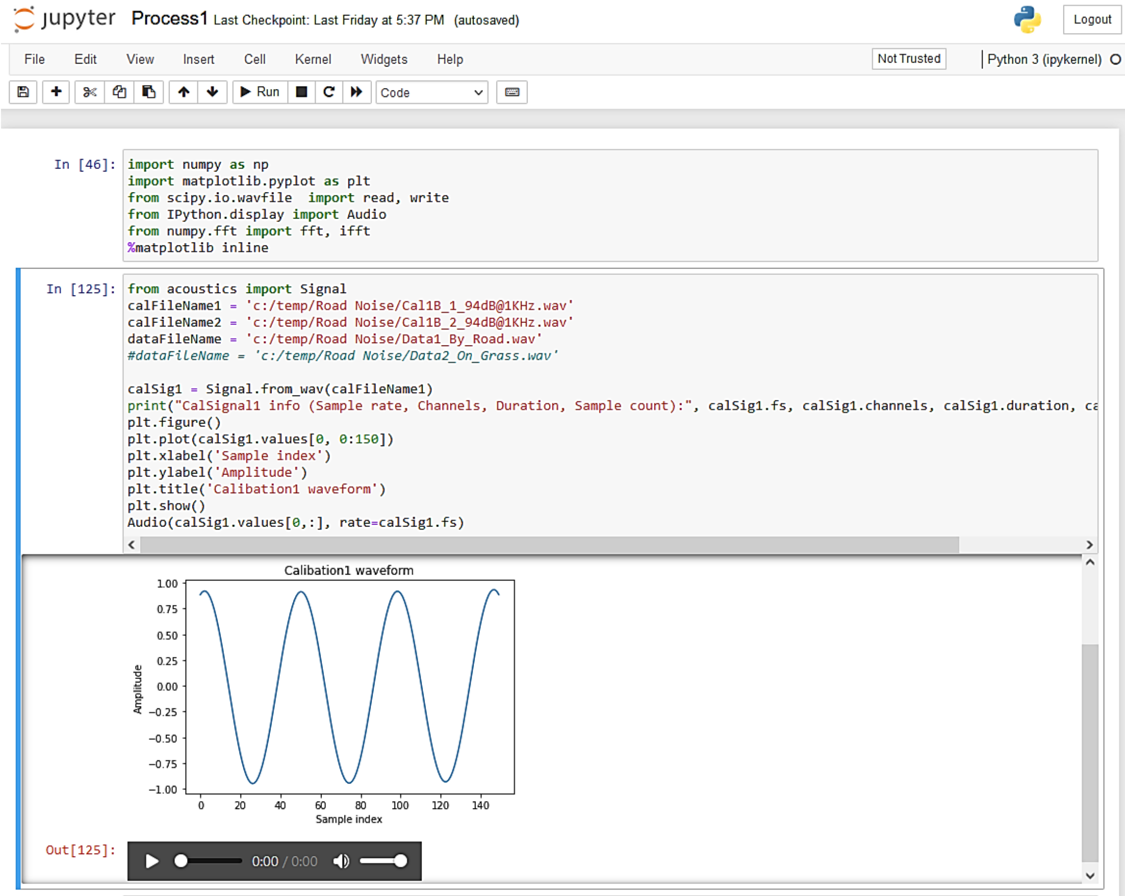

Figure 3. Snapshot of the top of the Jupyter notebook and the Python code to process the recordings

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 ($L_{Aeq,1s}$)
8. Calculate the maximum F-time weighted levels (L_{AFmax}) 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.

Table 1. Microphone inclination results

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

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.

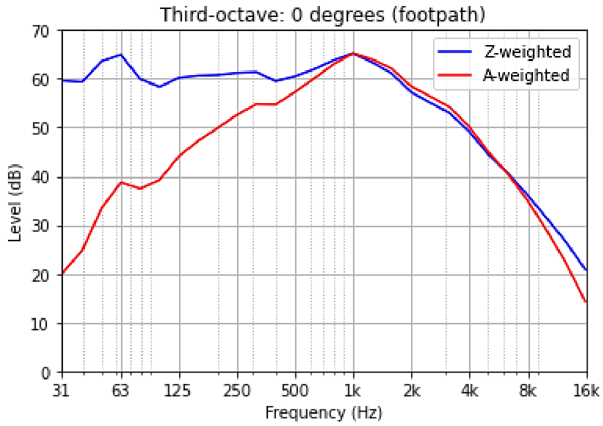

Figure 4. Third-octave spectrum of the 0° inclined microphone for traffic noise taken from the footpath

Figure 5 shows the difference between the (calibrated) third-octave spectrums for the 30° inclined upwards microphone and the horizontal (0°) microphone, for both measurement locations. The spectrum has been limited to 5 kHz, as above this there is little (about 20 dB lower than at 1 kHz) sound energy from the traffic noise.

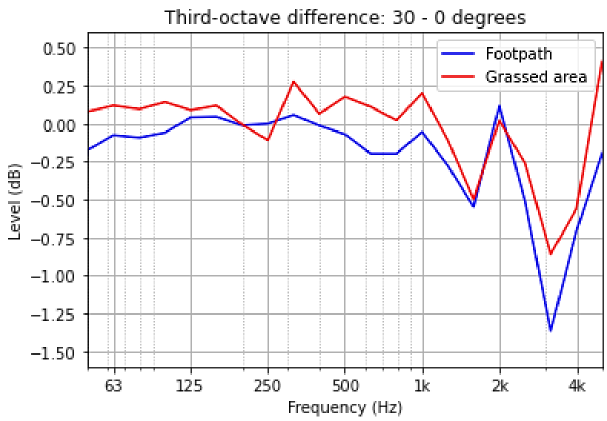

Figure 5. Third-octave spectrum difference between microphones for the traffic noise

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.

Table 2. Microphone inclination results for quiet sections

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 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 1.5 m.
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 1.5 m and the lower at 1.2 m off the ground

The previously developed Python scripts were used to process the recordings. The difference between the two microphone calibration factors for before and after recordings of the calibration tone was the same at 0.20 dB and there was only 0.04 dB difference between the before and after calibration factors.

Table 3. Microphone height results, 1.5 m – 1.2 m height

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 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.

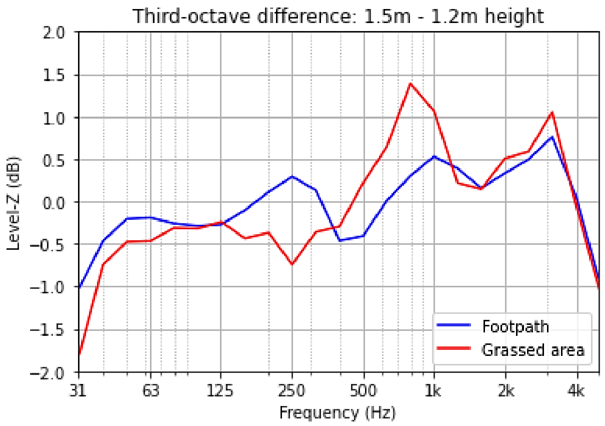

Figure 7. Third-octave spectrum difference between 1.5 and 1.2 m microphones for the traffic noise

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:

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

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 $L_{AF(t)}$ at observation time t can be represented by

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

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:

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

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.

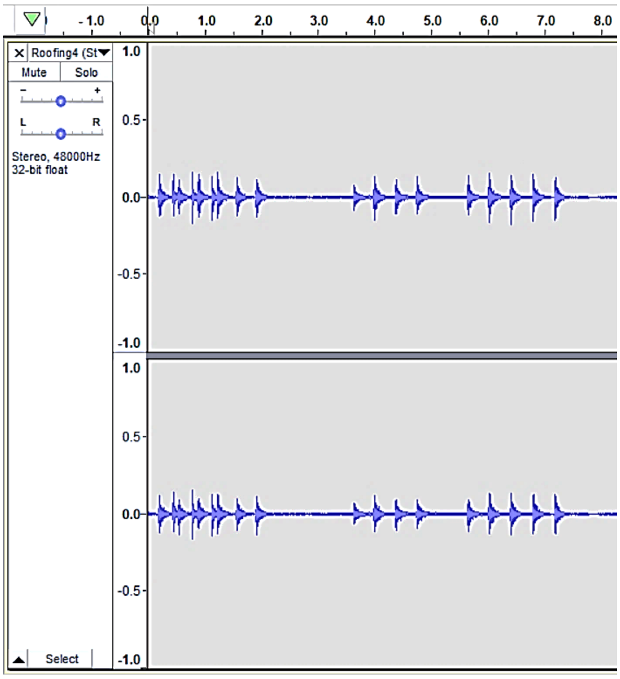

Figure 7. A section of the audio recording showing the impulsive stapling bursts

Results

Python scripts were developed to process the audio recordings to calculate L_{AFmax} 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.

Table 4. Roofing noise Lmax descriptor difference

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

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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