ANNOYANCE THRESHOLDS OF TONES IN NOISE AS RELATED TO BUILDING SERVICES EQUIPMENT

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SYMBOLS

N_Z	Stationary Zwicker Loudness
$N_{M\&N}$	Stationary Moore and Glasberg Loudness
dBA	A-weighted Sound Pressure Level
dBC	C-weighted Sound Pressure Level
TNR	Tone-to-Noise Ratio
PR	Prominence Ratio
ΔL_{ta}	Tonal Audibility
T	Aures Tonality
$S_{vB\ Z}$	von Bismarck Sharpness based on Zwicker Loudness
$S_{vB\ M\&G}$	von Bismarck Sharpness based on Moore and Glasberg Loudness
$S_{A\ Z}$	Aures Sharpness based on Zwicker Loudness
$S_{A\ M\&G}$	Aures Sharpness based on Moore and Glasberg Loudness
R	Roughness
PNL	Perceived Noise Level
SQI	Sound Quality Indicator
OP	Oppressive Penalty

ABSTRACT

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Rotating machinery (e.g., fans, motors, compressors, etc.) is widely used in building services equipment, especially in HVAC applications. It is a primary tonal noise contributor to building noise. Sounds with these tonal components often sounds more annoying than those without tones. There is a recent trend to develop and produce more energy-efficient equipment, and this always comes with a price of poor sound quality. Higher output power and higher rotation speed would inevitably result in not only stronger tonal components but also higher frequency tones that people tend to be more sensitive to.

Current noise criteria guidelines that are widely used in the industry do not account for the tonal sounds. Reducing the limit for tonal sounds ('rules of thumb') sometimes is inadequate. There is not enough data exist to provide guidelines for tonal building noise.

The goal of this research is to determine human annoyance threshold for tonal building noise with different tonal levels, with different tonal frequencies, with or without harmonics, and with different broadband background noise. Three sets of subjective tests (all rating tests) were conducted. In the first test, sounds with different duration were tested to investigate whether subjects' annoyance ratings to short sounds can be used to predict annoyance for longer exposures. The second test consisted of a 5-second sound session and a 2-minute sound session. Sounds with similar background noise were used to investigate how tonal levels and harmonics affect annoyance ratings and the annoyance acclimation for tonal sounds with similar neutral background noises. The third test consisted of four 5-second sound sessions and two 2-minute sound sessions. The test contains a variety of sounds to learn how tonal levels, frequencies, the presence of harmonics, and broadband affect annoyance ratings. Annoyance acclimation for tonal sounds with different level broadband noises was studied. From the test results, sound quality models were developed to relate sound characteristics to perceived annoyance under different time exposures.

1. INTRODUCTION

Tonal sounds are a particular problem of concern in building environments, arising from the widely used rotating machinery (e.g., compressors, fans, motors, transformers, etc.). In the recent trend of designing and manufacturing high-performance building mechanical systems, higher output power and higher rotation speed are pursued, this inevitably results in a more severe noise problem, since the equipment noise not only becomes louder but also shifts to a higher frequency region (which, in most cases, results in a poorer sound quality due to the shift in spectral balance and tonal components moving into the frequency regions where people are most sensitive to tones). Tonal sounds from rotary machines can be annoying, even at relative low levels.

Currently, noise criteria guidelines in Chapter 48 of the ASHRAE HVAC Applications Handbook [1] can be used to design the building mechanical system, but this does not apply well for tonal noise. Reducing the limit for noise with perceptible tones is one common strategy in the industry. However, it's not adequate for some cases, over-design in others. Thus, an adequate understanding of the annoyance threshold of tonal noises associated with building services equipment is valuable technical information not only in the design and manufacture of machines but also in the development of noise regulations related to building services equipment.

This research aims to develop a sound quality model that cooperates with sound level and tonalness and relates tonal building noises to the perceived annoyance.

1.1 Review of Literature

Level metric (e.g., A-weighted sound pressure level, Zwicker Loudness [2]) is known to be important for predicting perceived annoyance of the sound [3, 4], but human perception of tonal sounds cannot be simply measured by existing level metrics (e.g., A-weighted sound pressure level, Loudness), since it is known to be significantly affected by various factors related to tonalness. For example, the frequency of the tone, the prominence of the tone above the broadband background noise, the amount of variation in the tonal component which affects the bandwidth of the tonal feature, the distribution of the harmonic tones, and the combination of other non-harmonic tones.

The importance of tonalness in annoyance perception has been studied extensively. Hellman [5,6] found that tonal components have an influence on annoyance, loudness and noisiness perception, both the frequency of the tone and the number of tones have an impact on perceived annoyance. For the more complex tonal components, Lee *et al*. [7,8] studied the perceived tonal strength of harmonic complex tones. The prominence of complex tones were quantified, the feasibility of replacing harmonic complex tones with a tonally equivalent simple sound was investigated. Hasting *et al*. [9] found that perceived tonalness can be affected by the bandwidth and roll-off rate of narrowband tonal features in spectra. While much work has been done on the tonal components, Ryherd and Wang [10] also examined the influence of tones in a neutral broadband background noise. They found the current indoor noise criteria (e.g., Noise Criteria (NC), Room Criteria (RC) [11]) couldn't fully reflect the perceptual changes because they do not typically account for the tonal characteristics. Thus, the effect of tonal components should be included in an annoyance prediction model for tonal sounds to describe how tonal noise is perceived. A number of tonality metrics (described in Chapter 2) have been developed to quantify the perceived strength of tonal components, such as Tone-to-Noise Ratio (TNR) [12], Prominence Ratio (PR) [13], Tonal Audibility (ΔL_{ta}) [14] and Aures Tonality (T) [15].

To date, in the building equipment related area, the Tone-to-Noise Ratio, the Prominence Ratio and the Tonal Audibility are the most commonly used tonal noise measurement methods. In Balant *et al*.'s study [16], a round robin test has been conducted to compare Tone-to-Noise Ratio and Prominence Ratio. Two metrics were found to correlate well for broadband noise with a single tone. Different from TNR and PR, the Tonal Audibility considers both the prominence of the tone and the frequency of the tone. A frequency weighting that grows as the frequency increases is included in the Tonal Audibility. Aures Tonality model is more complicated in calculation as it incorporates more factors such as frequency and bandwidth for each tone, prominence of each tone to the broadband content around it, and the loudness of all the tone components relative to the entire signal. It is of both theoretical and practical value to study how do the predictions from these existing tonalness measurement and noise rating methods (e.g., Noise Criteria (NC), Room Criteria (RC)) agree with the subjective test results.

One way to compensate for the negative effects of prominent tones in some applications is to develop tonal penalties. For example, Danish environmental noise standard [14] corrects A-weighted sound pressure level by applying a penalty computed from on ΔL_{ta}. In the AHRI/ANSI 1140 [17], a tone penalized loudness metric for refrigeration equipment is suggested (Sound Quality Index - SQI). This method computes loudness metric from tone-adjusted the one-third octave band spectrums. These standards are developed under specific applications, and therefore the correction may be inaccurate when applied to other noises. A more recent study conducted by Oliva *et al*. [18] against the opinion of using fixed penalty values. He found that the penalty is depending on tonal frequency, tonal audibility and overall level. Another way to treat tonal components is to include a tonalness metric in the annoyance model. Lee *et al*. [19] investigated how tonal mechanical equipment sounds with varying degrees of prominent tones affect subjective annoyance perception and task performance. A linear annoyance prediction model was proposed with tonal audibility and Moore and Glasberg Loudness. In Lee *et al*.'s later study [20], a logistic regression equation for %Complaint was developed with same metrics for tonal building equipment noises. In both of his studies, sound attributes show no effects on the accuracy performed in the tasks. Apart from the perception of sound with a signal tone, models describing the tonal sensation of complex tones were also developed in previous studies by Lee *et al.* [7, 8].

Apart from developing metrics from subjects' responses, there is some concern on their annoyance adaptation to tonal noises. People are usually exposed to tonal office noise for a relatively long time, but long-duration exposures limit the variety of sounds that can be played in subjective tests, which, in turn, limits annoyance model development. There are very few researches have been done with the annoyance adaptation for tonal office noise which includes both broadband and tonal components. Most annoyance adaptation studies focuses on different broadband noises. Poulsen [21] found for traffic and gunfire noise, exposure time (ranges from 1 min to 30 min) does not have a significant effect on judged annoyance. Ryherd and Wang [22] found that compared with different broadband sounds with longer exposure time (240-minute), sounds with shorter exposure time (20-minute) were rated louder but not more annoying. In terms of the pure tones, some work [23, 24] on loudness adaptation had also been done. Most loudness adaptation occurs within the first 3 minutes of exposure and mostly with high-frequency tones (>4,000 Hz). A small but significant loudness enhancement would appear for higher level tones.

1.2 Objectives of this Research

The objectives of the research includes:

- Determine what the human annoyance thresholds are of tones in noise, across the most common tonal frequencies found in building services equipment.

- Study how the annoyance threshold is affected by various factors, such as: the frequency of the tone, other harmonic tones, the broadband background noise characteristics, etc. For example, when does a certain tonalness above a low background noise level demonstrate the same annoyance as that same tonalness above a higher background noise level?

- Investigate how the obtained annoyance threshold results correlate with the commonly used noise measurement standards and rating criteria in building related areas, such as Tone-to-Noise Ratio, Prominence Ratio, Noise Criteria, Room Criteria, etc.

1.3 Approach and Thesis Outline

The focus of this study is to develop a metric or a model to assess the annoyance due to the tonal office noise. Chapter 2 describes the properties of sounds measured in office, the process to characterize the tonal building noises, important sound quality metrics and sound evaluation criteria. Methods to decompose, simulate and modify tonal office noises are presented in Chapter 3. Recorded broadband and tonal components have first been studied to simulate a natural tonal building noise. Modifications have then been applied to the sounds to include a variety of sounds with different sound attributes and to generate a group of sounds with desired attributes. The simulated sounds were used in the subjective tests to examine the effect of tonality, the presence of harmonics, broadband level, and broadband spectral balance. Chapter 4 focuses on the setup (the office environment and the electrical system) of the test. The locations of the loudspeakers are carefully selected to avoid room mode. The inverse equalization filter design process is proposed and refined to compensate for the room and speaker response. The process of designing an inverse equalization filter for the room and speaker is refined. In Chapter 5, three sets of tests are described. Test results and analysis of results are also given. Two types of Global Annoyance Models are proposed, refined and validated in Chapter 6. Finally, conclusions of this research project and suggestions for future works are presented in Chapter 7.

2. CHARACTERIZATION OF BUILDING NOISE

The first step in the research was to gather typical building noise recordings with tones and understand their compositions. 36 measurements were provided by the ASHRAE advisory team and 27 additional recordings were made on campus. In this chapter, an overview of the measurements is given, and the spectral and temporal characteristics are described. Sound quality and other acoustical measures were conducted and the results are given. An overview of annoyance models used or developed by other researchers are also described. This includes a method developed to characterize the effects of low-frequency components in the perception of the sounds. This chapter ends with a summary of the finding from our analysis of the sounds and the findings of other researchers on tonal building noise.

2.1 Sound Measurements

ASHRAE (the sponsor of the research) identified tonal frequencies that should be examined in the research. They are: 29.5, 60, 120, 240, 500, 750, and 1000 Hz. There was also interest in perception of sounds with additional harmonics.

Measurements were taken at various locations on Purdue University West Lafayette Campus and the advisory team provided other measurements. The location of the measurements are described in Table A.1.

The measurements were taken using a 1/2” PCB PIEZOTRONICS microphone, calibrated using a Sound Calibrator (Type 4231). The microphone was placed around 1 m from the ground and signals were acquired using a HEAD ACOUSTICS 4 channel SQUADRIGA (Code 1369) acquisition system. The analog to digital counter is 16 bits and sampling rate is fixed at 48,000 samples/second. The analog filters in the acquisition systems are 1^{st} order high-pass filter with a cut-off frequency 2 Hz. The noise was recorded for 20 – 150 seconds for each measurement.

Most measurements measured on the campus have a stationary neutral broadband component and some have strong tones. The other measurements were provided by the ASHRAE Advisory includes condenser, cooling tower, fan, chiller and compressor sounds with strong tones and multiple harmonics. Measurements details are listed in Table A.2.

From Table 2.1, tones from 29.5 Hz to 1000 Hz with/without harmonics can be found in the building measurements. This indicates that tone frequencies listed in Request-for-Proposal worth investigation.

Table 2.1. Request-for-Proposal listed tones and corresponding tones identified from measurements.

Listed tonal frequencies	**Strong frequencies in the measurement**	**Location**
29.5 Hz	29.5 Hz	HERL, Student Lounge (Open Space)
29.5 Hz with harmonics	32 Hz, 63 Hz, 95 Hz	Fan-centrifugal-airfoil (from ASHRAE)
60 Hz	60 Hz	HLAB, High-Bay Area (Laboratory)
60 Hz with harmonics	59.5 Hz, 120 Hz, 240 Hz	HLAB, High-Bay Area (Laboratory)
120 Hz	120 Hz	HLAB, Student Open Area
120 Hz with harmonics	120 Hz, 240 Hz	HLAB, High-Bay Area (Laboratory)
240 Hz	240 Hz	HLAB, High-Bay Area (Laboratory)
240 Hz with harmonics	240 Hz, 480 Hz	HLAB, High-Bay Area (Laboratory)
500 Hz	489 Hz	PFSB, Chiller Noise (in the office)
500 Hz with harmonics	489 Hz, 978.5 Hz	PFSB, Chiller Noise (in the office)
750 Hz	723.5 Hz	HLAB, High-Bay Area (Laboratory)
750 Hz with harmonics	764 Hz, 1528 Hz	Compressor Tone (from ASHRAE)
1000 Hz	948 Hz	HERL, Student Lounge (Open Space)
1000 Hz with harmonics	1058 Hz, 2116 Hz	Compressor Tone (from ASHRAE)

2.2 Frequency Analysis of Measured Sounds

Narrow-band spectra were estimated for all the measurements. MATLAB's *pwelch* program was used for estimation of power spectral densities. A Hann window with 50% overlap was used. Energy loss due to the window type was compensated for the estimation. Spectra with different resolutions were generated to determine a resolution where the power and frequency of individual tones could be accurately estimated, and to identify whether the tonal feature was a pure tone (Figure 2.2) or contained some frequency and/or amplitude modulation (Figure 2.3).

Sound Quality metrics used to quantify the strength of tone presence (tonalness) usually derived from narrowband spectra are sensitive to frequency resolution. In this research, it was found that frequency resolutions less than 5 Hz are desirable. Note that because we are estimating power spectral densities, the height of the feature (peak) in the spectrum, due to the tone presence will change with different frequency resolutions, but the broadband components will stay at the same level. The strength of tones was estimated by integrating the power spectral density around the peak and compensation for the contribution of the noise floor. The averaging of results from each segment is also important and it was found that more than 10 segments (with the 50% overlap) are desirable, while keeping within the desired frequency resolution range.

The one-third octave spectrum was estimated from the narrow-band power spectral density using one-third octave filter frequency response described in [25].

Shown in Figure 2.1, 2.2 and 2.3 are three examples. The pressure time history, power spectral density and one-third octave spectra are shown in each of figures.

Shown in Figure 2.4, 2.5 are the power spectral densities of 12 recordings measured on Purdue University's West Lafayette campus and provided by ASHRAE advisory team. Stronger and more complex tones can be observed from sounds provided by ASHRAE advisory team.

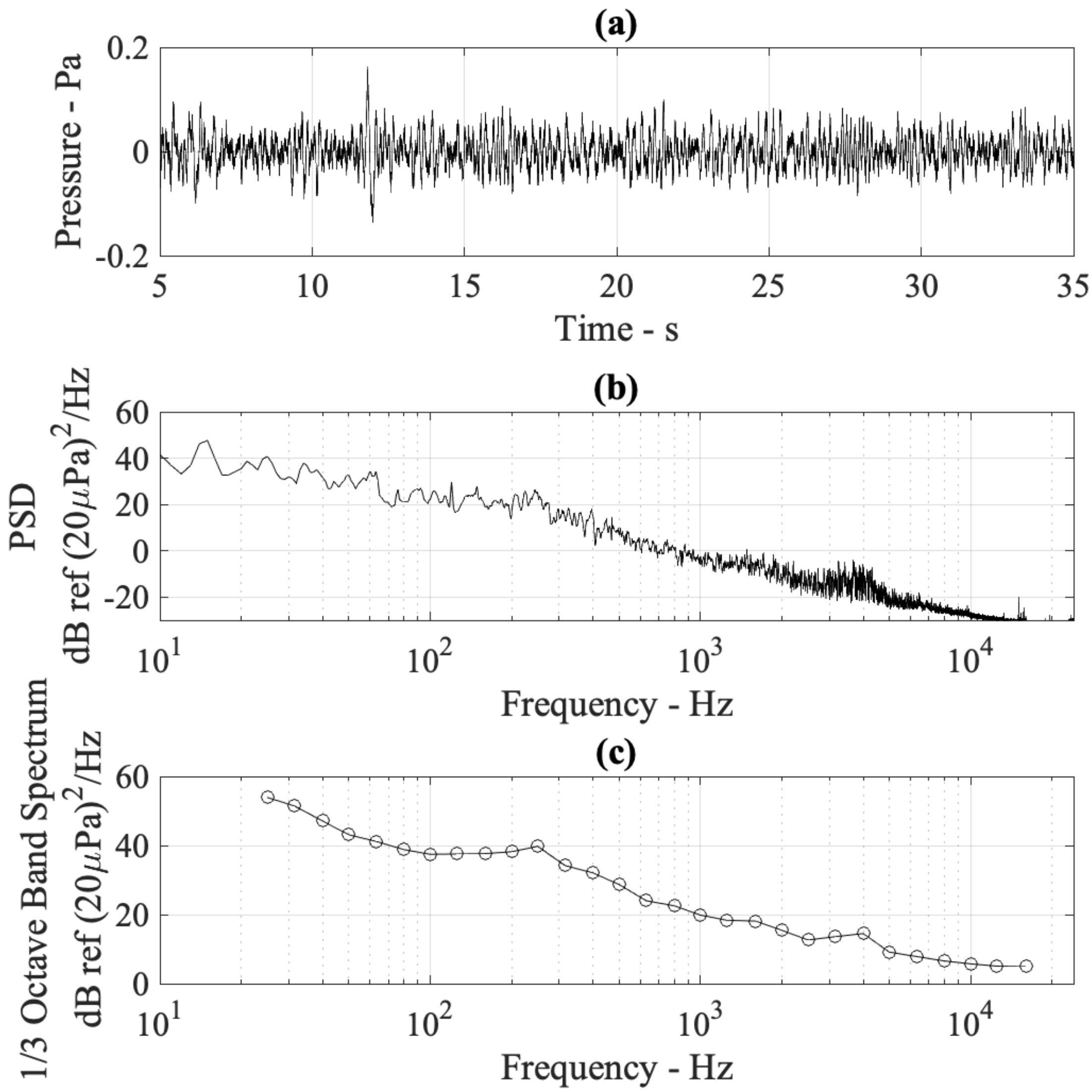

Figure 2.1. Frequency analysis of Sound 011 (conference room recording) in Table A.1: (a) pressure time history, (b) power spectral density, (c) 1/3-octave band spectrum

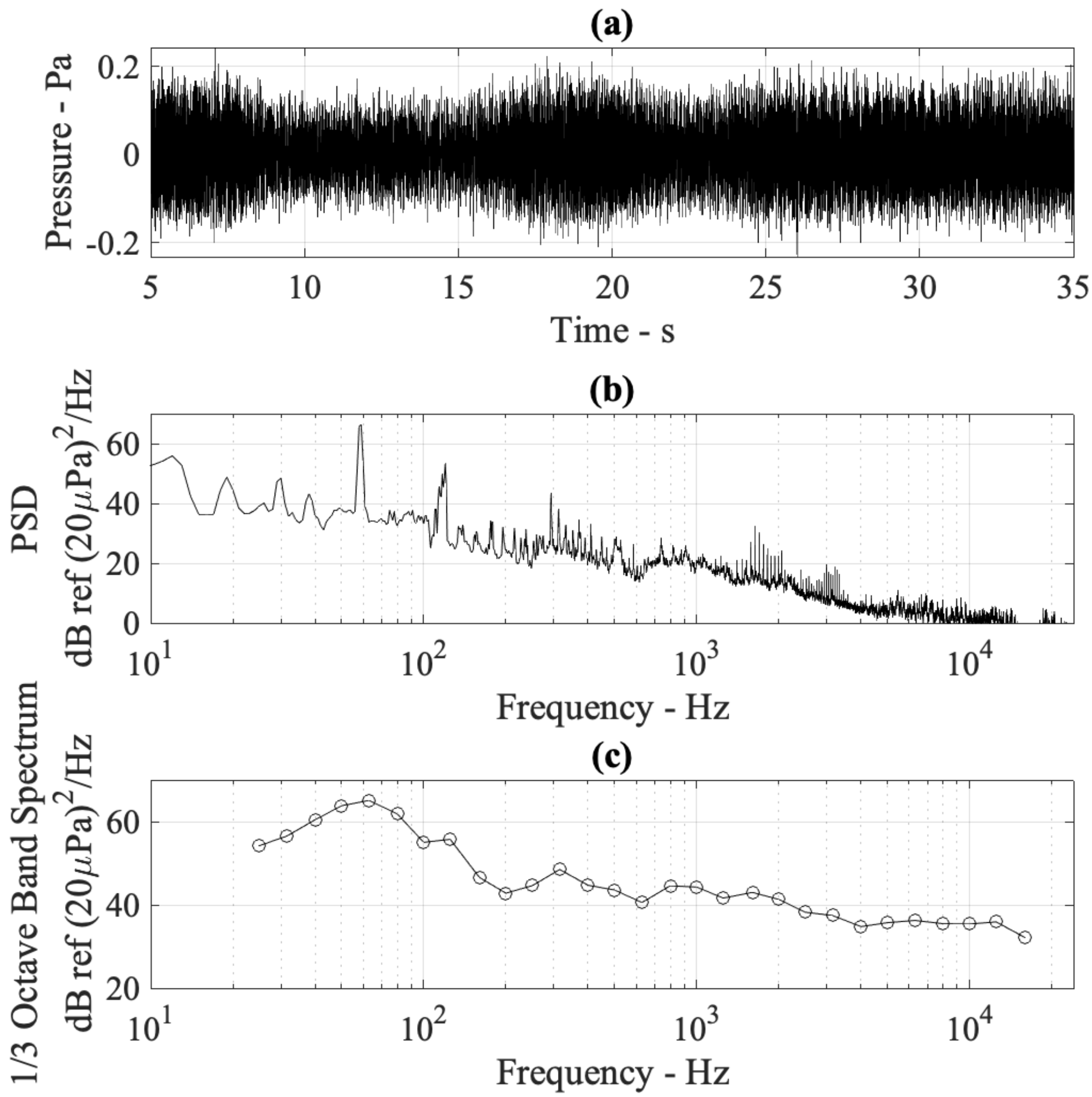

Figure 2.2. Frequency analysis of Sound 002 (heat pump tones) in Table A.2: (a) pressure time history (uncalibrated), (b) power spectral density (uncalibrated), (c) 1/3-octave band spectrum (uncalibrated)

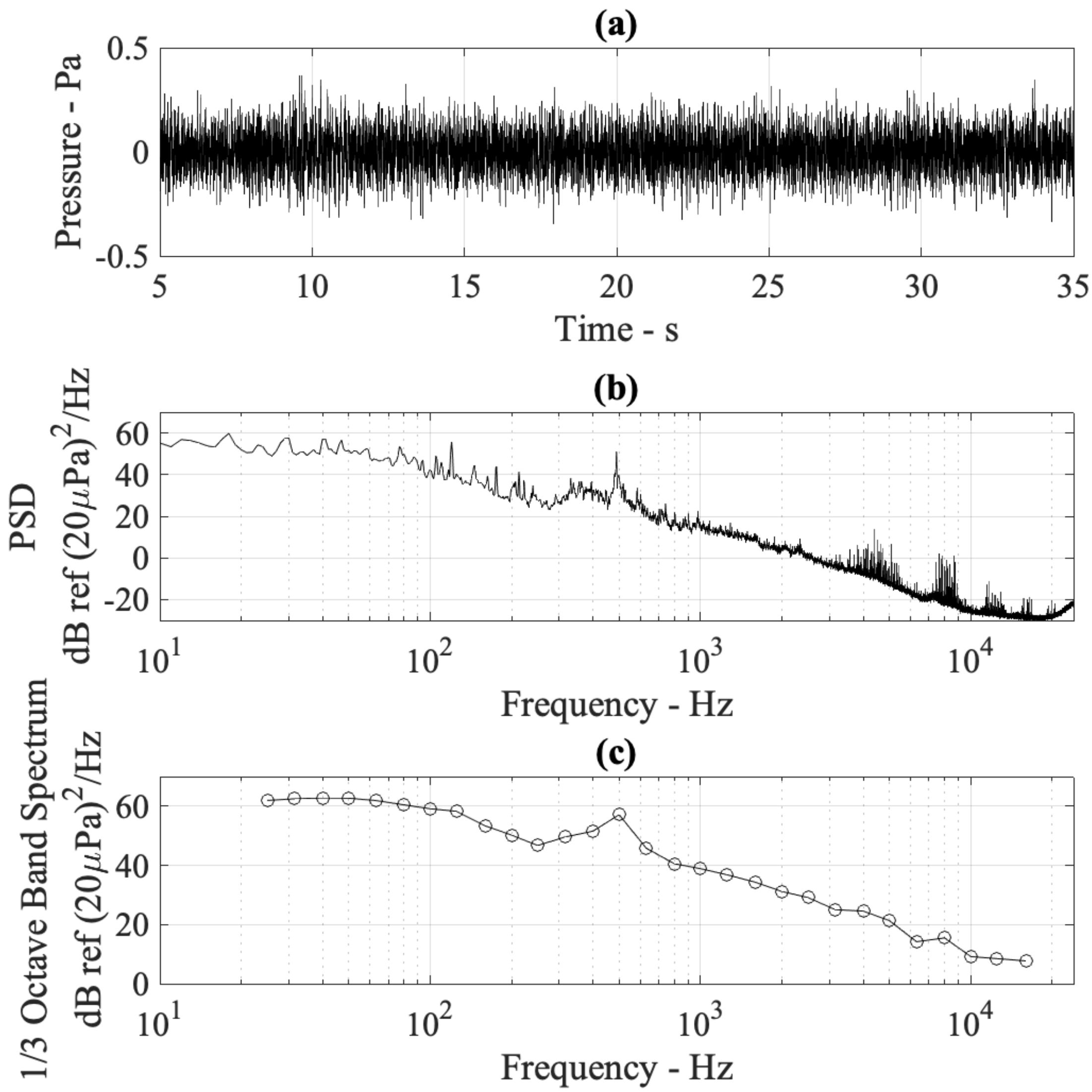

Figure 2.3. Frequency analysis of Sound 026 (indoor chiller noise) in Table A.1: (a) pressure time history, (b) power spectral density, (c) 1/3-octave band spectrum

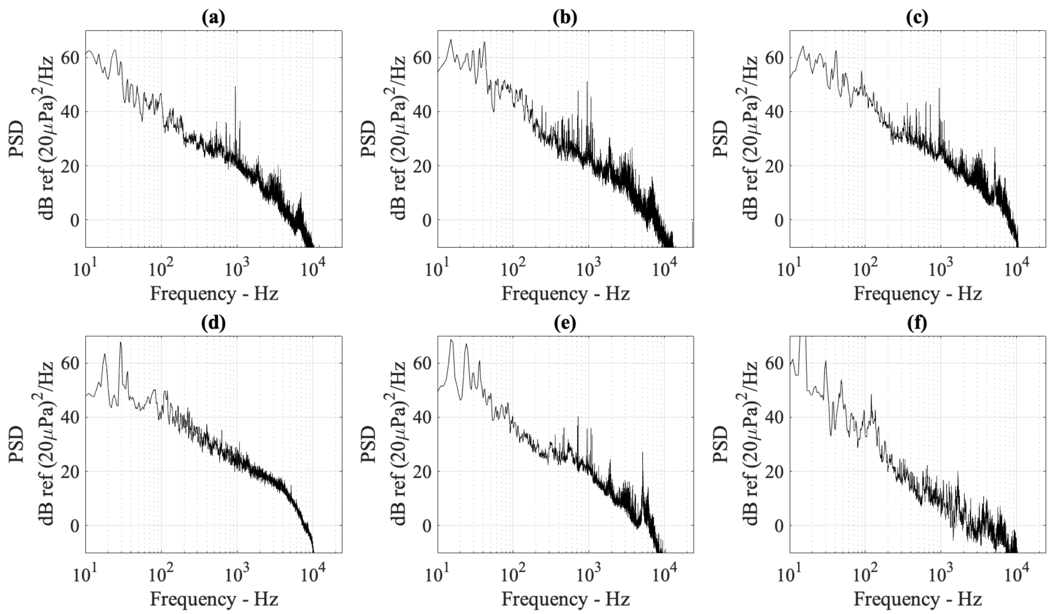

Figure 2.4. Power spectral densities of 6 on-campus measurements (Sound 001-006 in Tabel A.1)

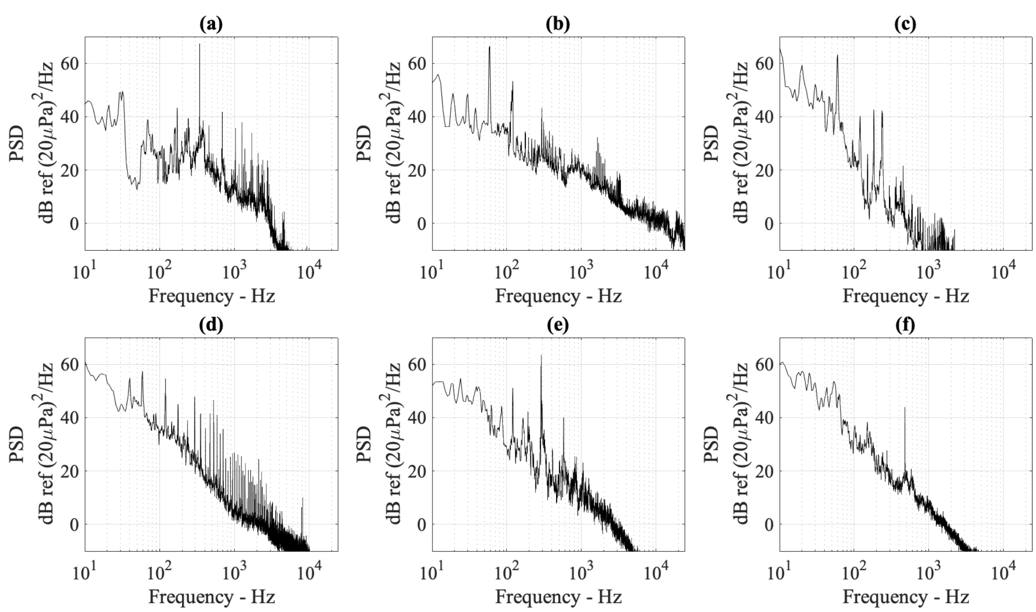

Figure 2.5. Power spectral densities (uncalibrated) of 6 measurements provided by ASHRAE advisory team (Sound 001-006 in Tabel A.2)

2.3 Temporal Properties of Measured Sounds

Some of the tonal components can be modeled as pure tones, while others come with some variations (frequency and/or amplitude modulations), which would introduce a narrow-band feature in the power spectral density. Thus, temporal properties of the tonal components need to be investigated. Both temporal and spectral properties are important to simulate natural building noises. For a narrow band tonal component, a method based on the Hilbert Transform is described below to extract sounds' instantaneous phase, frequency and amplitude. The amount of amplitude and frequency modulation (AM & FM) can be analyzed follow on.

2.3.1 Instantaneous Phase, Frequency, Amplitude

For a narrowband tonal component with AM & FM, it can be modeled as:

$$y(t) = A(t)cos(\phi(t)) = A(t)cos(2\pi f(t)t + \phi(0)) \tag{2.1}$$

$$y_h(t) = A(t)sin(\phi(t)) = A(t)sin(2\pi f(t)t + \phi(0)) \tag{2.2}$$

In the Equation 2.1 and 2.2, $y(t)$ and $y_h(t)$ correspond to the original narrow band tonal signal and the Hilbert transform of the signal. $A(t), \phi(t), f(t)$ represent the instantaneous amplitude, phase and frequency of the tonal component, respectively. Instantaneous frequency is related to the time derivative of instantaneous phase. $A(t)$ and $f(t)$ correspond to the amplitude and frequency modulation of the sound. The instantaneous amplitude, phase and frequency information can be extracted by:

$$A(t) = \sqrt{y^2(t) + y_h^2(t)}, \phi(t) = tan^{-1}\frac{y_h(t)}{y(t)}, f(t) = \frac{1}{2\pi}\frac{d\phi(t)}{dt} = \frac{1}{2\pi}\frac{y_h'(t)y(t) - y'(t)y_h(t)}{A^2(t)} \tag{2.3}$$

where $y'(t)$ and $y_h'(t)$ are the time derivatives of $y(t)$ and $y_h(t)$.

In order to perform Hilbert transform, a finite impulse response (FIR) filter was designed by specifying 2048 points in the frequency domain. To avoid sharp changes in spectrum, transient regions with quarter sinusoidal shape were designed for frequencies below $\frac{f_s}{200}$ and above $\frac{f_s}{2} - \frac{f_s}{200}$. The Differential filter was designed in time domain by scaling and windowing the theoretical impulse response with a 201 points Hann window. Filter performs as a differentiator below $\frac{99}{200}f_s$ and as a low-pass filter when it get close to $\frac{f_s}{2}$. With a down sampled $f_s = 4000$ Hz, both filter work for all the tonal components covered in the research.

2.3.2 Amplitude and Frequency Modulation

In real-world measurements, sounds often come with modulations, different metrics are designed to quantify them (e.g. Roughness, Fluctuation Strength, etc.). A pure tone, with a form of $A_0 sin(2\pi f_0 t)$, is not enough to model a real world tonal component (sometimes comes with a narrow band spectrum). Amplitude and frequency modulations are introduced to the tonal component as Equation 2.1. The amplitude and frequency modulation can be visualized with instantaneous amplitude ($A(t)$) and instantaneous frequency ($f(t)$) of the tonal component (shown in Figure 2.6). In this case, both amplitude and frequency modulation were controlled by low-passed random noises.

2.4 Sounds Pressure Evaluation Criteria

There are several sound pressure criteria to evaluate noise. The criteria attempt to consider people's responses to the overall spectrum and level. Criteria like A and C-weighted sound pressure levels are used in a variety of applications, while Noise Criteria and Room Criteria are mostly used for indoor spaces.

2.4.1 A and C-weighted Sounds Pressure Levels

Two most commonly used criteria are A and C-weighted sound pressure levels (dBA, and dBC). They are developed based on the equal loudness contours [26]. A-weighting is derived from the 40-phon equal-loudness contour. It has a roll off for both low-frequency and high-frequency contents. Although A-weighting is only

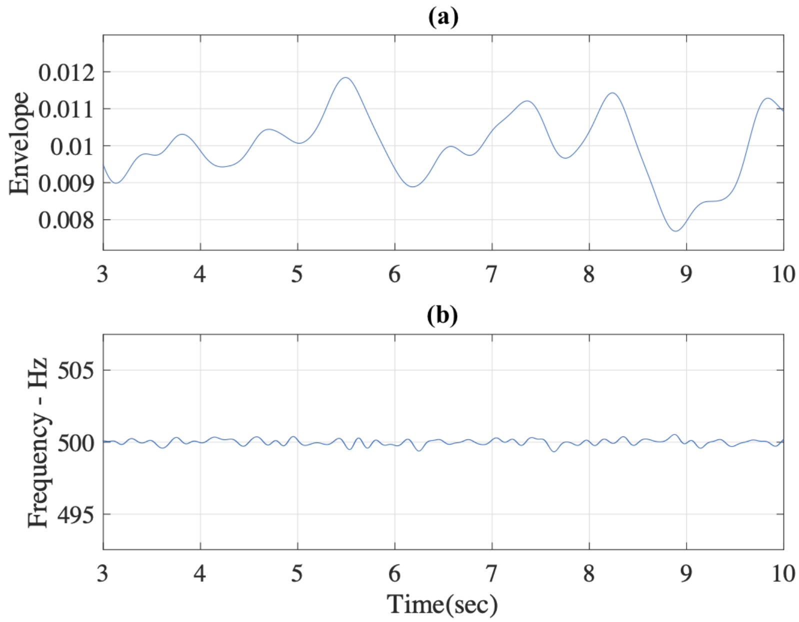

Figure 2.6. Temporal properties of an example tonal component: (a) instantaneous amplitude, (b) instantaneous frequency.

designed for relatively low level sounds, it's widely used for community noise measurement. Different from A-weighting, C-weighting takes more low-frequency content into account as the response of the human ear is flatter at a higher level (100 dB and above). C-weighing is designed to assess the higher level sounds.

2.4.2 Noise Criteria (NC)

Noise Criteria and NC curves [11] (shown in Figure 2.7) are the most widely used criteria for indoor spaces. It takes into account the human response to sound pressure level in different octave bands. The NC curves were published, revised and extended by Beranek [27, 28]. In low frequencies, the expanded NC curves have relatively high levels, and the differences between NC curves are smaller. In this research, the NC tangent method was used for evaluating octave band noise spectrum of the sound for simplicity and universality. This method picks the highest NC rating in which the octave levels lie. Another method for evaluating sounds' noise criteria is based on Speech Interference Level (SIL). The SIL-based NC rating is the average of the levels in the 250, 500, 1000, and 2000 Hz octave bands. Two methods would give different NC ratings for a sound.

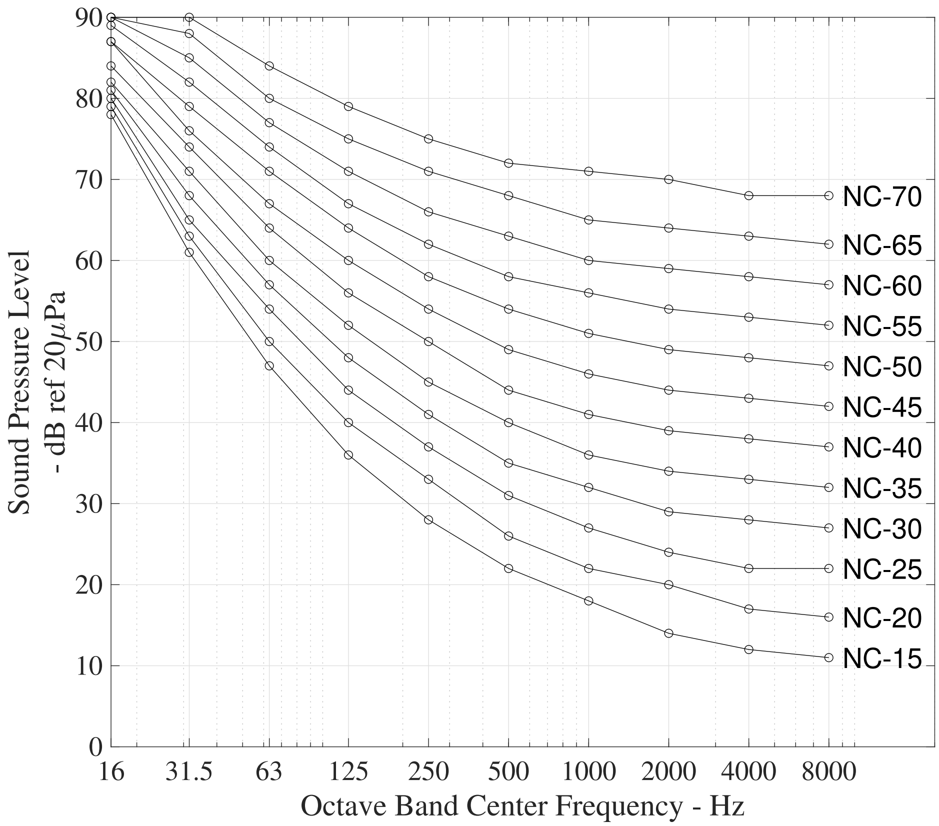

Figure 2.7. Noise criteria (NC) curves.

2.4.3 Room Criterion (RC)

The room criterion (RC) [11] curves (shown in Figure 2.8) were developed to guide the design of heating, ventilating, and air-conditioning (HVAC) systems in building. The RC curves were derived from a study of the noise in 68 offices where there were no complaints by the occupants about the HVAC noise [29]. A-weighted sound pressure levels of these office noises were 40 – 50 dBA. The slopes of RC curves are -5 dB per octave band except for the low-frequency curves. The levels of RC curves are designed lower for the low-frequency components.

The RC value is the average of 500, 1000, 2000 Hz bands. An additional R indicating rumble will be added to the RC value if the level in any of the octave bands at and below 500 Hz exceeds the corresponding RC curve more than 5 dB. An additional H indicating hiss will be added to the RC value if the level in any of the octave bands at and above 1000 Hz exceeds the corresponding RC curve more than 3 dB. A letter N for neutral will be used if the sound is neither labeled rumble or hiss.

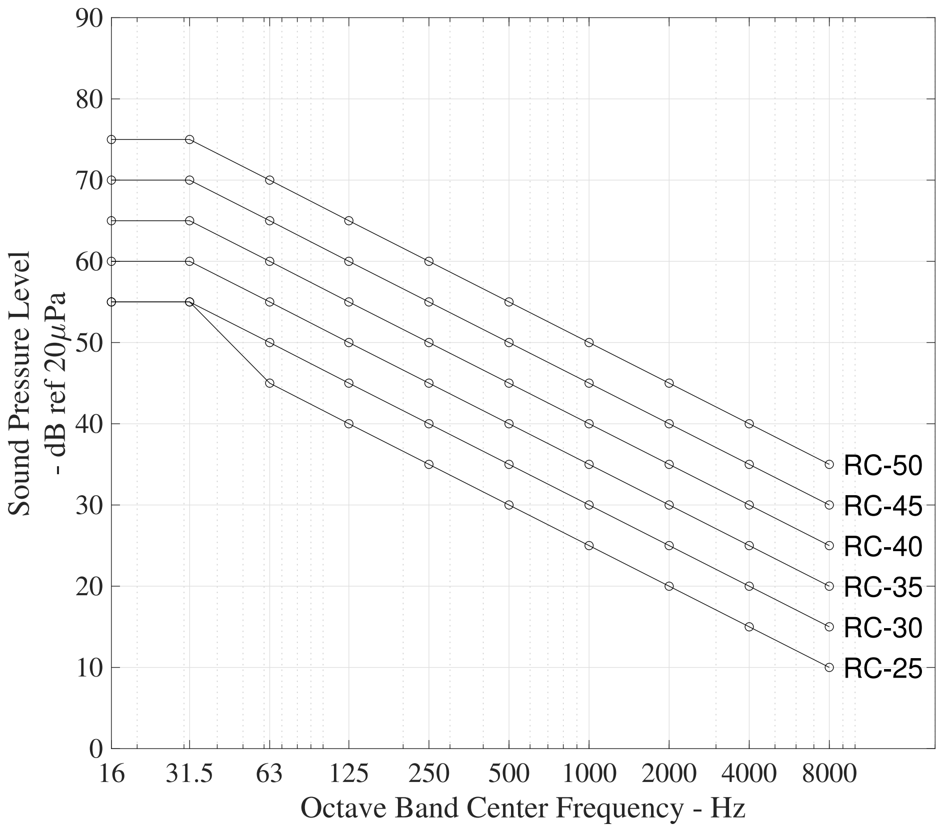

Figure 2.8. Room criteria (RC) curves.

2.5 Sound Quality Metrics

Sound quality metrics such as loudness, tonality, sharpness and roughness are developed to quantify different sound attributes. The metric values can be used as a criterion or as a predictor for annoyance model. Details of sound quality metrics used in the research are described in this section.

2.5.1 Loudness

Loudness is a metric related to sound strength perceived by a person. The perception of the loudness grows logarithmic with the intensity of a sound. Steven's power law states the relationship between the stimuli intensity (I) and perceived loudness (L):

$$L = kI^p \tag{2.4}$$

k, p are constant. Based on this relationship, several models, such as Zwicker Model [30] and Moore & Glasberg Model [31] have been proposed to predict loudness better by considering frequency selectivity, frequency masking and temporal masking. Zwicker's model and Moore & Glasberg's model to predict stationary loudness are standardized in ISO 532-1 [2] and ISO 532-2 [32]. Zwicker's model has a constant low frequency bandwidth while the low frequency bandwidth of Moore & Glasberg's model decreases. The low frequency bandwidth of Moore & Glasberg loudness model follows a similar trend as $1/3^{rd}$ octave bands. Two loudness models are slightly different at higher frequencies as well.

2.5.2 Tonality

The perception of a tonal component is not easy to measure. It's related to various factors such as the frequency of the tone, the prominence of the tone above the broadband noise, the bandwidth of the tonal feature, the distribution of the harmonic tones, and the combination of other non-harmonic tones. Tonal metrics have been developed to describe how tonal noise is perceived. Tone-to-Noise Ratio (TNR) [12], Prominence

Ratio (PR) [13], Tonal Audibility (ΔL_{ta}) [14] are three simple and commonly used tonality model. Different from TNR, PR and ΔL_{ta}, Aures tonality model (T) [15] is more complex as it considers the effect of bandwidth, center frequency, prominence of the tone and the ratio of additional loudness due to tone. TNR, PR and ΔL_{ta} are calculated separately for all identified tones, but only the highest tonality value is used, while Aures tonality model sums the contribution of all identified tones. This section describes four commonly used tonality metrics.

Tone-to-Noise Ratio

Tone-to-Noise Ratio (TNR) is the ratio of tone power to the masking noise power in the critical band centered on that tone in decibels:

$$TNR = 10log_{10}(W_t/W_n)\ dB \tag{2.5}$$

where W_t is the power of the tone, W_n is the power of the masking noise. The power of the masking noise is determined by removing the power of tone from the total power within the critical band:

$$W_n = (W_{tot} - W_t)\frac{\Delta f_c}{\Delta f_{tot} - \Delta f_t} \tag{2.6}$$

where W_{tot} is the total power within the critical band centered on the tone, Δf_{tot} is the width of frequency used to compute W_{tot}, Δf_c is the critical bandwidth of the tonal component. Δf_c is defined by:

$$\Delta f_c = 25.0 + 75.0\left[1.0 + 1.4\,(f_t/1000)^2\right]^{0.69}\ \ Hz \tag{2.7}$$

The theoretical critical bandwidth (Δf_c) is calculated from equation, while the actual critical bandwidth (Δf_{tot}) is slightly different. Δf_{tot} is determined by the frequencies in spectrum that are closest to the start and the end of the critical band f_1 and f_2:

$$f_1 = -\frac{\Delta f_c}{2} + \frac{\sqrt{(\Delta f_c)^2 + 4f_t^2}}{2} \tag{2.8}$$

$$f_2 = f_1 + \Delta f_c \tag{2.9}$$

If multiple tones exist in a critical band, the power of tones would be removed from masking noise. The power of tones would be added if the tones are sufficiently close. A tone is considered prominent if its Tone-to-Noise Ratio is greater than 6 dB.

Prominence Ratio

Prominence Ratio (PR) is the ratio of power contained in the critical band centered on the tone to the average power contained in the two adjacent critical bands in decibels:

$$PR = 10log_{10} \frac{W_M}{(W_L + W_U)/2} \ dB \tag{2.10}$$

where W_M is the power in the critical band centered on the tone, W_L and W_U are the power in two adjacent critical band. The start (f_1) and the end (f_2) of the critical band centered on the tone is computed with Equation 2.7, 2.8, 2.9, while the start and the end for upper and lower critical band are given as:

$$f_{1,L} = \frac{f_1^2}{f_2}; \ f_{2,L} = f_1 \tag{2.11}$$

$$f_{1,U} = f_2; \ f_{2,U} = \frac{f_2^2}{f_1} \tag{2.12}$$

A tone is considered prominent if its Prominence Ratio is greater than 7 dB. When multiple tones exist in the sound, the Prominence Ratio would be calculated for each tone individually. The highest Prominence Ratio would be chosen to use.

Tonal Audibility (Joint Nordic Method)

Tonal Audibility (ΔL_{ta}) assess the prominence of tones by calculating the audibility of tones within a critical band centered at the frequency of the tone:

$$\Delta L_{ta} = L_{pt} - L_{pn} + 2 + log_{10} \left(1 + \left(\frac{f_c}{502} \right)^2 \right) \ dB \tag{2.13}$$

where f_c is the frequency of the identified tone, L_{pt} is the total sound pressure level of tones in the critical band centered on f_c, L_{pn} is the total sound pressure level of the masking noise in the critical band. The term related to the f_c can be considered as a frequency weighting in the tonality model. If there are multiple tones exist in the critical band, the total sound pressure level of tones L_{pt} is computed by:

$$L_{pt} = 10log_{10}\Sigma 10^{\frac{L_{pti}}{10}}\ dB \tag{2.14}$$

where L_{pti} is the sound pressure level of i^{th} tone in the critical band. The masking noise level (L_{pn}) is computed with the average sound pressure level within the band:

$$L_{pn} = L_{pn,avg} + 10log_{10}\left(\frac{CBW}{EAB}\right) \tag{2.15}$$

where EAB is the effective analysis bandwidth which is 1.5 times the frequency resolution if a Hann window is applied to estimate the spectrum. CBW is the critical band width. It depends on the center frequency of the critical band. CBW is 100 Hz for center frequencies (f_c) ranges from 50 to 500 Hz and is $0.2f_c$ for center frequency above 500 Hz.

Aures Tonality

Aure proposed a tonality model based on psychoacoustic perceptions. The model is a function of bandwidth, center frequency, prominence of the tone and the ratio of additional loudness due to tone. The component corresponding to bandwidth is:

$$w_1(\Delta z) = \frac{0.13}{\Delta z + 0.13} \tag{2.16}$$

where Δz is the bandwidth of the tonal component in Bark. The component corresponding to center frequency is:

$$w_2(f) = \left(\frac{1}{\sqrt{1 + 0.2\left(\frac{f}{700} + \frac{700}{f}\right)^2}}\right)^{0.29} \tag{2.17}$$

where f is the frequency of identified tonal component in Hz. This term can be considered as a frequency weighting in the model. This weighting rolls off at low frequencies and at high frequencies, and is a maximum at 700 Hz, which is the frequency where people are most sensitive to the tonalness of a sound. The component corresponding to the prominence of the tone is:

$$w_3(\Delta L) = \left(1 - e^{-\frac{\Delta L}{15}}\right)^{0.29} \tag{2.18}$$

where ΔL is the excess level of the tonal component in dB. The excess level of the ith tonal component with center frequency f_i is:

$$\Delta L_i = L_i - log_{10}\left\{\left[\Sigma_{k \neq i}^{n} A_{Ek}(f_i)\right] + E_{Gr}(f_i) + E_{Hs}(f_i)\right\}\ dB \tag{2.19}$$

where $A_{Ek}(f_i)$ is the excitation level produced by ith tonal component, $E_{Gr}(f_i)$ is the noise intensity present in the critical band centered on ith tonal component, and $E_{Hs}(f_i)$ is the corresponding hearing threshold at f_i. The overall tonal weighting w_T is computed by:

$$w_T = \sqrt{\Sigma_{i=1}^{N}\left[w_1(\Delta z_i) w_2(f_i) w_3(\Delta L_i)\right]^{\frac{2}{0.29}}} \tag{2.20}$$

Tonal weighting w_T integrates the contribution of all identified tonal components. The last term based on the ratio of additional loudness due to tone is:

$$w_{Gr} = 1 - \frac{N_{Gr}}{N} = \frac{N - N_{Gr}}{N} \tag{2.21}$$

where N_{Gr} is the loudness of the tone removed noise component, and N is the total loudness of the sound. Aure Tonality (T) is then defined by:

$$\begin{aligned} T &= c \cdot w_T^{0.29} \cdot w_{Gr}^{0.79} \\ &= c \cdot \left\{\sqrt{\Sigma_{i=1}^{N}\left[w_1(\Delta z_i) w_2(f_i) w_3(\Delta L_i)\right]^{\frac{2}{0.29}}}\right\}^{0.29} \cdot w_{Gr}^{0.79} \end{aligned} \tag{2.22}$$

where c is a calibration factor that ensures Aures Tonality (T) equals 1 for a 1 kHz, 60 dB pure tone.

2.5.3 Sharpness

Sharpness is a measure of spectrum balance. Sounds with more high-frequency content would sound sharper. Sharpness metrics can be computed from the specific loudness. The standard sharpness computation only supports specific loudness input from Zwicker method. Swift have proposed a method to take specific loudness from Moore and Glasberg method as input for sharpness calculation [33]. Sharpness formulation with Zwicker Loudness input is given by:

$$S_Z = C\frac{\int_0^{24} N_Z'(z_1)g(z_1)z_1dz_1}{N_Z}\ acum \tag{2.23}$$

where z_1 is the critical band rate in Bark, N_Z is overall Zwicker loudness, N_Z' is specific Zwicker loudness, and $g(z_1)$ is a weighing factor. c is the constant that ensures a narrow band noise with 1 kHz center frequency, 160 Hz bandwidth and with a sound pressure level 60 dB has a sharpness of 1 acum. Weighting factors ($g(z_1)$) are different for different sharpness models. In order to use specific loudness input from Moore and Glasberg method, the ERBN-number scale in Cams z_2 (used in Moore and Glasberg method) has to be transformed into the critical band rate in Bark scale z_1 (used in Zwicker method). Sharpness formulation can be rewritten with Moore and Glasberg Loudness:

$$S_{M\&G} = C\frac{\int N_{M\&G}'(z_2)g(z_1(z_2))z_1(z_2)dz_2}{\int N_{M\&G}'(z_2)dz_2}\ acum \tag{2.24}$$

where z_2 is the critical band rate in Cams, $N_{M\&G}'(z_2)$ is specific loudness from Moore and Glasberg Loudness, and $z_1(z_2)$ is a mapping from frequency in Cams (z_2) to frequency in Bark (z_1). The weighting factor for the von Bismark Sharpness model [34] is:

$$g(z) = \begin{Bmatrix} 1 & for\ z \leq 16 \\ 0.066e^{0.171z} & forz\ \ > 16 \end{Bmatrix} \tag{2.25}$$

Aures Sharpness [15] is another type of model that emphasizes high frequencies. The weighting factor for the Aures Sharpness model is:

$$g(z) = 0.078\frac{e^{0.0171z}}{z} \cdot \frac{N}{ln(0.05N+1)} \tag{2.26}$$

Compared with von Bismark sharpness, Aures' Sharpness is more correlated with sound level (loudness).

2.5.4 Roughness

Roughness (R) is a function of rapid changes in loudness that cannot be tracked. The roughness sensation reaches a maximum if the rate of fluctuation is around 60 to 70 times per second. Zwicker proposed a roughness model [35]:

$$R = 0.3 f_{mod} \int_0^{24} \Delta L(z) dz \ asper \tag{2.27}$$

where z is critical band rate in Bark, f_{mod} is the modulation frequency in kHz, and $\Delta L(z)$ is the modulation depth of specific loudness at critical band rate z after the temporal filtering. $\Delta L(z)$ can be approximated by:

$$\Delta L(z) = 20log_{10}\left(\frac{N'_{max}(z)}{N'_{min}(z)}\right) \ or \ \Delta L(z) = 20log_{10}\left(\frac{N'_1(z)}{N'_{99}(z)}\right) \tag{2.28}$$

where $N'_{max}, N'_{min}, N'_1, N'_{99}$ are maximum specific loudness, minimum specific loudness, specific loudness exceeds 1% of the time, and specific loudness exceeds 99% of the time. Roughness for a 1 kHz, 60 dB tonal sound with 100%, 70 Hz amplitude modulation is 1 asper.

2.6 Annoyance Models

In order to properly account for the effect of tonality, penalties for tonal sounds were introduced to level metrics such as loudness and A-weighted sound pressure level. Some models were developed or modified with tonality metrics. Penalty based models and annoyance models with tonal metrics are described in this section.

2.6.1 Penalty Based Models

Sound Quality Indicator

Sound Quality Indicator (SQI) [17] is a tone penalized level metric. It is designed for air-conditioning and refrigeration equipment. The SQI calculation is based on the Perceived Noise Level calculation procedure and Zwicker loudness. It takes one-third octave band data as input. Whenever the level of one-third octave band data exceeds more than 1.5 dB to the average of two adjacent bands, the level would be adjusted by:

$$L' = L - P + 10log_{10}\left(10^{D+B} + 1\right) \tag{2.29}$$

where L' is the tone adjusted sound level, L is the measured sound level, and P is the projection above the average of tow adjacent bands.

$$\begin{aligned} &D = log_{10}(10^{P/10} - 1) \\ &B = 76.28 - 75.74Y + 29.98Y^2 - 6.14Y^3 + 0.69Y^4 - 0.04Y^5 + 0.001Y^6 \\ &Y = ln(F) \end{aligned} \tag{2.30}$$

where F is the octave band center frequency. Adjusted levels are converted to rating indices based on the conversion table. The formula for SQI is:

$$\begin{aligned} &SQI = K + 10log\Sigma_{i=100Hz}^{10000Hz} I_i \\ &K = 11.83888 - 4.94569lnX + 0.614812(lnX)^2 \\ &X = \frac{\Sigma I_i}{I_M} \end{aligned} \tag{2.31}$$

where I_m is the maximum rating index from 100 to 10000 Hz. SQI is a level metric computed from tone corrected one-third octave band. It can be calculated with either sound power or sound pressure input.

Danish Environmental Standard

Danish Environmental Standard [14] assumes that the perception of tonalness starts at a low prominence level and saturates at a high prominence level. Tonal Audibility is used as a measure of tonalness in this method. The A-weighted sound pressure level is corrected for tonal sounds by a penalty k:

$$\begin{aligned} &k = 0\ dB && for\ \Delta L_{ta} < 4\ dB \\ &k = \Delta L_{ta} - 4\ dB && for\ 4\ dB \le \Delta L_{ta} \le 10\ dB \\ &k = 6\ dB && for\ \Delta L_{ta} > 10\ dB \end{aligned} \tag{2.32}$$

2.6.2 Psychoacoustic Annoyance Model

Zwicker and Fastl proposed a psychoacoustic annoyance model used in transportation noise. The model incorporates loudness, roughness, fluctuation strength and sharpness. The model is given by:

$$PA = N_5 \left[1 + \sqrt{w_x^2 + w_{FR}^2} \right] \tag{2.33}$$

where

$$w_s = \begin{Bmatrix} 0.25(S - 1.75)log_{10}(N_5 + 10) & for & S > 1.75 \\ 0 & for & S < 1.75 \end{Bmatrix} \tag{2.34}$$

$$w_{FR} = \frac{2.18}{(N_5)^{0.4}}(0.4F + 0.6R) \tag{2.35}$$

N_5 is loudness exceeded 5% of the time, S is sharpness, F is fluctuation strength, and R is roughness. In Equation 2.34, the model introduced a lower limit to sharpness based on the assumption that the sharpness metric does not play a role in annoyance below a certain value. The Zwicker's model doesn't contain a metric related to tonalness which may lead to an under-estimation to some tonal sounds. More and Davies [36] have modified the Psychoacoustic Annoyance by considering tonality metric in the model:

$$PA_{mod} = N_5 \left[1 + \sqrt{\gamma_0 + \gamma_1 w_s^2 + \gamma_2 w_{FR}^2 + \gamma_3 w_T^2} \right] \tag{2.36}$$

where the additional tonality term w_T is defined as:

$$w_T^2 = \left[\left(1 - e^{-\gamma_4 N_5}\right)^2 \left(1 - e^{-\gamma_5 T_5}\right)^2 \right] \tag{2.37}$$

where T_5 is Aures Tonality exceeded 5% of the time. The coefficients were developed based on the subjective tests results.

2.7 Tokita and Nakamura's Low Frequency Noise Characterization

Nakamura and Tokita have studied the perception of low-frequency sounds in the laboratory environment [37]. They developed contour maps based on levels in $1/3^{rd}$

octave bands from people's responses to low frequency sounds. The contours (shown in Figure 2.9) represent boundaries between, e.g., detectable and annoying, annoying and displeasing, displeasing and oppressive/detect vibration. These low-frequency threshold contours were used to identify outliers in the annoyance in Chapter 6.

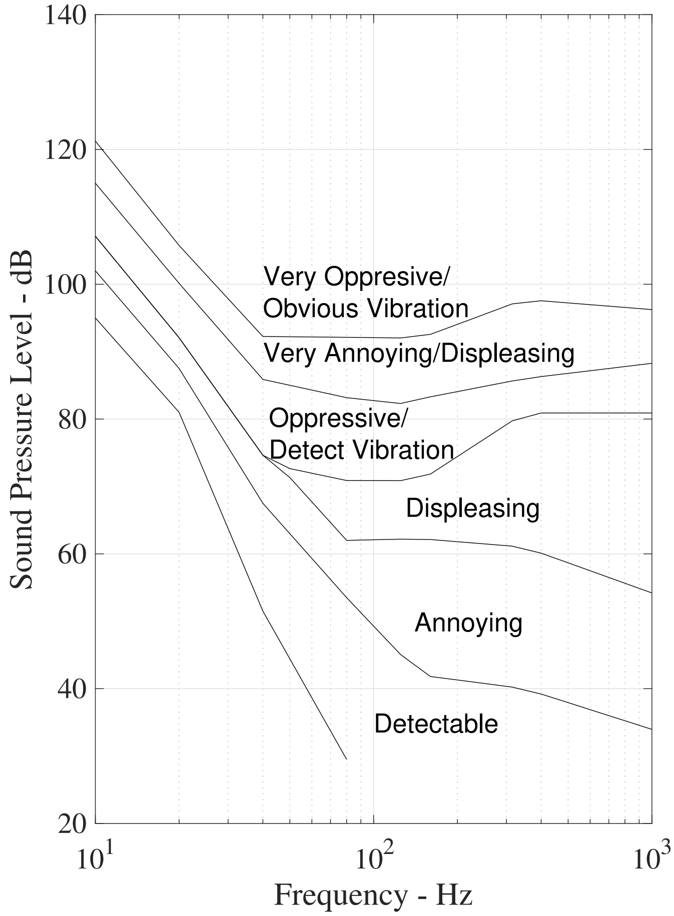

Figure 2.9. Nakamura and Tokita's low frequency noise threshold curves with different regions of feelings.

2.8 Summary of Office Noise Characterization

A number of sound attributes have been described including spectral and temporal properties of the tonal office noise, sound pressure evaluation criteria, sound quality metrics, annoyance models, and low-frequency noise threshold curves. The methodologies to simulate building noises with different sound attributes are described in next chapter.

3. OFFICE NOISE SIMULATION

Sound signals simulated based on recording in typical commercial building environments are used in subjective tests (described in Chapter 5) instead of direct recordings. The main advantages of using simulated signals are: (1) the signals played in subjective tests are not limited by the recording length, (2) it is easier to modify signal characteristics so that sounds with a wide range of values in different psychoacoustic metrics can be obtained. In the simulation, the broadband and tonal components are extracted and simulated separately. The realism of combined sounds was checked by the ASHRAE advisory team of the project. Two more methods were developed to modified tonal sounds to have required sound attributes.

3.1 Measurements Used in Simulation

Simulated sounds were obtained based on 27 recordings in various office spaces on the campus of Purdue University and 36 more from the ASHRAE advisory team of the project. The details of these measurements are described in Appendix A.1. Among these measurements, some sounds with natural broadband or dominant tonal components were chosen for necessary analysis in generating simulated sounds. Four typical measurements' power spectral densities (PSDs) are presented in Figure 3.1 with 0.5 Hz frequency resolution, Hann window was applied to each segment with 50% overlap between adjacent segments.. The signal lengths are between 17 to 111 seconds. The PSD of tonal sound from the advisory team was estimated with 18 segments averaged, while PSDs of other sounds were estimated with 110 segments averaged. Figure 3.1 (a) illustrates the spectrum of a typical office broadband noise. The sound was measured in an office in the HERL Building on Purdue University campus, which was later selected as the reference to simulate the broadband components. The signal presented in Figure 3.1 (b) was measured in an open 36’ by 87’ high-bay area in the Ray W. Herrick Lab on Purdue campus. A set of harmonics (120 + 240 Hz) could be identified from the spectrum. Sounds with strong tonal components can also be observed from other recordings, two examples are shown in Figure 3.1 (c) and (d). In Figure 3.1 (d).

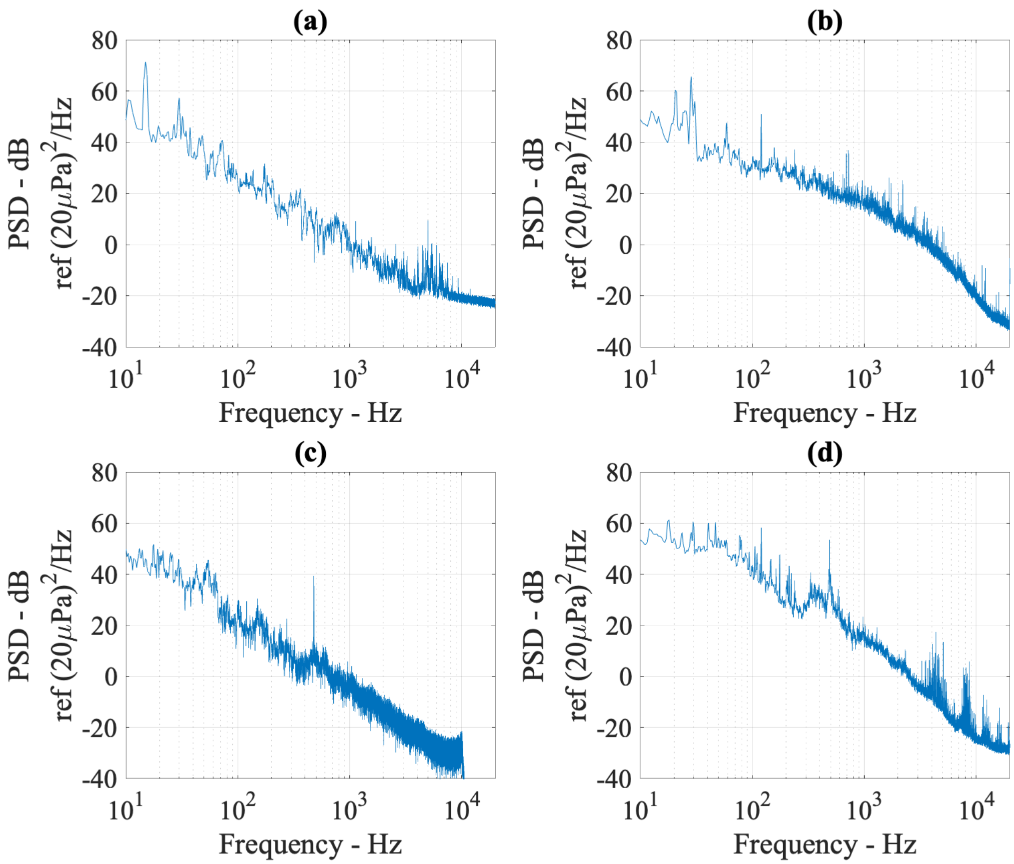

Figure 3.1. Power spectral densities of four building measurements. (a) office noise in HERL Building, (b) general lab area noise in HLAB Building, (c) tonal recording provided by ASHRAE advisory team , (d) indoor chiller noise in PFS Building.

3.2 Sound Decomposition and Broadband Simulation

In the signal simulations, broadband and narrowband components were extracted and simulated separately. The component extraction method and broadband simulation are described in this section. Based on the measured tonal sounds, tonal components are removed by first modeling their contributions to signal and then removing them from the original signal. The method of tonal components removal applied in the current work is similar to that developed by Sung [38]. The remaining component of this process is the broadband background noise. After the extraction of the broadband component, a method was developed to simulate the broadband component so that the signals played in the subjective tests are not limited by recording time.

3.2.1 Approximated Removal of Tonal Components

To remove tonal components from recordings, their contributions need to be modeled first. This is done by extracting and analyzing the temporal properties (i.e. time varying amplitudes and frequencies of the tones) of tonal components. Figure 3.2 (a) illustrates the tonal components extraction process, and Figure 3.2 (b) shows the procedures to extract instantaneous amplitude $A(t)$ and instantaneous phase $\phi(t)$ of a tonal component. The detailed calculation is described in Chapter 2.3.1.

The sampling frequency of the recording is 48 kHz. As most tonal components from measurements are in the 0 to 2000 Hz range, a 6^{th} order Butterworth low pass filter with a 2 kHz cut-off frequency was implemented. Implementing low-pass filter prevented the aliasing effect when the signal was down-sampled to 8 kHz. Down sampling was conducted with a factor of 6 to ease the bandpass filter design. A series of 8^{th} order Bandpass Butterworth filters with 10 Hz bandwidth were then applied to extract tonal components. All filters were implemented by forward and backward filtering so that no additional phase components were introduced. Tonal components were up-sampled to original sampling frequency (f_s = 48 kHz) at the end of this process.

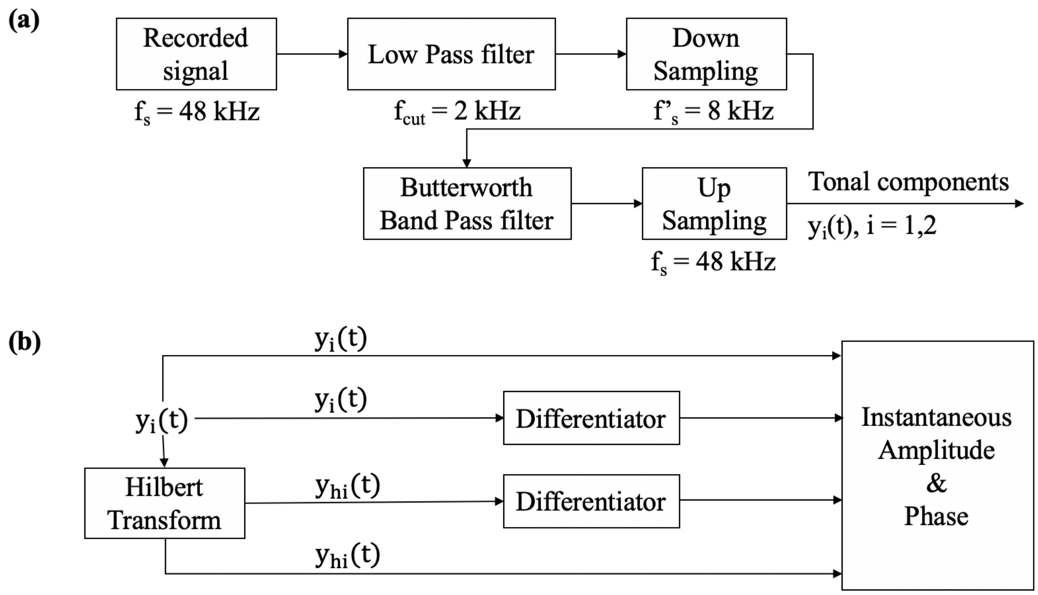

Figure 3.2. Flowchart of the tonal components extraction process and instantaneous amplitude, phase and frequency calculation.

As there was no phase change introduced in the process of extracting tonal components, different from the approach used by Sung in HVAC&R noise decomposition, the extracted tonal components in the current work could be simply modeled as $y_i(t) = A_i(t)cos(\phi_i(t))$ instead of $y_i(t) = A_i(t)cos(\phi_i(t)) + B_i(t)sin(\phi_i(t))$, where $A_i(t)$ and $\phi_i(t)$ are the instantaneous amplitude and phase of the ith tonal component (described in Section 2.3.1). In a typical building noise recording, the amplitude of a tonal component doesn't change much through time. Thus, a constant amplitude was used in modeling the tonal component. Instantaneous phase information turns out to be important to model some slight changes in tonal component's frequency. As the building noise can be considered as a combination of tonal and broadband components, the measured sound pressure history is expressed as:

$$\begin{pmatrix} p(t_1) \\ p(t_2) \\ ... \\ p(t_N) \end{pmatrix} = \begin{pmatrix} cos(\phi_1(t_1)) & cos(\phi_2(t_1)) & \cdots & cos(\phi_{np}(t_1)) \\ cos(\phi_1(t_2)) & cos(\phi_2(t_2)) & \cdots & cos(\phi_{np}(t_2)) \\ \vdots & \vdots & \ddots & \vdots \\ cos(\phi_1(t_N)) & cos(\phi_2(t_N)) & \cdots & cos(\phi_{np}(t_N)) \end{pmatrix} \begin{pmatrix} A_1 \\ A_2 \\ ... \\ A_{np} \end{pmatrix} + \begin{pmatrix} n(t_1) \\ n(t_2) \\ ... \\ n_{np}(t_N) \end{pmatrix} \quad (3.1)$$

In Equation 3.1, np is the number of tonal components, N is the number of time samples in the recording, $p(t_i)$ and $n(t_i)$ are the measured sound pressure history and broadband noise at time t_i, $\phi_m(t_i)$ represents the instantaneous phase of mth tonal component at time t_i. The amplitudes of tonal components could then be estimated by the least square error method.

After the removal of tonal component from the recording, a natural broadband component is obtained. Some examples of tone extraction results are shown in Figure 3.3. It is observed that all prominent tones can be successfully removed from the signals recorded in different building environments, and the broadband components can be extracted.

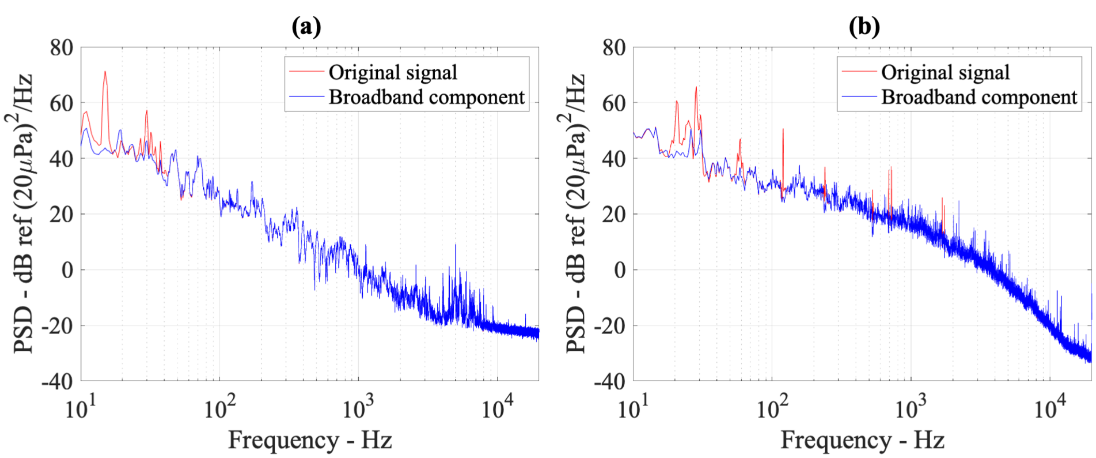

Figure 3.3. Power spectral densities of the recording (blue line) and the tone-removed recording (red line). Recordings were measured in (a) office in HERL Building, (b) general lab area in HLAB Building.

3.2.2 Modify and Simulate Broadband Components

With the extracted broadband component from the recording, a method was developed to simulate and modify the broadband components. Figure 3.4 describes the procedures to simulate broadband noises.

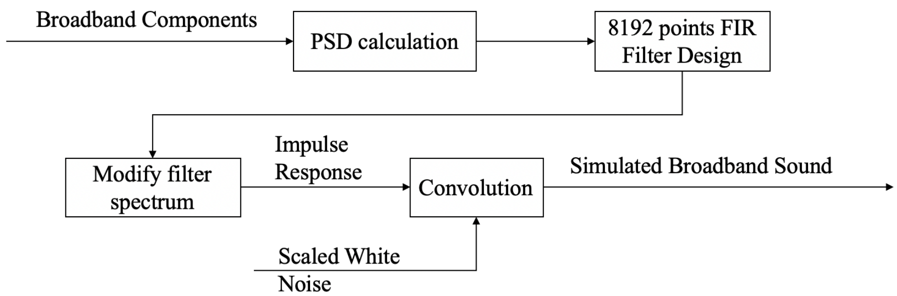

Figure 3.4. Flowchart of broadband simulation process based on a neutral broadband noise.

The PSD of the broadband was first estimated with an 8196-point Hann window and 50% overlap between adjacent segments. An 8196-point zero-phase filter was then designed by specifying the filter frequency response to be the square root of the broadband signal PSD. Based on the feedback from the advisory team of the project, an 8196-point filter is long enough to reproduce a similar broadband sound. Some modifications were applied to the filter frequency response to remove identified artifacts. An example of the designed filter properties was shown in Figure 3.5. There are some non-stationary equipment sounds at around 4 kHz, some narrow band features in the middle frequencies (around 200 Hz), and some quantization noises at the high-frequency end. The filter frequency response was modified to remove these undesired artifacts in recordings. The impulse response of the designed filter was obtained by a direct inverse Fourier transform. A Hann window was applied to the filter response to ensure two tails of the filter impulse response decay to zero.

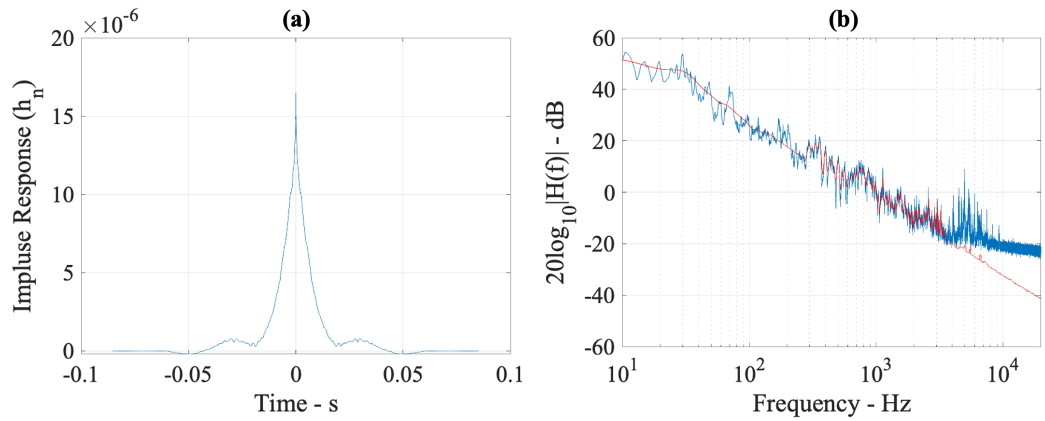

Figure 3.5. Properties of the broadband filter. (a) the impulse response of the filter. (b) the frequency response function of the filter (red line) and the power spectral density of the reference broadband sound (blue line). Reference broadband recording was measured in the office in HERL Building.

The filter was then implemented to shape a white noise. Four broadband noises were simulated based on different indoor recordings. Both simulated and recorded broadband noises were scaled to different levels (35, 45, 55 dBA), mixed together, listened and evaluated by the advisory team of the project. The simulated broadband noises were rated almost as natural as the recorded signals. The simulated broadband noise based on a recording in an office in the HERL Building (shown in Figure 3.6) was selected to be used as a baseline of broadband signal for the test sounds as it is rated, among all simulated recordings, to be the closest to atypical office background noise, and it contains no significant undesired components. The selected broadband has a roll-off rate of 5.6 dB/Octave.

Although the simulated signals might sound slightly different from the recorded one, they are still used in subjective tests, mainly because they are not limited by the recording length when playing in subjective tests. Thus, ambient background noises could be played throughout the test session so that subjects could acclimatize to the background level. This is felt to be preferred because the PBE Lab background noise levels are extremely low, even though the replayed sound is at the desired NC level, it

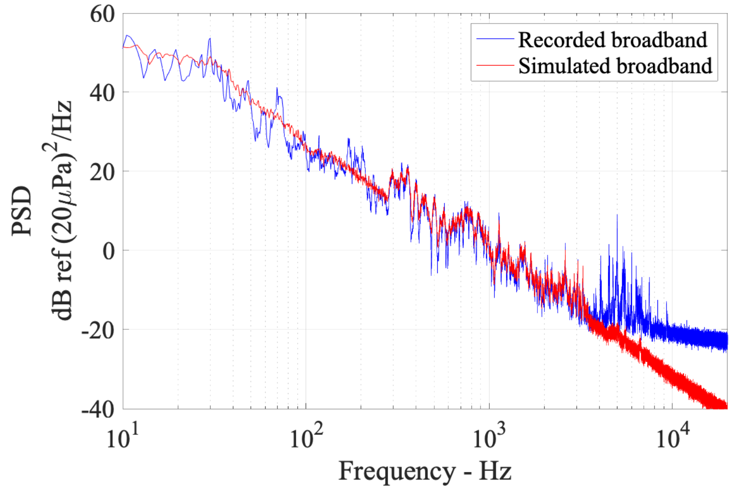

Figure 3.6. Power spectral densities of the broadband component from recording (blue line) and the simulated broadband component (red line).

sounds loud in contrast. For example, a 90-minute NC-20 ambient broadband noise was generated for Test 2. Simulated broadband noises are also easier to be modified to get rid of artifacts and generate signals with different spectral balance.

3.3 Simulate Tonal Components in Recordings

When simulating tonal components, tones with constant amplitudes and frequencies sound artificial because they are too steady. To resolve this issue, temporal and spectral characteristics of recorded tonal components were first analyzed to investigate what modifications are needed. Two amplitude and frequency modulated (AM & FM) tonal models were developed to add some slight modulations to tonal components. The parameters of modulation were adjusted to match the simulated tone's temporal and spectral characteristics with those of the recorded tone. Based on advisory team's feedback from listening to simulated signals, one modulation model with random amplitude and frequency modulation was used to simulate the tonal components.

3.3.1 Temporal and Spectral Characteristics of Recorded Tones

For signals with strong tones, tonal components were first extracted with a Butterworth bandpass filter. Instantaneous analysis was performed to the tonal components (described in Section 2.3.1). One example of the extracted tonal component and its instantaneous characteristics were shown in Figure 3.7 and 3.8. From the results presented in these figures, no significant periodic modulations can be observed in recorded signals. This suggests that random (non-periodic) modulations may need to be included in simulations to generate signals with similar tone characteristics as the recordings.

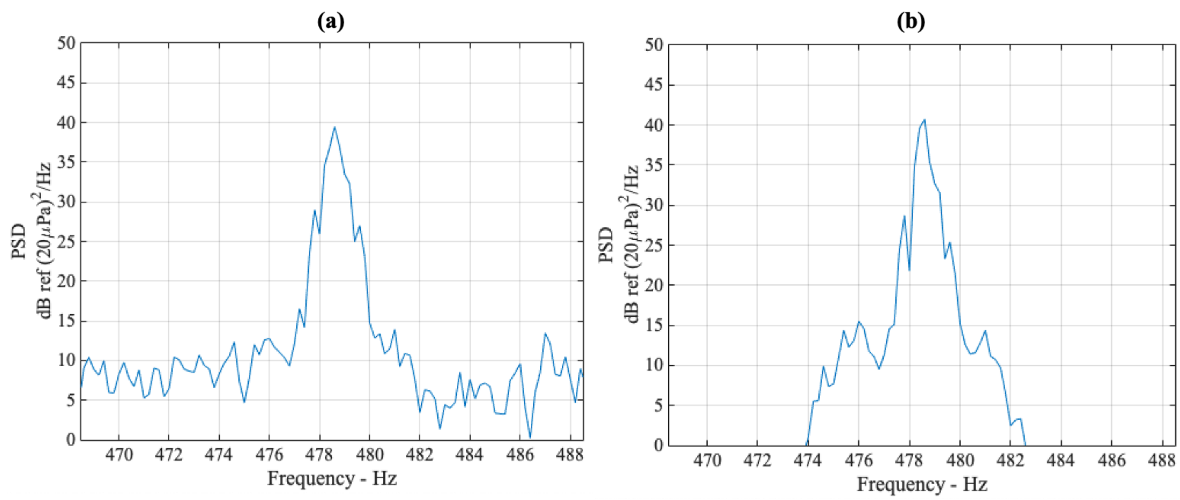

Figure 3.7. Extracted tonal components (with a center frequency 478 Hz) from the recording provided from advisory team. (a) power spectral density of actual signal, (b) power spectral density of band passed actual signal.

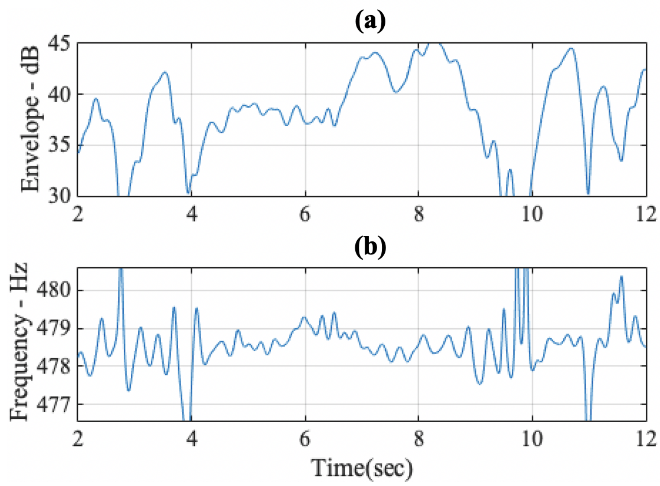

Figure 3.8. Temporal properties of the extracted tone: (a) instantaneous amplitude, (b) instantaneous frequency. Recording provided by the advisory team.

3.3.2 Develop Models to Synthesize Tonal Components

Two AM & FM Model was developed to match both the temporal and spectral properties of the tonal component.

Tonal Model 1

In the Tonal Model 1, low-passed random noises were added directly to both amplitude and frequency to the amplitude and frequency of each tone. It is expressed as:

$$y(t) = (A_0 + A_1 r_A(t))\, sin\left(2\pi f_0 t + 2\pi f_1 \int_0^t r_f(t)dt\right) \tag{3.2}$$

where $r_A(t)$ and $r_f(t)$ are low-passed uniform distributed random signals, $A_1(t)$ and $f_1(t)$ control the amount of amplitude and frequency modulations added to a steady tone. The values of parameters were chosen to match both spectral (shown in Figure 3.9) and temporal characteristics (shown in Figure 3.10) obtained from the analysis of recorded signals. In this example, $A_0 = 0.017, A_1 = 120A, f_0 = 478.6Hz, f_1 = 48Hz,$

the cut-off frequency of two low-passed random signals were chosen to be are 5 Hz for $r_A(t)$ and 2 Hz for $r_f(t)$.

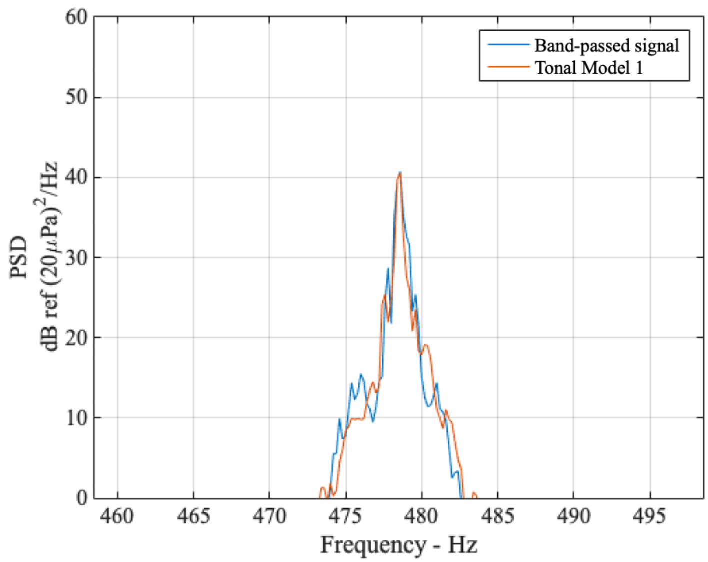

Figure 3.9. Power spectral densities of the band-passed recording (blue line) and the Tonal Model 1 simulated tonal component (orange line).

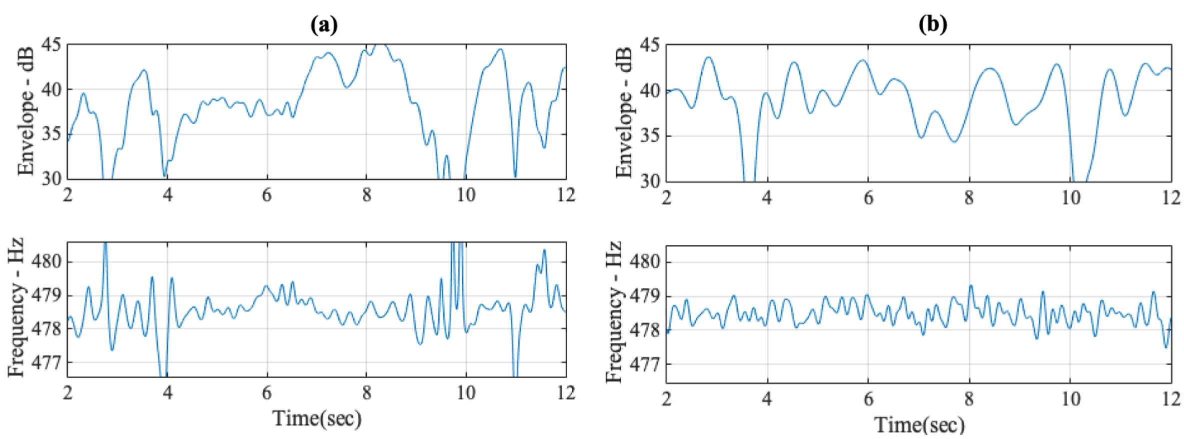

Figure 3.10. Temporal properties of (a) band-passed recording, (b) simulated tonal component (Tonal Model 1).

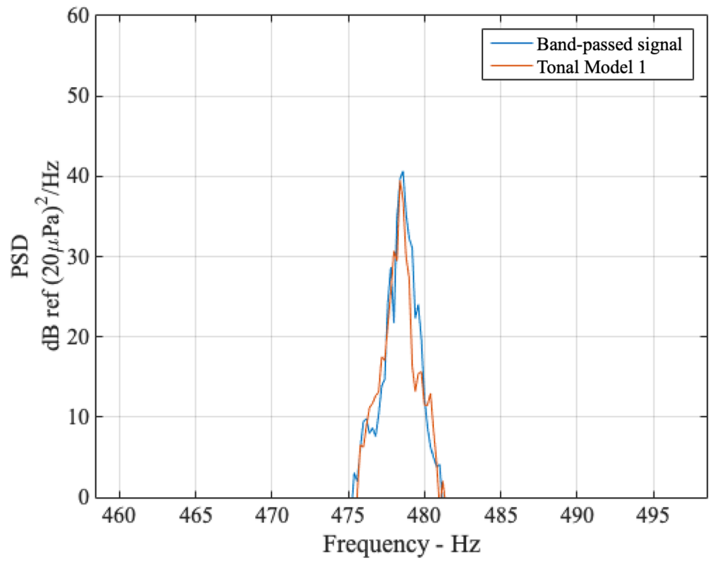

Figure 3.11. Power spectral densities of the band-passed recording (blue line) and the Tonal Model 2 simulated tonal component (orange line).

Tonal Model 2

Different from Tonal Model 1 where random modulation was directly added to the amplitude and frequency, in Tonal Model 2, the added amplitude modulation is a roughly sinusoidal signal, while the added frequency modulation was still controlled by the low-passed random noises. Tonal Model 2 is expressed as:

$$y(t) = \left(A_0 + A_1 sin\left(2\pi f_{A_0} t + 2\pi f_{A_1} \int_0^t r_A(t)dt\right)\right) sin\left(2\pi f_0 t + 2\pi f_1 \int_0^t r_f(t)dt\right) \tag{3.3}$$

where $r_A(t)$ and $r_f(t)$ are low-passed uniform distributed random signals, $A_1(t)$, f_{A_0}, f_{A_1} and $f_1(t)$ control the amount of modulation.The values of parameters were chosen to match both spectral (shown in Figure 3.11) and temporal characteristics (shown in Figure 3.12) of the recorded signals. In this example, $A_0 = 0.017, A_1 = 0.45A, f_{A_0} = 0.4Hz, f_{A_1} = 80Hz, f_0 = 478.6Hzm, f_1 = 48Hz$, the cut-off frequencies of two random noise are both 2 Hz.

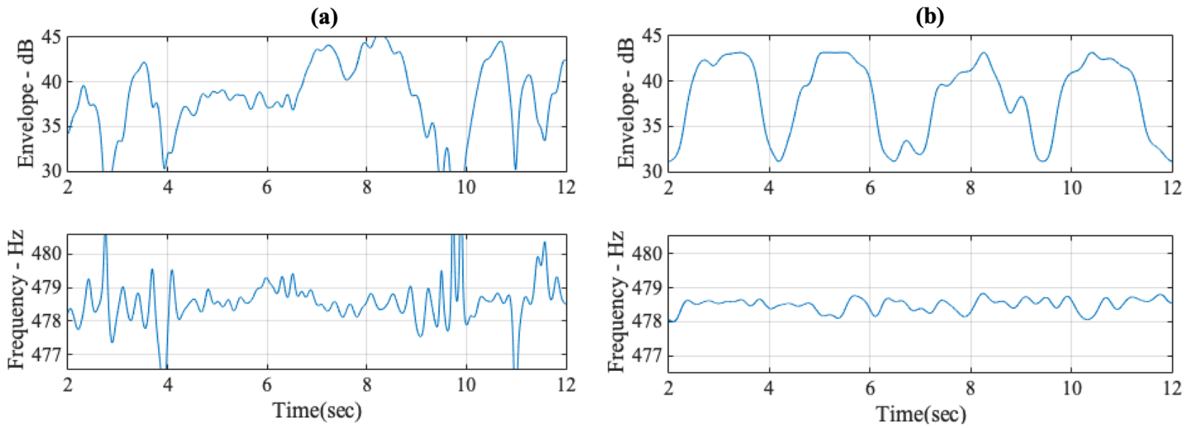

Figure 3.12. Temporal properties of (a) band-passed recording, (b) simulated tonal component (Tonal Model 2).

3.3.3 Model Selection and Realism Check

The realism of simulated tonal components with modulations was checked by the ASHRAE advisory team. The resulting subjective evaluations were used to choose one tonal model and the modulation parameters. Based on the feedback, Tonal Model 1 was selected to simulate tonal components because the fluctuations in Tonal Model 1 are less periodic. The similar process had been gone through for another sound with a strong tonal component (a recording of bathroom fan), compared with two results, the amount of the frequency modulation was at a similar level, while the amount of the amplitude modulation varied. Thus, tonal components with five levels of AM were simulated and evaluated by the advisory team (in Table 3.1). The second case was preferred based on the feedback.

3.4 Combine Broadband and Tonal Components

After successfully generating realistic broadband background and tonal signal components, they were then combined to simulate ready-to-play signals for subjective tests. The broadband compont was scaled to different levels (NC-20, NC-30, NC-40) to

Table 3.1. Simulated tonal components (Tonal Model 1) with different amounts of modulations.

Parameters for Tonal Model 1	$\mathbf{A_1}$	$\mathbf{f_1}$	$\mathbf{f_{cut_A}}$	$\mathbf{f_{cut_f}}$
Case 1 (Bathroom Fan)	$20A_0$	$0.1f_0$	1.5 Hz	$f_0/100$
Case 2	$45A_0$	$0.1f_0$	1.5 Hz	$f_0/100$
Case 3	$70A_0$	$0.1f_0$	1.5 Hz	$f_0/100$
Case 4	$95A_0$	$0.1f_0$	1.5 Hz	$f_0/100$
Case 5 (478 Tonal component)	$120A_0$	$0.1f_0$	1.5 Hz	$f_0/100$

investigate the broadband level effect on annoyance. Prominence ratio (PR) was selected as a criterion to generate signals with different tonal levels. The level of the tonal component was adjusted to achieve a designed prominence ratio within an error of 0.05 dB. For example, the power spectral density of a tonal sound consists of an NC-30 broadband and a 1000 Hz tone with Prominence Ratio 19.0 dB is shown in Figure 3.13.

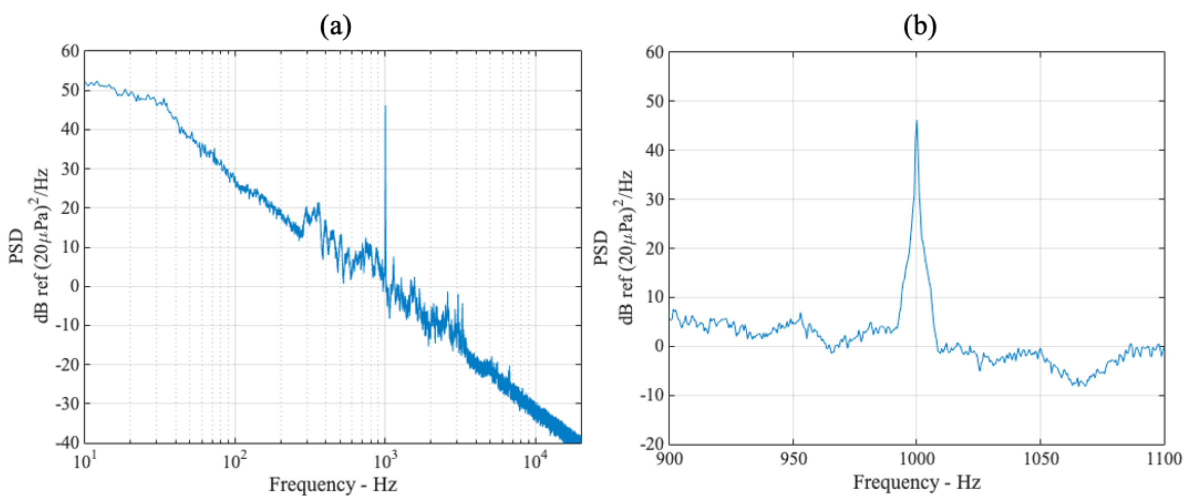

Figure 3.13. Power spectral densities of the simulated sound with NC-30 broadband and a 1000 Hz tone with prominence ratio 19.0 dB. Power spectral density is plotted (a) from 10 - 20000 Hz, on a log scale, (b) from 900 - 1100 Hz, on a linear scale.

In addition to signals with a single tone, harmonic sounds were simulated based on Tonal Model 1 as well. To limit the number of signals played in subjective tests to be within a manageable range, all signals with harmonics in the current work only have one harmonic tone at the twice frequency of the fundamental tone. For the harmonic sounds, the amplitude of the 2^{nd} harmonic is a scale of the amplitude of the fundamental tone. The phase of the 2^{nd} harmonic is twice the phase of the fundamental tone. Tonal sounds with different levels of 2^{nd} harmonic were tested to investigate how presence of harmonics affects annoyance perception.

The ASHRAE advisory team helped to check whether the sounds with the tonal components were too extreme or artificial sounding, which set a tonality range generated test signals.

3.5 Signal Modification

Test signals are modified mainly for two purposes: (1) to slightly change the broadband spectral balance levels (2) to generate a group of sounds with desired metrics (loudness, tonalness, and roughness level). Two sound modification methods used in the research are described in this section.

3.5.1 Broadband Components with Different Spectral Balance Levels

There is some concern that the spectral balance might play a role in people's annoyance perception, NC-30 broadband with a different spectral balance (tilted NC-30 broadband) was simulated. The procedures to generate a tilted broadband follows the broadband simulation procedure (shown in 3.4) except that the designed frequency response of 8192-point broadband filter is tilted in the spectrum to produce a different roll-off rate. Compared with original NC-30 broadband with a 5.6 dB/Octave roll-off rate, this tilted NC-30 broadband signal's roll-off rate is 4.4 dB/Octave. The roll-off rates of both broadband components are typical in a building environment. The spectrums of two NC-30 broadband components are shown in Figure 3.14.

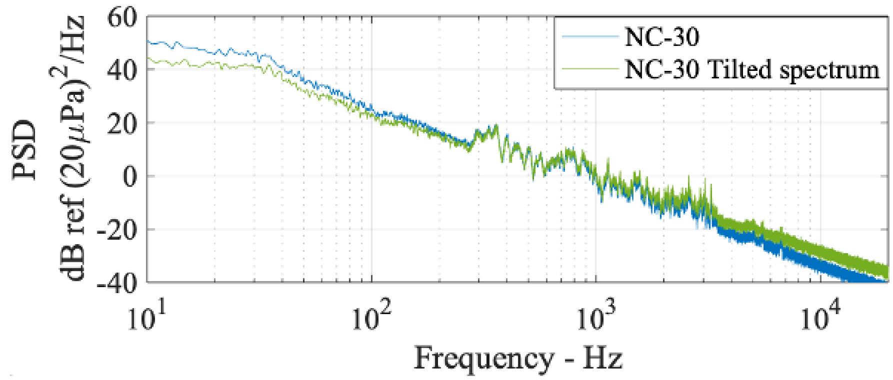

Figure 3.14. Power spectral densities of NC-30 broadband (blue line) and titled NC-30 broadband (green line).

3.5.2 Equal Loudness and Roughness Tonal Sounds

Loudness and roughness are known to be factors affecting annoyance perception, while the scope of this research is more on tonality's effect. To investigate how tonality alone affects annoyance perception, a group of sounds with Zwicker loudness and roughness unchanged, but increasing tonality was generated. Tonality, loudness, and roughness are not independent with each other, change in one metric value would result in changes in the rest two metric values. In Table 3.2, three ways are presented to change tonality, loudness and roughness of a sound.

Table 3.2. Three ways used to modify sounds and corresponding changes of sound attributes (tonality, loudness, roughness).

	Add tones	**Scale down the sound**	**Add modulations to broadband**
Tonality	↑	Not changed	Slightly changed
Loudness	↑	↓	Slightly changed
Rougness	Slightly changed	↓	↑

Figure 3.15 illustrated the procedures to generate a group of sounds with equal loudness and roughness but increasing tonality. Start with a neutral broadband sound, adding tones would result in an increase in both tonality and loudness. Normalizing sounds to have equal loudness would lead to the change of the broadband contribution. Sounds with quieter broadband components sound less rough. Thus, some modulations are added to the broadband to increase the roughness. The process would be gone through iteratively to ensure the sound has an almost equal loudness and roughness.

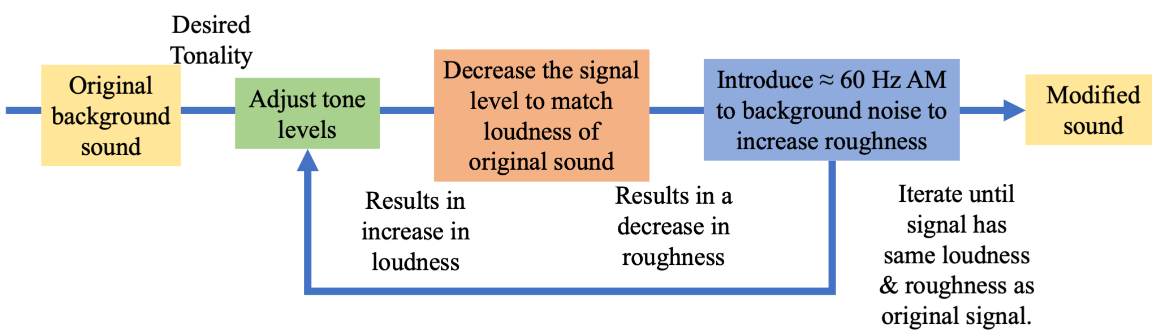

Figure 3.15. An iterative process generating the equal loudness and roughness sounds.

In order to introduce modulations to the broadband sound, random noise is filtered by a 20^{th} order low-pass filter with a cut-off frequency 2 Hz. The noise is then scaled to have a standard deviation of 5 Hz. As roughness is most sensitive to modulations in 50 – 80 Hz, the modulated frequency (f_{50-80}) was produced by summing the 65 Hz center frequency and the low-passed noise. This modulated frequency was later used to control the amplitude modulation of the broadband background noise.

A group of equal loudness and roughness sounds were tested in the subjective Test 2 with an NC-20 ambient broadband noise. The procedures to generate these sounds were conducted by changing C_0, C_1, C_2 in Equaltion 3.4:

$$\begin{aligned} Signal = {} & Broadband_1 + C_0 \left(1 + C_1 sin\left(2\pi f_{50-80} t\right)\right) Broadband_2 \\ & + C_2 \ Tonal\ component \end{aligned} \tag{3.4}$$

$Broadband_1$ is an NC-20 ambient broadband noise, $Broadband_2$ is a neutral broadband uncorrelated with $Broadband_1$. C_0, C_1, C_2 correspond to the green, orange, and blue boxes in Figure 3.15. Three parameters are designed for loudness, roughness, and tonality control. The parameters are adjusted iteratively to simulate equal loudness and roughness sounds.

3.6 Summary of Office Noise Simulation

Broadband noises with different levels and spectral balance were simulated based on an office recording measured in the HERL Building. A tonal model with amplitude and frequency modulation was used to generate tonal components. The realism of simulated sounds was confirmed and the range of the tonalness metric of signals played in subjective tests was determined by ASHRAE advisory team's listening tests. The test sounds were generated by gradually changing the prominence ratio from -1 to 19 dB. In addition to these sounds, a set of sounds with equal loudness and roughness were tested in second subjective test (described in Chapter 5) to investigate how tonalness affect annoyance.

4. TEST ENVIRONMENT AND SOUND REPRODUCTION

All of the subjective tests in this research were performed in the Perception-based Engineering Lab (PBE Lab) at Herrick Labs. PBE Lab is a space where the temperature, light and humidity can be accurately controlled, and it is also acoustically isolated from the outside environment and other parts of the Laboratories. This highly controllable and quiet environment ensures that the sounds are reproduced accurately. A small office mock-up was set up in the lab to create a natural testing environment. The playback system was set up with two loudspeakers to reproduce the simulated sounds (described in Chapter 3) at the subject's listening location. The loudspeaker locations of them were carefully selected to avoid room modes in the lab. A finite impulse response filter was then designed to equalize the spectral shaping caused by the loudspeaker responses and the room. The quality of the reproduced sounds was verified by comparing the power spectral density functions of the reproduced sounds at the listening location and that of the simulated sounds.

4.1 Office Environment and Playback System Setup

A small office mock-up was set up in the east-south corner of the lab. Furniture such as desks, chairs, bookshelves, carpets and a whiteboard was set up in the PBE Lab. Partitions (Kick Panel system with fabric surfaces manufactured by Steelcase, with height $\geq$ 5'6") were installed in the office environment to prevent subjects from seeing the loudspeakers on their way entering the office environment. Based on discussion with the advisory team members, it was decided to bring some natural light in the space so that the space can be perceived more natural. Privacy window films were then applied to let natural light into the space while avoiding distracting subjects by the outside view. A camera (without recording) was installed to monitor test activities. The subject was told to wave at the camera if she or he wants to pause the test. The amplifier (Furman SP20AB) and test computer used for playing test sounds and interacting with subjects were located in the control room. The test computer was connected to a screen, a mouse and a keyboard in the office environment so that subjects can interact with the sound rating software without being distracted by computer noise, and the researcher can monitor their responses in the control room via another computer screen. The amplifier was connected to two loudspeakers (ALTEC N1201-8A) in the lab (Speaker 1 and 2 in Figure 4.1). Speaker 1 was used to continuously play the ambient broadband noise throughout the test, and Speaker 2 was used to produce tonal components and some additional broadband noises. Speaker 1 was playing all the time so that subjects can acclimatize to the background level before the test begins. This is desirable because the PBE Lab background noise levels are extremely low and when the background noise is turned on, it sounds loud in contrast, even though it is at the desired NC level. Extension cables were connected to the earphone and a pushbutton of the audiometer. During the hearing test, instead of standing right behind the subject, the researcher sat on the other side of the partition to avoid making noises. Figure 4.1 shows a floor plan of the lab.

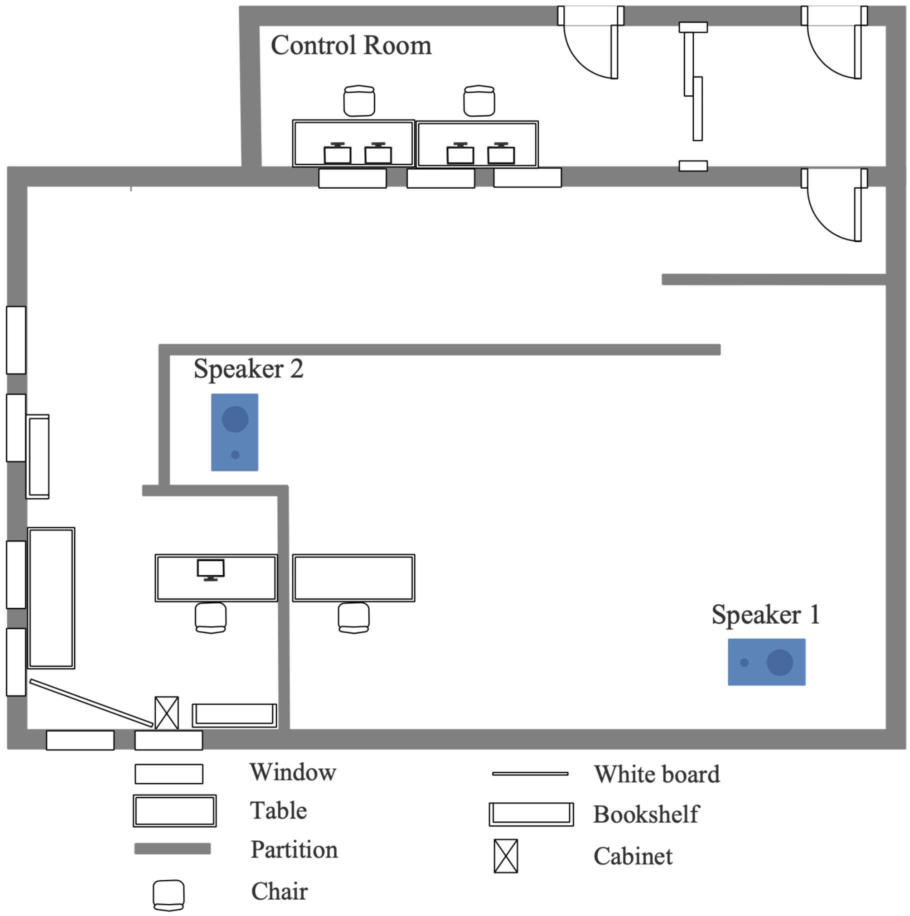

Figure 4.1. Floor plan of the Perception-based Engineering Lab.

Some photos of the test setup are shown in Figure 4.2. In Figure 4.2 (a), two monitors were placed in the control room, one for streaming the camera view, the other one for monitoring subject's noise rating. In Figure 4.2 (b) is a photo of the office environment. With the privacy window film, some natural light is included in the test. A headphone for the hearing test was hung on the partition. Figure 4.2 (c) shows an overview of the office and the lab setup.

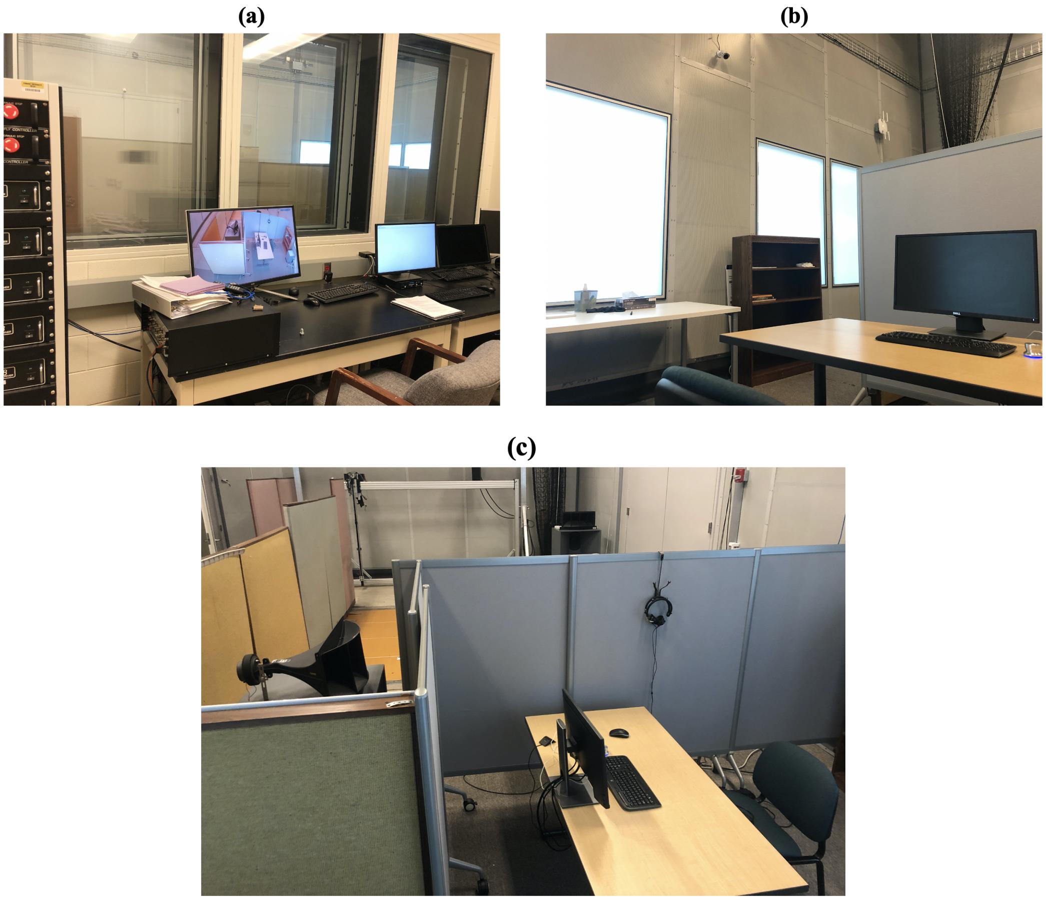

Figure 4.2. Photos of the Perception-based Engineering Lab taken in (a) control room, (b) office environment (at subjects' test location), (c) office environment (with a different view).

Figure 4.3 shows how the sound is reproduced in the subjective test. The sounds were played through the test computer. The playback system consists of a LynxONE sound card, a Furman SP20AB amplifier, and two ALTEC loudspeakers (Speaker 1 & Speaker 2). Speaker 1 was located in the north-east corner of the lab, it was continuously playing an ambient broadband noise throughout the test. Speaker 2 stands behind the partition, playing tonal components and some additional broadband noise to the subject. The locations of two speakers are carefully selected for two purposes: (1) avoid the cases that subjects can easily identify the source of the sound (directivity issue) (2) avoid the cases that the playback system have trouble in producing some of the low-frequency contents due to room modes. An equalization filter was then designed to compensate for the effect of the playback system. A nominal listening location is defined for measuring and calibrating the reproduced sounds. The nominal listening location is next to the desk, facing the monitor, at the height of 4 feet.

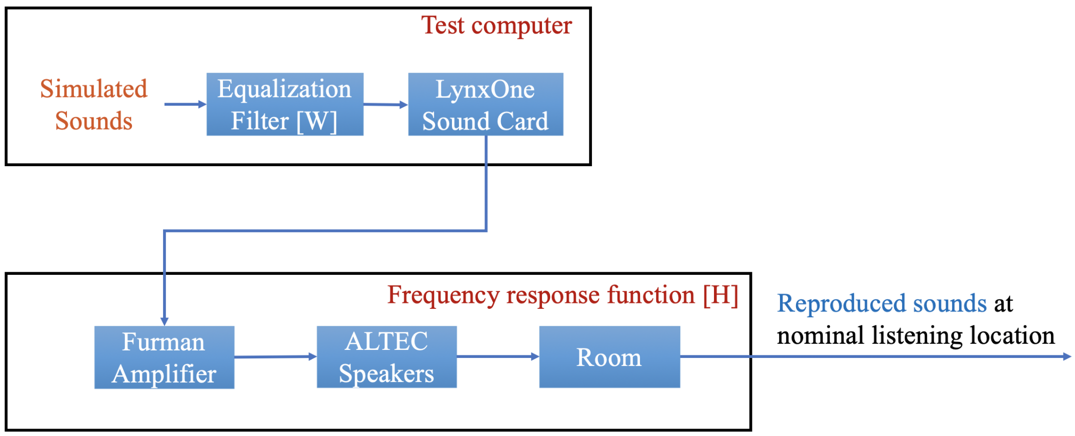

Figure 4.3. A system diagram of the playback system, starting from the simulated sounds in the computer to the replayed sounds at the nominal listening location.

4.2 Room and Loudspeaker Response Equalization

Equalization has to be performed to accurately reproduce the simulated sound at the subject's listening location. This is not only because the frequency response functions of two loudspeakers are not flat, but also because the room environment would bring notches and peaks to the sound transfer path response from loudspeaker to the listening location.

For the tonal component equalization, pure tones were played and measured at the nominal listening location. Based on the sound pressure level difference between the desired and measured sounds, scaling factors were introduced to compensate the differences at those frequencies. For the broadband component equalization, an inverse filter was designed to compensate for the speaker and room effects. To do this, the frequency response function relating computer output to sound pressure at the listening location was first measured. A complex smoothing process [39] was then applied to smooth out the detailed spectral behavior at higher frequencies while keeping more details at low frequencies. This is because people are not sensitive to high-frequency notches. Then, an inverse filter design method was developed to design the equalization filter. Measurements were conducted to verify the playback system performance.

4.2.1 Estimated Frequency Response Function

The output signal of the test computer's sound card and the sound pressure at the nominal listening location were measured. The output of the computer was considered as the input signal of the playback system, and the measured sound pressure at the listening location was considered as the output signal of the playback system. The noise in the input signal (electric noise of the sound card) is negligible, while the noise in the output signal (acoustic background noise in the PBE Lab) is much louder. The noise in output is uncorrelated with input signal. Thus, the frequency response function could be estimated by H1 estimation:

$$H_1 = \frac{S_{xy}(f)}{S_{xx}(f)} \tag{4.1}$$

where $S_{xx}(f)$ is the auto power spectral density of input signal, $S_{xy}(f)$ is the cross power spectral density of input and output signals.

To avoid the directivity issue, Speaker 1 was located at the north end of the Lab. From the frequency response function, a relatively low response at around 20 Hz could be observed due to the room mode (shown in Figure 4.4). Measurements were carried out at 11 different loudspeaker locations. The choice of the loudspeaker at the north-east corner of the room was used since it resulted in the least notch phenomenon at low frequencies in the frequency response. Similar procedures were conducted to determine the location of Speaker 2 as well.

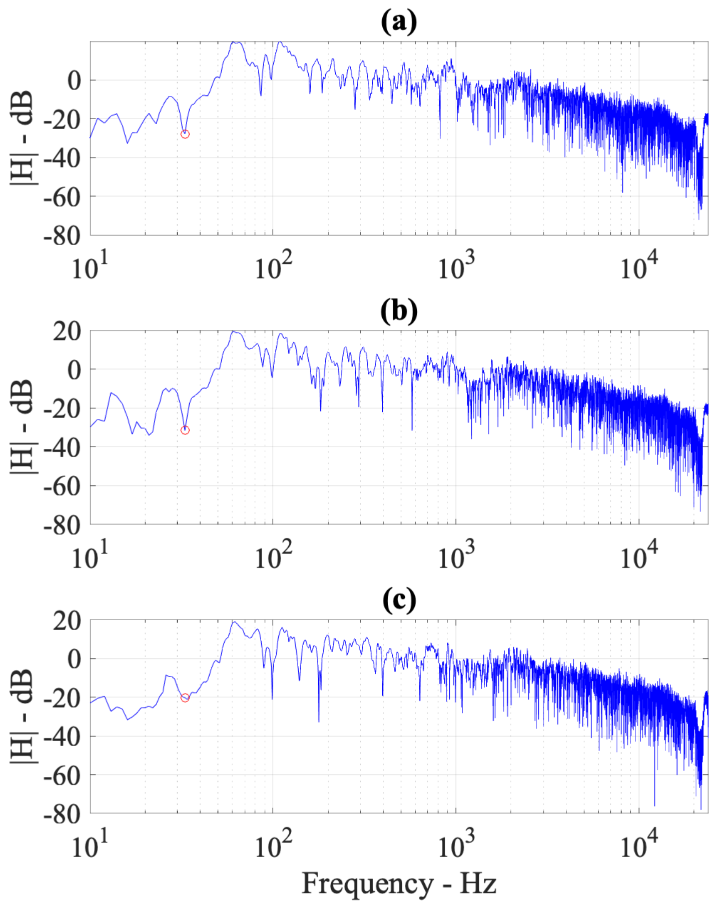

Figure 4.4. Example frequency response functions of Speaker 1 at different locations. (a) At the northwest corner, (b) in the middle of the north end, (c) at the northeast corner. The red circle corresponds to one notch in the frequency response due to room modes.

4.2.2 Smooth Frequency Response Function

As the resolution of human ear is nonlinear and non-uniform with frequency, a non-uniform frequency resolution [40] was used in smoothing the spectrum for a better equalization performance. In this research, the equalization involves complex smoothing process [39]. This is based on the recognition that due to slight head movement, the fine details of the frequency response function between the loudspeaker and the listening position will change at high frequencies but this phenomenon is not obvious at low frequencies. The general trend in frequency response at high frequencies will not change significantly. This method applies nun-uniform smoothing independently on the magnitude and the phase of the frequency response function. A pointwise smooth was implemented with a moving Hann window which covers a one-third octave band centered on the frequency. Thus, the frequency response function was smoothed on a logarithmic frequency scale. The smoothed amplitude and phase are calculated as:

$$A_{sm}(f_i) = \Sigma_{k=-N_i}^{N_i} W_i(k) A(f_i + k) \tag{4.2}$$

$$\phi_{sm}(f_i) = \Sigma_{k=-N_i}^{N_i} W_i(k) \phi(f_i + k) \tag{4.3}$$

where$A(f), \phi(f)$ are amplitude and phase of the measured frequency response, and $A_{sm}(f), \phi_{sm}(f)$ are smoothed amplitude and phase. W_i is a Hann window normalized to have a sum of 1. Window size 2N$_i$+1 roughly covers an one-third octave band of the center frequency f_i and is different for different frequencies. An example complex smoothed spectrum was shown in Figure 4.5.

4.2.3 Equalization Filter Design

A finite impulse response (FIR) equalization filter was designed based on the complex smoothed frequency response function. Fine details in low frequencies require a longer impulse response to compensate for, while the general shape for the high-frequency contents only requires a relatively short filter. Thus, the filter is designed with two

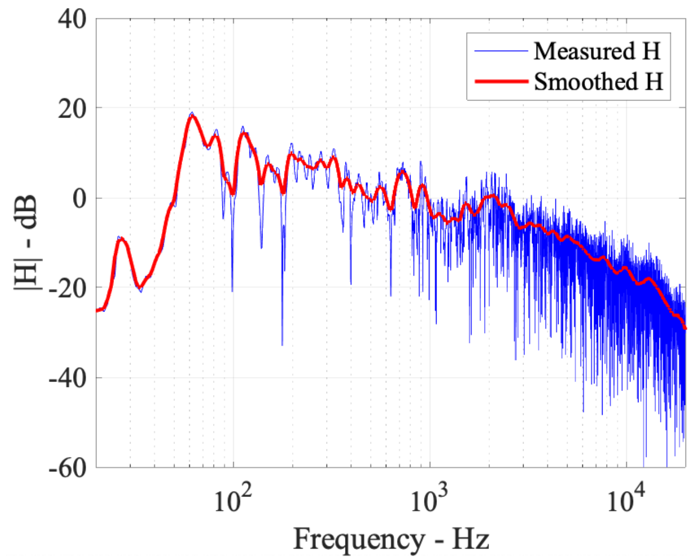

Figure 4.5. Measured frequency response function of the system (blue line) and corresponding complex smoothed frequency response function (red line).

sub-filters. The first sub-filter only performs low-frequency equalization which is a 1600-point FIR filter with a sampling frequency of 2 kHz. The second sub-filter is designed to equalize the response for the whole frequency range. It is a 5000-point FIR filter with a sampling frequency of 48 kHz. The overall equalization filter is a combination of the upsampled low-frequency filter and the broadband filter.

The filter structure is shown in Figure 4.6, the error signal is defined as the differences between the desired output signal (input signal with a suitable choice of time delay) and the actual output signal (input signal going through equalization filter W and the playback system H). Based on this relationship, the power spectrum density of the error signal was derived with the time delay, auto and cross power spectral densities of input and actual output signals. The equalization filter is designed by minimizing the total power of the error signals. Convex optimization is implemented to estimate the impulse responses of two filters.

$$S_{ee} = \lim_{t\to\infty} E\left(\frac{E_T^* E_T}{T}\right) = |H|^2\,|W|^2\,S_{xx} - 2RE\{HWe^{j\omega m\Delta}\}S_{xx} \tag{4.4}$$

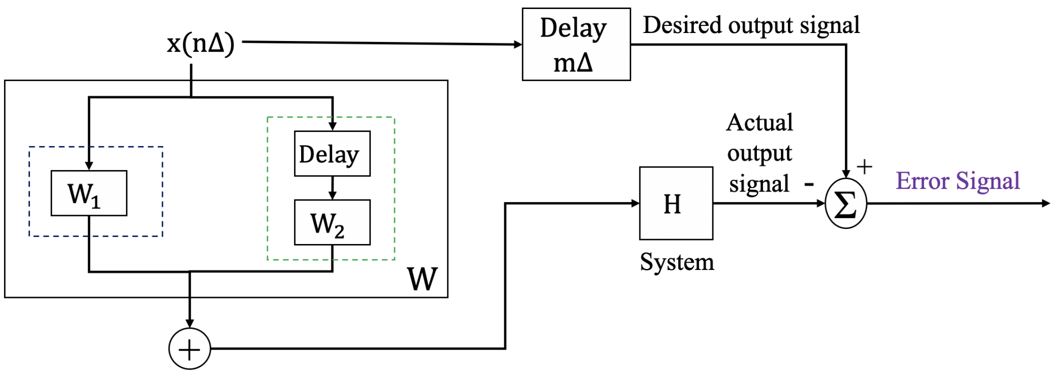

Figure 4.6. System diagram of the implementation of the designed equalization filter.

Figure 4.7 shows the impulse responses and the performance of two sub-filters. Ideally, the combination of the equalization filter (W) and the complex smoothed spectrum (H) would have a flat spectrum, $|WH| = 0$ dB. Results show that actual error was within 1 dB across all the frequencies of interest.

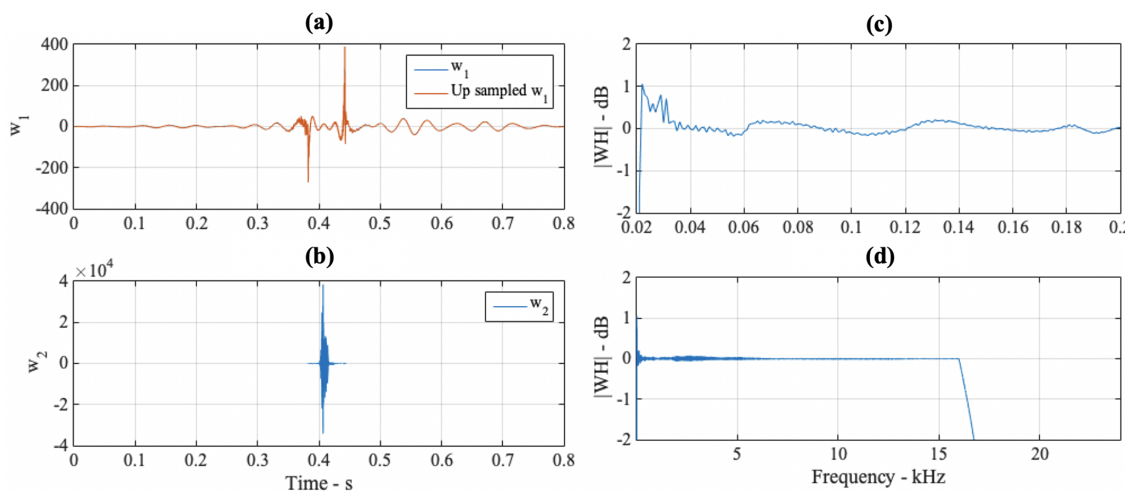

Figure 4.7. The impulse responses and the performance of the designed equalization filter. (a) Impulse response of the low-frequency filter, (b) impulse response of the broadband filter, (c) equalization filter's performance from 20 to 200 Hz, on a linear scale, (d) equalization filter's performance from 0 to 24 kHz, on a linear scale.

4.2.4 Examples of reproduced sounds

The equalization filter performance was verified by comparing the power spectral densities of the desired broadband sound and the measured reproduced broadband sounds in the room. The reproduced broadband sound was measured at four locations: one at the nominal listening location, the other three locations are 10 cm right to, left to and behind the nominal listening location. The power spectral densities of the desired broadband noise and the measured broadband noises are compared to visualize the performance of the equalization filter in Figure 4.8.

There is some concern about tonal amplitudes changing as the subject moves around at the table, a problem at and above 500 Hz. However, in discussions with the advisory team, this was considered as a realistic situation in an office environment,

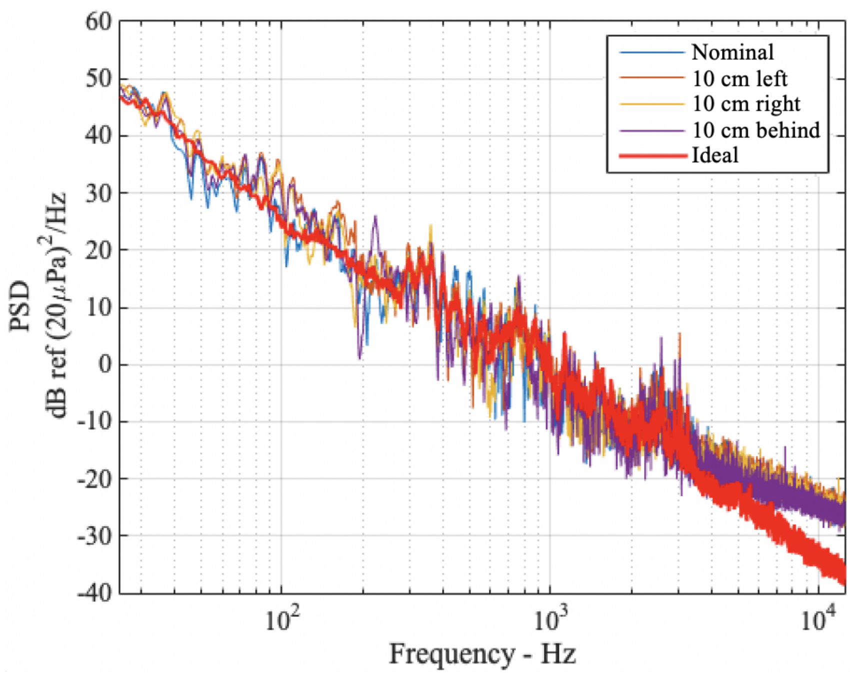

Figure 4.8. Power spectral densities of the desired broadband noise (red line) and the replayed broadband noises measured at nominal listening location (blue line), 10 cm right to the listening location (orange line), 10 cm left to the listening location (yellow line), and 10 cm behind the listening location (purple line).

so calibrating signals to produce desired tonal levels at the nominal listener location should be sufficient for setting up the test.

4.3 Summary of Setup Used in Subjective Tests

To conclude, an office environment and a sound reproduction system were set up in Lab space. Different speaker locations have been tested, and the configuration is shown in Figure 4.1 was selected. Speaker 1 was used to continuously play the ambient broadband noise throughout the test, and Speaker 2 was used to produce tonal components and some additional broadband noises. An equalization filter was designed to ensure that the playback system reproduces the desired sounds at listening location accurately. The performance of the designed equalization filter was verified by comparing the reproduced sound at the listening location and the desired sound.

5. SUBJECTIVE TESTS

With the wide use of rotating machinery in building equipment, tonal sounds turn out to be a common issue in buildings. The perception of tonal noise cannot be measured by loudness alone. It is known to be affected by various factors, such as the exposure time [references], the frequency and prominence of the tone [23], the existence of harmonics [6], and the characteristics of the broadband noise [41]. Three sets of subjective tests that were conducted to explore the influence of these factors on the perception of tonal building noise are described in this Chapter.

Purdue's Institutional Review Board (IRB) approved the use of human subjects in this research, Purdue IRB Protocol Number: 1811021317. In each test conducted, subjects listened to sounds and rated them on an annoyance scale. Test 1 included tonal sounds with different durations ranging from 5 seconds to 4 minutes. The goals were to identify whether subjects' ratings of tonal sounds changed with exposure times, to quantify that change, and determine a reasonable exposure time for sounds used in the later tests. Test 2 was designed to investigate the effect of different tonal levels and the presence of a harmonic at twice the fundamental frequency on tonal noise perception. Five groups of tonal sounds were played to subjects. All the tonal sounds used in this test have a broadband component with an NC value equal or close to 30. In Test 3, various sounds with different tonal levels, different tone frequencies, with or with a harmonic, and different level broadband components were played to subjects. The test was split into 6 parts The goals for Test 3 were to understand how these various factors affected annoyance ratings, examine the correlation between sound metrics and responses, and to gather sufficient data to develop an annoyance model for tonal building noise. The annoyance model development is described in Chapter 6. A brief summary of 3 subjective tests and the number of sounds used in each test part is given in Table 5.1. A detailed description of the test signals is given in Appendix A. The average responses from subjects are given in Appendix. Details and results of three subjective tests are described in this chapter.

Table 5.1. The summary of subjective tests and signals

		Signals
Test 1		22 sounds with NC-30 broadband - With a duration from 10 seconds to 4 minutes - Test with NC-30 ambient
Test 2	Part A	144 5-second sounds - 2 broadband sounds, 70 tonal sounds played twice - Test with NC-20 ambient.
	Part B	29 2-minute sounds - 1 broadband sound, 28 tonal sounds - Test with NC-20 ambient.
Test 3	Part A	90 5-second sounds with NC-30 broadband - 5 broadband sounds (from NC-30 to NC-38) - 85 tonal sounds (42 with harmonics) - Test with NC-30 ambient.
	Part B	50 5-second sounds with NC-20 broadband - 5 broadband sounds (from NC-20 to NC-28), 45 tonal sounds - Test with NC-20 ambient.
	Part C	9 2minute sounds with NC-20 broadband - 1 broadband sound, 8 tonal sounds - Test with NC-20 ambient.
	Part D	43 5-second sounds with Tilted NC-30 broadband - 1 broadband sound, 42 tonal sounds - Test with Tilted NC-30 ambient.
	Part E	50 5-second sounds with NC-40 broadband - 1 broadband sound, 38 tonal sounds - Test with NC-40 ambient.
	Part F	9 2-minute sounds with NC-40 broadband - 1 broadband sound, 8 tonal sounds - Test with NC-40 ambient.

5.1 General Test Procedures and Noise Sensitivity

Just before the subject arrived, the window blinds were opened, the room configuration was checked. The temperature in the lab was checked to be close to 70° (usually 69° - 71°), and the lighting level was measured at the test table. The playback system was calibrated separately with a 36.7 dBA broadband signal for Speaker 1 and a 50 dBA, 240 Hz calibration tone for speaker 2. Both calibration and test sounds were .wav files stored on the test computer and played through two loudspeakers. A Brüel and Kjær Type 2250 handheld sound level meter was mounted on a tripod, placed at the nominal listening location (described in Section 4.1). The microphone on the sound level meter was first calibrated by using the Brüel and Kjær type 4231 calibrator. The sound level meter was set to record fast averaging of A-weighted Sound Pressure Level for 15 seconds. Then, calibration sounds for Speaker 1 and Speaker 2 were played one at a time, Each of the two-channel outputs of the Lynx soundcard were adjusted on the test computer until the A-weighted Sound Pressure Level was within 0.5 dB of the expected value. The playback system was checked by comparing the A-weighted sound pressure level of 5 - 6 of the sounds that would be used in the test with the expected values. These sounds were chosen to span the range of levels of sounds used in the test. Differences between the measured and expected values less than 1.0 dB were deemed acceptable. The sound level meter was then removed. A warning sign ("Do Not Enter, Testing in Progress") was hung on the entrance to the test area.

When the subject arrived, he (or she) would be first given a brief description of the test procedures. Then, a consent form was given to read. If the subject decided to continue and signed the consent form, he (or she) then completed a background questionnaire (including the age, racial group, work experience related to the noise, etc.). A preliminary hearing check was then conducted with a MAICO MA 25 audiometer. The audiometer was placed behind the partitions, connected with a headphone and a press-button with extension cables. Continuous tones ranging from 125 Hz to 8 kHz were used as stimuli. Hearing screening was handled carefully to avoid distracting subjects. This was to ensure the subject had a normal hearing (with hearing thresholds of 25 dB or below from 125 Hz to 8 kHz for both ears). If the subject didn't have a normal hearing, the test would end, and the subject would be awarded $5. Subjects with normal hearing would continue to read the test instructions, in which a scenario was given to subjects, "It may be helpful, while you are listening, imagining yourself in your office." The test started with a familiarization session. Subjects would listen to 5 to 6 sounds without rating them. Sounds with different levels of tonalness and broadband sounds were included in this familiarization session. Then, 2 to 6 sounds were played again, and the subject was asked to practice rating the sound on the scale. If the subject felt comfortable with the test, then the main part of the test was conducted. For tests with more than one part, additional familiarization and practice sessions would be included in each part, and an optional break may be given between parts. After the main test, the subject would be asked to fill a comment sheet and a Noise-Sensitivity-Questionnaire (NoiSeQ) [42] (only in Test 2 & 3). A second hearing test was given to check the hearing threshold levels had not changed. The subject would be paid $10 per hour for participation.

After the subject had completed the test and been paid, the playback system was recalibrated with two calibration sounds to ensure the difference in A-weighted sound pressure levels were within 1.0 dB. Temperature and lighting level were remeasured.

The NoiseQ is a 35-item, self-assessment questionnaire consisting of five 7-question subscales covering sensitivity to noise at home, during leisure, in communication, at work, and during sleep. Items consist of a statement such as: I need peace and quiet in order to do difficult work. It is rated on a 5-point scale (Strongly Agree = 5, Agree = 4, Neutral = 3, Disagree = 2, Strongly Disagree = 1). Each question is counted as a number from 1 to 5, and 9 of 35 questions are scored backward. The overall NoSeQ score is the average score of all the responses. The detail of NoSeQ is described in Appendix C.3.

5.2 Subjective Test 1: Test on Different Exposure Time

People are usually exposed to tonal office noise for a relatively long time, but long-duration exposures limit the variety of sounds that can be played in subjective tests, which, in turn, limits annoyance model development. Investigating annoyance adaptation over time may allow translation of predictions of annoyance from models developed with responses to shorter duration stimuli to annoyance predictions due to longer exposures. The results also helped in determination of a reasonable exposure time for later subjective tests.

5.2.1 Stimuli, Test Procedures and Subjects

In Test 1, test sounds were a combination of an NC-30 broadband component and different tonal components. 2 different Prominence Ratios (3.0, 11.0 dB) were selected for a low frequency (60 Hz), a middle frequency (240 Hz), and a high frequency (1000 Hz) tone. The lengths of the 22 test sounds ranged from 10 seconds to 4 minutes. Frequencies, Prominence Ratios and durations of the stimuli used in Test 1 are given in Table 5.2.

Table 5.2. Sounds used in Test 1. All sounds had an NC-30 broadband component.

Frequency	**Prominence Ratio**	**Sound duration**
60 Hz	3 dB	10s, 60s, 120s
60 Hz	11 dB	10s, 60s, 120s
240 Hz	3 dB	10s, 30s, 60s, 120s, 240s
240 Hz	11 dB	10s, 30s, 60s, 120s, 240s
1000 Hz	3 dB	10s, 60s, 120s
1000 Hz	11 dB	10s, 60s, 120s

In Test 1, six 5-second sounds were used for familiarization and 2 more for practice. During the test, subjects were asked to do some typical office work (mostly reading and writing). Subjects were not allowed to use personal electronic equipment during the test. The background noise levels in the laboratory are very low, and to avoid the sudden changes in broadband sound levels as stimuli ended and started, which might drive people to rate sounds as more annoying, ambient NC-30 background noise was played throughout the test through Speaker 2. Speaker 1 played additional tonal components in the test. For each subject, sounds were played in a different random order. Due to the concern that the subject may not notice the sound is over (especially for sounds longer than 1-minute), a blinking red box is added to the right-bottom corner of the screen after each sound was played. In the analysis, the five equally-spaced points labeled on the scale (shown in Figure 5.1) are assigned numerical values of 2, 3.5, 5, 6.5, 8, and the end-points of the scale correspond to 1 and 9, respectively. Subjects could place the cursor on any part of the line.

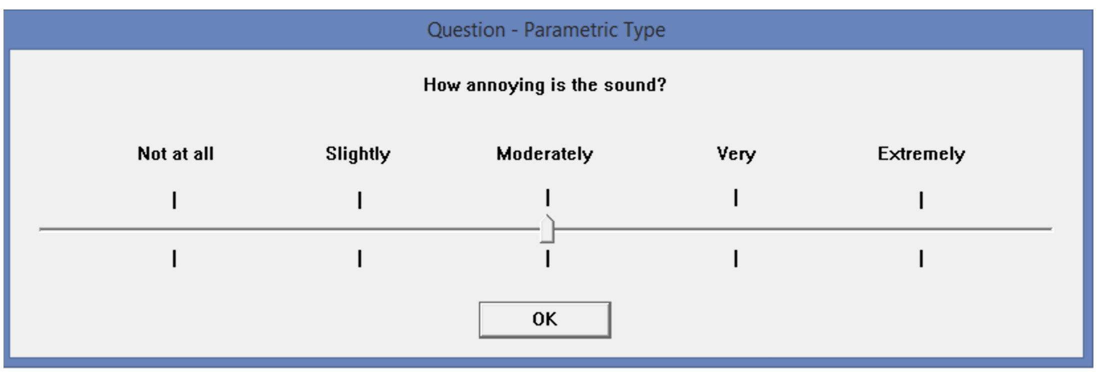

Figure 5.1. Graphical user interface used by subjects when rating sounds in the test.

Twenty subjects (12 males, 8 females) participated in Test 1, aged 20-42. Average age across all subjects was 26.7 years with a standard deviation of 6.8, and a median age was 25 years.

5.2.2 Data Analysis

The average of the ratings of the first sound heard (different sound for each subject), the second sound heard, the third sound heard, etc. was studied to see whether subjects were still adapting to the rating system at the beginning of the test. If the familiarization and practice rating test were sufficient, we would not expect to see any significant trends in these average ratings. Subjects' averaging rating on first several played sounds has been studied to see whether subjects rated all the sounds under a constant standard. In Figure 5.2, the blue line is the average annoyance ratings of n^{th} played sound plotted against the sequence of the sounds in the test. The red line corresponds to the average of n^{th} played sound's average annoyance rating plotted against the sequence of the sounds. The error bars are standard deviation of the estimated mean. The difference in average annoyance rating is not significant if two error bars overlap.

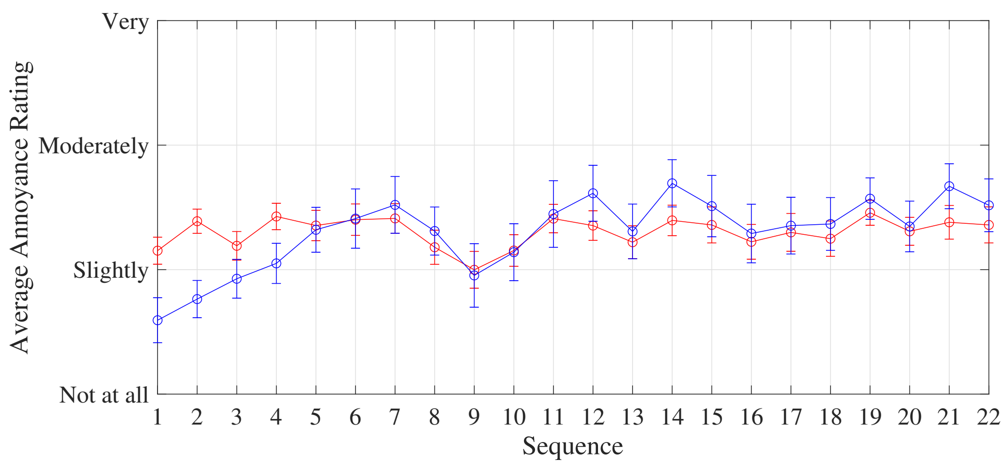

Figure 5.2. Average annoyance ratings of n^{th} played sounds (blue line) and the average of average annoyance ratings for n^{th} playd sounds (red line) plotted by the sequence of sounds played. Error bars are +/− the standard deviation of the estimated mean.

As the sounds were played in different random orders for different subjects, the variation in the red line should not be significant. This is confirmed in Figure 5.2. The difference between 2 lines is significant for the first four sounds, which indicates a learning pattern exists. This learning pattern infers that it may not be enough to only includes two sounds in the practice session, at least four more sounds should be added to the practice session to avoid this problem.

Average annoyance ratings for different duration sounds are shown in Figure 5.3. The test results illustrate that higher tonality signals are perceived as being more annoying. Subjects' average annoyance ratings for 11 dB Prominence Ratio signals are at a similar level, while their ratings for 3 dB Prominence Ratio signals are slightly different. Signals with 60 Hz tones are rated less annoying than those with 240 and 1000 Hz tones. Besides, there is no obvious annoyance adaptation for six different tonal office signals in the first 2 minutes. There may be some rating differences for two 240 Hz tonal office noises from 2 minutes to 4 minutes, but they are not significant.

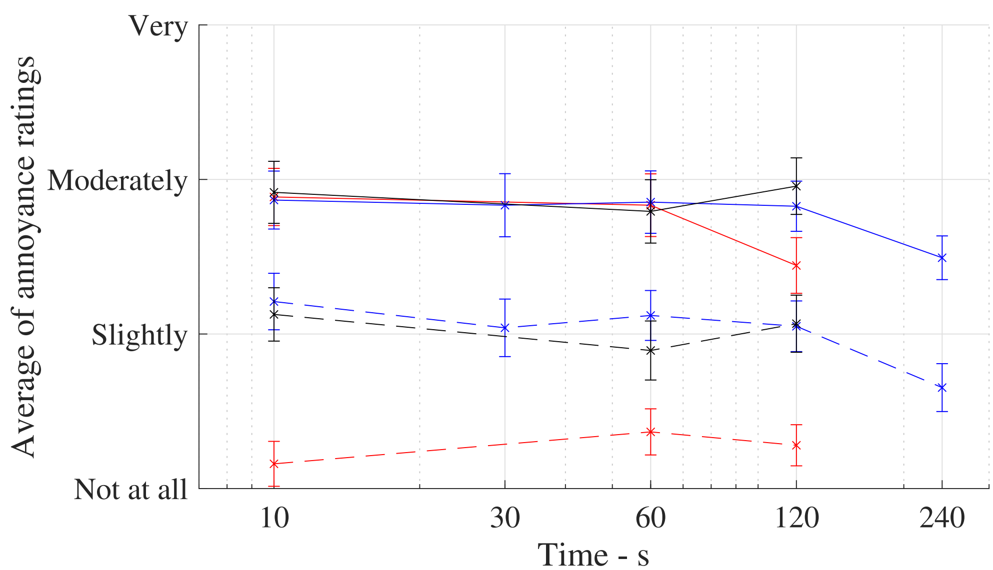

Figure 5.3. Averaged annoyance ratings for different duration sounds in Test 1. 60 Hz (red line), 240 Hz (blue line), and 1000 Hz (black line). Prominence Ratio = 11 dB (solid line) and = 3 dB (dashed line). Error bars are +/− the standard deviation of the estimated mean.

5.2.3 Test 1 Summary

From Test1 results, exposure time does not appear to be a significant factor affecting the trends of the average annoyance ratings over time. For the 240 Hz sounds (blue lines in Figure 5.3), there may be an annoyance decrease beyond 2 minutes. One reason for this could be that the subjects were working in the office-like environment, and for the longer sounds the interruption for the rating at the end of the exposure time is perhaps less intrusive than when interrupted after a shorter duration sound.

5.3 Subjective Test 2: Test on NC-30 Based Sounds

Due to the concern that interrupting the subjects for ratings with the shorter duration signals may have caused an increase in annoyance for those sounds, Test 2 was designed with two parts: a short exposure time test (five groups of sounds of 5 seconds duration) in which subjects are focusing on rating the sounds (no accompanying activity); and a long exposure time test (two groups of sounds of 2 minutes duration) in which subjects are doing some office work (e.g,. reading, writing, etc.). In total, 72 building noises were included in Test 2 to investigate how tonal levels and harmonics affect annoyance ratings. An overview of Test 2 was shown in Table 5.1.

5.3.1 Stimuli and Test Procedures

2 broadband sounds with different spectral balance (NC-30 & tiled NC-30 broadbands) and 70 tonal sounds were used as stimuli in Test 2. Tonal sounds included one of two commonly-found frequencies in office recordings (240 Hz, 500 Hz), with 7 different Prominence Ratios (PRs), ranging from 3 dB to 15 dB. Details of the test sounds in Test 2 were given in Appendix A.2. Following groups of tonal signals were used in Test 2:

- Single tone signals with NC-30 broadband

- Single tone signals with equal loudness and roughness as NC-30 broadband, consist of a modulated broadband and a tonal component.
- Single tone signals with tilted NC-30 broadband
- Signal with fundamental and one harmonic with NC-30 broadband (Compared with PR of fundamental tone, PR of 2^{nd} harmonic is almost same for 240 Hz, around 6 dB lower for 500 Hz.)
- Single tone (fundamental only) signals with same loudness as the signal with the fundamental and the harmonic

The test started with a short exposure time (5-second sounds) test. All 72 unique sounds were tested twice in this test. In total, 144 sounds were played in a different random order for each subject. Subjects were asked to focus on rating the sounds in this part. A 5-minute break for water and restroom would be given between the short and long exposure time tests. In the long exposure time (2-minute sounds) test, due to the limit of testing time, only 29 sounds with the same characteristics as some of 5-second duration sounds (NC-30 broadband sound and first 2 groups of tonal sounds) were tested, and subjects heard them once. Subjects were doing some typical office work (reading, writing, grading, etc.) while the sounds were being played. The main test started after a familiarization session with 6 sample sounds and a practice session with 6 more sounds.

5.3.2 Subject

Test 2 involved 37 subjects (22 males, 15 females), aged 20-44. Average age was 26.3 years with a standard deviation of 6.2. A Noise-Sensitivity-Questionnaire (NoiSeQ) [42] was included. The average of NoiSeQ score was 3.06 with a standard deviation of 0.51 (5 – most sensitive, 1 – least sensitive).

Subject-to-Group Correlation Analysis

Subject-to-group correlation was analyzed to ensure no one used scale completely different from others (with 0 or negative subject-to-group correlation). For subject i:

$$(Subject - to - group\ correlation)_i = correlation(Rating_i, \frac{1}{36}\sum_{j=1, j\neq i}^{37} Rating_j) \quad (5.1)$$

$Rating_i$ is a column vector that consists of ratings of subject i. In Figure 5.4 and 5.5, most subjects' ratings have a high correlation with the average of other subjects' annoyance ratings. This infers that no one used the scale backward. Although five subjects (number 2, 5, 24, 29, 31) have a lower correlation in some particular sessions (lower than 0.25), they are still positive. Thus, all 37 subjects' ratings are included in the statistical analysis presented below.

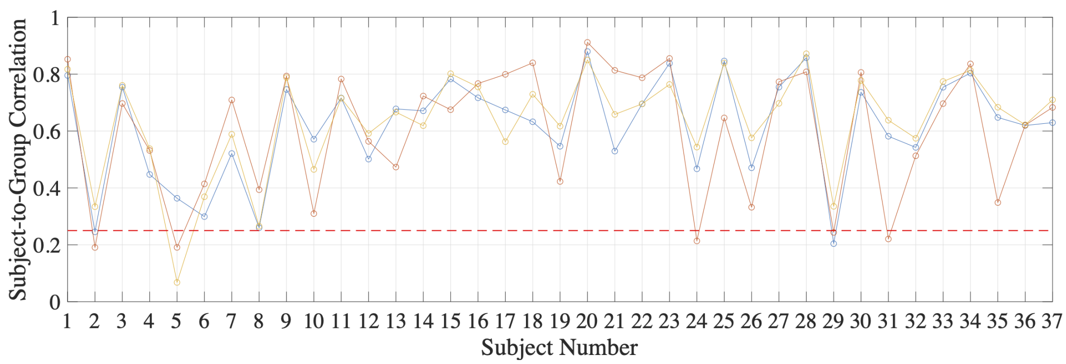

Figure 5.4. 37 subjects' subject-to-group correlation in the short exposure time test (blue line), in the long exposure time test (orange line), and in both tests (yellow line). The horizontal dashed red line is at correlation coefficient = 0.25.

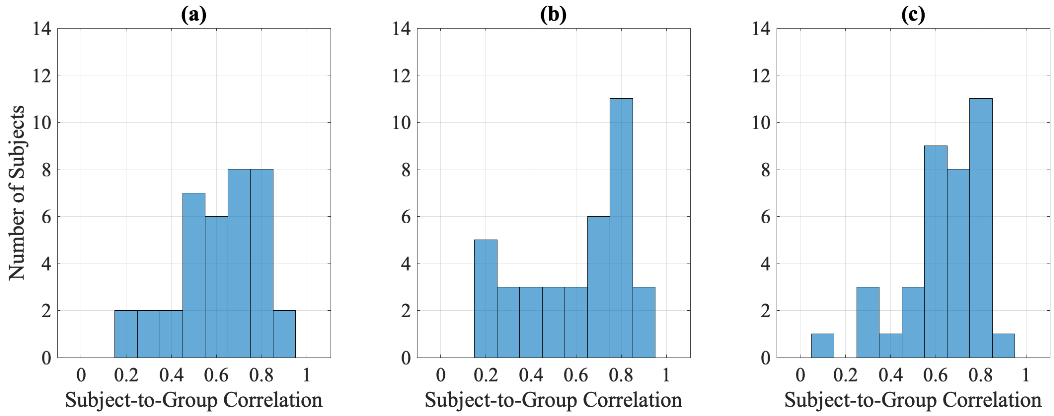

Figure 5.5. Histogram for Subject-to-group correlation coefficients. (a) Short exposure time test, (b) long exposure time test, (c) both short and long exposure time tests.

Individual Response (Short Exposure Time Test)

In terms of the short exposure time results, each subject's individual annoyance ratings and the average annoyance ratings are shown in Figure 5.6. As expected, there are a lot of distributions in individual ratings. But more tonal sounds are, on average, perceived as being more annoying.

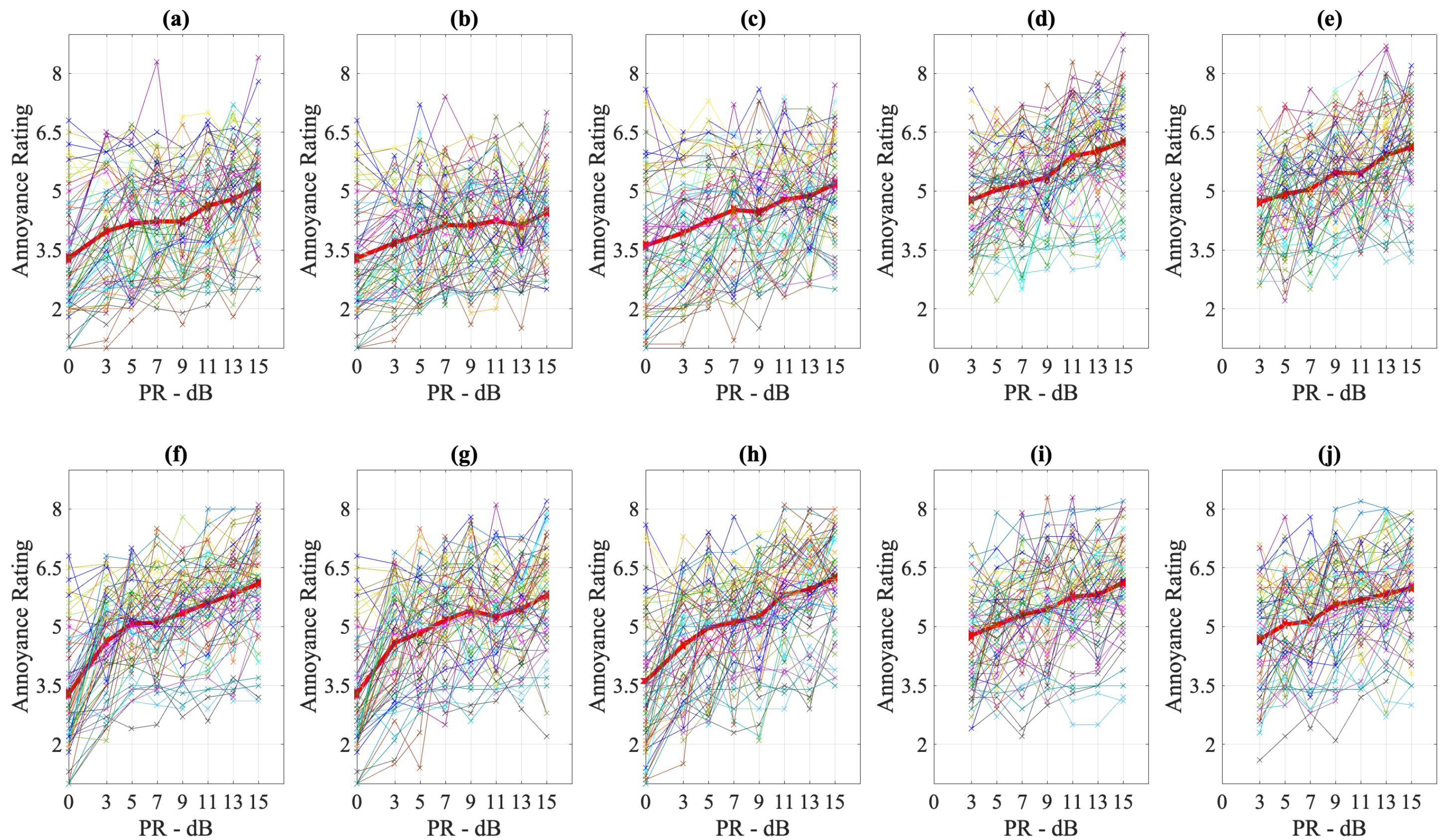

Figure 5.6. Subjects' individual and average annoyance ratings for 5 groups of sounds (described in Section 5.3.1) against Prominence Ratio. Each color is associated with a particular subject's responses. The red line is the average of 74 ratings for each of the signals. (a) – (e) ratings for 5 groups sounds with 240 Hz (or 240 + 480 Hz) tonal components, (f) – (i) ratings for 5 groups sounds with 500 Hz (or 500 + 1000 Hz) tonal components. Number corresponds to: "Not At All Annoying" = 2, "Slightly Annoying" = 3.5, "Moderately Annoying" = 5, "Very Annoying" = 6.5, "Extremely Annoying" = 8.

5.3.3 Data Analysis

Short Exposure Time Test Result

In the short exposure time test, distributions in ratings can be seen from Figure 5.6. As 72 unique sounds were tested twice, sound Group 1 (sound number 1 – 72) is exactly the same as sound Group 2 (sound number 73-144). Two groups of sounds were tested together in random order. Figure 5.7 shows the average rating of Group 1 sounds (red line) and Group 2 sounds (blue line). As most of the error bars are overlapping, the average rating difference between the two groups sounds is not significant. This confirms that subjects' ratings are consistent in the test.

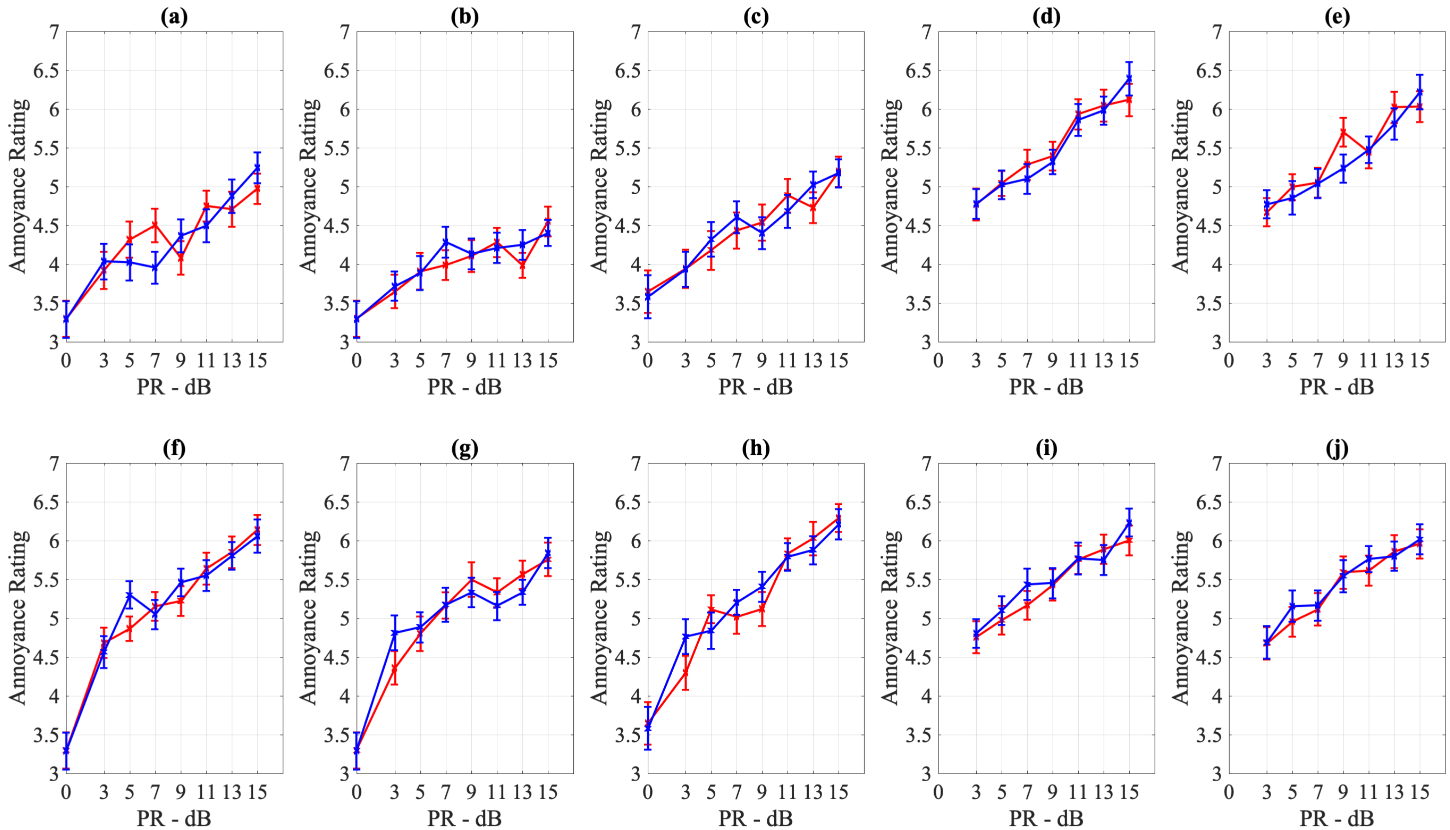

Figure 5.7. Average annoyance ratings of group 1 signals (signal number 1-72) and Group 2 signals (signal number 73 – 144). Error bars are +/− the standard deviation of the estimated mean. (a) – (e) average ratings for 5 groups sounds with 240 Hz (or 240 + 480 Hz) tonal components, (f) – (i) average ratings for 5 groups sounds with 500 Hz (or 500 + 1000 Hz) tonal components.

From Figure 5.8, the 500 Hz tonal sounds are, on average, more annoying than the 240 Hz sounds without harmonics. Average ratings of the NC-30 tonal sounds (blue line) and equal loudness and roughness sounds (red line) are not significantly different at lower Prominence Ratios ($\leq$ 9 dB) and are significantly different at higher Prominence Ratios ($\geq$ 11 dB). Therefore, tonality is the main driver to perceived annoyance in the less tonal sounds, while other metrics (e.g., loudness) appear to be playing a role in more tonal sounds. The 240 Hz signals with the additional harmonic at 480 Hz were rated as a lot more annoying than the other signals with a single tone at 240 Hz. One reason could be that we have introduced a tone at 480 Hz close to 500 Hz with similar Prominence Ratio as the tone at 240 Hz. Previous research on tonality has shown that people's perception of tonality changes with frequency, peaking at around 700 Hz, rolling off quickly at much lower frequencies and rolling off more gently at higher frequencies. So, the similarity between the results for the 240 Hz harmonic signals and the 500 Hz single tone signals may be due to the role that the 480 Hz harmonic component is playing.

Loudness is known to be the main driver of annoyance. In Figure 5.9, average annoyance rating is plotted against Zwicker Loudness (N_Z), five groups of sounds use the same color as Figure 5.8. The figure shows that the loudness only model is not enough to predict annoyance. The fact that higher tonality sounds are always rated more annoying infers that tonality is playing a role in subjects' annoyance perception.

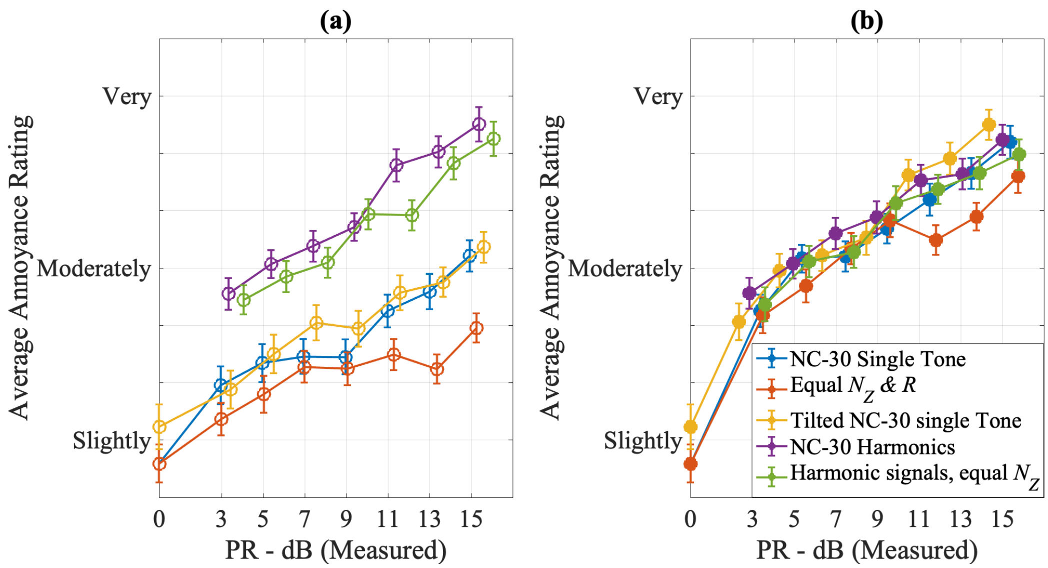

Figure 5.8. Average annoyance ratings of the sounds in short exposure time test. The results are plotted versus the Prominence Ratios calculated from measurements of the sounds close to the subjects' listening location. (a) average rating of sounds with 240 Hz (or 240 + 480 Hz) tonal component (unfilled markers), (b) average rating of sounds with 500 Hz (or 500 + 1000 Hz) tonal component (solid markers).

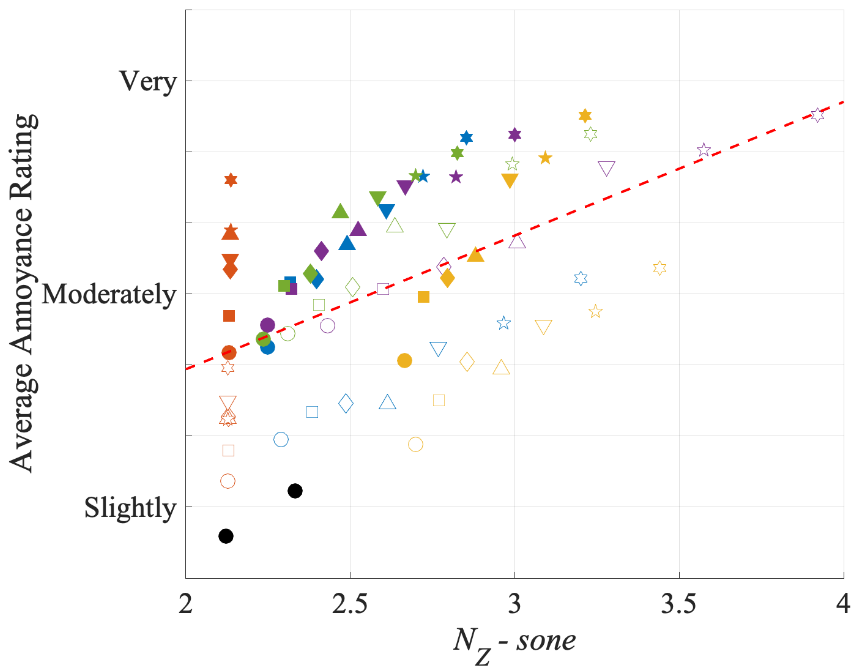

Figure 5.9. Relationship between average annoyance ratings and Zwicker Loudness(N_Z). The red dashed line corresponds to model predicted annoyance with N_Z only, R^2= 0.293. Symbols: fundamental frequency at 240Hz (open), 500 Hz (filled); NC-30+one tone (•), NC30+single tone normalized to have same Loudness and Roughness as neutral NC-30 broadband (•), tilted NC-30+one tone (•), NC-30+tone+harmonic (•), harmonic signal normalized to be the same Loudness as the NC30+single tone at the same Prominence Ratio (•); Prominence Ratio 3 (circle), 5 (square), 7 (diamond), 9 (upward-pointing triangle),11(downward-pointing triangle), 13 (five-pointed star), 15 (six-pointed star) dB.

Long Exposure Time Test Result

Compared with the short exposure time test results, average annoyance ratings for tonal sounds in the long exposure time test are rated consistently lower (in Figure 5.10 & 5.11). Average annoyance ratings for tonal sounds in short exposure time test are, on average, rated 0.57 higher than those in the long exposure time test. By using this relationship, long-term annoyance could be predicted from the short exposure time test results.

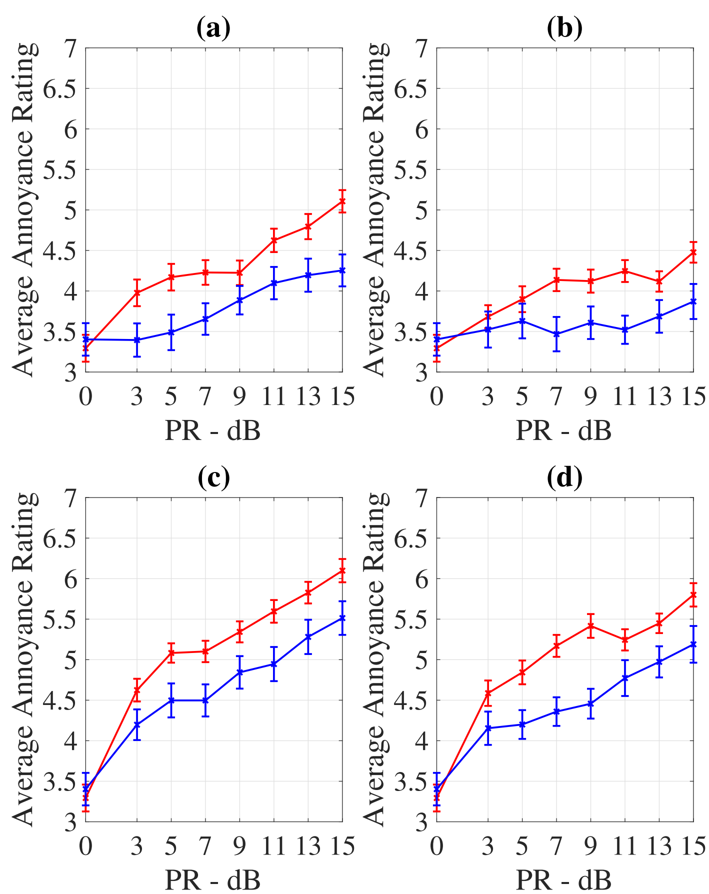

Figure 5.10. Average of annoyance ratings for 5-second sounds (red line) and corresponding 2-minute sounds (blue line). (a) NC-30 broadband with a 240 Hz tone, (b) Equal N_Z & R sounds with a 240 Hz tone, (c) NC-30 broadband with a 500 Hz tone, (d) Equal N_Z & R sounds with a 500 Hz tone.

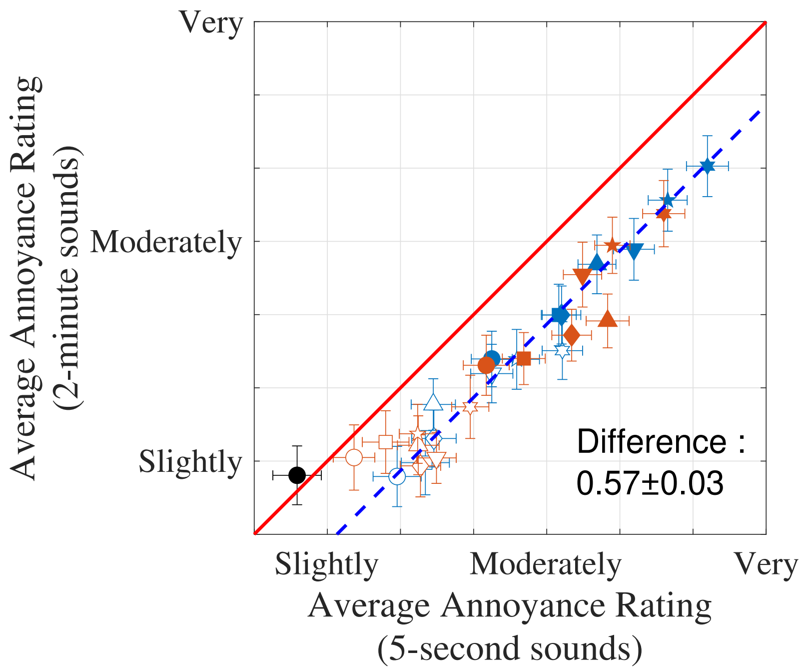

Figure 5.11. Session length effect, average of 5-second sounds annoyance ratings (horizontal axis) against average of corresponding 2-minute sounds annoyance ratings (vertical axis). Unfilled markers represent 240 Hz signals, solid markers represent 500 Hz signals. Color coding (for different groups of sounds) and marker shapes (for different Prominence Ratios) are same as Figure 5.9. Dashed line is the estimated offset of tonal sounds.

5.3.4 Test 2 Summary

Based on Test 2 results, it appears that longer exposure time annoyance can be predicted from short exposure time annoyance. But in Test 2, only tonal sounds with NC-30 broadband are investigated, subjects' annoyance acclimation for tonal sounds with NC-20, NC-40 background sounds still needs investigation.

For low Prominence Ratio signals ($PR \leq 9$ dB), annoyance appears to be mainly due to tonality. For the higher Prominence Ratio sounds ($PR \geq 11$), other metrics (e.g., loudness) start to play a role in annoyance perception. This points towards using a nonlinear function of the measure of tonality in the annoyance model, and this is consistent with tonal penalties used in several applications, e.g., $EPNL$ for aircraft certification [43], and the Danish Environmental Noise Standard.

The presence of the harmonic in sounds with a fundamental of 240 Hz resulted in a large difference in the average annoyance ratings, even when sounds were normalized to have equal loudness, but for the 500 Hz sound, the presence of the 1000 Hz harmonic resulted in very small changes in the average ratings. It may be related to the need for a frequency weighting for individual tonal components, such as the one used in the tone audibility metric or the one used in Aures' tonality model. This needs further investigation. Whether to pick the tonality based on the maximum value of all tones examined in a signal or to add up contributions from all tonal components (as is done in Aures' model of tonality) is also under investigation.

5.4 Subjective Test 3: Test on All Different Type Sounds

Test 3 consists of 6 parts (shown in Table 5.1): four 5-second sound parts that consist of tonal sounds with NC-20, NC-30, tilted NC-30, and NC-40 broadband separately; two 2-minute sound parts for tonal sounds with NC-20, NC-40 broadbands. In total, 222 tonal sounds with different broadbands, different level harmonics, different levels, and frequencies were tested with a purpose to investigate how tonal levels, frequencies, presence of harmonics, and broadband affect annoyance ratings, and whether subjects' annoyance acclimation changes with different broadband levels.

5.4.1 Pre-Test Analysis

To generate a set of harmonic sounds with different levels of 2^{nd} harmonic, a quick internal test was conducted. Three internal members took the test and were asked to rate the dominant tone in the harmonic sounds on a five-point scale ("Low Frequency", "Mostly Low Frequency", "Both", "Mostly High Frequency", "High Frequency").

The internal test signals consisted of an NC-30 broadband component and three harmonic sets (120 + 240 Hz, 240 + 480 Hz, 500 + 1000 Hz), the Prominence Ratios of fundamental tone were either 7 or 15 dB. The Prominence Ratios of 2^{nd} harmonic were changed from 8 dB lower than fundamental to 4 dB higher than fundamental, with an interval 2 dB. Sounds with 7 dB fundamental Prominence Ratio were tested first, then, sounds with 15 dB fundamental Prominence Ratio were tested. All the sounds were tested in order. For each set of harmonics (e.g., 120 + 240 Hz), the test starts with the lowest 2^{nd} harmonic level and gradually increase the level.

Based on this internal harmonic test results, 42 of harmonic sounds were simulated for Test 3. Some details of the harmonic sounds were shown in Table 5.3.

Table 5.3. Prominence ratios of harmonic sounds used in Test 3 (in total 42 sounds)

120 + 240 Hz		**240 + 480 Hz**		**500 + 1000 Hz**	
Fundamental PR	2^{nd} harmonic PR	Fundamental PR	2^{nd} harmonic PR	Fundamental PR	2^{nd} harmonic PR
3 dB	-1, 3, 7, 11 dB	7 dB	-1, 3, 7 dB	7 dB	-1, 3, 7, 11 dB
7 dB	3, 7, 11, 15 dB	11 dB	3, 7, 11 dB	11 dB	3, 7, 11, 15 dB
11 dB	7, 11, 15, 19 dB	15 dB	7, 11, 15 dB	15 dB	7, 11, 15, 19 dB
15 dB	11, 15, 19 dB	19 dB	11, 15, 19 dB	19 dB	11, 15, 19 dB

5.4.2 Stimuli and Correlation Analysis

In Test 3, four 5-second-duration-signals test parts were designed with 4 background noises: NC-30 (Part A), NC-20 (Part B), tilted NC-30 (Part D), NC-40 (Part E). There was some concern that the rating difference for tonal sounds with 5-second and 2-minute duration found in Test 2 might change with different level background noise. Thus, two long exposure time parts were designed, with NC-20 (Part C), NC-40 (Part F) broadbands. The test includes sounds with 7 commonly found frequencies in office noise, as listed in Request-for-Proposal, with different Prominence Ratios, ranging from -1 dB to 19 dB. The test consists of 6 parts. Some sounds were not used due the low frequency limits of the speakers. The test parts were:

1. Part A (5-second duration session): NC-30 broadband + tonal components
    - 5 broadband sounds ranging from NC-30 to NC-38.
    - 43 tonal sounds with a single tone (3 of 43 tonal sounds were not tested for first 5 subjects).
    - 42 harmonic sounds (120 + 240 Hz, 240 + 480 Hz, 500 + 1000 Hz), listed in Table 5.3.
2. Part B (5-second duration session): NC-20 broadband + tonal components
    - 5 broadband sounds ranging from NC-20 to NC-28.
    - 45 tonal sounds with a single tone (3 of 45 tonal sounds were not tested for first 5 subjects).
3. Part C (2-minute duration session): NC-20 broadband + tonal components
    - NC-20 broadband sound.
    - 8 tonal sounds with a single tone.
4. Part D (5-second duration session): NC-30 broadband with different spectrum balance + tonal components

- Tilted NC-30 broadband sound.
- 42 tonal sounds with a single tone (3 of 42 tonal sounds were not tested for first 5 subjects).

5. Part E (5-second duration session): NC-40 broadband + tonal components
   - NC-40 broadband sound.
   - 38 tonal sounds with a single tone (1 tonal sounds was not tested for first 5 subjects).
6. Part F (2-minute duration session): NC-40 broadband + tonal components
   - NC-40 broadband sound.
   - 8 tonal sounds with a single tone.

Prior to the test, the relationship between Loudness, Tonalness, and Sharpness metrics was checked to ensure none of them were highly correlated (shown in Figure 5.12). Testing with sounds with decorrelated metrics would result in a more robust annoyance model. Figure 5.12(a) illustrates the difference of predicted loudness between Zwicker model (N_Z) and Moore and Glasberg model ($N_{M\&G}$). The difference is most obvious for sounds with strong low-frequency tonal components, in these cases, N_Z predicts these sounds louder than $N_{M\&G}$. For each individual part (e.g., NC-30 part), there is a correlation between loudness and tonalness. But for all 222 sounds, the relationship between Moore and Glasberg Loudness ($N_{M\&G}$), Tone-to-Noise Ratio (TNR), von Bismark Sharpness calculated from Moore and Glasberg Loudness ($S_{vB\ M\&G}$) is not clear.

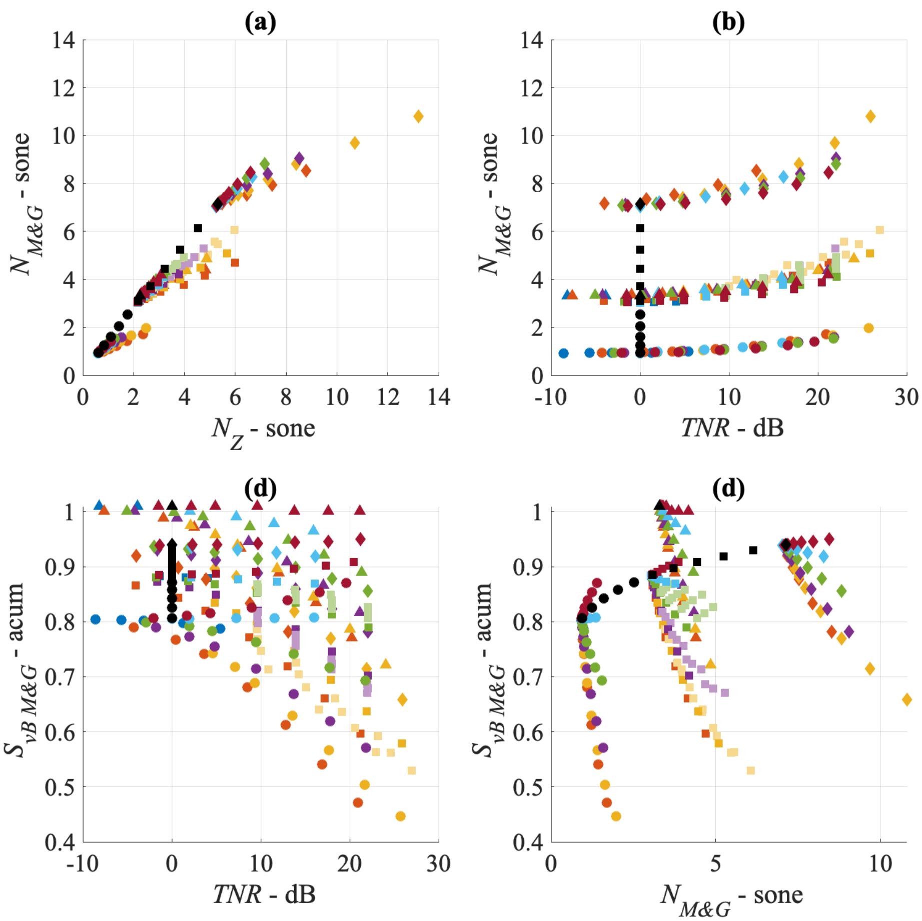

Figure 5.12. Relationship between Loudness, Tonality, and Sharpness metrics. The markers represent tonal sounds with NC-20 broadband (○), with NC-30 broadband (□), with tilted NC-30 broadband (△), with NC-40 broadband(◇). The colors indicates 29.5 Hz (●), 60 Hz (●), 120 Hz (●), 240 Hz (●), 500 Hz (●), 750 Hz (●), 1000 Hz (●), neutral broadband (●)

5.4.3 Test Procedures

Test 3 is a 2-hour test. During the test, the subject was asked to focus on rating in the shorter duration test and to do some typical office work (e.g. reading, writing, etc.) in the longer duration test. In order to even out the ordering effect, half of the subjects (Group 1) took the NC-20 test part ahead of the NC-40 test part, while the other half the subjects (Group 2) took the NC-40 test part earlier. Following are two orders:

- NC-30 (Part A)→ NC-20 (Part B & C) → Tilted NC-30 (Part D)→ NC-40 (Part E & F)
- NC-30 (Part A)→ NC-40 (Part E & F) → Tilted NC-30 (Part D)→ NC-20 (Part B & C)

Subjects were asked to leave the lab when the ambient broadband noise changed. In Test 3, ambient broadband noise is the lowest broadband in the test part (e.g. NC-30 for Part A). An optional 5-minute break was given before the part with tilted NC-30 broadband. Test Parts A, B, D, E started after a familiarization session with 6 sample sounds and a practice session with 6 more sounds.

5.4.4 Subjects

In Test 3 involved 30 subjects (13 males, 17 females), aged 19-58. Average age across all subjects was 26.4 years with a standard deviation of 8.0. A Noise-Sensitivity-Questionnaire (NoiSeQ) [42] was included. The average of NoiSeQ score (5 – most sensitive, 0 – least sensitive) was 3.06 with a standard deviation of 0.51. A profile of subject is shown in Figure 5.13.

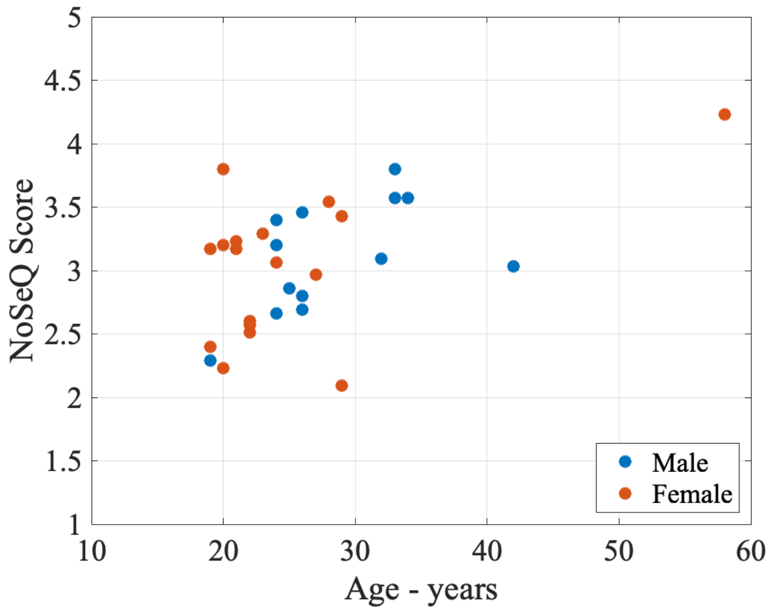

Figure 5.13. Profile of subjects' gender, age, and NoSeQ score.

5.4.5 Data Analysis

The average annoyance ratings for Group 1 and Group 2 subjects are plotted against each other in Figure 5.14. Most annoyance ratings were close to the 45-degree red line, thus, different parts' ratings are not significantly affected by different testing order.

Although people with different test orders rated sounds, on average, similarly, people with different gender and NoSeQ score did rate tonal sounds differently. Figure 5.15(a) illustrates the difference in rating between female and male subjects. Female subjects tended to rate high-frequency tonal sounds more annoying and to rate low-frequency tonal sounds as annoying as male subjects. Figure 5.15(b) infers that low tonality sounds were rated similarly regardless of the NoSeQ score, while high tonality sounds tended to be rated more annoying by the subject with higher NoSeQ scores.

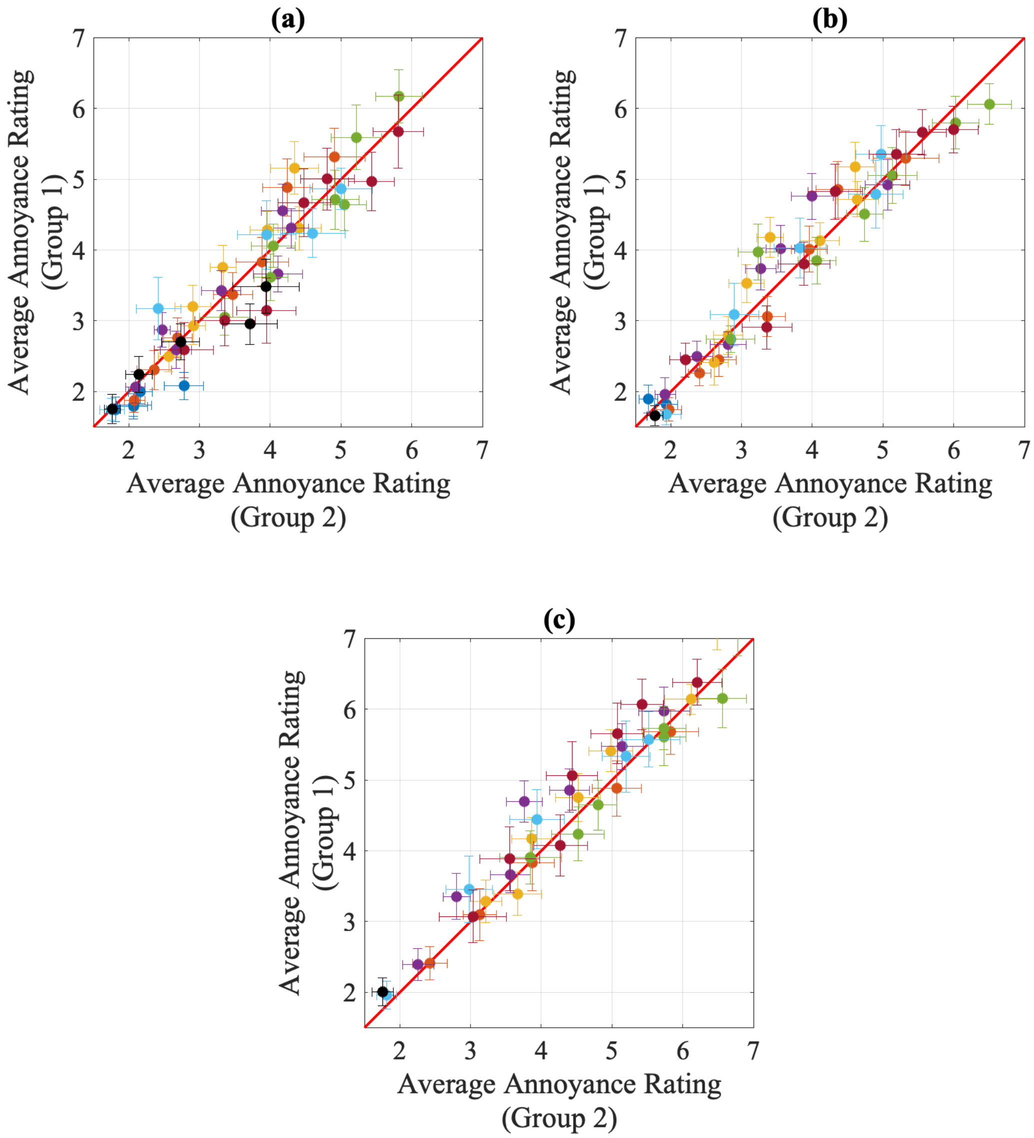

Figure 5.14. Comparison of Group 1 (vertical axis) and Group 2 (horizontal axis) subjects' average annoyance ratings. (a) NC-20 test part, (b) tilted NC-30 test part, (c) NC-40 test part. Group 1: NC-30→NC-20→Tilted NC-30→NC-40, Group 2: NC-30→NC-40→Tilted NC-30→NC-20. The colors indicates 29.5 Hz (●), 60 Hz (●), 120 Hz (●), 240 Hz (●), 500 Hz (●), 750 Hz (●), 1000 Hz (●), neutral broadband (●).

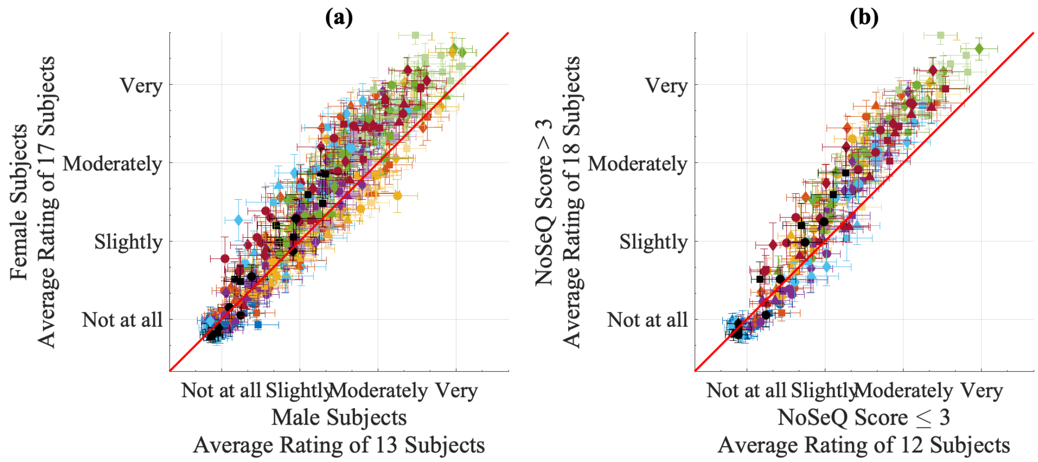

Figure 5.15. Gender, self-reported noise sensitivity effect. (a) difference of average ratings between female subjects and male subjects, (b) difference of average ratings between subjects with different NoSeQ scores. The markers indicate tonal sounds with NC-20 broadband (○), NC-30 broadband (□), tilted NC-30 broadband (△), NC-40 broadband(◇). The color coding is same as Figure 5.14.

From Figure 5.11 & 5.16, acclimation effects present in all sessions: NC-20, NC-30 (Test 2) and NC-40. In all 3 tests, the broadband sounds were rated almost the same while tonal sounds were rated consistently more annoying in the shorter test. The average difference in ratings between short and long duration signals is greater as broadband NC level increases: NC-20: 0.44 ± 0.07; NC-30: 0.56 ± 0.03; NC-40: 0.71 ± 0.11. It appears that average annoyance ratings of 5-second sounds could be used to predict annoyance for 2-minute sounds.

As for the tonality metrics, PR, TNR, and ΔL_{ta} would come up with multiple values for multiple tones, while Aures Tonality adds up contributions for all tonal components. Thus, for sounds with harmonics, correlation between PR, TNR, ΔL_{ta} and average annoyance ratings were calculated based on the maximum value or summed value through the following formula:

$$Summed\ Tonality = 10 \times log_{10} \sum_{i=1}^{n} 10^{\frac{Tonality_i}{10}} \tag{5.2}$$

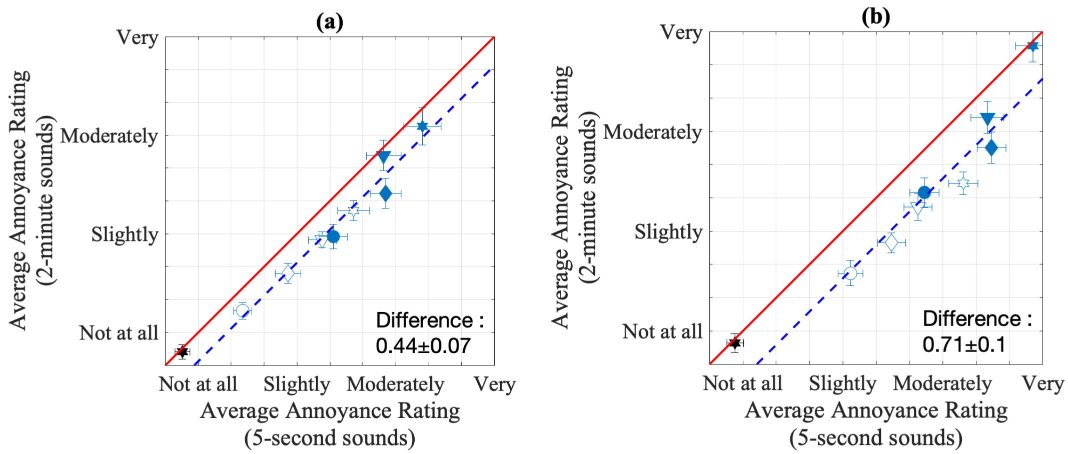

Figure 5.16. Session length effect, average of 5-second sounds annoyance ratings (horizontal axis) against average of 2-minute sounds annoyance ratings (vertical axis). (a) Tonal sounds with NC-20 broadband. (b) Tonal sounds with NC-40 broadband. Markers and color coding are same as Figure 5.9. Dashed lines are estimated offsets of tonal sounds.

The tonal sounds used in the test were generated based on PR, but TNR, and ΔL_{ta} turned out to be more correlated to ratings (shown in Table 5.4). TNR was chosen to illustrate data as it performs better in the metrics model.

Table 5.4. Correlation between different Tonality metrics and annoyance ratings.

Correlation with ratings	PR	TNR	ΔL_{ta}	Aures Tonality
Maximum Tonlaity	0.771	0.823	0.863	/
Summed Tonality	0.780	0.834	0.860	0.7784

For TNR, integrating contributions from different tones works better than simply picking the maximum value. In Figure 5.17, ratings for harmonic sounds were closer to each other when they are plotted against the summed TNR. Although the loudness effect is not considered in this case, it provides a potential reason why summed TNR

works better in the annoyance model. A summed TNR was used for harmonic sounds in later analysis.

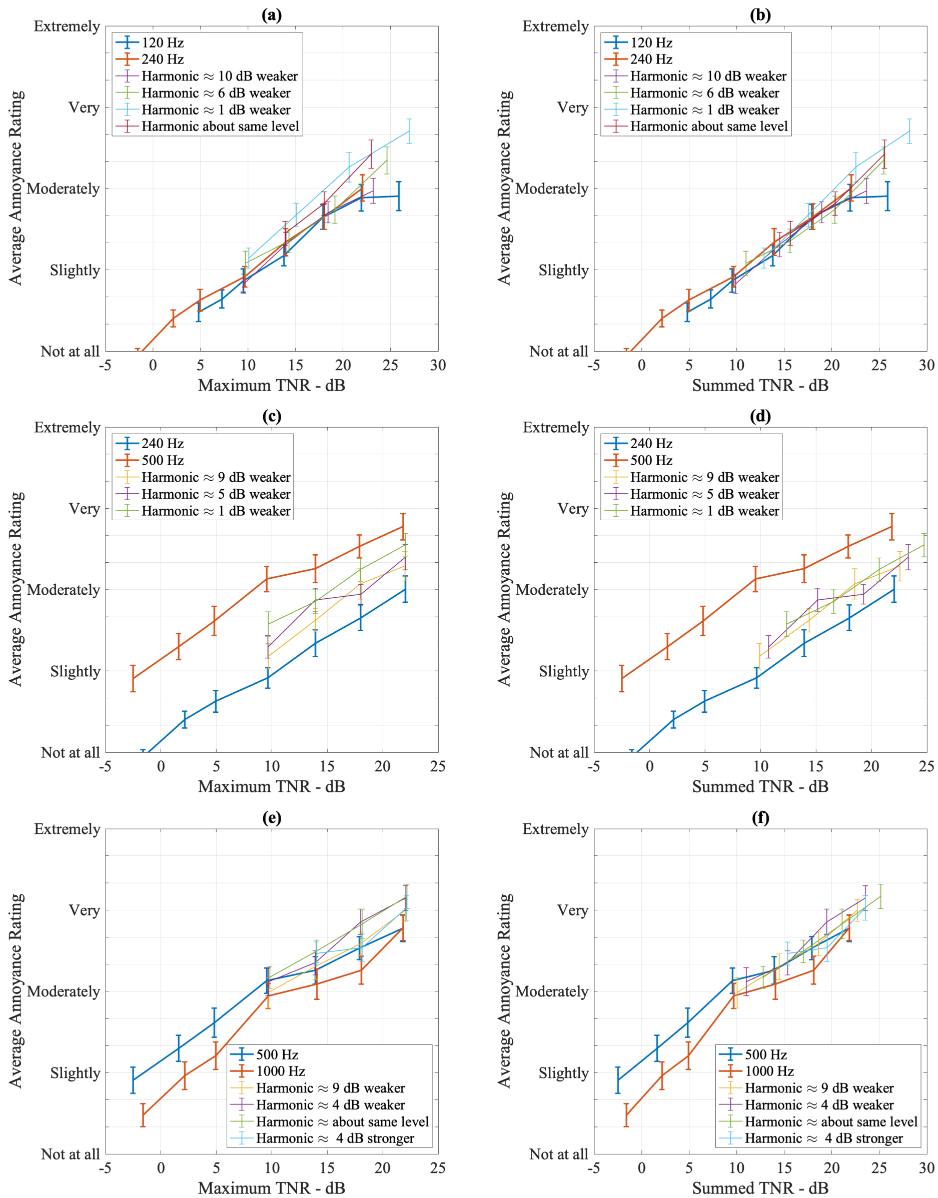

Figure 5.17. Average of annoyance ratings of harmonic sounds and corresponding single tone sounds plot against maximum TNR (left) and summed TNR (right).(a)-(b) NC30 broadband with 120 + 240 Hz tonal components, (c)-(d) with 240 + 480 Hz tonal components, (e)-(f) with 500 + 1000 Hz tonal components.

Most of the average annoyance ratings follow a general trend when plotting versus TNR, while in NC-20 session, sounds with 29.5 Hz tonal were rated less annoying than other tonal sounds (shown in Figure 5.18). One reason is in this session, 29.5 Hz tonal sounds were close to the hearing threshold at around 29.5 Hz.

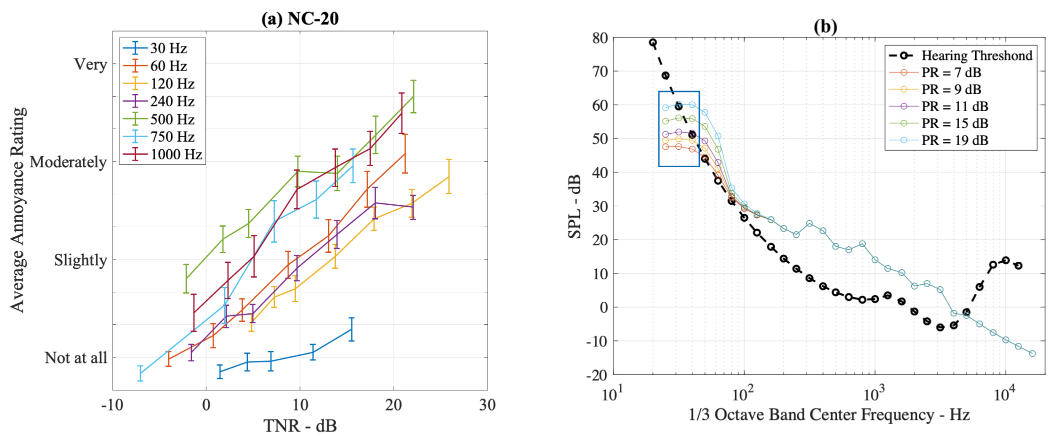

Figure 5.18. Average annoyance ratings and spectrums for some tonal sounds with NC-20 broadband. (a) average annoyance ratings of 5-second tonal sounds with neutral NC-20 broadband, (b) 1/3-Octave band plot of 29.5 Hz outlier sounds (plot with hearing threshold).

To compromise this issue, $Modified\,TNR$ was proposed. Instead of using masking noise alone, it introduced the hearing threshold term E_H as Aures Model did.

$$Modified\,TNR = 10 \times log_{10}\frac{W_t}{W_n + E_H}\ dB \tag{5.3}$$

$Modified\,TNR$ fixed some outliers at 29.5 Hz. The improvement by introducing the hearing threshold to TNR can be seen from Figure 5.19. Tonal sounds with higher frequency tones are rated more annoying. This consisting tendency drove us to look at tonality metrics' frequency weighting. In the NC-40 test part, 60 Hz sounds were rated more annoying, possibly due to the larger loudness and higher tonal component level.

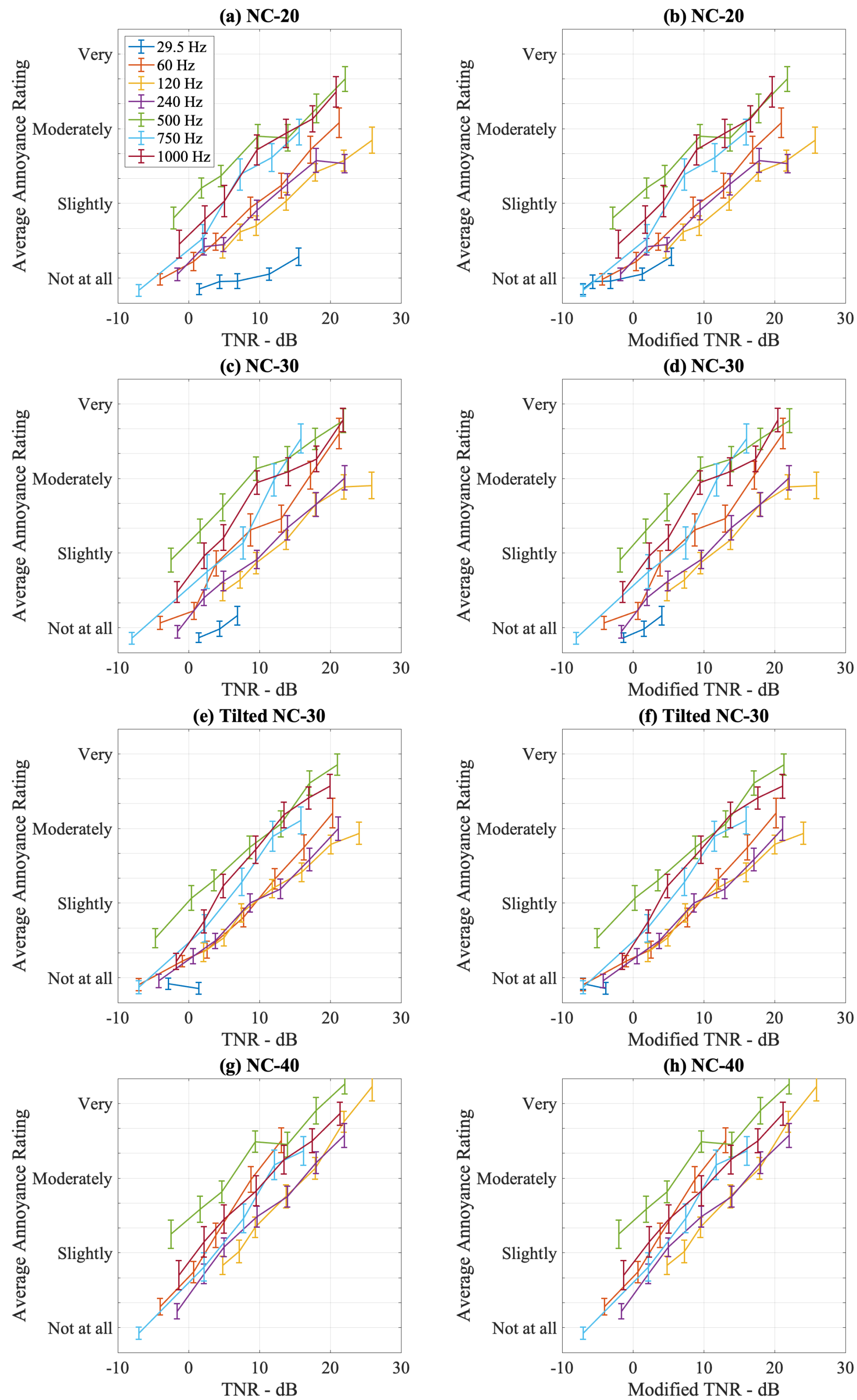

Figure 5.19. Averaged of annoyance ratings of 5-second test parts plot against TNR (left) and $Modified\ TNR$ (right).

Across the test, subjects used scale differently in different parts (see Figure 5.20). In Test 3, the background level in the room when subjects entered for a test part was set to the lowest level broadband non-tonal sound, so perhaps subjects just rated sounds relative to the background level in the room, rather than using an absolute scale. Thus, a global scale needs to be introduced because if broadband sounds were played at different NC levels, based on many studies in the literature, the perceived annoyance should go up with NC level. This problem needs to be addressed in the annoyance model that works for sounds in all different parts.

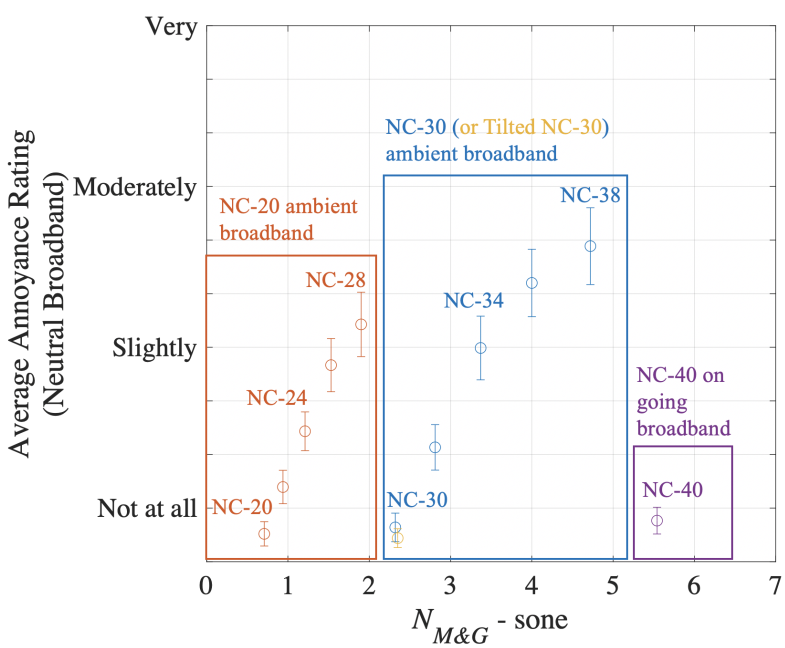

Figure 5.20. Average annoyance ratings of neutral broadband sounds tested in different short exposure time tests plot versus Moore and Glasberg Loudness ($N_{M\&G}$).

5.4.6 Test 3 Summary

For tonal office sounds, TNR appears to be a better tonality metric to predict changes in annoyance due to the presence of tones. For the tonal sounds with harmonics, annoyance is a combined result of multiple tones. Summing up $TNRs$ of all harmonics in a signal on an unlogged scale (and logging the result) is a better predictor of tonal contributions to subjects' annoyance responses. Introducing hearing threshold would also help predict low-level tonal sounds.

As subjects were rating sounds with different baseline in different parts, the global model needs to comfort this effect. The model would be able to explain loudness and tonality's effect on annoyance more correctly. Differences between ratings in Test 2, with an NC-20 background in the room, but the stimuli had a baseline of NC-30, and the ratings in the NC-30 background part of Test 3, where the background level in the room was NC-30, appear support this hypothesis.

5.5 Summary

Test results show the broadband sounds were rated almost as annoying in short and long sessions, while tonal sounds were rated consistently more annoying in the shorter test.

Subjects' perception of tones is dependent on both the prominence of the tone and the frequency. As tonality increases, the average annoyance increases. Tonal sounds with high frequencies (500, 750, 1000 Hz) were rated more annoying. The presence of harmonics would make sounds more annoying. The annoyance of the harmonic sounds sometimes is mainly driven by one dominant tone, sometimes is due to the combined effect of the whole harmonic set. Picking the maximum tonality metric value is not enough to examine the tonality of harmonic sounds, a summed version of tonality metric is used. To fix some outliers (29.5 Hz tonal sounds in NC-20 test), a modification to TNR was implemented by introducing the hearing threshold to masking noise term, as is done in the Aures Model.

A tonal sound with similar tonality would be perceived more annoying with NC-40 broadband than NC-30 broadband. But due to the inconsistencies in ratings, broadband's contributions to annoyance is hard to judge without a global model.

6. DEVELOPMENT OF ANNOYANCE MODELS

In this chapter, two methods would be presented to develop global annoyance models. Both methods aim to solve the subjects' inconsistencies in Ratings. Several ways to improve model performance would be discussed. With the studied annoyance acclimation pattern for tonal sounds with different broadbands (NC-20, NC-30, NC-40), annoyance ratings for 5-second sounds from Test 3 have been transferred to those for 2-minute sounds. Annoyance models (built with either 5-second or 2-minute sounds' ratings) would be validated by Test 2 ratings.

6.1 Apparent Inconsistencies in Ratings

Figure **??** illustrates inconsistencies in ratings. The average annoyance rating for NC-32 neutral broadband tested with NC-30 ambient broadband was rated less annoying than the average annoyance rating of NC-28 tested with NC-20 ambient broadband. Subjects rated the annoyance relative to the ambient background noise rather than rate just based on the annoyance scale. The rating scale was used differently in different parts. Thus, it's not desirable to use a direct linear regression with metrics. Inconsistencies need to be resolved in the global model.

6.2 Global Annoyance Models (5-second Ratings)

Two global annoyance models were proposed to deal with the inconsistencies in ratings. Global Annoyance Model 1 maps different parts ratings to one global scale, while Global Annoyance Model 2 incorporates the ambient background noise (broadband) noise used in that part of the test.

6.2.1 Global Annoyance Model 1

One straightforward idea was to deal with this issue by mapping results (average annoyance ratings) from different parts' to one global scale, then develop a global annoyance model with the adjusted Ratings. Based on the selected metrics (Moore and Glasberg Loudness and the Modified Tone-to-Noise Ratio), a regression method was derived to estimate an offset (intercepts: $\alpha_A, \alpha_B, \alpha_D, \alpha_E$) and a scaling (gradients: $\beta_A, \beta_B, \beta_D, \beta_E$) for the ratings from 5-second test parts A, B, D, and E, respectively. The goal in the adjustment was to ensure that broadband Ratings align, as it would be expected that annoyance would grow with increased loudness with these neutral sounds if they all played within a single test.

Global Scale Mapping: with Offset

To start with, the mapping to the global annoyance scale is done by adding offsets to different parts' ratings:

$$Modified\ Rating_{i,j} = Rating_{i,j} + \alpha_i \tag{6.1}$$

i donates Part A, Part B, Part D or Part E, j is the signal number. For a group of sounds with very small tonality (39 sounds with $Modified\ TNR \leq 0$), based on the studies from literature, perceived annoyance would be mainly driven by loudness. Thus, the modified ratings can be modeled as a linear model with Moore and Glasberg Loudness ($N_{M\&G}$) only.

$$Modified\ Rating_{i,j} = Rating_{i,j} + \alpha_i = c_0 + c_1 N_{M\&G_{i,j}} \tag{6.2}$$

Offsets are estimated by solving the following equations in a least square error method.

$$\begin{pmatrix} Rating_{A,j_{A,1}} \\ \vdots \\ Rating_{A,j_{A,N_A}} \\ Rating_{B,j_{B,1}} \\ \vdots \\ Rating_{B,j_{B,N_B}} \\ Rating_{D,j_{D,1}} \\ \vdots \\ Rating_{D,j_{D,N_D}} \\ Rating_{E,j_{E,1}} \\ \vdots \\ Rating_{E,j_{E,N_E}} \end{pmatrix} = \begin{pmatrix} -1 & 0 & 0 & 0 & N_{M\&G_{A,,j_{A,1}}} \\ \vdots & \vdots & \vdots & \vdots & \vdots \\ -1 & 0 & 0 & 0 & N_{M\&G_{A,j_{A,N_A}}} \\ 0 & -1 & 0 & 0 & N_{M\&G_{B,j_{B,1}}} \\ \vdots & \vdots & \vdots & \vdots & \vdots \\ 0 & -1 & 0 & 0 & N_{M\&G_{B,j_{B,N_B}}} \\ 0 & 0 & -1 & 0 & N_{M\&G_{D,j_{D,1}}} \\ \vdots & \vdots & \vdots & \vdots & \vdots \\ 0 & 0 & -1 & 0 & N_{M\&G_{D,j_{D,N_D}}} \\ 0 & 0 & 0 & -1 & N_{M\&G_{E,j_{E,1}}} \\ \vdots & \vdots & \vdots & \vdots & \vdots \\ 0 & 0 & 0 & -1 & N_{M\&G_{E,j_{E,N_E}}} \end{pmatrix} \begin{pmatrix} \alpha_A - c_0 \\ \alpha_B - c_0 \\ \alpha_D - c_0 \\ \alpha_E - c_0 \\ c_1 \end{pmatrix} + \begin{pmatrix} \epsilon_{A,j_{A,1}} \\ \vdots \\ \epsilon_{A,j_{A,N_A}} \\ \epsilon_{B,j_{B,1}} \\ \vdots \\ \epsilon_{B,j_{B,N_B}} \\ \epsilon_{D,j_{D,1}} \\ \vdots \\ \epsilon_{D,j_{D,N_D}} \\ \epsilon_{E,j_{E,1}} \\ \vdots \\ \epsilon_{E,j_{E,N_E}} \end{pmatrix} \tag{6.3}$$

N_A, N_B, N_D, N_E are the number of sounds with small tonality ($Modified\ TNR \leq 0$) in Parts A, B, D and E, $j_{A,k}$ corresponds to k^{th} small tonality sound in Part A. Assume $\alpha_B = 0$ (mapping ratings to Part B (NC-20) ratings), different offsets are estimated to solve the inconsistencies in rating.

Table 6.1. Modify ratings with offset to align ratings of Parts A, B, D, E. Offsets are estimated by tonal sounds with $Modified\ TNR \leq 0$.

Part A (NC-30)	$Modified\ Rating_{A,j} = Rating_{A,j} + 1.75$
Part B (NC-20)	$Modified\ Rating_{B,j} = Rating_{B,j}$
Part D (tilted NC-30)	$Modified\ Rating_{D,j} = Rating_{D,j}$ +2.09
Part E (NC-40)	$Modified\ Rating_{E,j} = Rating_{E,j} + 4.72$

The modified ratings are then scaled so the lowest average modified annoyance rating matched the lowest rating (unadjusted) over all 4 parts of the test and the highest average modified annoyance rating matched the highest average annoyance rating. The performance of a two metrics annoyance model (with Moore and Glasberg Loudness and Tone-to-Noise Ratio modified to include the hearing threshold) is presented in Figure 6.1. Average ratings for neutral broadband sound align in the global scale.

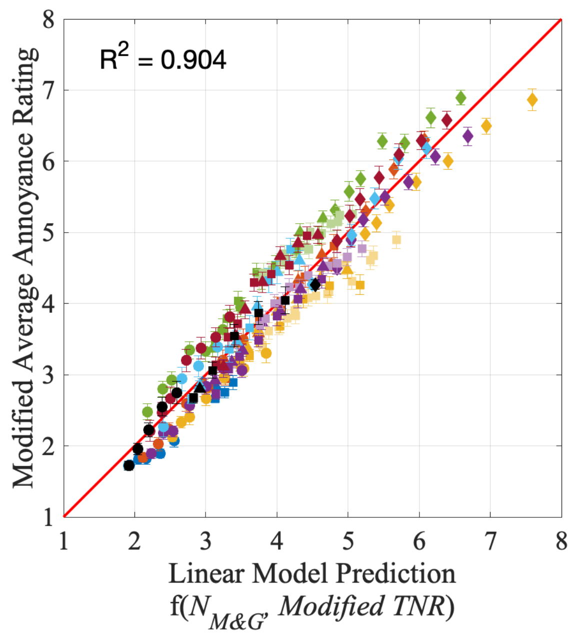

Figure 6.1. Average annoyance prediction (horizontal axis) from a linear model (offset only Global Model 1 with 5-second ratings) with two metrics: Moore and Glasberg Loudness, Modified Tone-to-Noise Ratio. Numerical values of 2, 3.5, 5, 6.5, 8 corresponds to not at all, slightly, moderately, very, extremely annoying. The markers indicate tonal sounds with NC-20 broadband (○), NC-30 broadband (□), tilted NC-30 broadband (△), NC-40 broadband(◇). The colors donate 29.5 Hz tonal sounds (•), 60 Hz tonal sounds (•), 120 Hz tonal sounds (•), 120 + 240 Hz tonal sounds(•), 240 Hz tonal sounds(•), 240 + 480 Hz tonal sounds (•), 500 Hz tonal sounds (•), 500 + 1000 Hz tonal sounds (•), 750 Hz tonal sounds (•), 1000 Hz tonal sounds (•), neutral broadband (•)

Global Scale Mapping: with Offset and Scaling

Apart from the offset, subjects may also potentially compress their ratings differently in different test parts. Thus, additional scaling is taking into account. In this method, all the test sounds are considered, and multiple metrics (e.g., $N_{M\&G}$ and $Modified\ TNR$) are used in building the global annoyance scale. As more sound quality metrics are considered in this method, ratings would be mapped based on a linear combination of different metrics, for example, $c_0+c_1N_{M\&G}+c_2Modified\,TNR$. The mapping considers both offset and scaling:

$$Modified\ Rating_{i,j} = \beta_i Rating_{i,j} + \alpha_i \tag{6.4}$$

Firstly, an offset and scaling are estimated with a least square method for different metrics (e.g. $N_{M\&G}$ or $Modified\ TNR$):

$$
\begin{pmatrix}
N_{M\&G_{A,1}} \\ \vdots \\ N_{M\&G_{A,90}} \\ N_{M\&G_{B,1}} \\ \vdots \\ N_{M\&G_{B,50}} \\ N_{M\&G_{C,1}} \\ \vdots \\ N_{M\&G_{C,43}} \\ N_{M\&G_{D,1}} \\ \vdots \\ N_{M\&G_{D,39}}
\end{pmatrix}
=
\begin{pmatrix}
1 & 0 & 0 & 0 & Rating_{A,1} & 0 & 0 & 0 \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\
1 & 0 & 0 & 0 & Rating_{A,90} & 0 & 0 & 0 \\
0 & 1 & 0 & 0 & 0 & Rating_{B,1} & 0 & 0 \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\
0 & 1 & 0 & 0 & 0 & Rating_{B,50} & 0 & 0 \\
0 & 0 & 1 & 0 & 0 & 0 & Rating_{C,1} & 0 \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\
0 & 0 & 1 & 0 & 0 & 0 & Rating_{C,43} & 0 \\
0 & 0 & 0 & 1 & 0 & 0 & 0 & Rating_{D,1} \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\
0 & 0 & 0 & 1 & 0 & 0 & 0 & Rating_{D,39}
\end{pmatrix}
\begin{pmatrix}
\alpha_A^N \\ \alpha_B^N \\ \alpha_D^N \\ \alpha_E^N \\ \beta_A^N \\ \beta_B^N \\ \beta_D^N \\ \beta_E^N
\end{pmatrix}
+
\begin{pmatrix}
\epsilon_{A,1}^N \\ \vdots \\ \epsilon_{A,90}^N \\ \epsilon_{B,1}^N \\ \vdots \\ \epsilon_{B,50}^N \\ \epsilon_{D,1}^N \\ \vdots \\ \epsilon_{D,43}^N \\ \epsilon_{E,1}^N \\ \vdots \\ \epsilon_{E,39}^N
\end{pmatrix}
\tag{6.5}
$$

$$
\begin{pmatrix}
Modfied\ TNR_{A,1} \\
\vdots \\
Modfied\ TNR_{A,90} \\
Modfied\ TNR_{B,1} \\
\vdots \\
Modfied\ TNR_{B,50} \\
Modfied\ TNR_{C,1} \\
\vdots \\
Modfied\ TNR_{C,43} \\
Modfied\ TNR_{D,1} \\
\vdots \\
Modfied\ TNR_{D,39}
\end{pmatrix}
=
\begin{pmatrix}
1 & 0 & 0 & 0 & Rating_{A,1} & 0 & 0 & 0 \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\
1 & 0 & 0 & 0 & Rating_{A,90} & 0 & 0 & 0 \\
0 & 1 & 0 & 0 & 0 & Rating_{B,1} & 0 & 0 \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\
0 & 1 & 0 & 0 & 0 & Rating_{B,50} & 0 & 0 \\
0 & 0 & 1 & 0 & 0 & 0 & Rating_{C,1} & 0 \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\
0 & 0 & 1 & 0 & 0 & 0 & Rating_{C,43} & 0 \\
0 & 0 & 0 & 1 & 0 & 0 & 0 & Rating_{D,1} \\
\vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots & \vdots \\
0 & 0 & 0 & 1 & 0 & 0 & 0 & Rating_{D,39}
\end{pmatrix}
\begin{pmatrix}
\alpha_A^{TNR} \\
\alpha_B^{TNR} \\
\alpha_D^{TNR} \\
\alpha_E^{TNR} \\
\beta_A^{TNR} \\
\beta_B^{TNR} \\
\beta_D^{TNR} \\
\beta_E^{TNR}
\end{pmatrix}
+
\begin{pmatrix}
\epsilon_{A,1}^{TNR} \\
\vdots \\
\epsilon_{A,90}^{TNR} \\
\epsilon_{B,1}^{TNR} \\
\vdots \\
\epsilon_{B,50}^{TNR} \\
\epsilon_{D,1}^{TNR} \\
\vdots \\
\epsilon_{D,43}^{TNR} \\
\epsilon_{E,1}^{TNR} \\
\vdots \\
\epsilon_{E,39}^{TNR}
\end{pmatrix}
\tag{6.6}
$$

To simplify the equations, Equation 6.5 and 6.6 are rewritten as: $\left[N_{M\&G}\right] = \left[R\right]\left[X^N\right] + \left[\epsilon^N\right]$, $\left[Modified\ TNR\right] = \left[R\right]\left[X^{TNR}\right] + \left[\epsilon^{TNR}\right]$.

$\left[R\right]$ is a matrix that consists of 4 test parts' average ratings. $\left[X^N\right]$, $\left[X^{TNR}\right]$ are the offsets and scaling estimated with least square method. $\left[\epsilon^N\right]$, $\left[\epsilon^{TNR}\right]$ are parts of the metrics value that can't be explained by shifting and compressing the ratings. By combining two equations:

$$\left[c_1 N_{M\&G} + c_2 Modified\ TNR\right] = \left[R\right]\left[c_1 X^N + c_2 X^{TNR}\right] + \left[c_1\epsilon^N + c_2\epsilon^{TNR}\right] \quad (6.7)$$

This is equivalent to:

$$\left(\begin{bmatrix} N_{M\&G} & Modified\ TNR \end{bmatrix} - \left[R\right]\begin{bmatrix} X^N & X^{TNR} \end{bmatrix}\right)\begin{bmatrix} c_1 \\ c_2 \end{bmatrix} = \left[c_1\epsilon^N + c_2\epsilon^{TNR}\right] \quad (6.8)$$

For $\left[A\right] = \begin{bmatrix} N_{M\&G} & Modified\ TNR \end{bmatrix} - \left[R\right]\begin{bmatrix} X^N & X^{TNR} \end{bmatrix}$, $\left[C\right] = \begin{bmatrix} c_1 \\ c_2 \end{bmatrix}$, the error $\left[c_1\epsilon^N + c_2\epsilon^{TNR}\right] = \left[A\right]\left[C\right]$.

For a given length $\left[C\right]$, the error is minimized when $\left[C\right]^t\left(\left[A\right]^t\left[A\right]\right)\left[C\right]$ is minimized. This indicates that $\left[C\right]$ is the eigenvector corresponding to smallest eigenvalue of $\left[A\right]^t\left[A\right]$. With given c_1, c_2, the offsets and scaling for this method can be calculated with:

$$\begin{pmatrix} \alpha_A - c_0 \\ \alpha_B - c_0 \\ \alpha_D - c_0 \\ \alpha_E - c_0 \\ \beta_A \\ \beta_B \\ \beta_D \\ \beta_E \end{pmatrix} = c_1 \begin{pmatrix} \alpha_A^N \\ \alpha_B^N \\ \alpha_D^N \\ \alpha_E^N \\ \beta_A^N \\ \beta_B^N \\ \beta_D^N \\ \beta_E^N \end{pmatrix} + c_2 \begin{pmatrix} \alpha_A^{TNR} \\ \alpha_B^{TNR} \\ \alpha_D^{TNR} \\ \alpha_E^{TNR} \\ \beta_A^{TNR} \\ \beta_B^{TNR} \\ \beta_D^{TNR} \\ \beta_E^{TNR} \end{pmatrix} \quad (6.9)$$

By assuming $\alpha_B = 0, \beta_B = 1$, all the ratings are mapped to Part B (test with NC-20 neutral broadband). The mapping is shown in Table 6.2. The scaling for Part E (test with NC-40 neutral broadband) is slightly larger than the other scaling.

Table 6.2. Modify ratings with offset and scaling to align ratings of Parts A, B, D, E. Parameters are estimated with $N_{M\&G}$ and $Modified\ TNR$.

Part A (NC-30)	$Modified\ Rating_{A,j} = 1.03\ Rating_{A,j} + 1.88$
Part B (NC-20)	$Modified\ Rating_{B,j} = 1.00\ Rating_{B,j}$
Part D (tilted NC-30)	$Modified\ Rating_{D,j} = 0.94\ Rating_{D,j} + 2.38$
Part E (NC-40)	$Modified\ Rating_{E,j} = 1.25\ Rating_{E,j} + 4.74$

The modified ratings are then scaled so the lowest and highest average modified annoyance rating matched the lowest and highest rating in all 4 parts of the test. The performance of a two metrics annoyance model is shown in Figure 6.2. By introducing additional scaling, the model achieves a slightly higher R^2 value.

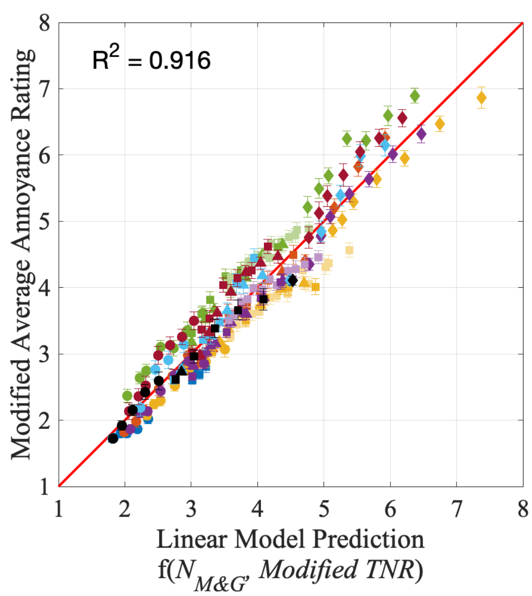

Figure 6.2. Average annoyance prediction (horizontal axis) from a linear model (offset and scaling Global Model 1 with 5-second ratings) with the metrics: Moore and Glasberg Loudness ($N_{M\&G}$) and Tone-to-Noise Ratio modified to include the hearing threshold ($Modified\ TNR$). Markers and color coding are same as Figure 6.1

6.2.2 Global Annoyance Model 2

Another idea, based on the idea that people acclimate to the ambient sound and then rate changes in the sounds heard relative to that ambient level, was to use ambient broadband characteristics in the model. Thus, instead of directly using the predicted loudness of the sound from the Moore and Glasberg Loudness model (as in Model 1), a relative loudness $\Delta N_{M\&G}$ was used. The relative loudness is the difference between the predicted loudness of the sound that was rated and the predicted loudness of the ambient sound. A relative sharpness was also introduced into the model to fix the overprediction caused by tonality, but this is a calculation relative to the broadband level of the sound with the tones removed. To calculate relative sharpness $\Delta S_{vB\ M\&G}$, the broadband spectrum was first extracted by detecting and removing tonal components in the power spectrum density. The relative sharpness was achieved by subtracting broadband sharpness from the tonal sound's sharpness. Relative loudness, Modified TNR, relative sharpness all turned out to be significant in the model (shown in Figure 6.3). A relatively high R^2 value ($R^2 = 0.872$) can be achieved when using this global model, but some outliers still can be identified.

6.2.3 Metrics Refinement

Frequency Weighting

The model over predicts the annoyance due to the 29.5 Hz tonal sounds, in Figure 6.3(c). One possible reason is the lack of frequency weighting in tonality metrics. By implementing a frequency weighting that is used in the well-established Aures Model this problem can be corrected. Including this in the $Modified\ TNR$ results in:

$$Modified\ TNR' = Modified\ TNR + 10log_{10}\frac{1}{\left(\sqrt{1 + 0.2\left(\frac{f}{700} + \frac{700}{f}\right)^2}\right)^{0.29}} \quad (6.10)$$

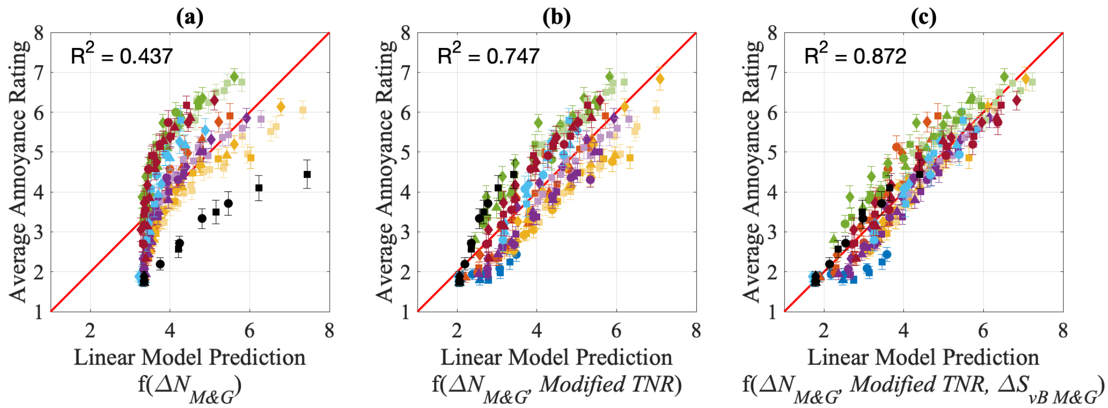

Figure 6.3. Global Model 2 (5-second rating) predictions with metrics: (a) only relative Loudness (relative to ambient loudness), (b) relative Loudness and Modified Tone-to-Noise Ratio, and (c) relative Loudness, Modified Tone-to-Noise Ratio, and relative Sharpness (relative to the Sharpness of the broadband component in the sound).

This weighting rolls off at low frequencies and at high frequencies, and is a maximum at 700 Hz, which is the frequency where people are most sensitive to the tonalness of a sound. The results before and after introducing the frequency weighting are shown in Figure 6.4.

Oppressiveness Penalty

In Figure 6.4(b), still some groups of sounds are under-predicted: 60 Hz sounds (orange markers) and 500 Hz sounds (green markers). Nakamura and Tokita's work on the perception of low-frequency sounds [37] were looked into due to the under-prediction at 60 Hz. They developed contour maps based on levels in 1/3rd Octave bands from people's responses to low frequency sounds (see Figure 6.5(a)). The contours represent boundaries between, e.g., detectable and annoying, annoying and displeasing, displeasing and oppressive/detect vibration.

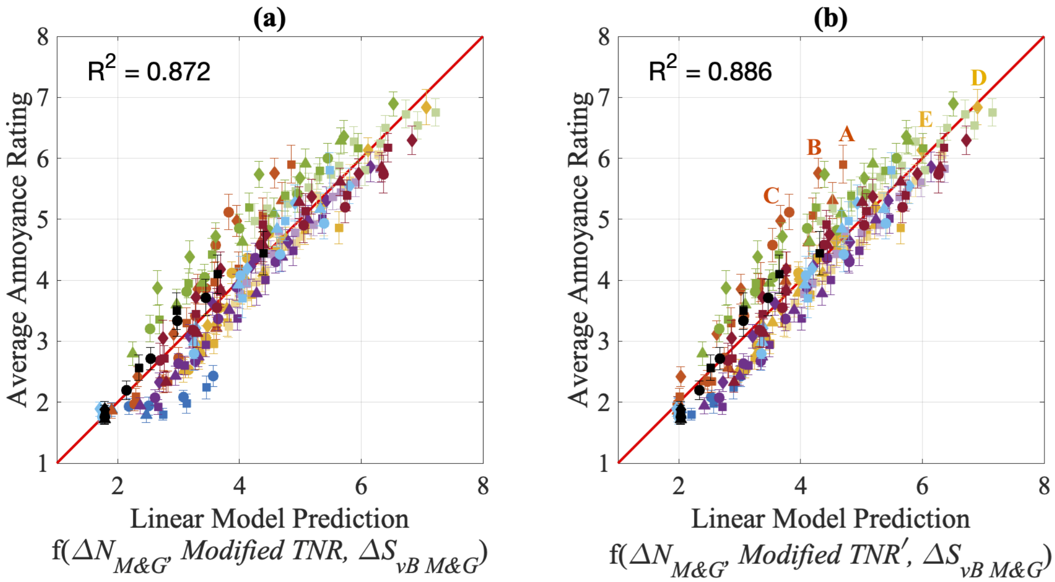

Figure 6.4. Effect of introducing frequency weighting. (Global Model 2 with 5-second ratings) (a) tonality without frequency weighting $R^2 = 0.872$, (b) tonality with frequency weighting $R^2 = 0.886$. Sounds with high level low-frequency tones are labeled from A-E.

From Figure 6.4 and 6.5, most low-frequency outliers had portions of their one-third octave spectra above the annoying - displeasing contour, and some had portions about the displeasing – oppressive/detect vibration contour. This inspired investigation of a possible oppressiveness penalty (OP). For example, adding a penalty if the 1/3-octave band spectrum exceeds the Annoying-Displeasing boundary. While this needs more research, a simple approach was adopted here. With reference to the regions identified in the right most plot in Figure 6.5(a), the suggested penalty is given in Table 6.3.

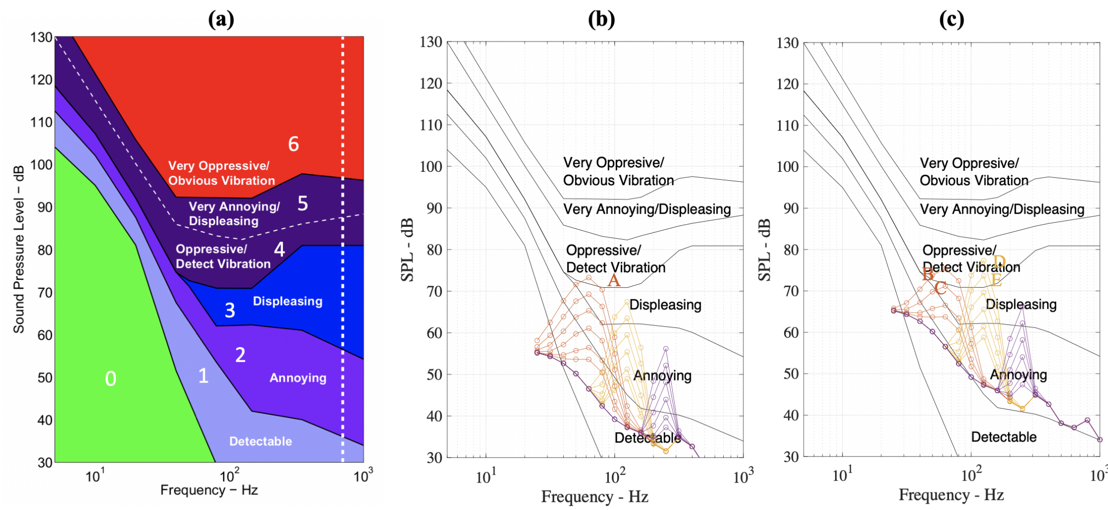

Figure 6.5. (a) Nakamura and Tokita's low-frequency noise contours. (b)-(c) One-third octave spectra of tonal sounds with NC-30, NC-40 broadband for identified outlier sounds plotted with Nakamura and Tokita's low-frequency noise contours. Sounds close to the displeasing and oppressive/detect vibration boundary were labeled from A-E (same as Figure 6.4(b)). The colors indicates 60 Hz tonal sounds (●), 120 Hz tonal sounds (●), 240 Hz tonal sounds (●).

Table 6.3. Suggested method for calculating an Oppressiveness Penalty (OP) based on highest number region (Figure 6.5) in which the one-third octave levels lie. With the sounds in our tests, this is the location of the peak in the spectrum.

Region of Peak Location	**Number of Sounds**	**Oppressiveness Penalty (OP)**
1&2	205	no penalty: OP = 0
3	9	OP is linearly interpolated from 0 (lower contour) to 1 (upper contour) of region 3 at peak location
4	8	OP is linearly interpolated from 1 (lower contour) to 2 (upper contour) of region 4 at peak location
5&6	0	Expect would be $>$ 2, needs further investigation.

Including OP in the model did increase the model performance a little as shown in Figure 6.6.

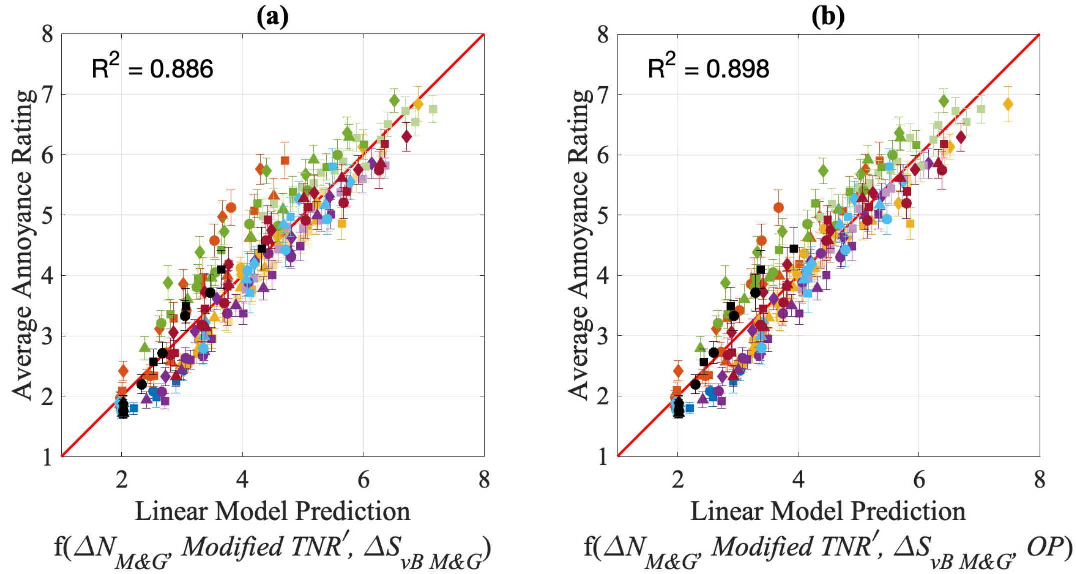

Figure 6.6. Global Model 2 (5-second rating) performance with and without the Oppressiveness Penalty (OP). Average annoyance plotted against the model predictions: (a) without the penalty $R^2 = 0.886$, and (b) with the penalty $R^2 = 0.898$

Limit Metric Contributions

Finally, metric contributions were limited. The lower limits are based on an assumption that below a certain value the sound characteristic measured by the metric does not play a role in annoyance. Upper limits are based on an assumption that the sound characteristic's role in annoyance saturates at some point. Having a lower limit on the Sharpness is consistent with the Unbiased Annoyance Model from Zwicker & Fastl [35], and limiting the tonality contribution is consistent with tonal penalties to level metrics (A-weighted sound pressure level or Loudness model predictions) used in several applications to assess annoyance, e.g., EPNL for aircraft certification [43], Danish environmental noise standard [14], Sound Quality Index for refrigeration equipment [17]. Thus, Sharpness's lower limit, tonality's higher and lower limits were optimized to achieve a higher model R^2 value. The variation in R^2 values with different lower and upper limits is shown in Figure 6.7. Relative Sharpness was calculated after limiting tonal sounds' and broadband sounds' Sharpness.

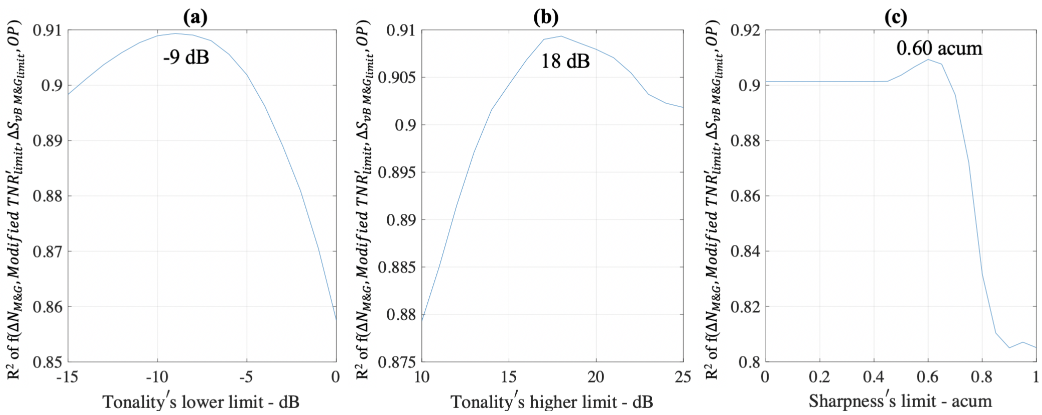

Figure 6.7. R^2 variation as a function of (a) the lower limit and (b) the upper limit for Modified TNR with frequency weighting (shortened to Tonality in the axis label), and (c) the lower limit for Sharpness. Sharpness was calculated from the Moore and Glasberg Loudness spectrum.

The final Global Model 2 for 5-second sounds is re-estimated with with limited metrics:

$$\begin{aligned} Annoyance^{5-sec}_{Global\ Model\ 2} =& 1.94 + 0.71\Delta N_{M\&G} + 0.137 Modified\ TNR'_{limit} \\ & + 7.49\Delta S_{vB\ M\&G_{limit}} + 0.65OP \end{aligned} \tag{6.11}$$

Figure 6.8 illustrates the Global Annoyance Model 2 performance with modified metrics. The model had an $R^2 = 0.909$, and there were no obvious outliers. 500 Hz sounds are still slightly under-predicted and need further investigation.

Global Model 1 was estimated with same modified metrics (i.e., including frequency weighting, hearing threshold, and limiting) and including the oppressiveness factor:

$$\begin{aligned} Annoyance^{5-sec}_{Global\ Model\ 1} =& 1.37 + 0.44N_{M\&G} + 0.059 Modified\ TNR'_{limit} \\ & + 3.65\Delta S_{vB\ M\&G_{limit}} + 0.24OP \end{aligned} \tag{6.12}$$

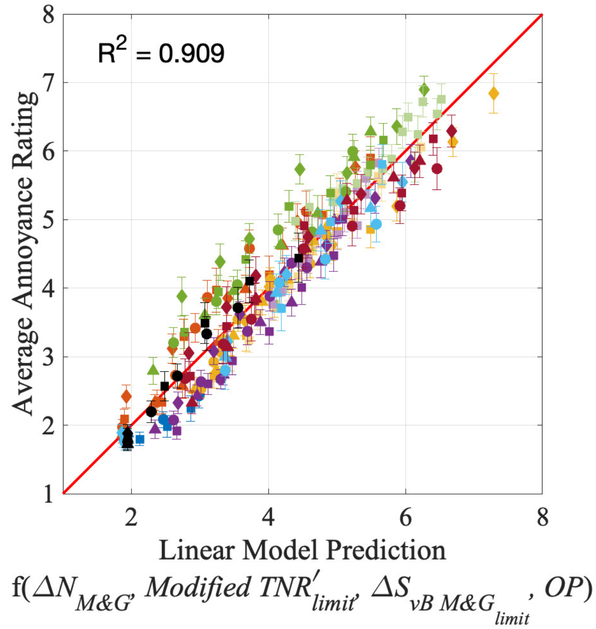

Figure 6.8. Global Model 2 (5-second rating) performance with the contributions of the tonalness and sharpness metrics limited.

The annoyance predictions for this model are shown in Figure 6.9. Recall that with Global Model 1 the average of annoyance ratings in each part of the test were modified with a test part specific offset and scaling to ensure that the broadband signals aligned with the general trend. This was based on the assumption that had all the broadband signals been played in one part of the test, the results would be highly correlated with loudness. This results in a compression of the ratings for each part of the test, and the assumption would be that subjects used the scales differently in each part of the test, i.e., expanded their use of the scale for each part. Because of this adjustment, which involved estimating 8 additional parameters, Global Model 2 was expected to have a higher R^2 value .

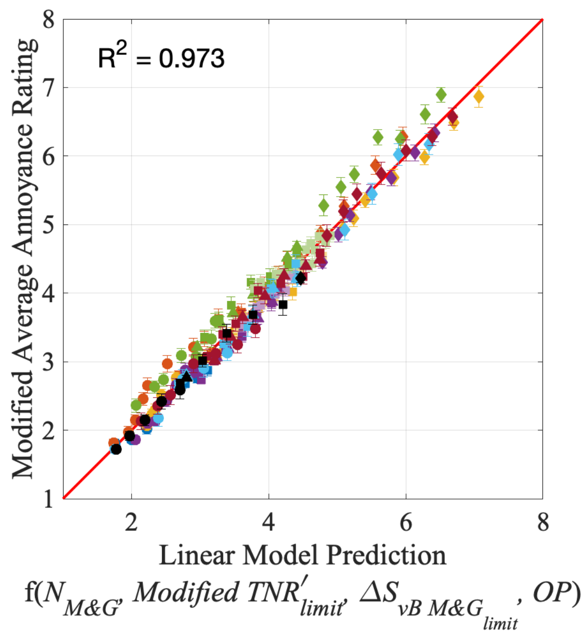

Figure 6.9. Global Model 1 (5-second rating) performance with modified metrics.

6.3 Global Annoyance Models (2-minute Ratings)

From Figure 5.11 and 5.16, acclimation effects exists in all sessions: NC-20 (Test 3), NC-30 (Test 2), and NC-40 (Test 3) sessions. The 5-second tonal sounds were rated consistently more annoying than the corresponding sounds in the 2-minute tests. The average difference in ratings between short and long duration signals is greater as the NC level increases (NC-20: 0.44±0.07, NC-30: 0.57± 0.03, NC-40: 0.71 ±0.11). Based on this relationship, annoyance ratings for 5-second sounds could be used to predict that for 2-minute sounds.

An offset was subtracted from 5-second ratings for each subject number i, sound number j to predict annoyance rating for 2-minute sounds. The offset is 0.44 for tonal sounds with NC-20 broadband, 0.57 for those with NC-30 or tilted NC-30 broadbands, 0.71 for those with NC-40 broadband. For subject i, the rating for tonal sound j is limited by his (or her) rating to the neutral broadband sound:

$$Predicted\ Annoyance_{i,j}^{2-min} = max\left(Annoyance_{i,j}^{5-sec} - offset_j, Annoyance_{i,broadband_j}^{5-sec}\right) \tag{6.13}$$

Same procedures were gone through to determine metrics limits. The lower limit for sharpness metric was still 0.60 acum, while the lower and higher limits for the tonality metric changed to -6 and 20 dB. Using the modified metrics, the annoyance prediction global models for 2-minute sounds were estimated. Equation 6.14 and 6.15.

$$\begin{aligned} Annoyance_{Global\ Model\ 1}^{2-min} =& 1.49 + 0.36N_{M\&G} + 0.054Modified\ TNR'_{limit} \\ &+ 3.48\Delta S_{vB\ M\&G_{limit}} + 0.27OP \end{aligned} \tag{6.14}$$

$$\begin{aligned} Annoyance_{Global\ Model\ 2}^{2-min} =& 2.04 + 0.59N_{M\&G} + 0.126Modified\ TNR'_{limit} \\ &+ 7.10\Delta S_{vB\ M\&G_{limit}} + 0.05OP \end{aligned} \tag{6.15}$$

The performance of two four-metric Global Models for 2-minute ratings was shown in Figure 6.10.

6.4 Annoyance Model Validation

As noted above, four global models (two for 5-second sounds, two for 2-minute sounds) were estimated with Test 3 ratings. Models were validated by checking how well these models predicted the average annoyance ratings for both 5-second and 2-minute sounds in Test 2 (which were not used in the estimation of the models' parameters). In Test 2, most test sounds had an NC-30 background component, with an ambient level in the room of NC-20. The results are shown in Figure 6.11. Sounds with 500 Hz tones are slightly underpredicted, this is consistent with some outliers in model development. For Global Model 2 results, with an NC-20 ambient broadband input, predictions almost lay on a 45-degree line. The predictions for Global Model 1 are compressed due to the adjustment of the annoyance ratings in the model development.

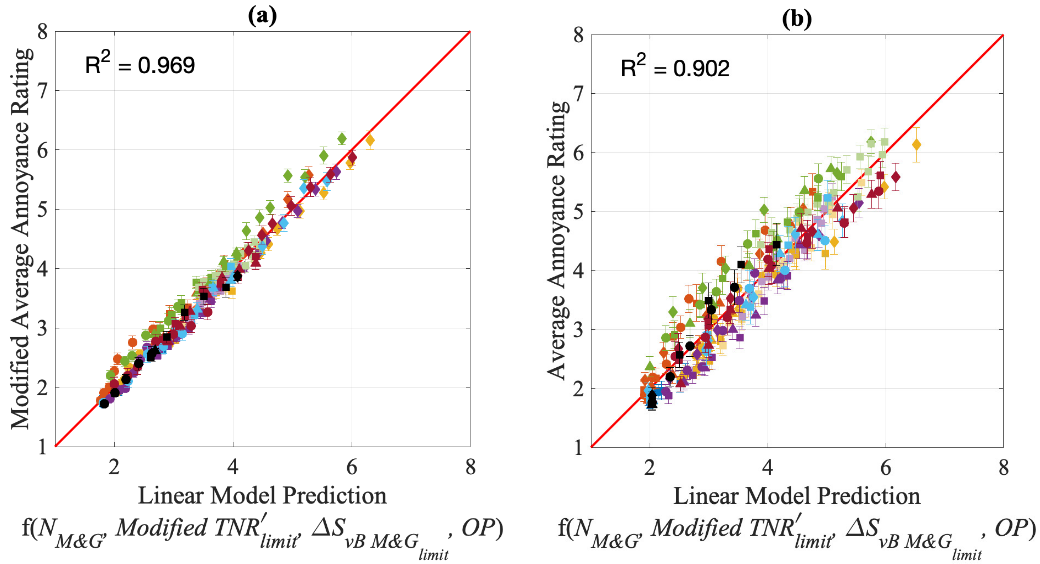

Figure 6.10. Global Model 1 and 2 (2-minute rating) performance with modified metrics.

This is consistent with the assumption that within a test subjects expand the range of their ratings to use more of the scale.

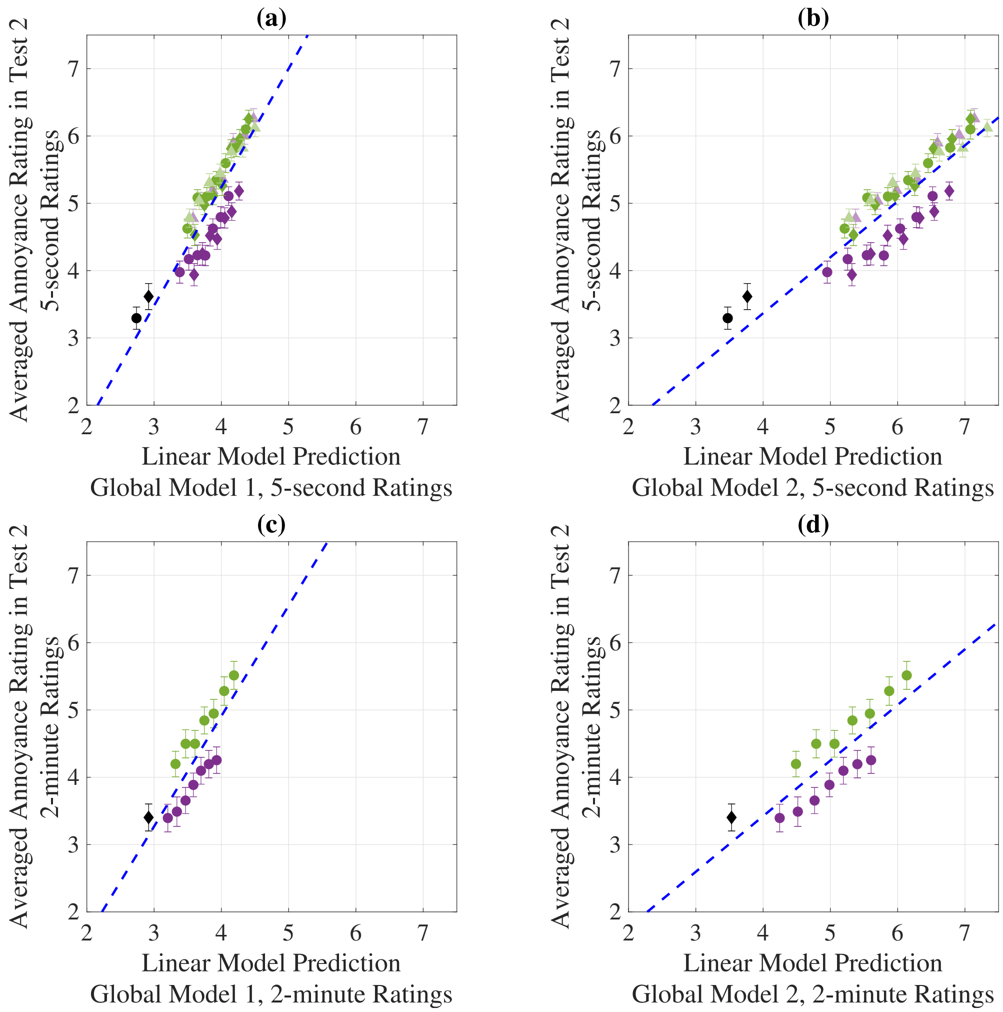

Figure 6.11. Performance of (a) Global Model 1 (5-second rating), (b) Global Model 2 (5-second rating), (c) Global Model 1 (2-minute rating) and (d) Global Model 2 (2-minute rating) in predicting results of Test 2 sounds. The global models were estimated from ratings in Test 3. The markers indicate different groups of tonal sounds: NC-30 broadband with single tone (○), NC-30 broadband with harmonics (△), tilted NC-30 broadband with single tone (◇). The colors indicates 240 Hz tonal sounds (•), 240 + 480 Hz tonal sounds (•), 500 Hz tonal sounds (•), 500 + 1000 Hz tonal sounds (•), neutral broadband (•).

6.5 Chapter Summary

Two global annoyance models have been developed for 5-second sounds and 2 more for 2-minute sounds. To improve the model performance, several modifications have been made to the metrics. Adding a hearing threshold to the TNR and limiting the metrics (Tonalness and Sharpness) contributions to annoyance are considered to be good to apply. Introducing frequency weighting improves the prediction for 29.5 Hz tonal sounds and maybe good to apply, but this needs further investigation because there is a consistent under prediction of annoyance for 500 Hz tonal sounds. The oppressive penalty looks useful in the model but still needs further investigation.

Global Model 1 predicts and NC-20 neutral broadband to be around "Not at all annoying". Increasing either sound level or prominence of tonal components would result in a higher predicted annoyance level. Global Model 2 is developed with an assumption that people would acclimatize to an ambient broadband background noise. Predicted annoyance from Global Model 2 is a function of difference s between the measured tonal sound and ambient broadband. Global Model 1 is more affected by the overall level, while Global Model 2 is more focused on the contributions due to the additional tonal components.

The difference between responses to 5-second and 2minute exposures have been examined. Responses to 2-minute exposure tended to be lower. Further research is required to see if further acclimation occurs.

7. CONCLUSIONS AND FUTURE WORK

The study focuses on determining what the human annoyance thresholds are of tones in noise and how those annoyance thresholds vary, depending on existence of harmonics and broadband background noise level. With an understanding of human annoyance thresholds to tones in noises, guidelines can be provided for building services equipment design, so occupants of offices will be less annoyed by building services noise. Results will also be helpful for people trying to improve other environments where tonal noises are a problem.

7.1 Research Summary and Contributions

Due to the conflict between signal exposure time and the amount of sounds included in the test, a preliminary test (Test 1) was designed to determine a reasonable exposure time. In Test 1, three different tone frequencies with two levels of tones were used as stimuli. The duration of the 22 test sounds ranged from 10 seconds to 4 minutes. It was observed that the average of subjects' ratings doesn't change much with signal duration. But there is some concern that asking subjects to focus on rating sounds with short exposure time may have caused an increase in annoyance for those sounds. Therefore, in Test 2, 29 tonal office sounds (with NC-30 broadband background noise) with similar sound attributes were tested in a 5-second duration test and a 2-minute duration test. People rated shorter tonal sounds, on average, as more annoying than longer sounds, and the differences between the two average ratings for all the tonal sounds were similar. Besides, Test 2 also examined how tonal levels and harmonics affect annoyance ratings. The 500 Hz tonal sounds are rated more annoying than the 240 Hz sounds without harmonics. For the sounds with Prominence Ratio $\leq$ 9 dB, tonality appears to be the main driver to annoyance. In Test 2, the ratio between the second harmonic and the fundamental tone is fixed, the annoyance to these harmonic sounds are either dominated by the fundamental tone or dominated by the second harmonics. This needs further investigation. With the results and concerns from Test 1 and Test 3, Test 3 was designed with a variety of tonal sounds to investigate how tonal levels, frequencies, presence of harmonics and broadband affect annoyance ratings. Test 3 was divided into four 5-second sound tests and two 2-minute sound tests. Subjects' annoyance acclimation for tonal sounds with NC-20 and NC-40 broadband is studied in this test. Based on the short duration test results, two types of global models are proposed. Global Model 1 is more affected by the overall level, while Global Model 2 is more focused on the contributions due to the additional tonal components. The refinements have been introduced to Tone-to-Noise Ratio to improve annoyance predictions. These modifications include: when harmonics are present, summing contributions to Tone-to-Noise Ratio rather than just picking the maximum; adding the hearing threshold to the Tone-to-Noise Ratio calculations; and limiting the metric's contributions to annoyance, so that it only plays a role when it exceeds a certain value, and its contribution to annoyance saturate at higher levels. Moore and Glasberg Loudness ($N_{M\&G}$), refined Tone-to-Noise Ratio ($Modified\ TNR'_{limit}$), limited sharpness all turn out to be significant in perceive annoyance. Oppressive Penalty (OP) was developed based on Nakamura and Tokita's low-frequency sounds contours. OP looks useful in the model but still needs further investigation. Based on the annoyance acclimation studied from Test 2 and Test 3, 5-second ratings in Test 3 were used to predict 2-minute ratings. Global annoyance models for 2-minute sounds are then re-estimated. Responses to 2-minute exposure tended to be lower. Further research is required to see if further acclimation occurs.

The annoyance models are developed to predict annoyance to tonal building noises with typical building background noises. Various factors such as tonal components with different frequencies and levels, the presence of harmonics and different background noises have been considered in the models.

The software has been developed with MATLAB to predict annoyance with developed models. The software takes estimated narrowband power spectrum densities or *.wav files with calibration A-weighted sound pressure level as input to predict annoyance. Functions were programmed to extract the broadband component of a sound, to calculate Oppressive Penalty, Modified TNR' (with the frequency weighting and hearing threshold), Moore and Glasberg Loudness, and Sharpness (based on the calculated Loudness spectrum). Annoyance was predicted using global models for 5-second sounds and 2-minute sounds with the calculated metrics. Details and guidelines for the software were described in Appendix B.

7.2 Recommendations for Future Work

The study covered many issues associated with human annoyance perception to tonal building noises, but there are some remaining issues that need to be investigated. Possible future works are:

1. Frequency weighting (or tone penalty):

   It was observed that by implementing a frequency weighting that is used in the well-established Aures Model, the outliers with 29.5 Hz tonal components are corrected. But this frequency weighting is not perfect, there is still a consistent under-prediction for sounds with 500 Hz tonal components. Subjective tests for sounds with more tonal frequencies are recommended. A better frequency weighting for the tonality metric or a tone penalty for the level metric can be developed to predict perceived annoyance more accurately.

2. Oppressive Penalty (OP):

   As annoyance for the sounds may increase if the level of sounds is close to or exceeds the displeasing – oppressive/detect vibration contour. A simple Oppressive Penalty is developed based on linear interpolation. OP turns out to be helpful in the model, but it needs more investigation. Some of the low-frequency outliers could be corrected with Oppressive Penalty, but the perception to low-frequency components is not linear. Subjective tests with more low-frequency sounds are needed to further investigate to help develop a proper Oppressive Penalty.

3. Acclimation beyond 2 minutes:

   Subjects' acclimation has been studied for tonal building sounds with 5-second duration and 2-minute duration, but people are usually exposed to tonal building sounds day and night. Further research is required to study further acclimation.

4. The application of models:

   The current research is conducted in an office-like environment. A scenario was given to the subjects to ask them to imagine themselves working in the office. Testing environment would have effect on the degree of annoyance, the predicted annoyance are only valid office measurements. Subject's responses in different scenarios needs to be examined to expand the application of models.

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REFERENCES

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APPENDICES

A. SUBJECTIVE TEST SIGNALS, METRICS AND SUBJECT'S RESPONSES

A.1 Sound Meaaurements

This section describes 27 recordings in different buildings on the campus of Purdue University and 36 more from the ASHRAE advisory team of the project. Some example measurements are used in Chapter 2.1 to identify tones listed in Request-for-proposal and Chapter 3 for simulate test sounds.

As for the on campus recording, the measurements were taken using a 1/2" PCB PIEZOTRONICS microphone, calibrated using a Sound Calibrator (Type 4231). The microphone was placed around 1 m from the ground and signals were acquired using a HEAD ACOUSTICS 4 channel SQUADRIGA (Code 1369) acquisition system. The analog to digital counter is 16 bits and sampling rate is fixed at 48,000 samples/second. The analog filters in the acquisition systems are 1^{st} order high-pass filter with a cut-off frequency 2 Hz. The noise was recorded for 20 – 150 seconds for each measurement. The details of these measurements are shown in Table A.1.

Table A.1. Details of on measurements on Purdue University's West Lafayette Campus.

Signal Number	**Signal Length**	**Building, Room Number**	**Interior/ Exterior**	**HVAC Source Nearby**
001	20s	HLAB Building	Interior	Ventilation noise
002	20s	HLAB Building	Interior	Pump noise
003	157s	HERL Building, South Acoustic Wing	Interior	Air-conditioning, ventilation noise
004	156s	HERL Building, Student Lounge (Open Space)	Interior	Air-conditioning, ventilation noise
005	156s	HERL Building,	Interior	Air-conditioning,

Table A.1. Continued from previous page.

Signal Number	**Signal Length**	**Building, Room Number**	**Interior/ Exterior**	**HVAC Source Noise**
		North Acoustic Wing		ventilation noise
006	159s	HERL Building, Room 247	Interior	Air-conditioning, ventilation noise
007	163s	HERL Building, Room 244	Interior	Air-conditioning, ventilation noise
008	168s	HERL Building, Room 254	Interior	Air-conditioning, ventilation noise
009	101s	HLAB Building, 2^{nd} floor, temporary office	Interior	Air-conditioning noise
010	120s	HERL Building, Acoustic Wing	Interior	Air-conditioning, ventilation noise
011	156s	HLAB Building, Conference Room	Interior	Air-conditioning noise
012	157s	HLAB Building, Student Open Area (North)	Interior	Air-conditioning noise
013	159s	HLAB Building, Student Open Area (South)	Interior	Air-conditioning noise
014	158s	HLAB Building, Stairway	Interior	Ventilation noise
015	156s	HLAB Building, High-Bay Area (Laboratory)	Interior	Air-conditioning, ventilation noise
016	188s	HERL Building, Student Lounge (Open Space)	Interior	Air-conditioning, ventilation noise
017	178s	HERL Building, Ford Motor Company Booth	Interior	/
018	160s	HERL Building	Exterior	DX Chiller (location 1)

Table A.1. Continued from previous page.

Signal Number	**Signal Length**	**Building, Room Number**	**Interior/ Exterior**	**HVAC Source Noise**
019	161s	HERL Building	Exterior	DX Chiller (location 2)

Table A.1. Continued from previous page.

Signal Number	**Signal Length**	**Building, Room Number**	**Interior/ Exterior**	**HVAC Source Noise**
020	152s	HLAB Building, PBE Lab	Interior	/
021	160s	HLAB Building, PBE Lab Control Room	Interior	Air-conditioning noise
022	151s	HERL Building, Outside PBE Lab Control Room	Interior	Air-conditioning noise
023	194s	HERL Building	Interior	Chiller noise
024	190s	PSF Building	Interior	Chiller 1 noise
025	181s	PSF Building	Interior	Chiller 1 noise, Location 2
026	115s	PSF Building	Interior	Chiller 2 noise
027	189s	HERL Building, Room 254	Interior	Air-conditioning, ventilation noise

Table A.2. Details of measurements provided by ASHRAE advisory team.

Signal Number	Signal Length	Sampling Rate	Filename
001	35s	24 kHz	Pump Tone.wav
002	70s	48 kHz	Heat Pump Tones.wav
003	49s	24 kHz	Digital Compressor Tones.wav
004	19s	24 kHz	Compressor Tones.wav
005	22s	24 kHz	Bathroom Exhaust Fan Tone.wav
006	19s	24 kHz	478 Hz Tone.wav
007	10s	44.1 kHz	Chiller-centrifugal-1000tons.wav

Table A.2. Continued from previous page.

Signal Number	Signal Length	Sampling Rate	Filename
008	25s	22 kHz	Chiller-recip.wav
009	10s	44.1 kHz	Chiller-screw-66load.mp3
010	10s	44.1 kHz	Chiller-screw-100load.mp3
011	10s	44.1 kHz	Condenser-air-cooled.mp3
012	10s	22 kHz	Coolingtower-counterflow-centrifugalfan.mp3
013	10s	22 kHz	Coolingtower-crossflow-axialfan.mp3
014	10s	44.1 kHz	Diffuser.mp3
015	10s	44.1 kHz	Fan-centrifugal-airfoil.mp3
016	10s	44.1 kHz	Fan-centrifugal-fc.mp3
017	10s	44.1 kHz	Fan-mixedflow.mp3
018	10s	44.1 kHz	Fan-plenum-supply.mp3
019	10s	44.1 kHz	Fan-radialblade-31p5hz.mp3
020	10s	44.1 kHz	Fan-tubeaxial.mp3
021	10s	44.1 kHz	Rooftop-ahu-noceiling.mp3
022	60s	44.1 kHz	Example.1-CE-30Hz.wav
023	60s	44.1 kHz	Example.1-CE-38Hz.wav
024	60s	44.1 kHz	Example.1-CE-47.6Hz.wav
025	22s	24 kHz	S1.wav
026	19s	24 kHz	S2.wav
027	19s	24 kHz	S3.wav
028	49s	24 kHz	S4.wav
029	70s	48 kHz	S5.wav
030	35s	24 kHz	S6.wav
031	52s	24 kHz	SR0.wav

Table A.2. Continued from previous page.

Signal Number	Signal Length	Sampling Rate	Filename
032	19s	16 kHz	Moonshot AC-14 on 11-10-2019.wav
033	313s	44.1 kHz	SEQ A IN CHAMBER.wav
034	310s	44.1 kHz	SEQ B IN CHAMBER.wav
035	310s	44.1 kHz	SEQ C IN CHAMBER.wav
036	311s	44.1 kHz	SEQ D IN CHAMBER.wav

A.2 Test 1

In the Test 1, test sounds were a combination of an NC-30 broadband component and different tonal components. Two different prominence ratios (3.0, 11.0 dB) were selected for a low frequency (60 Hz), a middle frequency (240 Hz), a high frequency (1000 Hz) tones. The lengths of the 22 test sounds ranged from 10 seconds to 4 minutes. Descriptions of Test 1 sounds and corresponding average annoyance rating with standard error are shown in Table A.3.

Table A.3. Test 1 signals and corresponding average annoyance ratings with standard error.

Signal Number	Frequency Hz	PR dB	Duration sec	Annoyance	Standard Error
001	1000	11	10	4.88	0.30
002	1000	11	120	4.93	0.27
003	1000	11	60	4.69	0.31
004	1000	3	10	3.69	0.26
005	1000	3	120	3.60	0.28
006	1000	3	60	3.34	0.29
007	240	11	10	4.80	0.28
008	240	11	120	4.74	0.24
009	240	11	240	4.24	0.21
010	240	11	30	4.75	0.31
011	240	11	60	4.78	0.30
012	240	3	10	3.81	0.27
013	240	3	120	3.58	0.25
014	240	3	240	2.98	0.23
015	240	3	30	3.56	0.28
016	240	3	60	3.68	0.24
017	60	11	10	4.83	0.28

Table A.3. Continued from previous page.

Signal Number	Frequency Hz	PR dB	Duration sec	Annoyance	Standard Error
018	60	11	120	4.17	0.27
019	60	11	60	4.75	0.30
020	60	3	10	2.24	0.22
021	60	3	120	2.42	0.20
022	60	3	60	2.55	0.22

A.3 Test 2

This section contains the test signals detail and results of Test 2, which consists of a short exposure time test (72 5-second sounds) and a long exposure time test (29 2-minute sounds). The results of Test 2 are discussed in Chapter 5.3. In Table A.4 and A.5, important sound quality metrics for 5-second and 2-minute sounds used in Test 2 are presented. Most metric values calculated by using Head Acoustics ArtemiS software, except for PR, TNR, ΔL_{ta}, and Aures Tonality. In Table A.6 are the average annoyance ratings and corresponding standard errors of 5-second sounds in Test 2. In Table A.7 are the average and standard error of annoyance ratings for 2-minute sounds and corresponding 5-second sounds in Test 2.

A.3.1 Test 2 Signals and Sound Metrics

Table A.4. Values of the main sound quality metrics for Test 2 5-second signals. For PR, TNR, ΔL_{ta}, the first columns corresponds to the tonality of fundamental tone, the second columns corresponds to the tonality of second harmonic.

Signal	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
001	37.9	54.8	2.1	2.8	0.01	0.56	0.75	10.5	/	13.1	/	14.4	/	0.36
002	43.1	57.9	3.3	4.2	0.01	0.60	0.76	11.1	11.0	13.7	12.2	14.8	14.6	0.47
003	41.3	56.1	2.8	3.6	0.01	0.58	0.74	11.1	11.0	13.7	12.2	14.8	14.6	0.49
004	40.9	57.7	2.7	3.6	0.01	0.60	0.79	10.8	/	13.3	/	14.4	/	0.34
005	40.6	54.1	2.9	4.0	0.01	0.74	0.92	11.0	/	13.4	/	14.3	/	0.28
006	38.6	54.3	2.1	2.6	0.01	0.53	0.72	12.6	/	15.2	/	16.6	/	0.41
007	44.9	58.2	3.5	4.5	0.01	0.58	0.74	13.1	13.0	15.7	14.2	16.9	16.6	0.52
008	42.7	56.1	3.0	3.8	0.01	0.56	0.72	13.1	13.0	15.7	14.2	16.9	16.6	0.55
009	42.3	58.0	2.9	3.8	0.01	0.59	0.77	12.8	/	15.4	/	16.5	/	0.38
010	41.9	54.6	3.1	4.2	0.01	0.71	0.90	13.0	/	15.5	/	16.4	/	0.32
011	39.3	53.9	2.1	2.5	0.01	0.50	0.69	14.5	/	17.2	/	18.6	/	0.45
012	46.6	58.7	3.9	4.8	0.01	0.57	0.71	15.1	15.0	17.7	16.2	19.1	18.5	0.57
013	44.3	56.4	3.2	4.0	0.01	0.55	0.69	15.1	15.0	17.7	16.2	19.1	18.5	0.60
014	43.9	58.4	3.1	3.9	0.01	0.57	0.75	14.7	/	17.4	/	18.5	/	0.42

Table A.4. Continued from previous page.

Signal Number	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
015	43.4	55.3	3.3	4.3	0.01	0.69	0.87	14.9	/	17.5	/	18.4	/	0.37
016	36.5	56.4	2.1	3.0	0.01	0.66	0.85	2.5	/	3.8	/	5.2	/	0.16
017	38.0	57.3	2.5	3.4	0.02	0.66	0.84	3.1	3.3	4.4	3.2	5.8	5.6	0.22
018	37.5	56.8	2.3	3.3	0.01	0.65	0.83	3.1	3.3	4.4	3.2	5.8	5.6	0.22
019	37.2	57.3	2.3	3.3	0.01	0.67	0.85	2.8	/	4.0	/	5.3	/	0.17
020	37.5	53.2	2.5	3.7	0.02	0.81	0.98	3.1	/	4.5	/	5.5	/	0.13
021	36.6	56.1	2.1	3.0	0.01	0.63	0.83	4.5	/	6.4	/	7.8	/	0.20
022	39.0	57.4	2.6	3.6	0.01	0.64	0.82	5.1	5.2	7.1	5.7	8.3	8.1	0.29
023	38.2	56.6	2.4	3.3	0.01	0.63	0.81	5.1	5.2	7.1	5.7	8.3	8.1	0.30
024	37.8	57.3	2.4	3.3	0.01	0.65	0.84	4.8	/	6.6	/	7.8	/	0.22
025	38.0	53.3	2.6	3.8	0.02	0.80	0.97	5.1	/	7.0	/	7.9	/	0.17
026	36.9	55.6	2.1	2.9	0.01	0.61	0.80	6.5	/	8.7	/	10.1	/	0.26
027	40.2	57.5	2.8	3.8	0.01	0.63	0.80	7.1	7.1	9.3	7.9	10.5	10.3	0.35
028	39.0	56.3	2.5	3.4	0.01	0.61	0.79	7.1	7.1	9.3	7.9	10.5	10.3	0.36
029	38.6	57.4	2.5	3.4	0.01	0.64	0.83	6.7	/	9.0	/	10.1	/	0.26
030	38.6	53.5	2.7	3.8	0.01	0.78	0.95	7.0	/	9.2	/	10.1	/	0.21
031	37.4	55.2	2.1	2.8	0.01	0.58	0.78	8.5	/	10.9	/	12.3	/	0.31

Table A.4. Continued from previous page.

Signal	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
032	41.6	57.6	3.0	4.0	0.01	0.61	0.78	9.1	9.1	11.6	10.1	12.7	12.5	0.41
033	40.1	56.2	2.6	3.5	0.01	0.60	0.77	9.1	9.1	11.6	10.1	12.7	12.5	0.43
034	39.6	57.5	2.6	3.5	0.01	0.62	0.81	8.8	/	11.2	/	12.3	/	0.30
035	39.5	53.7	2.8	3.9	0.01	0.76	0.93	9.0	/	11.4	/	12.3	/	0.24
036	38.4	55.2	2.1	3.0	0.01	0.64	0.81	11.6	/	14.1	/	17.9	/	0.34
037	40.6	57.3	2.7	3.7	0.01	0.68	0.84	11.0	2.9	13.3	5.2	17.5	5.6	0.35
038	40.2	57.0	2.6	3.6	0.01	0.68	0.84	11.0	2.9	13.3	5.2	17.5	5.6	0.36
039	40.6	57.4	2.6	3.6	0.01	0.68	0.84	11.3	/	13.4	/	17.8	/	0.35
040	40.2	53.3	2.8	4.0	0.01	0.81	0.96	9.9	/	12.4	/	15.2	/	0.26
041	39.1	54.7	2.1	2.9	0.01	0.62	0.79	13.6	/	16.0	/	19.9	/	0.38
042	42.0	57.4	2.8	3.9	0.01	0.68	0.83	13.0	4.5	15.4	7.2	19.5	7.7	0.40
043	41.5	56.9	2.7	3.7	0.01	0.68	0.83	13.0	4.5	15.4	7.2	19.5	7.7	0.41
044	42.0	57.4	2.7	3.8	0.01	0.67	0.82	13.3	/	15.7	/	19.9	/	0.39
045	41.5	53.5	2.9	4.1	0.01	0.80	0.95	11.9	/	14.4	/	17.3	/	0.30
046	4/	54.2	2.1	2.8	0.01	0.61	0.77	15.6	/	18.0	/	22.0	/	0.44
047	43.5	57.5	3.0	4.1	0.01	0.68	0.82	14.9	6.3	17.4	9.2	21.5	9.7	0.45
048	42.8	56.8	2.8	3.9	0.01	0.68	0.82	14.9	6.3	17.4	9.2	21.5	9.7	0.46

Table A.4. Continued from previous page.

Signal Number	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
049	43.6	57.5	2.9	3.9	0.01	0.67	0.81	15.3	/	17.7	/	21.9	/	0.42
050	42.9	53.7	3.0	4.2	0.01	0.79	0.93	13.9	/	16.4	/	19.3	/	0.33
051	36.6	56.6	2.2	3.1	0.01	0.69	0.87	3.4	/	4.9	/	8.8	/	0.18
052	37.1	57.2	2.3	3.3	0.02	0.69	0.87	3.2	-1.9	4.1	-3.1	8.5	/	0.21
053	37.1	57.2	2.3	3.2	0.02	0.69	0.87	3.2	-1.9	4.1	-3.1	8.5	/	0.21
054	37.1	57.2	2.3	3.2	0.02	0.69	0.87	3.3	/	4.5	/	8.7	/	0.21
055	37.4	53.1	2.5	3.6	0.02	0.84	1.00	2.2	/	3.4	/	6.1	/	0.13
056	36.8	56.3	2.2	3.1	0.01	0.68	0.86	5.5	/	7.4	/	11.3	/	0.23
057	37.6	57.2	2.4	3.3	0.02	0.69	0.86	5.0	-1.1	6.5	-1.4	10.9	-1.0	0.24
058	37.6	57.1	2.3	3.3	0.02	0.69	0.86	5.0	-1.1	6.5	-1.4	10.9	-1.0	0.24
059	37.7	57.2	2.3	3.3	0.02	0.69	0.86	5.3	/	7.0	/	11.2	/	0.25
060	37.8	53.1	2.5	3.7	0.02	0.84	0.99	4.0	/	5.8	/	8.6	/	0.16
061	37.2	56.0	2.2	3.0	0.01	0.67	0.84	7.5	/	9.8	/	13.6	/	0.25
062	38.4	57.3	2.4	3.4	0.01	0.69	0.86	7.0	/	8.8	0.8	13.2	1.3	0.26
063	38.2	57.1	2.4	3.4	0.01	0.69	0.86	7.0	/	8.8	0.8	13.2	1.3	0.26
064	38.4	57.3	2.4	3.4	0.01	0.68	0.85	7.3	/	9.2	/	13.5	/	0.28
065	38.4	53.2	2.6	3.8	0.02	0.83	0.98	6.0	/	8.1	/	10.9	/	0.19

Table A.4. Continued from previous page.

Signal	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
066	37.7	55.6	2.1	3.0	0.01	0.65	0.83	9.5	/	12.0	/	15.7	/	0.29
067	39.4	57.3	2.5	3.6	0.01	0.68	0.85	9.0	1.3	11.0	2.9	15.4	3.5	0.30
068	39.1	57.1	2.5	3.5	0.01	0.68	0.85	9.0	1.3	11.0	2.9	15.4	3.5	0.31
069	39.4	57.3	2.5	3.5	0.01	0.68	0.85	9.3	/	11.4	/	15.7	/	0.32
070	39.2	53.2	2.7	3.9	0.01	0.82	0.97	7.9	/	10.3	/	13.1	/	0.23
071	36.8	53.1	2.4	3.5	0.02	0.70	1.01	/	/	/	/	/	/	0.00
072	36.3	57.2	2.2	3.1	0.01	0.85	0.89	/	/	/	/	/	/	0.01

Table A.5. Values of the main sound quality metrics for Test 2 2-minute signals.

Signal Number	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
001	37.9	54.8	2.1	2.8	0.01	0.56	0.75	10.5	/	13.1	/	14.4	/	0.36
002	40.9	57.7	2.7	3.6	0.01	0.60	0.79	10.8	/	13.3	/	14.4	/	0.34
003	38.6	54.3	2.1	2.6	0.01	0.53	0.72	12.6	/	15.2	/	16.6	/	0.41
004	42.3	58.0	2.9	3.8	0.01	0.59	0.77	12.8	/	15.4	/	16.5	/	0.38
005	39.3	53.9	2.1	2.5	0.01	0.50	0.69	14.5	/	17.2	/	18.6	/	0.45
006	43.9	58.4	3.1	3.9	0.01	0.57	0.75	14.7	/	17.4	/	18.5	/	0.42
007	36.5	56.4	2.1	3.0	0.01	0.66	0.85	2.5	/	3.8	/	5.2	/	0.16
008	37.2	57.3	2.3	3.3	0.01	0.67	0.85	2.8	/	4.0	/	5.3	/	0.17
009	36.6	56.1	2.1	3.0	0.01	0.63	0.83	4.5	/	6.4	/	7.8	/	0.20
010	37.8	57.3	2.4	3.3	0.01	0.65	0.84	4.8	/	6.6	/	7.8	/	0.22
011	36.9	55.6	2.1	2.9	0.01	0.61	0.80	6.5	/	8.7	/	10.1	/	0.26
012	38.6	57.4	2.5	3.4	0.01	0.64	0.83	6.7	/	9.0	/	10.1	/	0.26
013	37.4	55.2	2.1	2.8	0.01	0.58	0.78	8.5	/	10.9	/	12.3	/	0.31
014	39.6	57.5	2.6	3.5	0.01	0.62	0.81	8.8	/	11.2	/	12.3	/	0.30
015	38.4	55.2	2.1	3.0	0.01	0.64	0.81	11.6	/	14.1	/	17.9	/	0.34
016	40.6	57.4	2.6	3.6	0.01	0.68	0.84	11.3	/	13.4	/	17.8	/	0.35
017	39.1	54.7	2.1	2.9	0.01	0.62	0.79	13.6	/	16.0	/	19.9	/	0.38

Table A.5. Continued from previous page.

Signal Number	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
018	42.0	57.4	2.7	3.8	0.01	0.67	0.82	13.3	/	15.7	/	19.9	/	0.39
019	4/	54.2	2.1	2.8	0.01	0.61	0.77	15.6	/	18.0	/	22.0	/	0.44
020	43.6	57.5	2.9	3.9	0.01	0.67	0.81	15.3	/	17.7	/	21.9	/	0.42
021	36.6	56.6	2.2	3.1	0.01	0.69	0.87	3.4	/	4.9	/	8.8	/	0.18
022	37.1	57.2	2.3	3.2	0.02	0.69	0.87	3.3	/	4.5	/	8.7	/	0.21
023	36.8	56.3	2.2	3.1	0.01	0.68	0.86	5.5	/	7.4	/	11.3	/	0.23
024	37.7	57.2	2.3	3.3	0.02	0.69	0.86	5.3	/	7.0	/	11.2	/	0.25
025	37.2	56.0	2.2	3.0	0.01	0.67	0.84	7.5	/	9.8	/	13.6	/	0.25
026	38.4	57.3	2.4	3.4	0.01	0.68	0.85	7.3	/	9.2	/	13.5	/	0.28
027	37.7	55.6	2.1	3.0	0.01	0.65	0.83	9.5	/	12.0	/	15.7	/	0.29
028	39.4	57.3	2.5	3.5	0.01	0.68	0.85	9.3	/	11.4	/	15.7	/	0.32
029	36.3	57.2	2.2	3.1	0.01	0.85	0.89	/	/	/	/	/	/	0.01

A.3.2 Test 2 Subject's Responses and Statistics

Table A.6. Average annoyance ratings and corresponding standard errors for 5-second sounds in Test 2.

Signal Number	Annoyance	Standard Error	Signal Number	Annoyance	Standard Error
001	4.25	0.14	037	5.76	0.14
002	5.90	0.14	038	5.69	0.13
003	5.46	0.13	039	5.60	0.14
004	4.62	0.14	040	5.81	0.13
005	4.78	0.15	041	5.45	0.12
006	4.12	0.13	042	5.82	0.13
007	6.01	0.14	043	5.83	0.14
008	5.91	0.14	044	5.83	0.13
009	4.79	0.16	045	5.95	0.14
010	4.88	0.13	046	5.80	0.14
011	4.48	0.13	047	6.12	0.13
012	6.26	0.15	048	5.99	0.13
013	6.13	0.15	049	6.10	0.14
014	5.11	0.14	050	6.25	0.13
015	5.18	0.13	051	4.59	0.16
016	3.68	0.14	052	4.78	0.14
017	4.77	0.14	053	4.68	0.15
018	4.72	0.13	054	4.62	0.14
019	3.98	0.16	055	4.53	0.16
020	3.94	0.17	056	4.84	0.15
021	3.90	0.16	057	5.04	0.13
022	5.03	0.12	058	5.06	0.14
023	4.92	0.14	059	5.08	0.12
024	4.17	0.16	060	4.98	0.15

Table A.6. Continued from previous page.

Signal Number	Annoyance	Standard Error	Signal Number	Annoyance	Standard Error
025	4.25	0.17	061	5.17	0.14
026	4.14	0.14	062	5.30	0.14
027	5.19	0.14	063	5.14	0.14
028	5.05	0.13	064	5.10	0.13
029	4.23	0.15	065	5.11	0.13
030	4.52	0.15	066	5.42	0.15
031	4.12	0.14	067	5.44	0.14
032	5.36	0.12	068	5.57	0.15
033	5.47	0.13	069	5.34	0.13
034	4.22	0.15	070	5.26	0.15
035	4.47	0.16	071	3.61	0.19
036	5.24	0.13	072	3.29	0.17

Table A.7. Average and standard error of annoyance ratings for 2-minute sounds and corresponding 5-second sounds in Test 2.

2-minute sounds			5-second sounds		
Signal Number	Annoyance	Standard Error	Signal Number	Annoyance	Standard Error
001	3.52	0.17	001	4.25	0.14
002	4.10	0.20	004	4.62	0.14
003	3.69	0.20	006	4.12	0.13
004	4.19	0.20	009	4.79	0.16
005	3.87	0.22	011	4.48	0.13
006	4.25	0.20	014	5.11	0.14
007	3.52	0.22	016	3.68	0.14
008	3.39	0.21	019	3.98	0.16
009	3.63	0.21	021	3.90	0.16
010	3.49	0.22	024	4.17	0.16
011	3.47	0.21	026	4.14	0.14
012	3.65	0.19	029	4.23	0.15
013	3.61	0.20	031	4.12	0.14
014	3.89	0.18	034	4.22	0.15
015	4.77	0.22	036	5.24	0.13
016	4.95	0.21	039	5.60	0.14
017	4.97	0.19	041	5.45	0.12
018	5.28	0.21	044	5.83	0.13
019	5.19	0.23	046	5.80	0.14
020	5.51	0.21	049	6.10	0.14
021	4.15	0.21	051	4.59	0.16
022	4.20	0.19	054	4.62	0.14
023	4.20	0.18	056	4.84	0.15
024	4.50	0.21	059	5.08	0.12

Table A.7. Continued from previous page.

2-minute sounds			5-second sounds		
Signal Number	Annoyance	Standard Error	Signal Number	Annoyance	Standard Error
025	4.36	0.18	061	5.17	0.14
026	4.50	0.20	064	5.10	0.13
027	4.46	0.18	066	5.42	0.15
028	4.84	0.20	069	5.34	0.13
029	3.40	0.20	072	3.29	0.17

A.4 Test 3

This section contains the test signals detail and results of Test 3, which consists of four short exposure time tests (Part A, Part B, Part D and Part E) and two long exposure time tests (Part C and Part F). The results of Test 3 are discussed in Chapter 5.4. Global models are further developed based on the annoyance ratings in Chapter 6. In Table A.8 and A.9 are important sound quality metrics for sounds used in Test 3 are presented. In Table A.10 are the average annoyance ratings and corresponding standard errors of 5-second sounds in Test 3. In Table A.11 are the average and standard error of annoyance ratings for 2-minute sounds and corresponding 5-second sounds in Test 3.

A.4.1 Test 3 Signals and Sound Metrics

Table A.8. Values of the main sound quality metrics for Test 3 5-second signals.

Signal	Test	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	Part	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
001	A	36.4	56.8	2.2	3.1	0.01	0.70	0.89	/	/	/	/	/	/	0.01
002	A	38.4	58.8	2.7	3.7	0.02	0.72	0.90	/	/	/	/	/	/	0.01
003	A	40.4	60.8	3.2	4.4	0.02	0.74	0.91	/	/	/	/	/	/	0.01
004	A	42.4	62.8	3.8	5.2	0.02	0.76	0.92	/	/	/	/	/	/	0.01
005	A	44.4	64.8	4.5	6.1	0.02	0.78	0.93	/	/	/	/	/	/	0.01
006	A	36.5	56.6	2.2	3.1	0.01	0.70	0.89	1.2	/	2.1	/	2.8	/	0.11
007	A	38.2	57.5	2.6	3.3	0.01	0.60	0.81	1.1	/	7.2	/	6.3	/	0.21
008	A	36.8	56.7	2.2	3.1	0.01	0.68	0.86	0.9	/	2.2	/	3.5	/	0.11
009	A	36.7	56.6	2.2	3.1	0.01	0.69	0.87	1.1	/	1.6	/	4.8	/	0.12
010	A	37.1	59.3	2.4	3.2	0.01	0.62	0.84	1.0	/	0.8	/	-5.1	/	0.01
011	A	36.7	56.6	2.2	3.1	0.01	0.71	0.89	3.2	/	4.9	/	5.6	/	0.14
012	A	39.2	58.1	2.8	3.4	0.01	0.58	0.79	3.2	/	9.5	/	8.6	/	0.26
013	A	39.3	58.1	2.8	3.5	0.01	0.58	0.79	3.1	-1.1	9.6	-1.6	8.8	0.6	0.25
014	A	39.9	58.2	2.8	3.6	0.01	0.57	0.77	3.4	3.1	9.7	4.9	8.9	6.3	0.26
015	A	40.7	58.4	2.9	3.7	0.01	0.56	0.75	3.1	7.2	10.1	9.6	9.1	10.8	0.31

Table A.8. Continued from previous page.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	PR dB	TNR dB	TNR dB	ΔL_{ta} dB	ΔL_{ta} dB	Aures Tonality
016	A	42.7	58.9	3.2	4.0	0.01	0.53	0.71	3.1	11.1	10.9	13.9	10.1	15.2	0.39
017	A	37.3	56.7	2.3	3.2	0.01	0.66	0.85	2.8	/	5.0	/	6.3	/	0.16
018	A	37.1	56.6	2.2	3.2	0.02	0.69	0.87	3.1	/	4.8	/	7.7	/	0.17
019	A	37.9	61.1	2.6	3.3	0.01	0.58	0.82	2.4	/	3.9	/	-2.0	/	0.12
020	A	36.2	56.6	2.1	3.0	0.01	0.70	0.88	/	/	-10.4	/	/	/	0.01
021	A	37.4	56.6	2.3	3.2	0.01	0.72	0.89	6.9	/	9.7	/	10.3	/	0.20
022	A	41.9	59.8	3.2	3.7	0.01	0.53	0.75	7.2	/	13.8	/	12.8	/	0.35
023	A	42.3	59.9	3.2	3.8	0.01	0.52	0.73	7.1	2.7	14.0	5.0	13.3	6.3	0.35
024	A	43.0	60.2	3.3	4.0	0.01	0.52	0.70	7.3	7.1	14.3	9.7	13.6	11.0	0.36
025	A	44.3	60.7	3.5	4.2	0.01	0.50	0.68	7.2	11.1	15.1	13.9	14.1	15.1	0.41
026	A	46.8	62.0	4.0	4.6	0.01	0.48	0.64	7.1	15.1	16.6	18.0	15.9	19.3	0.48
027	A	38.8	56.9	2.5	3.4	0.01	0.63	0.82	6.9	/	9.7	/	11.0	/	0.24
028	A	38.9	56.9	2.5	3.4	0.01	0.63	0.82	7.1	-0.1	9.7	-3.2	11.0	-0.8	0.24
029	A	39.2	56.9	2.6	3.5	0.01	0.63	0.81	7.2	2.7	9.6	4.2	11.0	6.4	0.28
030	A	40.1	56.9	2.8	3.6	0.01	0.63	0.80	6.9	6.6	9.7	8.9	10.9	11.1	0.33
031	A	36.4	60.7	2.2	3.0	0.01	0.69	0.88	3.8	/	1.5	/	-1.8	/	0.01
032	A	38.4	56.7	2.4	3.3	0.01	0.68	0.85	7.1	/	9.5	/	12.4	/	0.24

Table A.8. Continued from previous page.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR		TNR		ΔL_{ta}		Aures Tonality
									dB	dB	dB	dB	dB	dB	
033	A	38.5	56.7	2.4	3.3	0.01	0.69	0.85	7.0	-0.9	9.7	-1.3	12.5	-0.6	0.24
034	A	38.6	56.7	2.5	3.4	0.01	0.69	0.86	6.9	2.7	9.7	5.1	12.3	5.5	0.26
035	A	39.1	56.7	2.6	3.5	0.01	0.70	0.86	6.8	6.9	9.7	9.8	12.3	10.4	0.30
036	A	40.2	56.7	2.7	3.6	0.01	0.72	0.87	7.0	11.1	9.6	14.0	12.4	14.7	0.36
037	A	40.2	64.9	3.1	3.5	0.01	0.51	0.77	6.5	/	8.8	/	2.9	/	0.25
038	A	37.1	56.6	2.2	3.1	0.01	0.70	0.88	7.2	/	2.6	/	4.7	/	0.15
039	A	36.5	63.2	2.2	3.1	0.01	0.69	0.88	9.2	/	4.4	/	1.4	/	0.04
040	A	36.3	56.6	2.1	3.1	0.01	0.70	0.89	-1.0	/	-1.6	/	-0.9	/	0.01
041	A	37.5	57.2	2.5	3.2	0.01	0.63	0.83	-0.8	/	4.8	/	3.8	/	0.15
042	A	36.5	56.6	2.2	3.1	0.01	0.69	0.87	-1.4	/	-1.6	/	0.6	/	0.00
043	A	36.4	56.6	2.2	3.1	0.01	0.70	0.88	-0.9	/	-2.5	/	0.8	/	0.01
044	A	36.5	57.7	2.3	3.1	0.01	0.66	0.87	-1.8	/	-4.0	/	/	/	0.02
045	A	38.9	56.7	2.5	3.4	0.01	0.73	0.90	10.7	/	14.1	/	14.6	/	0.27
046	A	45.2	62.4	3.8	4.0	0.01	0.47	0.69	11.0	/	17.9	/	16.9	/	0.43
047	A	46.0	62.8	3.9	4.3	0.01	0.47	0.66	10.9	6.8	18.4	9.7	17.7	11.0	0.44
048	A	47.1	63.5	4.1	4.6	0.01	0.46	0.64	11.1	11.2	19.2	13.9	18.5	15.3	0.46
049	A	48.9	64.7	4.4	4.9	0.01	0.45	0.61	11.0	15.1	20.6	18.0	19.8	19.2	0.50

Table A.8. Continued from previous page.

Signal	Test	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	Part	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
050	A	51.8	67.0	5.2	5.6	0.01	0.42	0.56	11.0	19.0	23.0	22.1	22.3	23.4	0.56
051	A	41.2	57.3	2.8	3.5	0.01	0.60	0.79	10.9	/	13.9	/	15.3	/	0.33
052	A	41.5	57.3	2.9	3.7	0.01	0.60	0.78	10.9	3.0	13.9	4.2	15.2	6.4	0.35
053	A	42.0	57.3	3.0	3.8	0.01	0.60	0.77	11.1	6.3	13.9	9.0	15.2	11.2	0.40
054	A	43.1	57.4	3.2	4.0	0.01	0.60	0.76	11.0	10.4	13.9	13.2	15.2	15.3	0.46
055	A	36.8	65.4	2.2	3.1	0.01	0.67	0.88	10.3	/	6.9	/	4.0	/	0.07
056	A	40.6	56.8	2.6	3.5	0.01	0.68	0.83	11.0	/	13.9	/	16.6	/	0.31
057	A	40.8	56.8	2.7	3.6	0.01	0.68	0.84	11.0	2.9	13.9	5.1	16.7	5.9	0.33
058	A	40.9	56.8	2.8	3.7	0.01	0.69	0.84	10.9	6.7	13.9	9.8	16.5	10.2	0.37
059	A	41.7	56.8	2.9	3.8	0.01	0.71	0.85	10.8	10.8	13.9	14.0	16.6	14.7	0.41
060	A	43.1	56.9	3.1	4.0	0.01	0.73	0.86	10.9	15.0	13.9	18.1	16.6	18.8	0.47
061	A	43.3	68.8	4.0	3.8	0.01	0.42	0.72	10.7	/	13.1	/	7.2	/	0.33
062	A	38.6	56.7	2.4	3.3	0.01	0.71	0.88	11.1	/	7.7	/	9.1	/	0.24
063	A	41.3	56.7	2.7	3.6	0.01	0.76	0.90	14.7	/	18.1	/	18.7	/	0.34
064	A	48.9	65.7	4.6	4.5	0.01	0.43	0.64	15.1	/	21.9	/	21.1	/	0.51
065	A	50.4	66.9	4.9	5.0	0.01	0.42	0.59	14.9	10.8	23.2	13.9	22.5	15.3	0.52
066	A	52.1	68.3	5.3	5.5	0.01	0.41	0.56	15.0	15.2	24.6	18.0	24.0	19.4	0.55

Table A.8. Continued from previous page.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
067	A	54.5	70.3	6.0	6.1	0.01	0.39	0.53	15.0	19.1	27.0	22.1	26.1	23.2	0.58
068	A	44.4	58.1	3.2	3.8	0.01	0.56	0.75	14.8	/	18.0	/	19.3	/	0.42
069	A	44.8	58.1	3.4	4.1	0.01	0.56	0.73	15.0	6.6	18.0	9.0	19.3	11.2	0.47
070	A	45.4	58.2	3.6	4.3	0.01	0.56	0.72	15.1	10.6	18.0	13.3	19.3	15.4	0.52
071	A	46.7	58.4	3.9	4.6	0.01	0.57	0.71	15.0	14.2	18.0	17.3	19.2	19.4	0.56
072	A	43.6	57.0	2.8	3.8	0.01	0.67	0.81	15.0	/	17.9	/	21.7	/	0.39
073	A	43.9	57.0	3.0	4.0	0.01	0.68	0.82	15.0	7.1	18.0	9.8	20.8	10.6	0.44
074	A	44.0	57.0	3.1	4.1	0.01	0.70	0.83	14.9	10.7	18.0	14.0	21.5	14.5	0.47
075	A	44.9	57.0	3.4	4.3	0.01	0.72	0.84	14.7	14.9	18.0	18.1	20.6	18.8	0.52
076	A	46.6	57.2	3.6	4.6	0.01	0.74	0.85	14.9	19.0	17.9	22.2	20.7	22.8	0.57
077	A	46.7	72.5	4.8	4.2	0.01	0.37	0.66	14.0	/	17.2	/	11.4	/	0.42
078	A	41.0	56.7	2.7	3.5	0.01	0.72	0.87	15.3	/	12.1	/	13.5	/	0.33
079	A	44.4	56.8	3.0	3.9	0.01	0.78	0.91	18.7	/	21.8	/	22.7	/	0.42
080	A	52.6	69.3	5.7	5.1	0.01	0.40	0.58	18.8	/	25.9	/	25.1	/	0.57
081	A	48.0	59.7	3.8	4.2	0.01	0.53	0.70	18.8	/	22.1	/	23.4	/	0.51
082	A	48.4	59.7	4.2	4.7	0.01	0.53	0.69	18.9	10.8	22.0	13.3	23.4	15.4	0.58
083	A	49.1	59.8	4.4	4.9	0.01	0.54	0.68	19.1	15.2	22.0	17.3	23.3	19.5	0.62

Table A.8. Continued from previous page.

Signal	Test	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	Part	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
084	A	50.5	60.1	4.8	5.3	0.01	0.54	0.67	19.0	18.9	22.0	21.4	23.3	23.4	0.66
085	A	47.1	57.5	3.2	4.1	0.01	0.66	0.79	19.0	/	21.9	/	25.8	/	0.47
086	A	47.2	57.5	3.5	4.4	0.01	0.69	0.81	19.0	11.2	22.1	14.1	25.6	14.7	0.54
087	A	47.7	57.5	3.7	4.7	0.01	0.70	0.82	19.0	15.1	22.1	18.1	25.6	18.7	0.58
088	A	48.6	57.6	4.0	4.9	0.01	0.73	0.83	18.8	19.2	22.1	22.2	25.6	22.8	0.62
089	A	50.5	76.5	6.0	4.7	0.01	0.32	0.60	18.4	/	21.2	/	15.4	/	0.48
090	A	44.1	56.9	3.0	3.8	0.01	0.73	0.87	19.2	/	15.8	/	18.4	/	0.42
091	B	26.4	46.8	0.6	0.9	0.01	0.59	0.81	/	/	/	/	/	/	0.02
092	B	28.4	48.8	0.8	1.2	0.01	0.62	0.82	/	/	/	/	/	/	0.02
093	B	30.4	50.8	1.1	1.6	0.01	0.64	0.84	/	/	/	/	/	/	0.02
094	B	32.4	52.8	1.4	2.0	0.01	0.66	0.86	/	/	/	/	/	/	0.01
095	B	34.4	54.8	1.8	2.5	0.01	0.68	0.87	/	/	/	/	/	/	0.01
096	B	26.5	46.6	0.6	1.0	0.01	0.60	0.81	0.9	/	2.3	/	3.0	/	0.16
097	B	28.2	47.5	0.8	1.1	0.01	0.48	0.72	/	/	7.3	/	6.4	/	0.30
098	B	26.8	46.7	0.7	1.0	0.01	0.56	0.77	0.7	/	2.2	/	3.5	/	0.18
099	B	26.6	46.6	0.6	1.0	0.01	0.59	0.79	1.2	/	1.8	/	4.7	/	0.19
100	B	27.1	49.4	0.7	1.0	0.01	0.51	0.77	-1.3	/	0.7	/	-5.0	/	0.01

Table A.8. Continued from previous page.

Signal	Test	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	Part	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
101	B	26.7	46.6	0.6	1.0	0.01	0.61	0.81	2.8	/	5.1	/	5.7	/	0.23
102	B	29.2	48.1	0.9	1.1	0.01	0.45	0.69	3.5	/	9.5	/	8.7	/	0.36
103	B	27.2	46.7	0.7	1.0	0.01	0.55	0.76	2.9	/	5.0	/	6.3	/	0.25
104	B	27.1	46.6	0.7	1.0	0.01	0.59	0.78	3.2	/	4.5	/	7.7	/	0.26
105	B	28.0	51.1	0.8	1.0	0.01	0.46	0.74	1.1	/	3.9	/	-1.8	/	0.16
106	B	26.2	46.6	0.6	0.9	0.01	0.59	0.81	/	/	-10.4	/	/	/	0.02
107	B	27.5	46.6	0.7	1.0	0.01	0.64	0.82	6.6	/	9.7	/	10.5	/	0.32
108	B	31.8	49.8	1.1	1.2	0.01	0.40	0.63	7.1	/	13.8	/	13.0	/	0.46
109	B	28.7	46.9	0.8	1.1	0.01	0.51	0.71	6.6	/	9.7	/	11.0	/	0.38
110	B	26.4	50.6	0.6	0.9	0.01	0.58	0.80	6.7	/	1.5	/	-1.8	/	0.02
111	B	28.3	46.7	0.7	1.1	0.01	0.59	0.76	7.2	/	9.8	/	12.4	/	0.37
112	B	30.2	54.9	1.0	1.1	0.01	0.38	0.68	3.4	/	8.8	/	3.1	/	0.33
113	B	27.1	46.6	0.7	1.0	0.01	0.61	0.81	6.9	/	2.0	/	4.1	/	0.23
114	B	26.5	53.2	0.6	0.9	0.01	0.58	0.80	8.4	/	4.4	/	1.4	/	0.01
115	B	26.3	46.6	0.6	0.9	0.01	0.60	0.81	-0.9	/	-1.3	/	-0.8	/	0.02
116	B	27.4	47.2	0.8	1.0	0.01	0.51	0.74	-1.3	/	4.8	/	4.0	/	0.23
117	B	26.5	46.6	0.6	0.9	0.01	0.58	0.79	-1.2	/	-1.6	/	0.5	/	0.00

Table A.8. Continued from previous page.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
118	B	26.4	46.6	0.6	0.9	0.01	0.59	0.80	-0.9	/	-2.1	/	0.7	/	0.02
119	B	26.6	47.7	0.6	0.9	0.01	0.56	0.79	-2.7	/	-4.0	/	/	/	0.02
120	B	29.0	46.7	0.8	1.1	0.01	0.68	0.84	10.5	/	13.8	/	14.7	/	0.42
121	B	35.1	52.4	1.5	1.4	0.01	0.36	0.57	11.0	/	17.9	/	17.2	/	0.55
122	B	31.1	47.2	1.0	1.2	0.01	0.47	0.67	10.9	/	13.9	/	15.3	/	0.49
123	B	26.7	55.3	0.6	0.9	0.01	0.57	0.80	9.4	/	6.9	/	4.0	/	0.01
124	B	30.5	46.8	0.9	1.2	0.01	0.59	0.74	11.2	/	14.0	/	16.6	/	0.47
125	B	33.3	58.8	1.3	1.2	0.01	0.32	0.61	9.9	/	13.1	/	7.4	/	0.42
126	B	28.6	46.7	0.8	1.1	0.01	0.64	0.81	11.4	/	7.2	/	9.3	/	0.37
127	B	31.4	46.7	0.9	1.2	0.01	0.73	0.85	14.4	/	17.5	/	19.2	/	0.51
128	B	38.8	55.6	1.9	1.6	0.01	0.33	0.50	15.1	/	21.9	/	21.2	/	0.61
129	B	34.3	48.1	1.2	1.4	0.01	0.45	0.62	14.7	/	18.0	/	19.3	/	0.59
130	B	27.6	59.6	0.7	0.9	0.01	0.55	0.80	14.3	/	11.4	/	8.6	/	0.11
131	B	33.5	47.0	1.0	1.3	0.01	0.59	0.72	15.0	/	18.1	/	21.7	/	0.57
132	B	36.9	62.8	1.7	1.4	0.01	0.27	0.54	14.9	/	17.2	/	11.5	/	0.48
133	B	30.9	46.7	0.9	1.2	0.01	0.67	0.81	15.0	/	11.7	/	13.7	/	0.49
134	B	34.6	46.9	1.1	1.4	0.01	0.77	0.87	18.2	/	20.8	/	23.3	/	0.60

Table A.8. Continued from previous page.

Signal	Test	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	Part	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
135	B	42.6	59.2	2.5	2.0	0.01	0.31	0.45	18.9	/	25.9	/	25.2	/	0.66
136	B	37.9	49.6	1.5	1.6	0.01	0.42	0.57	18.8	/	22.1	/	23.4	/	0.66
137	B	29.1	63.7	0.7	1.0	0.01	0.50	0.79	17.7	/	15.5	/	12.8	/	0.17
138	B	37.0	47.5	1.2	1.5	0.01	0.59	0.69	18.9	/	22.1	/	25.7	/	0.65
139	B	40.7	66.7	2.4	1.7	0.01	0.23	0.47	16.8	/	21.2	/	15.5	/	0.54
140	B	34.1	46.9	1.1	1.4	0.01	0.70	0.81	19.2	/	15.6	/	17.8	/	0.59
141	D	36.3	52.7	2.2	3.3	0.02	0.84	1.01	/	/	/	/	/	/	0.00
142	D	36.6	52.7	2.3	3.4	0.02	0.85	1.01	0.5	/	2.2	/	2.7	/	0.09
143	D	37.1	53.6	2.5	3.5	0.02	0.77	0.96	0.9	/	5.0	/	4.6	/	0.14
144	D	36.6	52.8	2.3	3.4	0.02	0.82	0.99	0.9	/	0.6	/	1.8	/	0.07
145	D	36.6	52.7	2.3	3.4	0.02	0.84	1.00	1.2	/	0.4	/	4.7	/	0.09
146	D	36.5	54.7	2.4	3.4	0.02	0.80	0.99	0.6	/	-0.9	/	-5.1	/	0.00
147	D	36.8	52.7	2.3	3.4	0.02	0.85	1.01	2.5	/	4.9	/	5.5	/	0.13
148	D	37.7	54.1	2.6	3.6	0.02	0.74	0.94	3.2	/	7.4	/	7.1	/	0.19
149	D	37.0	52.9	2.4	3.4	0.02	0.81	0.98	3.2	/	3.8	/	4.9	/	0.13
150	D	37.0	52.8	2.3	3.4	0.02	0.83	0.99	3.2	/	3.6	/	7.8	/	0.14
151	D	36.9	56.4	2.5	3.4	0.02	0.76	0.97	2.8	/	2.6	/	-1.6	/	0.00

Table A.8. Continued from previous page.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
152	D	36.3	52.7	2.2	3.3	0.02	0.84	1.01	/	/	-12.2	/	/	/	0.00
153	D	37.7	52.8	2.4	3.5	0.02	0.85	1.01	6.3	/	9.5	/	10.2	/	0.19
154	D	39.4	55.9	3.0	3.8	0.01	0.68	0.90	7.2	/	11.8	/	11.5	/	0.27
155	D	38.1	53.1	2.5	3.6	0.02	0.77	0.94	7.4	/	8.7	/	9.8	/	0.21
156	D	38.1	52.8	2.5	3.5	0.02	0.82	0.97	6.8	/	8.6	/	12.8	/	0.21
157	D	37.9	6/	2.8	3.6	0.01	0.69	0.93	7.2	/	7.8	/	3.5	/	0.17
158	D	37.2	52.8	2.4	3.4	0.02	0.84	1.00	7.6	/	2.3	/	4.1	/	0.14
159	D	36.3	54.9	2.2	3.3	0.02	0.84	1.01	8.1	/	-2.8	/	/	/	0.00
160	D	36.4	52.7	2.3	3.3	0.02	0.85	1.01	-1.5	/	-1.7	/	-1.1	/	0.00
161	D	36.7	53.2	2.4	3.4	0.02	0.80	0.97	-0.7	/	2.1	/	1.8	/	0.06
162	D	36.4	52.7	2.3	3.3	0.02	0.84	1.00	-1.1	/	-4.2	/	/	/	0.00
163	D	36.4	52.7	2.3	3.3	0.02	0.84	1.00	-1.3	/	-4.7	/	/	/	0.00
164	D	36.3	53.1	2.3	3.3	0.02	0.83	1.00	-2.1	/	-7.6	/	/	/	0.00
165	D	39.4	52.8	2.6	3.7	0.02	0.86	1.00	10.4	/	13.5	/	15.2	/	0.26
166	D	42.1	58.6	3.4	4.0	0.01	0.62	0.84	11.2	/	16.0	/	15.7	/	0.36
167	D	40.1	53.7	2.8	3.8	0.01	0.73	0.91	11.4	/	13.0	/	14.1	/	0.29
168	D	36.3	57.3	2.3	3.3	0.02	0.84	1.01	12.0	/	1.5	/	2.6	/	0.00

Table A.8. Continued from previous page.

Signal	Test	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	Part	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
169	D	40.2	53.0	2.7	3.7	0.01	0.80	0.95	10.7	/	13.0	/	17.1	/	0.28
170	D	39.8	63.8	3.2	3.8	0.01	0.61	0.89	10.8	/	12.2	/	7.8	/	0.26
171	D	38.9	52.8	2.5	3.6	0.02	0.84	0.99	11.7	/	7.5	/	8.6	/	0.23
172	D	42.0	53.0	2.8	3.9	0.02	0.87	1.00	14.2	/	17.0	/	19.3	/	0.33
173	D	45.4	61.9	4.1	4.4	0.01	0.55	0.79	15.0	/	20.1	/	19.8	/	0.44
174	D	43.0	55.0	3.1	4.0	0.01	0.68	0.87	15.3	/	17.1	/	18.3	/	0.38
175	D	43.1	53.4	2.9	4.0	0.01	0.78	0.92	14.9	/	17.1	/	21.2	/	0.36
176	D	42.5	67.7	3.8	4.0	0.01	0.53	0.83	14.1	/	16.3	/	12.0	/	0.34
177	D	41.5	53.0	2.8	3.8	0.02	0.84	0.98	15.5	/	11.9	/	13.0	/	0.32
178	D	45.2	53.3	3.1	4.2	0.01	0.88	1.00	18.2	/	2/	/	23.3	/	0.41
179	D	49.1	65.6	4.9	4.8	0.01	0.50	0.72	18.9	/	24.1	/	23.8	/	0.51
180	D	46.4	57.0	3.6	4.3	0.01	0.63	0.82	19.3	/	21.1	/	22.6	/	0.46
181	D	46.5	54.4	3.2	4.3	0.01	0.76	0.89	18.8	/	21.0	/	25.2	/	0.44
182	D	45.9	71.6	4.8	4.4	0.01	0.45	0.77	18.7	/	20.3	/	16.0	/	0.42
183	D	44.8	53.4	3.1	4.1	0.01	0.84	0.96	19.5	/	15.9	/	17.4	/	0.41
184	E	46.4	66.8	5.3	7.1	0.02	0.79	0.94	/	/	/	/	/	/	0.01
185	E	46.5	66.6	5.3	7.1	0.02	0.79	0.94	0.4	/	2.2	/	3.0	/	0.09

Table A.8. Continued from previous page.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
186	E	48.1	67.5	6.1	7.5	0.02	0.70	0.88	1.2	/	7.2	/	6.3	/	0.18
187	E	46.8	66.7	5.4	7.2	0.02	0.77	0.92	0.8	/	2.2	/	3.5	/	0.09
188	E	46.6	66.6	5.3	7.2	0.02	0.79	0.93	0.9	/	1.6	/	4.7	/	0.09
189	E	47.1	69.3	5.8	7.3	0.02	0.71	0.90	1.1	/	0.7	/	-4.9	/	0.00
190	E	46.7	66.6	5.4	7.2	0.02	0.79	0.94	2.4	/	5.0	/	5.8	/	0.11
191	E	49.3	68.1	6.5	7.7	0.02	0.67	0.86	3.2	/	9.4	/	8.6	/	0.23
192	E	47.3	66.7	5.5	7.3	0.02	0.76	0.91	2.8	/	5.0	/	6.3	/	0.13
193	E	47.0	66.6	5.4	7.2	0.02	0.78	0.93	3.0	/	4.7	/	7.6	/	0.13
194	E	48.0	71.2	6.5	7.5	0.02	0.66	0.88	3.4	/	3.9	/	-1.8	/	0.12
195	E	46.2	66.6	5.2	7.0	0.02	0.79	0.94	/	/	-10.4	/	/	/	0.01
196	E	47.4	66.6	5.5	7.4	0.02	0.80	0.94	6.2	/	9.6	/	10.4	/	0.16
197	E	51.9	69.9	7.3	8.2	0.01	0.62	0.82	7.1	/	13.8	/	13.0	/	0.30
198	E	48.8	66.9	5.9	7.6	0.02	0.73	0.89	6.7	/	9.7	/	11.0	/	0.20
199	E	48.3	66.7	5.6	7.5	0.02	0.77	0.91	7.0	/	9.4	/	12.3	/	0.18
200	E	50.3	74.9	7.5	8.0	0.01	0.58	0.83	7.5	/	8.8	/	3.1	/	0.23
201	E	47.1	66.6	5.4	7.2	0.02	0.79	0.94	7.8	/	2.2	/	4.1	/	0.11
202	E	46.3	66.6	5.3	7.1	0.02	0.79	0.94	-1.1	/	-1.4	/	-0.8	/	0.01

Table A.8. Continued from previous page.

Signal	Test	SPL_A	SPL_C	N_Z	$N_{M\&G}$	R	$S_{vB\ Z}$	$S_{vB\ M\&G}$	PR		TNR		ΔL_{ta}		Aures
Number	Part	dBA	dBC	sone	sone	asper	acum	acum	dB	dB	dB	dB	dB	dB	Tonality
203	E	47.5	67.2	5.9	7.4	0.02	0.72	0.90	-0.9	/	4.8	/	3.9	/	0.13
204	E	46.5	66.6	5.3	7.1	0.02	0.78	0.93	-1.3	/	-1.6	/	0.5	/	0.00
205	E	46.4	66.6	5.3	7.1	0.02	0.79	0.94	-1.0	/	-2.5	/	0.7	/	0.01
206	E	46.5	67.7	5.5	7.2	0.02	0.75	0.92	-1.2	/	-4.0	/	/	/	0.00
207	E	48.9	66.7	5.8	7.6	0.02	0.81	0.94	10.2	/	13.5	/	14.8	/	0.21
208	E	55.3	72.5	8.4	8.8	0.01	0.57	0.77	11.2	/	17.9	/	17.1	/	0.38
209	E	51.2	67.3	6.5	7.9	0.02	0.69	0.86	10.7	/	13.9	/	15.3	/	0.28
210	E	50.5	66.8	6.0	7.8	0.02	0.76	0.90	11.0	/	13.9	/	16.5	/	0.25
211	E	53.4	78.8	8.8	8.5	0.01	0.51	0.78	11.8	/	13.1	/	7.4	/	0.32
212	E	48.6	66.7	5.7	7.5	0.02	0.79	0.93	11.6	/	7.8	/	9.2	/	0.18
213	E	51.2	66.7	6.1	8.0	0.02	0.81	0.95	14.4	/	17.5	/	18.9	/	0.26
214	E	59.0	75.8	10.7	9.7	0.01	0.50	0.71	15.2	/	21.9	/	21.1	/	0.48
215	E	54.4	68.1	7.3	8.4	0.01	0.65	0.82	14.8	/	18.0	/	19.4	/	0.37
216	E	53.4	67.0	6.5	8.2	0.01	0.75	0.88	15.2	/	18.0	/	20.6	/	0.31
217	E	51.0	66.7	6.1	7.8	0.02	0.79	0.93	15.4	/	12.1	/	13.6	/	0.25
218	E	54.3	66.8	6.6	8.4	0.02	0.83	0.95	18.7	/	21.4	/	23.4	/	0.33
219	E	62.6	79.2	13.2	10.8	0.01	0.46	0.66	19.2	/	25.9	/	25.1	/	0.55

Table A.8. Continued from previous page.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
220	E	58.0	69.7	8.5	9.1	0.01	0.61	0.78	18.8	/	22.0	/	23.4	/	0.46
221	E	57.0	67.5	7.2	8.8	0.01	0.73	0.86	19.2	/	22.1	/	25.6	/	0.39
222	E	54.1	66.9	6.7	8.3	0.01	0.80	0.92	19.2	/	16.2	/	17.7	/	0.33

Table A.9. Values of the main sound quality metrics for Test 3 2-minute signals.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR		TNR		ΔL_{ta}		Aures Tonality
									dB	dB	dB	dB	dB	dB	
001	C	26.4	46.8	0.6	0.9	0.01	0.59	0.81	/	/	/	/	/	/	0.02
002	C	27.2	46.7	0.7	1.0	0.01	0.55	0.76	2.9	/	5.0	/	6.3	/	0.25
003	C	27.1	46.6	0.7	1.0	0.01	0.59	0.78	3.2	/	4.5	/	7.7	/	0.26
004	C	28.7	46.9	0.8	1.1	0.01	0.51	0.71	6.6	/	9.7	/	11.0	/	0.38
005	C	28.3	46.7	0.7	1.1	0.01	0.59	0.76	7.2	/	9.8	/	12.4	/	0.37
006	C	31.1	47.2	1.0	1.2	0.01	0.47	0.67	10.9	/	13.9	/	15.3	/	0.49
007	C	30.5	46.8	0.9	1.2	0.01	0.59	0.74	11.2	/	14.0	/	16.6	/	0.47
008	C	34.3	48.1	1.2	1.4	0.01	0.45	0.62	14.7	/	18.0	/	19.3	/	0.59
009	C	33.5	47.0	1.0	1.3	0.01	0.59	0.72	15.0	/	18.1	/	21.7	/	0.57
010	F	46.4	66.8	5.3	7.1	0.02	0.79	0.94	/	/	/	/	/	/	0.01
011	F	47.3	66.7	5.5	7.3	0.02	0.76	0.91	2.8	/	5.0	/	6.3	/	0.13
012	F	47.0	66.6	5.4	7.2	0.02	0.78	0.93	3.0	/	4.7	/	7.6	/	0.13
013	F	48.8	66.9	5.9	7.6	0.02	0.73	0.89	6.7	/	9.7	/	11.0	/	0.20
014	F	48.3	66.7	5.6	7.5	0.02	0.77	0.91	7.0	/	9.4	/	12.3	/	0.18
015	F	51.2	67.3	6.5	7.9	0.02	0.69	0.86	10.7	/	13.9	/	15.3	/	0.28
016	F	50.5	66.8	6.0	7.8	0.02	0.76	0.90	11.0	/	13.9	/	16.5	/	0.25
017	F	54.4	68.1	7.3	8.4	0.01	0.65	0.82	14.8	/	18.0	/	19.4	/	0.37

Table A.9. Continued from previous page.

Signal Number	Test Part	SPL_A dBA	SPL_C dBC	N_Z sone	$N_{M\&G}$ sone	R asper	$S_{vB\ Z}$ acum	$S_{vB\ M\&G}$ acum	PR dB	dB	TNR dB	dB	ΔL_{ta} dB	dB	Aures Tonality
018	F	53.4	67.0	6.5	8.2	0.01	0.75	0.88	15.2	/	18.0	/	20.6	/	0.31

A.4.2 Test 3 Subject's Responses and Statistics

Table A.10. Average annoyance ratings and corresponding standard errors for 5-second sounds in Test 3.

Signal Number	Test Part	Annoyance	Standard Error	Signal Number	Test Part	Annoyance	Standard Error
001	A	1.8	0.1	2.0	A	2.57	0.21
003	A	3.5	0.3	4.0	A	4.10	0.31
005	A	4.4	0.4	6.0	A	3.45	0.26
007	A	3.0	0.2	8.0	A	2.60	0.16
009	A	3.9	0.2	10.0	A	2.34	0.17
011	A	3.8	0.3	12.0	A	3.30	0.22
013	A	3.2	0.2	14.0	A	3.62	0.22
015	A	3.7	0.2	16.0	A	4.17	0.22
017	A	2.9	0.2	18.0	A	4.42	0.27
019	A	3.3	0.3	20.0	A	1.79	0.12
021	A	4.9	0.2	22.0	A	3.77	0.19
023	A	4.0	0.2	24.0	A	4.03	0.22
025	A	4.5	0.2	26.0	A	4.72	0.22
027	A	3.4	0.2	28.0	A	3.77	0.23
029	A	3.9	0.2	30.0	A	4.36	0.23
031	A	1.8	0.1	32.0	A	5.20	0.23
033	A	5.0	0.3	34.0	A	5.18	0.26
035	A	5.3	0.2	36.0	A	5.70	0.21
037	A	4.0	0.3	38.0	A	3.13	0.34
039	A	2.0	0.2	40.0	A	2.72	0.21
041	A	2.7	0.2	42.0	A	1.92	0.13
043	A	3.4	0.2	44.0	A	2.10	0.13
045	A	5.1	0.3	46.0	A	4.47	0.23
047	A	4.6	0.2	48.0	A	4.61	0.25

Table A.10. Continued from previous page.

Signal Number	Test Part	Annoyance	Standard Error	Signal Number	Test Part	Annoyance	Standard Error
049	A	5.4	0.3	50.0	A	5.63	0.26
051	A	4.0	0.2	52.0	A	4.43	0.22
053	A	4.8	0.2	54.0	A	4.78	0.22
055	A	2.2	0.2	56.0	A	5.39	0.25
057	A	5.5	0.2	58.0	A	5.53	0.23
059	A	5.7	0.2	60.0	A	5.80	0.25
061	A	4.2	0.3	62.0	A	3.70	0.32
063	A	5.4	0.3	64.0	A	4.83	0.25
065	A	5.0	0.2	66.0	A	5.52	0.25
067	A	6.1	0.2	68.0	A	4.48	0.24
069	A	5.1	0.3	70.0	A	4.92	0.19
071	A	5.4	0.2	72.0	A	5.80	0.21
073	A	5.9	0.2	74.0	A	6.28	0.24
075	A	6.2	0.3	76.0	A	6.54	0.23
077	A	5.1	0.3	78.0	A	4.97	0.32
079	A	6.2	0.2	80.0	A	4.86	0.27
081	A	5.0	0.2	82.0	A	5.44	0.27
083	A	5.6	0.2	84.0	A	5.82	0.22
085	A	6.2	0.2	86.0	A	6.50	0.20
087	A	6.7	0.2	88.0	A	6.75	0.23
089	A	5.9	0.3	90.0	A	5.80	0.29
091	B	1.8	0.1	92.0	B	2.20	0.16
093	B	2.7	0.2	94.0	B	3.33	0.25
095	B	3.7	0.3	96.0	B	3.18	0.27
097	B	2.9	0.2	98.0	B	2.63	0.17
099	B	3.8	0.2	100.0	B	2.33	0.18

Table A.10. Continued from previous page.

Signal Number	Test Part	Annoyance	Standard Error	Signal Number	Test Part	Annoyance	Standard Error
101	B	3.5	0.3	102.0	B	3.05	0.20
103	B	2.7	0.1	104.0	B	4.05	0.21
105	B	2.7	0.2	106.0	B	1.75	0.11
107	B	4.6	0.3	108.0	B	3.55	0.18
109	B	3.4	0.2	110.0	B	1.78	0.11
111	B	4.8	0.2	112.0	B	3.42	0.21
113	B	2.8	0.3	114.0	B	1.93	0.13
115	B	2.7	0.3	116.0	B	2.53	0.13
117	B	2.1	0.1	118.0	B	3.20	0.22
119	B	2.0	0.1	120.0	B	4.91	0.29
121	B	4.1	0.2	122.0	B	3.88	0.21
123	B	1.9	0.1	124.0	B	4.82	0.26
125	B	3.9	0.2	126.0	B	4.08	0.32
127	B	5.2	0.3	128.0	B	4.36	0.21
129	B	4.4	0.2	130.0	B	2.08	0.12
131	B	5.4	0.3	132.0	B	4.58	0.27
133	B	4.4	0.3	134.0	B	5.74	0.31
135	B	4.8	0.3	136.0	B	4.30	0.18
137	B	2.4	0.2	138.0	B	6.00	0.25
139	B	5.1	0.3	140.0	B	4.93	0.26
141	D	1.7	0.1	142.0	D	3.13	0.23
143	D	2.8	0.2	144.0	D	2.43	0.15
145	D	3.6	0.2	146.0	D	2.33	0.12
147	D	3.8	0.2	148.0	D	3.30	0.18
149	D	2.7	0.2	150.0	D	3.96	0.21
151	D	2.6	0.2	152.0	D	1.81	0.13

Table A.10. Continued from previous page.

Signal Number	Test Part	Annoyance	Standard Error	Signal Number	Test Part	Annoyance	Standard Error
153	D	4.6	0.3	154.0	D	3.79	0.18
155	D	3.5	0.2	156.0	D	4.62	0.23
157	D	3.2	0.2	158.0	D	2.99	0.28
159	D	1.9	0.1	160.0	D	2.33	0.16
161	D	2.5	0.2	162.0	D	1.94	0.13
163	D	2.8	0.2	164.0	D	1.86	0.12
165	D	5.3	0.3	166.0	D	4.12	0.18
167	D	3.8	0.2	168.0	D	1.79	0.12
169	D	5.1	0.3	170.0	D	3.99	0.20
171	D	3.9	0.3	172.0	D	5.61	0.23
173	D	4.7	0.2	174.0	D	4.38	0.23
175	D	5.9	0.2	176.0	D	4.62	0.26
177	D	4.8	0.3	178.0	D	5.85	0.24
179	D	4.9	0.2	180.0	D	4.99	0.23
181	D	6.3	0.2	182.0	D	5.31	0.30
183	D	5.2	0.3	184.0	E	1.88	0.12
185	E	3.7	0.3	186.0	E	3.53	0.22
187	E	3.1	0.2	188.0	E	4.38	0.26
189	E	3.1	0.2	190.0	E	4.18	0.28
191	E	4.0	0.2	192.0	E	3.62	0.19
193	E	4.7	0.2	194.0	E	3.85	0.25
195	E	1.9	0.1	196.0	E	4.75	0.30
197	E	4.6	0.2	198.0	E	4.23	0.21
199	E	5.7	0.2	200.0	E	4.97	0.26
201	E	3.2	0.3	202.0	E	3.05	0.30
203	E	3.2	0.2	204.0	E	2.33	0.15

Table A.10. Continued from previous page.

Signal Number	Test Part	Annoyance	Standard Error	Signal Number	Test Part	Annoyance	Standard Error
205	E	3.9	0.3	206.0	E	2.42	0.16
207	E	5.4	0.3	208.0	E	5.20	0.21
209	E	4.6	0.2	210.0	E	5.67	0.25
211	E	5.8	0.2	212.0	E	4.19	0.29
213	E	5.8	0.2	214.0	E	6.13	0.21
215	E	5.3	0.2	216.0	E	6.36	0.26
217	E	5.3	0.3	218.0	E	6.29	0.23
219	E	6.8	0.3	220.0	E	5.86	0.24
221	E	6.9	0.2	222.0	E	5.55	0.29

Table A.11. Average and standard error of annoyance ratings for 2-minute sounds and corresponding 5-second sounds in Test 3.

2-minute sounds				5-second sounds			
Signal Number	Test Part	Annoyance	Standard Error	Signal Number	Test Part	Annoyance	Standard Error
001	C	1.70	0.11	091	B	1.76	0.11
002	C	2.33	0.13	103	B	2.67	0.14
003	C	3.45	0.19	104	B	4.05	0.21
004	C	2.90	0.16	109	B	3.37	0.19
005	C	4.11	0.23	111	B	4.85	0.24
006	C	3.41	0.12	122	B	3.88	0.21
007	C	4.69	0.23	124	B	4.82	0.26
008	C	3.86	0.15	129	B	4.36	0.24
009	C	5.14	0.28	131	B	5.41	0.29
010	E	1.82	0.14	184	F	1.88	0.12
011	E	2.87	0.19	192	F	3.62	0.19
012	E	4.08	0.22	193	F	4.72	0.22
013	E	3.33	0.15	198	F	4.23	0.21
014	E	4.75	0.24	199	F	5.73	0.21
015	E	3.86	0.20	209	F	4.63	0.21
016	E	5.21	0.24	210	F	5.67	0.25
017	E	4.22	0.17	215	F	5.31	0.22
018	E	6.28	0.24	216	F	6.36	0.26

B. SOFTWARE AND GUIDELINES FOR METRIC CALCULATION

B.1 Software Details

The software has been developed with MATLAB to calculate important metrics and predict annoyance based on developed models. All the metrics used in the software are computed from narrow band power spectral density (PSD). The software takes estimated narrowband power spectrum densities or *.wav files with calibration information (A-weighted sound pressure level) as input to predict annoyance. For the *.wav file, after calibrating the level, narrow band power spectral density is estimated with the time history file. Functions were programmed to extract the broadband component of a sound, to calculate Oppressive Penalty, Modified TNR' (with the frequency weighting and hearing threshold), Moore and Glasberg Loudness, and Sharpness (based on the calculated Loudness spectrum). Annoyance was predicted using global models for 5-second sounds and 2-minute sounds with the calculated metrics. Figure B.1 illustrates the structure the software.

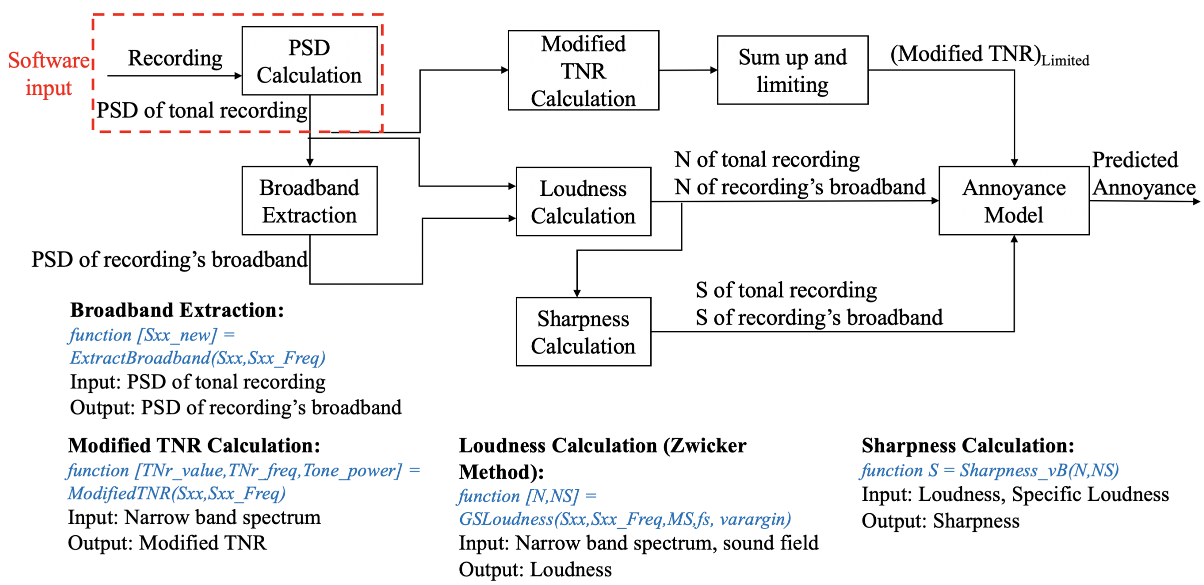

Figure B.1. Structure diagram of the software.

An example office recording was examined with the software. This sound is measured in an open office area under a rooftop unit. Figure B.2 shows the output of the software and a graphic interface of the software.

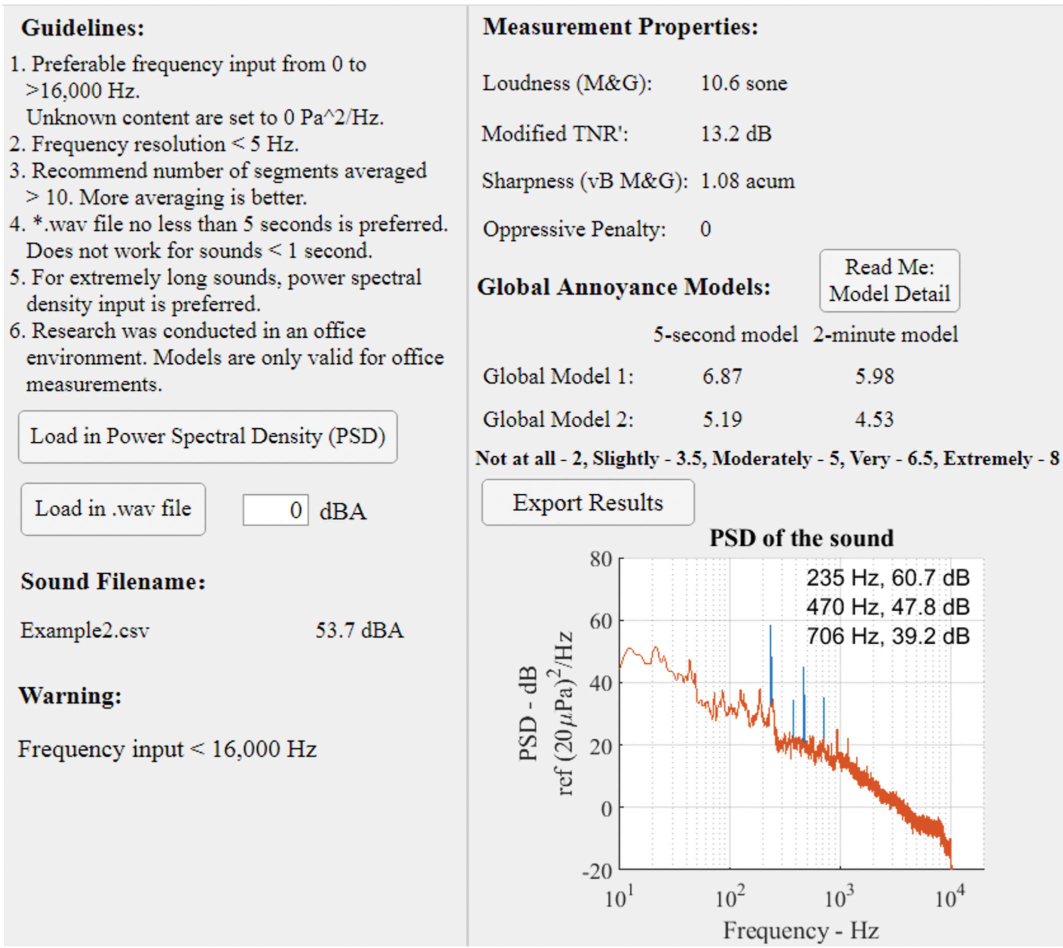

Figure B.2. User interface of the software. Example sound is an office recording under a rooftop unit.

The corresponding exported *.txt file contains all information in the user interface. It has a form of:

```
************************************************
File Name: Example2.csv


Power of prominent tones:
```

Frequency	Tone Power	Tonal Audibility
235 Hz	60.7 dB	12.1 dB
470 Hz	47.8 dB	10.1 dB
706 Hz	39.2 dB	0.6 dB
378 Hz	38.7 dB	-1.8 dB

Sound attributes:

A-weighted sound pressure level: 53.7 dBA

Loudness (M&G): 10.6 sone

Modified TNR with frequency weighting: 13.2 dB

Sharpness (vB M&G): 1.08 acum

Oppressive Penalty: 0

Model predicted annoyance:

	5-second sounds	2-minute sounds
Global Model 1:	6.87	5.98
Global Model 2:	5.19	5.19

****

Three columns represent:

1. Frequency - Hz 2. PSD - Pa^2/Hz 3. Tone removed PSD - Pa^2/Pa

0	2.639900e-08	2.639900e-08
1	7.410300e-07	7.410300e-07
2	6.128000e-06	6.128000e-06
3	2.844200e-05	2.844200e-05
4	5.795400e-05	5.795400e-05
5	4.466100e-05	4.466100e-05
6	3.264600e-05	3.264600e-05
7	3.673800e-05	3.673800e-05
8	2.417000e-05	2.417000e-05
9	1.195400e-05	1.195400e-05
10	1.093100e-05	1.093100e-05
11	3.025400e-05	3.025400e-05
...	...	...

Global Model 1 predicts and NC-20 neutral broadband to be around 2 (not at all annoying). Increasing either sound level or prominence of tonal components would result in a higher predicted annoyance level. Global Model 2 is developed with an assumption that people would acclimatize to an ambient broadband background noise. Predicted annoyance from Global Model 2 is a function of difference s between the measured tonal sound and ambient broadband. Global Model 1 is more affected by the overall level, while Global Model 2 is more focused on the contributions due to the additional tonal components.

The difference between responses to 5-second and 2minute exposures have been examined. Responses to 2-minute exposure tended to be lower. Further research is required to see if further acclimation occurs.

B.1.1 Software Guidelines

Some software guidelines are provided here for the proper use of the software:

1. As for the power spectral density, the preferable frequency input is from 0 to > 16 kHz. The unknown contents are set to 0 Pa^2/Hz. Similar to the *.wav file input, the preferable sampling frequency is > 32 kHz.

2. A fine frequency resolution (< 5 Hz) is required for the power spectral density so that an accurate tonality metric can be obtained.

3. For the power spectral density, recommend number of segments averaged is > 10. More averaging is better. For the *.wav file, sounds with duration > 5 seconds are preferred. The software does not work for sounds < 1 second.

4. For the extremely long sounds, power spectral density input is preferred.

5. Research was conducted in an office environment. Models are only valid for office measurements.

B.2 Metric Calculations

B.2.1 Loudness and Sharpness from Moore and Glasberg Method

This function in this example is called *MG_Loudness*. It was written by S. Hales Swift, and revised by G. Song. It takes narrow band power spectral density as input to compute the stationary Moore and Glasberg Loudness and corresponding von Bismark Sharpness.

```
function results=MG_Loudness(Sxx,Sxx_f,Fs)
%This version will calculate stationary loudness.
% Input Variables:
% Sxx, Sxx_f  = PSD of the sounds
%
% Output Variables:
% results     = Moore and Glasberg loudness, specific loudness and corresponding sharpness


f3o=[20 25 31.5 40 50 63 80 100 125 160 200 250 315 400 500 630 750 800 ...
1000 1250 1500 1600 2000 2500 3000 3150 4000 5000 6000 6300 8000 ...
9000 10000 11200 12500 14000 15000 16000 18000 20000];

farray=[0 0 0 0 0 0 0 0 .1 .3 .5 .9 1.4 1.6 1.7 2.5 2.7 2.6 2.6 3.2 5.2 ...
6.6 12 16.8 15.3 15.2 14.2 10.7 7.1 6.4 1.8 -.9 -1.6 1.9 4.9 2 ...
-2 2.5 2.5 2.5]+[-39.6 -32.0 -25.85 -21.4 -18.5 -15.9 -14.1 -12.4 -11.0 ...
-9.6 -8.3 -7.4 -6.2 -4.8 -3.8 -3.3 -2.9 -2.6 -2.6 -4.5 -5.4 -6.1 -8.5 ...
-10.4 -7.3 -7.0 -6.6 -7.0 -9.2 -10.2 -12.2 -10.8 -10.1 -12.7 -15.0 -18.2 ...
-23.8 -32.3 -45.5 -50.0];%2007 values

O2C = interp1(f3o,farray,Sxx_f,'spline');
O2C(Sxx_f>20000) = farray(end);
O2C(Sxx_f<20) = farray(1);
df=Sxx_f(2) - Sxx_f(1);
avgSpectra = 10.^(O2C./10).*Sxx/(20e-6).^2*df;
sumN = ceil(10/df);
N = floor(length(Sxx)/sumN);
avgSpectra_rough = zeros(N,1);
for i = 1:sumN
  avgSpectra_rough = avgSpectra_rough + avgSpectra(i:sumN:i+(N-1)*sumN);
end

dCam=0.1;
results.excitation_pattern=auditoryfilters(avgSpectra_rough',df*sumN,dCam);
cams=dCam:dCam:39.0;
```

```
if dCam==.1
cams=dCam:dCam:38.9;
activeIdxs=find(cams>=1.8&cams<=38.9);
elseif dCam==.25
activeIdxs=find(cams>=1.75&cams<=39.0);
else
activeIdxs=find(cams>=1.75&cams<=39.0);
end
cams=cams(activeIdxs);
% Thus ep is now a vector containing the excitation pattern in ERB increments

% Convert from excitation pattern to specific loudness pattern
M=results.excitation_pattern;
[Mm,Nn]=size(M);
M2=zeros(Mm,Nn);
% for nn=18:Nn;
for nn=1:Nn;
  M2(:,nn)=ex2spec2007(M(:,nn),cams(nn)*10);
end
results.specific_loudness=M2;

S=.2*results.specific_loudness*ones(Nn,1);
results.loudness_sones=S;
results.loudness_phons=sones2phons(S);
%% Calculate sharpness using method from S. Hales Swift, and Kent L. Gee
% Extending sharpness calculation for an alternative loudness metric input
% Citation: The Journal of the Acoustical Society of America142, EL549 (2017);
% doi: 10.1121/1.5016193
cf=(10.^(cams/21.366)-1)/.004368; % .1-ERBN# center frequencies
z=13*atan(0.00076*cf)+3.5*atan((cf/7500).^2); % calculate critical band rate values associated with
     .1-ERBN# filter center frequencies
g=sharpWeights(z,'bismarck',[]); % calculate sharpness weighting factors for each filter center
     frequency
LSharp=.2*results.specific_loudness*(g'.*z');
results.sharpness=0.11229*LSharp/S;

%*******************************************************************
%*******************************************************************
% function sharpWeights
%*******************************************************************
% subroutine for sharpness calculation
%*******************************************************************
function g=sharpWeights(z,type,N)
g=zeros(1,length(z));
```

```
switch type
  case 'standard'
    g(z<15.8)=1;
    g(z>=15.8)=0.15*exp(0.42*(z(z>=15.8)-15.8))+0.85;
  case 'bismarck'
    g(z<15)=1;
    g(z>=15)=0.2*exp(0.308*(z(z>=15)-15))+0.8;
  case 'aures'
  for nt=1:length(N)
    g(nt,:)=0.078*exp(0.171*z)./z*N(nt)/log(0.05*N(nt)+1);
  end
end

%*******************************************************************
%*******************************************************************
% function sones2phons
%*******************************************************************
% subroutine to convert sone to phone
%*******************************************************************
function [phons]=sones2phons(sones)
% This function converts a signal from sones to phon according to the
% relatiosnhip described in ANSI S3.4-2005.
s=[0 1 2 3 4 5 7.5 10 15 20 25 30 35 40 45 50 55 60 65 70 75 80 85 90 95 100 105 110 115 120];
p=[.00117 .00188 .00295 .00454 .00673 .00919 .01741 .02957 ...
.07082 .14283 .25616 .42409 .66411 1.00001 1.46418 2.10144 ...
2.97424 4.17094 5.81918 8.10880 11.33444 15.99175 22.93840 ...
33.20565 48.19039 70.23295 103.07866 152.42529 227.23890 341.14794];
phons=interp1(p,s,sones,'spline');
end

%*******************************************************************
%*******************************************************************
% function auditoryfilters
%*******************************************************************
% subroutine to calculates the shape and response of the auditory filters
%*******************************************************************
function [excitation] = auditoryfilters(matrox,df,dCam)
%% auditoryfilters
% The auditory filters function and calculates the shape and
% response of the auditory filters to the power spectrum arriving at the
% cochlea spaced in frequency increments of df. According to ANSI S3.4-2007
% df should be 10 Hz for the standard implementation.
% dCam is also adjustable, but .1 is used in ISO 532-2/ANSI S3.4-2007 and
% .25 in ISO 532-3 as well as Glasberg and Moore 2002
```

```
%% Set up various numerical arrays needed for the computation
[M,N]=size(matrox); % Determine number of time steps M and frequency bins N
ind=1:N;
fvector=ind*df; % gives the frequencies of the incoming spectral components
cams=dCam:dCam:39.0;
if dCam==.1
activeIdxs=find(cams>=1.8&cams<=38.9);
cams=dCam:dCam:38.9;
elseif dCam==.25
activeIdxs=find(cams>=1.75&cams<=39.0);
else
activeIdxs=find(cams>=1.75&cams<=39.0);
end
cams=cams(activeIdxs);
cf=(10.^(cams/21.366)-1)/.004368; % list of auditory filter center frequencies
invcf=1./cf; % gives the multiplicative inverse of the filter center frequencies

%% Determine level dependent filter elements to speed up processing
idx=repmat(cf',1,N)>repmat(fvector,length(cf),1);

%% Calculate static filter centered at each input component and associated power
g=abs(1./fvector'*fvector-1); % calculate normalized deviation from center frequency
p0=0.162120536618976./(0.004368+1./fvector);% find p value for initial calculation (ANSI S3.4-2007
    equation 1)
p0g=repmat(p0',1,N).*g; % calculate product pg from ANSI equation 2
W00=(1+p0g).*exp(-p0g); % calculate static filter shape (ANSI equation 2)
W00(g>4)=0;
epsilon=1e-20; % avoids problems with logarithm function in power calculation
X=10*log10(epsilon+matrox*W00'); % calculate power through each filter at each time

%% Calculate output of static part of dynamic filter
g2=abs(invcf'*fvector-1); % calculate normalized deviation from center frequency
g2(g2>4)=4;
p02=0.162120536618976./(0.004368+invcf);% find static p value for initial calculation (ANSI S3.4-2007
    equation 1) (4*cf)/ERBN=(4*cf)/24.673/(0.004368*cf+1)
p0g2=repmat(p02',1,N).*g2; % calculate product pg
W=(1+p0g2).*exp(-p0g2); % calculate static filter shape (ANSI equation 2);
W0=zeros(size(W)); % allocate space for static filter
W0(~idx&g2<4)=W(~idx&g2<4); % load in non-zero non-level-dependent part of filters
% W0=sparse(W0); % use sparse matrix operations
o2ea=W0*matrox'; % calculate excitation from level-independent partition

%% Prepare for level-dependent calculation
c1=0.257939336618976;c2=0.0018788;c3=0.004368; % constants
```

```
beta1=c1*g2./(c3+repmat(invcf',1,N)); % additive constant matrix term used in pg
beta2=c2*g2./(c3+repmat(invcf',1,N)); % multiplicative constant matrix term used in pg

%% Determine limiting filter shapes in order to avoid unnecessary calculations
% maxX=max(X);
% pg=beta1-beta2.*repmat(maxX,length(cams),1);
% Wlim=zeros(size(W));
% Wlim(idx)=(1+pg(idx)).*exp(-pg(idx));
% dBdown=-50;
% idx2=10*log10(Wlim(idx))>dBdown;
% idx=idx(idx2); % Trim all filter elements that are always dBdown dB below or more

% Prepare sparse matrices to broadcast and condense the arrays
test1=1:N;
test2=1:length(cams);
A=repmat(test1,length(cf),1);
B=repmat(test2',1,N);
a=A(idx);
b=B(idx);
% broadcast=sparse(1:length(a),a,ones(1,length(a)),length(a),N);
% condense=sparse(b,1:length(b),ones(1,length(b)),length(cams),length(b));


%% Calculate dynamic filter output as a sum of both level-dependent and level-independent parts
beta1idx=beta1(idx);
beta2idx=beta2(idx);
% PG=repmat(beta1idx,1,M)-repmat(beta2idx,1,M).*(broadcast*X'); % calculate PG projected
% condense product and add contribution from level-independent filter
% excitation=(condense*((broadcast*matrox').*(1+PG).*exp(-PG))+o2ea)';

PG=repmat(beta1idx,1,M)-repmat(beta2idx,1,M).*(X(a)'); % calculate PG projected
cond_broad_matrox = condense(b,(matrox(a)').*(1+PG).*exp(-PG),length(cams));
excitation = (cond_broad_matrox + o2ea)';

end

function [condense_ma] = condense(b,ma,N)
condense_ma = zeros(N,1);
for i = 1:length(b)
condense_ma(b(i)) = condense_ma(b(i)) + ma(i);
end
end
```

Modified Tone to Noise Ratio with Frequency Weighing

This function in this example is called *ModifiedTNR*. It was written by K.H. Lee, and revised by G. Song. It computes the Tone-to-Noise Ratio with hearing threshold and frequency weighting from the narrowband power spectral density input.

```
% Program to calculate Modified Tone-to-Noise Ratio
% ANSI S1.13 (1995)
%
% Author:   Kyoung Hoon Lee, Herrick Labs, Purdue University
% Date:     8/7/2002
%
% Modified by Guochenhao Song
% Date of last modification:
%           07/01/2020
%
% Syntax:
% [TNr_value,TNr_freq, Tone_power]=ModifiedTNR(Sxx,Sxx_Freq)
%
% Input Variables:
% Sxx, Sxx_f        = PSD of the sounds
%
% Output Variables:
% TNr_value     = A vector of TNR values
% TNr_freq        = A vector of frequencies corresponding to the TNR values
% Tone_power  = A vector of power of identified tones

function [TNr_value,TNr_freq, Tone_power]=ModifiedTNR(Sxx,Sxx_Freq)
peak_dist=8;
Y = Sxx; Freq = Sxx_Freq;

%Power Spectrum in Decibels
P_ref_sq=(2*10^-5)^2;
Y_dB=10*log10(Y/P_ref_sq);

% Calculate the moving average values of PSD
ave_width=20;
half_width = round(ave_width/2);
num_spec = size(Freq);

Y_dB_flr_temp=filter(ones(1,ave_width)/ave_width,1,Y_dB);
Y_dB_flr=Y_dB_flr_temp(1+half_width:end);
Y_dB_flr(length(Y_dB)-half_width+1:length(Y_dB))= ...
Y_dB(length(Y_dB)-half_width+1:length(Y_dB));
```

```
clear Y_dB_flr_temp;

% Peak detection
peak_width = 3;
peak_count = 0;

[peak_det,amp,peak_count]=findpeak(Y_dB,Y_dB_flr,Freq,peak_dist, ...
peak_width,num_spec(1));

% Remove peaks to calculate updated noise floor
Ynew=Y;

for (ii=1:peak_count)
  Pstart=0;
  Pend=0;
  Pcenter=peak_det(ii);
  while (Ynew(Pcenter-Pstart-1)<Ynew(Pcenter-Pstart))
    Pstart=Pstart+1;
  end
  while (Ynew(Pcenter+Pend)>Ynew(Pcenter+Pend+1))
    Pend=Pend+1;
  end
  slope=(Ynew(Pcenter+Pend)-Ynew(Pcenter-Pstart))/(Pstart+Pend);

  for (jj=-Pstart:Pend)
    Ynew(Pcenter+jj)=Ynew(Pcenter-Pstart)+slope*(Pstart+jj);
  end
end

Ynew_dB=10*log10(Ynew/P_ref_sq);

% Update Noise Floor
Y_dB_flr_temp=filter(ones(1,ave_width)/ave_width,1,Ynew_dB);
Y_dB_flr=Y_dB_flr_temp(1+half_width:end);
Y_dB_flr(length(Ynew_dB)-half_width+1:length(Ynew_dB))= ...
Ynew_dB(length(Ynew_dB)-half_width+1:length(Ynew_dB));
clear Y_dB_flr_temp;

Y_flr = P_ref_sq*10.^(Y_dB_flr/10);

delf= Freq(2) - Freq(1);
num_TNr=0;

TNr_value = []; TNr_freq = []; Tone_power = [];
```

```
for ii=1:peak_count
  f0=Freq(peak_det(ii));                  % Center freq. of CB
  [fc,f1,f2]=getCB(f0);
  [fidx,fnum]=findfmax(f1,f2,peak_det,amp,Freq,ii,peak_count);

  s1=peak_det(ii);
  s2=s1;
  while (Freq(s1) > f1)&(s1 > 1), s1=s1-1;, end
  while (Freq(s2) < f2)&(s2 < num_spec(1)), s2=s2+1;, end
  s1=s1+1;
  s2=s2-1;

  ftot=(s2-s1+1)*delf;
  wtot=sum(Y(s1:s2));

  wtot=getpower(f1,f2,Y,Freq,num_spec(1),delf);

  if (Freq(peak_det(fidx(1)))==f0)
    wt=0;
    wrm=0;
    ft=0;
    for jj=1:fnum
      if (jj==1)
        [p11,p12]=getRange(peak_det(fidx(1)),Y,Y_flr,num_spec(1));
        ft=(p12-p11+1)*delf;
        wt=sum(Y(p11:p12));
      elseif (jj==2)
        [p1,p2]=getRange(peak_det(fidx(jj)),Y,Y_flr,num_spec(1));
        if (f0<1000)
          fd=abs(f0-Freq(peak_det(fidx(jj))));
          if (p1 == p11)
            wt = wt;
          elseif (fd<21*10^(1.2*abs(log10(f0/212))^1.8))
            wt=wt+sum(Y(p1:p2));
            ft=ft+(p2-p1+1)*delf;
          else
            wrm=wrm+sum(Y(p1:p2));
            ft=ft+(p2-p1+1)*delf;
          end
        else
          wt=wt+sum(Y(p1:p2));
          ft=ft+(p2-p1+1)*delf;
```

```
        end
      end
    end

    if (ft/fc > 0.15)
      msg=sprintf('Warning: Too wide tone width, %.1f Hz',f0);
      disp(msg);
      msg=sprintf('fc : %.1f, ft : %.1f\n',fc,ft);
      disp(msg);
    end

    if wt<0
      wt = eps;
    end

    % Hearing Threshold
    Lth=3.64*(f0/1000).^(-0.8)-6.5*exp(-0.6*((f0/1000)-...
    3.3).^2)+(10^(-3))*(f0/1000).^4;
    Ehs=10.^(Lth/10)*4e-10 /delf;
    % Frequency Weighitng
    FreqWeight = 10*log10((1+0.2*(f0./700+700./f0).^2).^(-0.29));
    wn=(wtot-wt-wrm)*fc/(ftot-ft);
    num_TNr=num_TNr+1;
    TNr_freq(num_TNr)=f0;
    % Limiting
    TNr_value(num_TNr) = 10*log10(wt/(wn+Ehs)) + FreqWeight;
    Tone_power(num_TNr) = 10*log10(wt*delf/4e-10);
  end
  [TNr_value,TNr_freq,Tone_power]=sortValue(TNr_value,TNr_freq,Tone_power,-1);
end

%*********************************************************************
%*********************************************************************
% function findpeak
%*********************************************************************
% subroutine for peak detection
%*********************************************************************

function [peak_det,amp,peak_count]= ...
findpeak(Y_dB,Y_dB_flr,Freq,peak_dist,peak_width,num_spec)
% Initialize local variables
num_count=0;
peak_count=0;
amp=0;
```

```
peak_det=0;
lowFreqCut=20;
lowC=0;
while (1)
  lowC=lowC+1;
  if (Freq(lowC) > lowFreqCut), break, end
end
for (count=lowC:num_spec-1)
  if (Y_dB(count)-Y_dB_flr(count)>=peak_dist)
    if (Y_dB(count)>Y_dB(count-1) & Y_dB(count)>Y_dB(count+1))
      num_count=num_count+1;
      peak_cand(num_count)=count;
    end
  end
end
if num_count == 0
  return
elseif num_count == 1
  peak_count=1;
  peak_det(peak_count)=peak_cand(1);
  amp(peak_count)=10^(Y_dB(peak_cand(1))/20)*2*10^(-5)*1.75;
  return
else
  for (dummy=1:num_count-1)
    if (Freq(peak_cand(dummy+1))-Freq(peak_cand(dummy))<peak_width)
      if (Y_dB(peak_cand(dummy+1))<Y_dB(peak_cand(dummy)))
        peak_count=peak_count+1;
        peak_det(peak_count)=peak_cand(dummy);
        amp(peak_count)=10^(Y_dB(peak_cand(dummy))/20)*2*10^(-5)*1.75;
      end
    else
      peak_count=peak_count+1;
      peak_det(peak_count)=peak_cand(dummy);
      amp(peak_count)=10^(Y_dB(peak_cand(dummy))/20)*2*10^(-5)*1.75;
    end
  end
end
if peak_count == 0
  peak_det = [];
  return
end
peak_count=peak_count+1;
peak_det(peak_count)=peak_cand(num_count);
amp(peak_count)=10^(Y_dB(peak_cand(num_count))/20)*2*10^(-5)*1.75;
```

```
return

%*******************************************************************
%*******************************************************************
% Subroutine findfmax
%*******************************************************************

function [fidx,fnum]=findfmax(f1,f2,peak_det,amp,Freq,ii,peak_count)
s1=ii;
s2=ii;
while (Freq(peak_det(s1)) > f1)&(s1 > 1), s1=s1-1;, end
while (Freq(peak_det(s2)) < f2)&(s2 < peak_count), s2=s2+1;, end
s1=s1+1;
s2=s2-1;
if (ii==1),s1=1;,end
if (ii==peak_count),s2=peak_count;,end
for (jj=s1:s2),fidx(jj-s1+1)=jj;,end
for ii=1:s2-s1
  for jj=1:s2-s1+1-ii
    if (amp(fidx(jj))<amp(fidx(jj+1)))
      temp_idx = fidx(jj);
      fidx(jj) = fidx(jj+1);
      fidx(jj+1) = temp_idx;
    end
  end
end
fnum=s2-s1+1;
return

%*******************************************************************
%*******************************************************************
% Subroutine getRange
%*******************************************************************

function [p1,p2]=getRange(f0,Y,Y_flr,num_spec)
p1=f0;
p2=f0;
while (1)
  if (Y_flr(p1) > Y(p1))|(p1==2), break, end
  p1=p1-1;
end
while (1)
  if (Y_flr(p2) > Y(p2))|(p2==num_spec-1), break, end
  p2=p2+1;
```

```
end
return

%*********************************************************************
%*********************************************************************
% Subroutine getpower
%*********************************************************************

function [power]=getpower(f1,f2,Y,Freq,num_spec,delf)
s1=min(ceil(f1/delf),length(Freq));
s2= min(ceil(f2/delf),length(Freq));
while (Freq(s1) < f1)&(s1 < num_spec), s1=s1+1;, end
while (Freq(s2) > f2)&(s2 > 1), s2=s2-1;, end
while (Freq(s1) > f1)&(s1 > 1), s1=s1-1;, end
while (Freq(s2) < f2)&(s2 < num_spec), s2=s2+1;, end
s1=s1+1;
s2=s2-1;
power=sum(Y(s1:s2));
return
```

B.2.2 Oppressive Penalty

This function in this example is called *Penalty_LF*. It was written by G. Song. It takes octave band spectral as input to compute the Oppressive Penalty.

```
function [Penalty,RegionNum] = Penalty_LF(LT)

Num = 17;
Penalty = 0;
RegionNum = zeros(Num,1);
LF_Threshold = [71.5000   81.0600   86.4500   86.4500   95.5400  101.4000
                61.6700   74.3900   80.6300   80.6300   90.7900   96.9000
                51.5000   67.5000   74.6300   74.6300   85.8800   92.2500
                44.4200   62.9900   71.3200   72.6500   85.0100   92.2100
                37.0800   58.3300   66.7400   71.7800   84.1100   92.1700
                29.5000   53.5000   62.0000   70.8800   83.1700   92.1300
                NaN   49.2800   62.0900   70.8700   82.7400   92.0800
                NaN   45.0600   62.1800   70.8600   82.3100   92.0300
                NaN   41.8000   62.1300   71.8400   83.3000   92.5700
                NaN   41.2900   61.8000   74.4400   84.0700   94.0500
                NaN   40.7700   61.4800   77.0400   84.8500   95.5400
                NaN   40.2300   61.1400   79.7400   85.6500   97.0900
                NaN   39.2000   60.1000   80.9000   86.3000   97.5500
```

```
                NaN   37.9300   58.6700   80.9000   86.7800   97.2300
                NaN   36.6000   57.1800   80.9000   87.2700   96.9000
                NaN   35.2400   55.6400   80.9000   87.7900   96.5600
                NaN   33.9600   54.2000   80.9000   88.2700   96.2400];

for i = 1:Num
  if i > 6
  level = 2;
else
  level = 1;
end
while level <= 6 & LT(i) > LF_Threshold(i,level)
  level = level + 1;
end
if level == 1
  RegionNum(i) = 0;
elseif level == 7
  RegionNum(i) = 6;
else
  exceed_level = (LT(i) - LF_Threshold(i,level-1))/(LF_Threshold(i,level) - LF_Threshold(i,level-1));
  RegionNum(i) = level - 1 + exceed_level;
end

if RegionNum(i) > Penalty
  Penalty = RegionNum(i);
end
end
```

A subroutine was written to compute one-third octave spectrum based on the power spectral density.

```
function [LT,F0] = myoct3(Sxx,ff,fs,varargin)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% Guochenhao Song
% May 2020
% This program calculates the ont-third octave spectrum of a signal, using Butterworth filters.
% Inputs:
%   Sxx  -    Narrow band spectrum of sound
%   ff   -    Frequency vector of PSD
%   fs   -    the sampling rate of the signal (Hz)
%   method -  this input specifies whether the filtering is applied in the
%             time domain or the frequency domain
%
% Outputs:
%   P  - Nth-octave spectrum of signal (dB)
```

```
%   F0 - vector of center frequencies of Nth-octave spectrum (Hz)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

pref = 2e-5; % Reference pressure (Pa)
filtord = 5;


F0=[25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400,...
500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000,...
5000, 6300, 8000, 10000, 12500, 16000].';

% Band-edge frequencies (cf. Christoph Couvrer's 'oct3dsgn' and
% ANSI S1.1-1986
f1 = F0/(2^(1/6));
f2 = F0*(2^(1/6));
Qr = F0./(f2-f1);
Qd = (pi/2/filtord)/(sin(pi/2/filtord))*Qr;
alpha = (1 + sqrt(1+4*Qd.^2))/2./Qd;
F1 = F0./alpha;
F2 = F0.*alpha;
P = noct_FFTbutter(Sxx,ff,filtord,F0,F1,F2,fs/(ff(2) - ff(1)),fs);
LT = 10*log10(P/pref^2); % dB
end

%********************************************************************
%********************************************************************
% Subroutine noct_FFTbutte
%********************************************************************
function P = noct_FFTbutter(Sxx,ff,filtord,F0,F1,F2,Nf,fs)
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% This subroutine generates a 12th-octave spectrum by implementing
% Butterworth filters in the frequency domain.
% Inputs:
%   Sxx   -   Power spectral density of signal (V^2/Hz)
%   ff    -   Frequency domain vector (Hz)
%   filtord - Filter order
%   F0    -   Filter center frequencies (Hz)
%   F1    -   Low band-edge frequencies (Hz)
%   F2    -   High band-edge frequencies (Hz)
%   Nf    -   Number of points in FFT
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
P = zeros(size(F0)); % Power vector (V^2)
for jj=1:length(F0)
  if F2(jj)>ff(end)
```

```
    P(jj) = eps;
    break
  end
  % Zeros and poles of filter
  [z,p,~]=butter(filtord,[F1(jj) F2(jj)]/(fs/2),'bandpass');

  % Gain of filter (MATLAB function has some errors)
  zz = exp(1i*2*pi*F0(jj)/fs);
  k = abs(prod(zz-p)/prod(zz-z));

  % Designate filter as 2nd-order system
  [sos,g]=zp2sos(z,p,k);

  % Filter coefficients of 2nd-order system
  n = size(sos,2);
  B = sos(:,1:n/2);
  A = sos(:,n/2+1:end);

  % Frequency resolution needed to get 10 points in filter pass-band
  %     Nf_new = 2^nextpow2(fs*10/(FL(jj+1)-FL(jj)));
  %     if length(Sxx)> Nf_new && Nf_new >= wlen
  %         Sxx = Sxx(1:2:end);
  %         ff = ff(1:2:end);
  %         Nf = Nf/2;
  %     end

  % Filter frequency response
  H = g;
  Z = exp(1i*2*pi*ff/fs);
  for ii=1:filtord
    H = H.*(B(ii,1)+B(ii,2)./Z+B(ii,3)./(Z.^2))./(A(ii,1)+A(ii,2)./Z+A(ii,3)./(Z.^2));
  end

  % PSD of filtered signal (V^2/Hz)
  Syy = Sxx.*abs(H).^2;

  % Integrate Syy over frequency domain to get power (Pa^2)
  P(jj) = fs/Nf*(Syy(1)/2+Syy(end)/2+sum(Syy(2:end-1)));
end
end
```

B.2.3 Other Codes used in Software

Function *ExtractBroadband* is also used in the software. It takes narrowband power spectral density as input. The function first identified the prominent tones in the spectrum, then extract a broadband power spectral density by removing tones.

```
function [Ynew] = ExtractBroadband(Y,Freq)
% Author:   Guochenhao Song
%
% Syntax:
% [Ynew] = ExtractBroadband(Y,Freq)
%
% Input Variables:
% Y, Freq           = PSD of the sound
%
% Output Variables:
% Y_new         = PSD of broadband sound

peak_dist=8;

%Power Spectrum in Decibels
P_ref_sq=(2*10^-5)^2;
Y_dB=10*log10(Y/P_ref_sq);

% Calculate the moving average values of PSD
ave_width=20;
half_width = round(ave_width/2);
num_spec = size(Freq);

Y_dB_flr_temp=filter(ones(1,ave_width)/ave_width,1,Y_dB);
Y_dB_flr=Y_dB_flr_temp(1+half_width:end);
Y_dB_flr(length(Y_dB)-half_width+1:length(Y_dB))= ...
Y_dB(length(Y_dB)-half_width+1:length(Y_dB));
clear Y_dB_flr_temp;

% Peak detection
peak_width = 3;
peak_count = 0;
[peak_det,amp,peak_count]=findpeak(Y_dB,Y_dB_flr,Freq,peak_dist, ...
peak_width,num_spec(1));

% Remove peaks to calculate updated noise floor
Ynew=Y;
for (ii=1:peak_count)
```

```
  Pstart=0;
  Pend=0;
  Pcenter=peak_det(ii);
  while (Ynew(Pcenter-Pstart-1)<Ynew(Pcenter-Pstart))
    Pstart=Pstart+1;
  end
  while (Ynew(Pcenter+Pend)>Ynew(Pcenter+Pend+1))
    Pend=Pend+1;
  end
  slope=(Ynew(Pcenter+Pend)-Ynew(Pcenter-Pstart))/(Pstart+Pend);

  for (jj=-Pstart:Pend)
    Ynew(Pcenter+jj)=Ynew(Pcenter-Pstart)+slope*(Pstart+jj);
  end
end
end
```

C. SUBJECTIVE TEST MATERIALS

C.1 Sample Test Checklist for Operator

Appendix C.1. Sample Checklist

Research Project: **Annoyance Due to Noise in Buildings**

Maximum number of subjects approved: ___200___

Current subject total: _________
This subject's number: ________
Task category: ______________

- ☐ **Single Subject**
  - Go to section A
- ☐ **Consecutive Subject**
  - Go to section B

A. Before Subject Arrival:

- ☐ Pick up sound level meter, light meter, microphone, and calibrator.
- ☐ Wash hands; make sure you have gloves available
- ☐ Turn on computer
- ☐ Turn on amplifier
- ☐ Check connections (to sound card, amplifier, loud speakers)
- ☐ Check the camera.
- ☐ Check amplifier settings
  - Speaker outputs on
  - Mode switch set to "Stereo"
  - Ground lift switch set to "Lift"
  - Amp:__________
- ☐ Check sound card settings
  - Open Lynx Mixer
  - Unlock Mixer
  - Play 01: __________ Play 02: __________
- ☐ Enter lab space where the office environment is set up
- ☐ Check the room and microphone configuration and mark the room configuration sheet.
  The sound level microphone is close to where a subject's head would be during the test.
  - Open the window blinds.
  - Check the loudspeaker at the right place.
  - Check the room configuration.
  - Check the microphone location.
- ☐ Check the temperature: __________ , check the lighting level: __________
- ☐ Calibrate sound level meter
  - Attach microphone to sound level meter and turn meter on
  - Place calibrator over microphone and turn it on
  - Using stylus (stored in the slot on the side of the meter), tap the menu button in the bottom left corner of the meter screen, and select 'Calibration' from the pop-up menu
  - Tap the 'Start calibration' button
  - When the dialog box pops up, check the numbers and tap 'Accept calibration' if the numbers look good (generally deviates by 0.03 dB or lower)
    - New sensitivity: __________mV/Pa

Initials _________
Date and Time _________

Appendix C.1. Sample Checklist

  - Deviation from last: __________dB
- When the screen displays the 'Exit calibration' button, tap it
- Remove and stow calibrator

☐ On the computer, Open program 'SubjTest'

☐ Click on calibration button, play calibration signal through loud speakers separately to calibrate playback.
- ***Important note:*** *the meter and earphone and coupler assembly is very sensitive to vibration or movement. When recording sounds, use a tripod to hold the meter at the listening location.*

  Broadband_calibration.wav
  Expected: __36.3 ± 1.0dBA__ Actual: ________
  Speaker 2:
  Expected: __50.0 ± 1.0dBA__ Actual: ________

☐ Sound card settings
- Lock Mixer
- Play 01: __________ Play 02: __________

☐ The instrumentation settings for this tone are the same as the settings for all the signals to be played in the test.

☐ Tick here if yes. If no, what are the new amplifier settings for playing the test signals? And confirm that they have been set.
New Settings: _________________________ Have Been Set? ☐

☐ Play 5 check signals through loud speaker. (These signals located in subdirectory "Signal check" under "LoudSpeakerStudy".)
- Press the Play/Pause button above the screen to begin recording, and to stop. Recorded levels will appear on the screen.
- Clear recordings by pressing the button to the left of Play/Pause.
- Play each signal, clearing the display after each signal.
- Record fast A-weighted levels below. Circle all that are acceptable:

Signal A :	Expected __36.3 ± 1.0dBA__ Observed ________
Signal B :	Expected __43.5 ± 1.0dBA__ Observed ________
Signal C :	Expected __44.1 ± 1 .0dBA__ Observed ________
Signal D :	Expected __46.6 ± 1.0dBA__ Observed ________
Signal E :	Expected __43.7 ± 1.0dBA__ Observed ________

☐ Maximize the software window

☐ Turn off sound level meter; return meter, and microphone to their cases; return light meter

☐ Wash hands

☐ Clean and wipe down testing area
- Mouse and Keyboard
- Desk or Tables
- Audiometer

Checklist, page 2/5 Initials _________
Date and Time _________

Appendix C.1. Sample Checklist

- □ Turn on audiometer, set up hearing test paper – green sheet
- □ Hang 'Do Not Enter, Testing In Progress' signs on entrances to the test area
- □ Get subject packet (consent form, questionnaire) and fill in the cover sheet, which includes a check on the number of subjects tested so far under this IRB test protocol (# 1811021317).
    - Check the subject number is on the consent form.

  Test stops here, if this subject would exceed the approved number for testing.

B. When subject arrives:

- □ Greet subject and give a brief test overview (outline major points of test procedure – green sheet)
- □ Obtain informed consent (Appendix B) – white sheet
    - Make sure participants initial each page and sign on the last page of consent form
    - Make sure researcher sign on the last page of consent form after getting participants' signature
- □ Have subject fill out the questionnaire (Appendix C) – pink sheet
- □ Put gloves on
- □ Test subject hearing – blue sheet
    - Explain how the hearing test works and what the subject should expect
    - Subject should be facing away from the audiometer so they can't see you working the machine
    - If HL > 25 dB, provide information on Audiology clinic □ (Test stops here)
    - Cards containing Audiology clinic info are located in drawer on top of audiometer
    - Mark the subject and close the software
- □ Give them a short rest for water and restroom and ask them to wait by the elevator.
- □ On the computer, select the directory: <u>U:\Song_Test\Song_Test_3</u>
- □ Load test file:
    - □ Song_Test_3_SessionA
    - □ Song_Test_3_SessionB
- □ Initialize the test run
    - Put in the subject number yourself
- □ Turn on the background noise.
- □ Go out to get the subject in.
- □ Ask subject to leave cell phone and other electronic devices in the control room.

C. During test:

- □ Wear gloves.
- □ Give subject test instructions (Appendix F) – orange sheet
- □ Explain how test will work: typically e.g., listening to 6 sounds; some practice rating with 6 sounds; and taking two session tests.
- First test session
    - □ Run Familiarization and Practice Session
        - Mention that they can have more control by clicking and dragging the slider
    - □ Check that subjects are comfortable, answer any questions
    - □ Tell them the test would be started after researcher's leaving.

 Initials _________
Date and Time _________

Appendix C.1. Sample Checklist

- ☐ Exit the test area and shut the doors carefully.
- ☐ Start the test.
- ☐ Enter the door after the test session.

☐ Give them a 5 minutes short rest for water and restroom

☐ Check data (subject's responses) under "View Report"

- ☐ Save report file under name "##Response_Session1"

- Second test session
  - ☐ Load test file:
    - ☐ Song_Test_3_SessionA
    - ☐ Song_Test_3_SessionB
  - ☐ Initialize the test run
  - ☐ Go out to get the subject in.
  - ☐ Ask subject to leave cell phone and other electronic devices in the control room.
  - ☐ Take work out of bag.
  - ☐ Check that subjects are comfortable, answer any questions
  - ☐ Tell them they can start their work, the test would be started after researcher's leaving.
  - ☐ Exit the test area and shut the doors carefully.
  - ☐ Start the test.
  - ☐ Enter the door after the test session.

1. **After Test:**

☐ Get general comments from the subject (part of Appendix C) – yellow sheet

☐ Ask subjects to fill a noise sensitivity form.

☐ Save test (click "Save")

☐ Check data (subject's responses) under "View Report"

- ☐ Save report file under name "##Response_Session2"

☐ Turn off the background noise.

☐ Retest hearing – If subject shows sign of threshold shift, provide info on Audiology clinic

☐ Escort subject to secretary in the HLAB administrative suite

☐ Pay subject

☐ Ask if subject would like a copy of the consent form

☐ Escort subject to the door

☐ Transfer report file to directory "Results"

☐ Place all subject material in an envelope, except this checklist.

☐ Pick up sound level meter, light meter, microphone, and calibrator.

☐ Check the temperature: __________ , check the lighting level: __________

☐ Calibrate sound level meter.

- ▪ See "Before Subject Arrival" for procedure

☐ Play calibration signal through loud speaker 1 to calibrate playback.

Expected: __36.3 ± 1.0dBA__ Actual: ________

☐ Play calibration signal through loud speaker 2 to calibrate playback.

Expected: __50.0 ± 1.0dBA__ Actual: ________

☐ Turn off sound level meter; return meter, and microphone to their cases; return light meter

 Initials _________

Date and Time _________

Appendix C.1. Sample Checklist

- Wipe down testing area again

2. If expecting another subject immediately afterwards: (write yes/no here: ________)

- Take a fresh copy of the subject package
- Look at the calibration results above and copy them into the “Before Subject Arrival” section of the new checklist
- Look at the temperature and lighting level above and copy them into the new checklist
- Put this checklist into the subject packet; set aside in cabinet in the control room
- Continue with the new checklist, check consecutive subject and start from Section B

3. If not expecting another subject immediately afterwards: (check here if so ·❑)

- Close program
- Power down audiometer
- Power down computer
- Power down amplifier
- Close the window blinds.
- Take down ‘Do Not Enter, Testing In Progress’ signs
- Deposit packet(s) in Prof. Davies’ office

Initials _________
Date and Time _________

C.2 Background Questionnaire and Post-Test Questions

QUESTIONNAIRE ON SUBJECT'S BACKGROUND

<table>
<tr><td colspan="2">NAME:
LEAVE BLANK, REFER TO CONSENT FORM WITH SUBJECT NUMBER, IF NEEDED</td><td>AGE:</td><td>SEX:</td><td colspan="2">ETHNIC OR RACIAL GROUP**:</td></tr>
<tr><td>SUBJECT NUMBER:</td><td colspan="2">TEST NAME:</td><td colspan="2">MASTER FILE NAME:</td><td>DATA FILE NAME</td></tr>
<tr><td colspan="3">DATE OF FIRST TEST (IN THIS SERIES)

DATE OF LAST TEST (IN THIS SERIES)</td><td colspan="3">NAMES OF OTHER TESTS SUBJECT HAS BEEN INVOLVED IN:</td></tr>
<tr><td>HEARING:

TEST DATE:</td><td colspan="2">☐ NORMAL
☐ ABNORMAL</td><td colspan="3">COMMENTS:</td></tr>
<tr><td>NOISE EXPOSURE:</td><td colspan="2">☐ worked in a noisy industry for __________years
Type of industry:
____
____
________________________</td><td colspan="3">☐ has been exposed to loud explosions

☐ is a regular firearms user</td></tr>
<tr><td colspan="2">SOUND EVALUATION, SOUND QUALITY AND NOISE CONTROL AWARENESS:</td><td colspan="2">☐ no awareness
☐ very little awareness
☐ some awareness
☐ moderately aware
☐ highly aware
☐ have taken noise control and/or acoustics courses</td><td colspan="2">☐ have been involved in noise or vibration control studies
☐ have been involved in tests to do with sound quality previously.
☐ have studied music and/or am involved in musical events/productions/activities
☐ regularly ☐ occasionally</td></tr>
</table>

** For IRB diversity reporting purposes (e.g. Hispanic; Latino; American Indian; Alaska Native; African American; Asian; White; Native Hawaiian; Pacific Islander; more than one race; other) You are under no obligation to supply this information, if you do not want to.

Post-Test Questions

Were instructions clear?

What type of work were you doing?

- ☐ Reading
- ☐ Writing
- ☐ Editing
- ☐ Others: ____________

How was your experience with the study?

Is there anything that you recommend we do differently?

Is there anything you want to say about the sounds?

May we contact you to ask you if you would like to participate in similar research in future? *If yes, please provide your email. It is OK to say No if you do not wish to be contacted, this is purely voluntary.*

C.3 Noise Sensitivity Form

Appendix C.3. Noise Sensitivity Questionaire
Project: Annoyance due to building noise Date ________

We would like your opinion concerning a variety of sounds. Please try to imagine the situation presented in each statement, and indicate the extent to which you agree or disagree with it. If you are unsure, please choose that option which comes closest to reflecting your opinion.

1. I find it hard to relax in a noisy environment
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

2. I need peace and quiet in order to do difficult work
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

3. In return for a quiet place to live, I would accept disadvantages
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

4. I am very sensitive to neighborhood noise
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

5. I find it hard to communicate while it is noisy
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

6. I have no problems doing routine work in a noisy environment
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

7. I become very agitated if I can hear someone talking while I am trying to fall asleep
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

8. When I am absorbed in a conversation I do not notice if it is noisy around me
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

9. I can fall asleep even when it is noisy
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

10. My performance is much worse in noisy places
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

11. Listening to loud music helps me relax after work
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

12. In a restaurant I have trouble concentrating on my conversation when people are talking loudly at other tables
O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

Appendix C.3. Noise Sensitivity Questionaire

13. When I am at home, I become accustomed to noise quickly
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

14. When people around me are noisy I have trouble getting my work done
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

15. I need an absolutely quiet environment in order to get a good night's sleep
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

16. Even the slightest noise can prevent me from falling asleep
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

17. I need quiet surroundings to be able to work on new tasks
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

18. It would not bother me to live on a noisy street
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

19. If I'm dancing I don't mind how loud the music is
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

20. If my workplace was noisy I would always try to find a way for me to change this
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

21. I find it very hard to follow a conversation when the radio is playing
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

22. I think music interferes with conversations
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

23. In the movie theater I am annoyed by other people whispering or rustling paper
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

24. When other people's children are noisy I would prefer that they not play in front of my house
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

25. On weekends I prefer quiet surroundings
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

26. I do not feel well rested if there has been a lot of noise the night before
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

27. The sound of loud thunder does not usually wake me up
   - Strongly Agree
   - Agree
   - Neutral
   - Disagree
   - Strongly Disagree

Appendix C.3. Noise Sensitivity Questionaire

28. Loud music in a restaurant makes me stop my conversation

O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

29. I can do complicated work even while background music is playing

O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

30. I wake up at the slightest noise

O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

31. I avoid leisure activities which are loud

O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

32. I don't like noisy activities in my residential area

O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

33. Noises from neighbors can be extremely disturbing

O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

34. When I am at home I find it uncomfortable if the radio or TV is left on in the background

O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree

35. High noise levels make it hard for me to concentrate on my conversation

O Strongly Agree O Agree O Neutral O Disagree O Strongly Disagree