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AUDITORY RESEARCH PLATFORM : TWENTY-FOUR HOUR IN-EAR DOSIMETRY

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RÉSUMÉ

La perte auditive induite par le bruit, est une lésion permanente et irréversible. Bien qu'elle soit évitable, par la mesure appropriée de la dose de bruit reçue et la limite de cette dernière, la perte auditive induite par le bruit reste l'une des maladies professionnelles les plus communes et un fléau touchant plusieurs centaines de millions d'individus dans les pays industrialisé. Elle représente également un enjeu de santé grandissant dans les milieux de vie modernes, notamment par le biais des activités de loisir bruyants, tels les sorties dans des concerts, bars ou discothèques, mais aussi pas le biais de l'utilisation d'outils motorisés bruyants ou encore par le biais de l'utilisation des appareils multimédia personnels. Les dosimètres acoustiques conventionnels, capables de mesurer la dose de bruit reçue par un individu donné, peuvent être utilisés, dans des approches de santé sécurité au travail (SST), afin de limiter les risques de pertes auditives induites par le bruit en milieu de travail. Cependant leur utilisation en conjonction avec des appareils multimédia personnels, des appareils de communication voire même de simples protecteurs auditifs, reste très difficile en pratique. De plus, ces dosimètres de bruit étant principalement adapté à des problématiques d'hygiène industrielle, ils ne sont pas conçus pour une mesure continue sur 24 heures de l'exposition au bruit : en SST, la législation actuelle se base sur un quart de travail de 8 heures, suivi d'un temps de récupération de 16 h dans un environnement « relativement calme », c'est à dire dont le niveau sonore n'excède pas 75 dB(A), ce qui n'est pas réaliste compte-tenu des activités de loisirs bruyantes citées précédemment auxquels des travailleurs sont régulièrement soumis. Il n'y a, dans la législation actuelle, aucune recommandation pour la prise en compte de cette exposition sonore récréative, ni pour la mesure en continue sur 24 heures de la dose de bruit reçue.

Aussi, pour limiter le fléau de la perte auditive professionnelle et récréative conviendrait-il de pouvoir mesurer d'une part l'exposition sonore effective d'un travailleur durant son quart de travail, et notamment en tenant compte de l'atténuation effective offerte par le port éventuel d'un protecteur auditif, mais d'autre part en mesurant également l'exposition sonore récréative induite par l'utilisation l'utilisation d'outils motorisés bruyants ou encore par le biais de l'utilisation des appareils multimédia personnels.

Ce projet de maîtrise a pour but le développement d'outils permettant la mise au point d'une telle méthode de mesure continue sur 24 heures de la dose de bruit, et compatibles tant avec la mesure de l'exposition professionnelle, incluant l'effet du port d'un protecteur auditif, qu'avec l'exposition récréatives, incluant la compatibilité avec l'utilisation des appareils multimédia personnels. À cette fin, un système embarqué, baptisé « Auditory Research Platform » (ARP), a été développé et consiste en un bouchon d'oreille sur mesure équipé de 2 microphones et 1 haut-parleur et raccordé, par fils, à un processeur numérique du signal (DSP) lequel est logé dans un boitier compact se portant à la ceinture et contrôlable, via

Bluetooth, par un téléphone ou une tablette fonctionnant sous Android. Le système ARP développé peut ainsi être utilisé pour la mesure de l'exposition sonore professionnelle, puisqu'il agit à la fois comme protecteurs auditifs sur mesure et également comme dosimètre intégrateur, mais il peut aussi mesurer l'exposition sonore induite par l'utilisation des appareils multimédia personnels, puisqu'il est précisément hébergé sur un tel système. Par ailleurs, une boite à outil Matlab intitulée « *Dosimetry Toolbox* » a été développée afin de permettre, en ayant recours à une application tierce, la mesure précise de la dose de bruit mais afin de permettre également la visualisation aisée des données personnalisés d'exposition au bruit. À titre illustratif, un algorithme de mesure continue sur 24 heures de la dose de bruit incluant un modèle simple de récupération de la fatigue auditive lors des périodes de non exposition a également été développé. Les sources logicielles de l'outils « Dosimetry Toolbox », de même que les spécifications matérielles de la plate-forme ARP sont distribués en libre-accès (« Open Source »), afin de permettre à la communauté scientifique de poursuivre ce travail sur la détermination d'un critère de risque pour la dosimétrie continue sur 24 heures mais aussi pour l'étude des mécanismes plus fondamentaux sous-jacents au problème de la perte auditive induite par le bruit, tels que la détermination du taux effectif et individualisé de récupération de la fatigue auditive, la détermination du niveau sonore de silence effectif, la prise en compte de la contribution sonore de la voix du porteur dans la dose voire même la détermination de la susceptibilité individuelle à la perte auditive induite par le bruit.

Mots-clés: santé et sécurité au travail, acoustique, bruit, exposition sonore, dosimetrie, sonométrie

AUDITORY RESEARCH PLATFORM: TWENTY FOUR HOUR IN EAR DOSIMETRY

Jakub MAZUR

ABSTRACT

Noise induced hearing loss (NIHL) is permanent and irreversible, yet completely preventable. It remains to be one of the most common health and safety diseases in the workplace and is an increasing concern in daily urban life and recreational activities, such as live music, bars & nightclubs, and even personal media players (PMPs). Conventional ‘personal’ noise dosimeters do not easily interface with PMPs, communication devices, and HPDs, nor are they meant for 24-hour use. Not only is it quite difficult to keep track of noise exposure accumulated over a 24-hour period; there are few guidelines to follow. Current legislation is designed for 8-hour work-shifts and based on a 16-hour recovery period in a ‘relative quite’ environment (<75 dB(A)). In an effort to increase hearing protection compliance in dangerously noisy environments two common issues are addressed: 24-hour tracking of effective individual noise exposure level, including communication or PMPs, and tracking proper fit of hearing protection devices (HPDs).

The objective of this thesis is assessing the feasibility of designing a device capable of monitoring comprehensive 24-hour in-ear noise dosimetry, while interfacing with personal music players (PMPs) or communication devices. This thesis also proposes a novel algorithm for 24-hour (personal/individual/in-ear) noise dosimetry including auditory recovery while providing an open-source framework of hardware and software tools to encourage further development in this area. A custom in-ear open-source hardware ‘Auditory Research Platform’ (ARP) and an open-source Matlab ‘Dosimetry Toolbox’ have been developed for this purpose. The ARP hardware is an embedded digital signal processing system contained within a palm-sized belt-pack ready to interface with PMP, communication devices, and custom HPDs, all controllable with an Android device. The Matlab Dosimetry Toolbox is compatible with a third party iOS application and is intended to facilitate the creation of a personal noise exposure database while visualizing the results.

This Master degree thesis discusses the development of the hardware and software while identifying instrumentation challenges and highlighting several key research questions related to the risk assessment of NIHL. Preliminary laboratory studies are presented and the real-life usability of such a platform is discussed. The results of this work can be used to enable further studies revisiting damage risk criteria and further research into the underlying mechanisms of NIHL such as: recovery rate, effective silence, own voice contribution, and ultimately individual susceptibility.

Keywords: dosimetry, noise-induced hearing loss, hearing protection device

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LIST OF ABREVIATIONS

ARP	Auditory Research Platform
ATS	Asymptotic Threshold Shift
BTE	Behind-the-Ear
dB	Decibel
DSP	Digital Signal Processor
EPA	Environmental Protection Agency
GPIO	General-Purpose-Input-Output
HATS	Head and Torso Simulator
HPD	Hearing Protection Device
HTD	Head and Torso Diffraction
IDE	Integrated Development Environment
IEM	Inner-Ear-Microphone
IL	Insertion Loss
MIRE	Microphone in Real Ear
MSHA	Mine Safety and Health Administration
NHCA	National Hearing Conservation Association
NIHL	Noise Induced Hearing Loss
NIOSH	National Institute for Occupational Safety and Health
NIPTS	Noise-Induced Permanent Threshold Shift
NITTS	Noise-Induced Temporary Threshold Shift
NRR	Noise Reduction Rating

OEM	Outer-Ear-Microphone
OER	Occluded Ear Canal Resonance
OHC	Outer Hair Cells
OSHA	Occupational Health and Safety Administration
PCB	Printed Circuit Board
PMP	Personal Media Players
PND	Psychoacoustic Noise Dosimeter
REAT	Real-Ear-at-Threshold
SDIO	Serial-Data-Input-Output
SPI	Serial-Peripheral-Interface
SPL	Sound Pressure Level
TF	Transfer Function
TFOE	Transfer Function of the Open Ear
TTS	Temporary Threshold Shift
TWA	Time Weighted Average

LIST OF SYMBOLS AND UNITS OF MEASUREMENTS

i	Increment in discrete time
K_r	Maximum auditory recovery rate
$L_{Aeq(IEM)}$	Estimated free-field equivalent level based on actual measured in-ear levels
$L_{Aeq(NR)}$	Estimated free-field equivalent level based on NR of well-fitted earplug
L_{Aeqi}	A-weighted equivalent level for the *ith* interval
L_c	Criterion level
L_p	Free-field sound pressure level
L_{pA}	A-weighted free-field sound pressure level
L_{pFF}	Free-field sound pressure level
L_{pIEM}	SPL measured inside the occluded ear canal using the IEM
L_{pOEM}	SPL measured outside of the user's HPD near the entrance of the ear canal
L_s	Effective silence level
L_t	Threshold level
N	Total number of discrete increments
$NR_{(FIELD)}$	Noise reduction estimated in situ
q	Normalized exchange rate
q_r	Normalized recovery exchange rate
T_c	Criterion time
t_i	Duration of the i increment

INTRODUCTION

Context

There is a fine line between noise and music; a trip to the local karaoke bar can serve as a quick reminder. Although the definition of noise varies based on personal taste, from an acoustic perspective, music or noise, recreational or occupational, is just pressure fluctuations about the mean atmospheric pressure. From a health and safety perspective, both can be equally harmful.

Noise induced hearing loss (NIHL) is the result of overexposure to noise (or music) as a function of: sound pressure level and frequency distribution, duration and temporal spacing, as well as an individual's own physiological susceptibility (Johnson et al. 2010). NIHL is permanent and irreversible, yet completely preventable (NIOSH 2001). It remains one of the most common health and safety diseases in the workplace and is an increasing concern in daily urban life and modern recreational activities, such as attending live music, bars & nightclubs (Santos et al. 2007) and even listening to personal music players (Portnuff & Fligor 2011).

Personal hearing protection devices (HPDs) are often the best defense against NIHL; however, their effectiveness is contingent upon two variables, the noise exposure level and the attenuation of the HPD. The noise exposure level varies based on environmental factors, while the attenuation of the HPD is in itself a function of HPD model, proper fit and the percentage of time it is worn. A proper HPD provides adequate protection by affording sufficient attenuation while avoiding overprotection. Overprotection hinders perception of speech and warning signals, which may lead the user to remove the HPD, in turn drastically reducing the effective attenuation of the HPD. Thus, in order to select a proper HPD, two important metrics are required, the attenuation of the HPD and individual exposure levels of the user (Voix & Laville 2005). Although methods for measuring the real-world attenuation of HPDs on individuals have been examined for over 40 years, their laboratory performance remains notoriously misrepresentative of actual field attenuation.

Background and Need

The popularity of personal media players (PMP) is hard to ignore, in recent years headphones have outsold loudspeakers in the consumer market (Levin Consulting, personal communication, 2011). The ability of these devices to cause permanent hearing damage has been well documented in literature, even leading to recently updated technical safety standards in the European Union, requiring devices to default to safe listening levels and warn the user if the level is estimated to exceed 85 dB(A) (EU, 2013). Although listening habits are highly inter-individual and hard to accurately objectively assess, research suggests that many individuals are at risk (Levey et al., 2011). The growing concern is that the combination of PMP use with loud recreational activities or occupations can amount to hazardous exposure levels.

The purpose of a noise dosimeter is to track and inform the user of their individual noise exposure, hopefully allowing corrective action to take place before incurring harmful consequences. Unfortunately, conventional 'personal' noise dosimeters do not easily interface with HPDs, PMPs, and communication devices (Portnuff et al., 2012), nor are they meant for 24-hour use. Not only is it quite difficult to keep track of noise exposure accumulated over a 24-hour period, there are few guidelines to follow. Current legislation is designed for 8-hour work-shifts and based on a 16-hour recovery period in a 'relatively quiet' environment (CSA 2006).

Recent research revisiting noise exposure and associated physiological effects, through the assessment of temporary-threshold-shifts (TTS), emphasizes that the current damage risk criteria used in noise dosimetry does not adequately represent the associated risk. In order to accurately assess the potential damage, the spectral and temporal characteristics of the noise exposure must be taken into account; yet current metrics fail to do this (Kostek et al. 2012; Strasser et al. 2008).

Objectives

The general objective of this thesis is assessing the feasibility of designing a device capable of monitoring comprehensive 24-hour in-ear noise dosimetry, while interfacing with personal music players (PMPs) or communication devices. The specific objectives of this thesis are:

- Designing and prototyping an open-source development platform (hardware & software) capable of collecting noise dosimetry field data and testing dosimetry algorithms;
- Adapting a noise dosimetry algorithm for 24-hour use, including auditory recovery;
- Designing a Matlab toolbox to analyze field-data, calculate standard noise dosimetry and test the proposed 24-hour dosimetry algorithm.

While the objectives are quite ambitious, the goal of the thesis regarding hardware development is to demonstrate the feasibility of developing such a device and carry out initial testing of the platform. The contributions described within this thesis consist of:

Technical Contributions

- An open-source hardware Auditory Research Platform (ARP);
- An open-source toolbox for Matlab;
- Proposed dosimetry algorithm for 24-hour use;
- Technical user manual for the ARP (CRITIAS 2015);
- Poster entitled "Noise Dose" (see APPENDIX I, p. 78).

Scientific Publications

- Paper entitled, "Implementing 24-hour in-ear dosimetry with recovery", presented at 2013 International Congress on Acoustics and published in the proceedings (see APPENDIX II, p. 80);
- Abstract entitled, "Development of an individual dosimetric hearing protection device", presented at 2012 Inter-Noise conference (see APPENDIX III, p. 79);
- Poster entitled "Smart Dosimetric Hearing Protection Device", presented at 2011 National Hearing Conservation Association (NHCA) conference (see APPENDIX IV, p. 77).

Patent Applications

- "Advanced communication earpiece device and method" (Voix et al. 2014).

Outline

Chapter 1 begins with a literature review including measurement of individual HPD attenuation, measurement of individual noise exposure, traditional dosimetry techniques, a brief history of legislation and standards, hearing science perspective, revisiting recent temporary-threshold-shift studies and an overview of a psychoacoustic noise dosimeter model. Chapter 2 describes the methodology for the development of the hardware, algorithm and Matlab toolbox, including the in-ear electroacoustic platform, correction factors and signal processing algorithms, and details of the electronic hardware platform and Matlab toolbox. Chapter 3 describes the results of the thesis, the status of the hardware and a demonstration of the dosimetry algorithm using the Matlab toolbox. Finally, a conclusion is presented followed by recommendations for future work.

CHAPTER 1

LITERATURE REVIEW

Before exploring traditional dosimetry equipment (Section 1.3) this chapter begins by summarizing two underlying topics: measurement of HPD attenuation (Section 1.1) and measurement of individual noise exposure (Section 1.2). This is followed by an introduction to the noise exposure metrics and limits currently used in legislation and standards (Section 1.4), and a brief perspective from hearing science regarding the effects of noise exposure on the auditory system (Section 1.5). A series of experiments examining an auditory side-effect to noise exposure, the temporary-threshold-shift (Section 1.6) is highlighted, leading up to the summary of a recently published novel approach to noise dosimetry: the psychoacoustic noise dosimeter (Section 1.7). Finally, a short summary is presented, tying these topics back into the objectives of this thesis (Section 1.8).

1.1 Measurement of individual HPD attenuation

All HPDs sold in the U.S. are required to be labeled with a Noise Reduction Rating (NRR). This is a single-number estimation of the amount of attenuation that a particular HPD provides (Johnson et al. 2010). Unfortunately, this NRR value is notoriously misrepresentative of actual field attenuation. Attempts to identify this relationship began in the 70's and comprehensive reviews are available (Berger et al. 1996; Gaudreau et al. 2008). Originally, measurement of HPD attenuation was performed on a group of subjects after being fit by a trained experimenter or "supervisor fit". More recent standards include a "trained-subject fit" method, where a trained experimenter instructs the subjects on proper HPD insertion techniques, and even an "inexperienced-subject fit" (ANSI S12.6 2008), where no guided instructions are provided. Discrepancies between laboratory and field attenuation values remain and after over 40 years, it has been well established that the NRR ratings exceeds noise reduction that is experienced in the field (Voix & Laville 2005). The Occupational Health and Safety Administration (OSHA) has implemented the practice of derating the NRR measurement by 50% for all types of hearing protectors. The National

Institute for Occupational Safety and Health (NIOSH) recommends scaling the NRR with a predefined amount 25%, 50% or 70%, depending on the HPD type (Johnson et al. 2010). As Voix and Laville have said “Even if such methods better predict the average group field performance, it is still impossible to relate this population-based, statistically-derived, laboratory-driven attenuation estimate to real individual field attenuation” (Voix & Laville 2009). One solution to this fundamental problem, as proposed by Berger, is to perform ‘individual fit testing’. This requires measuring the real-world attenuation performance of HPDs on individual workers. Several methods to obtain these measurements exist and fall into two categories, psychophysical and objective (Casali et al. 1995).

The Real-Ear-at-Threshold (REAT) method, often referred to as the ‘gold standard’, is a psychophysical measurement involving the difference between the occluded (with HPD) and the unoccluded (without HPD) detection thresholds of a subject. The average REAT value, over a group of subjects, can then be used to generate an NRR. For labeled NRRs, the US Environmental Protection Agency (EPA) mandates a correction factory subtracting two standard deviations from the mean value, in order to estimate the minimum noise reduction theoretically achieved by 98% of the laboratory subjects (Berger et al. 1996; Casali et al. 1995). Although REAT takes into account all relevant sound paths to the inner ear, not only is it time-consuming and very sensitive to the ambient background noise, making it often impractical for field usage, but it is also hampered by two limitations: the first one is the well-known low-frequency masking error caused by physiological noise, which leads to an overestimation of the low-frequency attenuation, and the second one is the variability of determination of hearing threshold levels (Voix & Laville 2009).

The Microphone in Real Ear (MIRE) method is an objective measurement involving the placement of a measurement microphone directly in the ear canal of the subject and performing a measurement with and without the HPD. The difference between these two measurements is termed insertion loss (IL), expressed in dB, and can be used to predict an individual based REAT equivalent value (Casali et al. 1995). Originally, this measurement procedure was complex, required delicate laboratory equipment and fitting of the subject by a professional. Recently, a modified MIRE method, dubbed field-MIRE (F-MIRE), using a probe tube and two microphones was developed by Voix and Laville (Voix 2006; Voix & Laville 2009). This method although originally developed for custom HPDs, has been refined and confirmed by Berger et al. (Berger 2005) for non-custom earplugs and even adapted for ear muffs (Nelisse et al. 2011). Considering the trade-offs between speed, accuracy, repeatability and practicality, F-MIRE appears to be the most effective means to conduct field HPD attenuation measurements.

This idea of ‘individual fit testing’ has spawned the recent commercialization of many fit verification tools: VeriPro by Sperian Protection, Sonopass by Sonomax, SafetyMeter by Phonak, Multi-Fit by NIOSH, HPD Well-Fit by NIOSH, QuickFit by NIOSH, INTEGRAFit by Workplace Integra, and EARFit by 3M (Voix 2010). Some of these tools use psychophysical methods and some the objective method to enable a one-time confirmation of HPD attenuation.

1.2 Measurement of individual noise exposure

The tracking of individual noise exposure is challenging as it requires the knowledge of the HPD attenuation and the monitoring of environmental sound pressure levels (SPLs), or the direct measurement of the effective (protected) levels, i.e. behind the HPD. Traditionally, dosimeters are used in tracking noise exposure; they are body-worn derivatives of integrating sound level meters and are usually worn on top of the shoulder (Berger 2003). Although they provide an effective measure of unprotected exposure levels, they cannot account for HPD use or proper fit. It is well understood that the performance of a HPD is directly affected by how well it is worn and what percentage of time it is worn for. As seen in Figure 1.1, removing a HPD for even a short amount of time, drastically reduces its effectiveness.

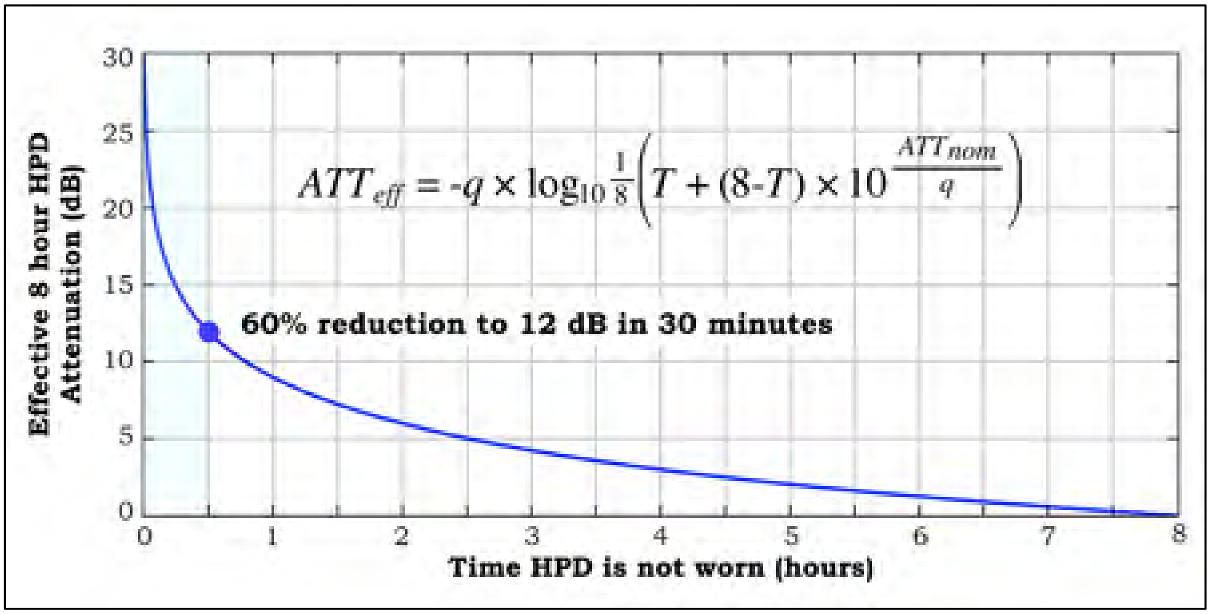

Figure 1.1 Effective 8-hour HPD attenuation vs. time HPD is not worn, during 8-hour shift, using 3 dB exchange rate. Where q is the normalized exchange rate, T is time HPD is not worn, ATT_{nom} is the nominal attenuation of the HPD and ATT_{eff} is the 8-hour effective attenuation. The highlighted point on the graph indicates an 8-hour effective attenuation of 12 dB after HPD removal of 30 minutes

The example HPD has a maximum attenuation of 30 dB, but when removed for just 30 minutes of an 8-hour work shift, its effective attenuation more than halves to just 12 dB (Voix & Laville 2005). Traditionally, questionnaires are used in assessing the percentage of time a HPD is worn, although recent research has shown that reported vs. measured results are victim to large subject reporting bias, especially in environments with unstable noise levels. In a study of HPD use among construction workers, Neitzel and Seixas found that workers who reported “always” using HPDs on a questionnaire, actually only wore them about 1/3 of the time [33% (±43)] when assessed via dosimetry, while being exposed above levels of 85 dB(A). Furthermore, combining the HPD attenuation levels with use time data dropped the effective HPD performance to less than 3 dB (Neitzel & Seixas 2005).

Noise exposure legislation is based on an 8-hour work shift, and 16-hour recovery period, usualy below 75 dB(A). For example if a worker’s off-duty quarters are noisy, this time is to be considered part of the work-shift (CSA Z107.56 2013). Currently, dosimeters are required to be manually reset back to a 0% dose which brings into question whether a subject has recovered back to 0% dose upon returning to work, either from exposure acquired during their previous work shift, or from recreational activities.

Recently, results of an F-MIRE-based study that measured the effective attenuation (i.e. behind the HPD) obtained over entire industrial work shifts show that it was indeed again lower than the labeled NRR. Furthermore, they saw variations in individual effective protected levels as a function of time, substantial left and right ear differences and significant inter-subject variability for workers wearing the same type of HPDs. These research findings further bring into question the effectiveness of HPDs in real-world environments and confirm the advantages of new instrumentation practices (Nelisse et al. 2011).

1.3 Traditional dosimetry equipment

Modern dosimeters are battery powered, body-worn derivatives of integrating sound level meters storing equivalent A-weighted levels and computing and presenting exposure levels in several fashions. The dosimeter is configured with the chosen damage risk criteria (Criterion Level, Criterion Time, Exchange Rate) and is meant for a subject to quickly assess their 8-hour 'noise dose'. As a convenient personal indicator this dose is often represented as a simple percentage, hiding the variables behind the calculation. For documentation purposes or comparisons L_{ex} is used, which is the L_{eq} adjusted for a specific time period, such as an 8-hour work shift (CSA Z107.56 2013). In the U.S., dosimeters are required to comply with the ANSI standards ANSI S1.43 and S1.25. The classification of these measurement systems falls into three categories, Type 0, 1 and 2, differing only in the tolerance limits allowed. As a rough guide, measurements with a Type 1 dosimeter will have errors not exceeding 1 dB, while Type 2 will not exceed 2 dB (American National Standards Institute Inc. 2007; American National Standards Institute Inc. 1991).

Dosimeters are traditionally worn on top of the shoulder, mid-way between shoulder and neck, with microphone pointing vertically upward. Although a study of the effects of body-worn dosimeter microphone placement concluded that in a diffused field the overall dB(A) errors were minor, in a direct field there were significant variations affected by both the position and supporting structure of the microphone and the localization of noise source and its spectrum. One of the suggestions was the use of a binaural system (Byrne & Reeves 2008). Although, dosimeters can currently provide an effective measure of unprotected noise exposure levels (in non-impulse dominant environments) (Kardous et al. 2005), they are still relatively simple devices in individual noise exposure assessment, as they cannot account for HPD attenuation or proper fit, nor do they take into account recovery of the human auditory system.

Recently, an untraditional in-ear dosimetry system has been commercialized, the Exposure Smart Protector™ by a startup company doseBusters™ which has now been acquired by Sperian Protection. It consists of a miniature microphone with a generic ear-tip adapter and is meant to insert into compatible HPDs. When the subject is wearing an HPD, the level measured by the dosimeter is the protected level and when the subject removes the HPD, the microphone continues to measure the level of exposure (unprotected) as a conventional dosimeter would (Byrne & Reeves 2008). This is meant to give the actual exposure of the subject, taking into account the performance of the protector as well as proper fit. Unfortunately, this system appears to have two main limitations. Firstly it does not provide any real-time data, which would allow the user to take corrective action before being over-exposed. Secondly, it does not provide insight as to why a particular subject is over his dose, e.g. was it an improperly fitted HPD, did they remove the HPD, or did the user simply spend too much time in an area that was too loud for the given device.

1.4 Legislation and standards history

One of the earliest references (more recently translated and re-published) to occupational NIHL dates back to 1713, describing noise-induced hearing loss and tinnitus (Ramazzini et al. 1964). In the 1800’s there were documentations of blacksmiths with hearing loss. By 1933, Temkin reported that after 15 years of work in metal industry, none of the men could hear a whisper 4m away. In 1944, McCoy published a study containing actual audiograms of shipyard workers. Subjects showed flat audiograms before starting employment, and extremely elevated threshold levels, 50 dB and more (with the highest notch at 4 kHz), after 7-hours of work. By 1950, Karl Kryter published an extensive report entitled "The Effects of Noise on Man", where he recommended a maximum safe level of 85 dB, for any frequency or narrow band of noise (Kryter 1950).

At this time overall SPLs were expressed in decibels (dB). Although it was understood quite early on that the frequency characteristics of noise should also be considered and octave band data was desirable, due to instrumentation limitations, this was simply not manageable. Throughout the 50's and 60's various organizations discussed this issue, but it was not until the Intersociety Committee in 1967 that the proposal of A-weighted sound levels became common practice. A-weighting filters low frequency content to better approximate the sensitivity of the human auditory system and remains in general use in hearing damage risk criteria to this day (Johnson et al. 2010).

In 1949 the first noise exposure regulation was issued, when the U.S. Airforce specified limits for noise exposure and required the provision of HPDs and audiometric testing for personnel exposed to high noise levels. The U.S. Air Force was also the first to introduce the equal energy principle in its regulation on Hazardous Noise Exposure in 1956 as part of the Armed Forces/National Research Council Committee on Hearing and Bioacoustics (CHABA). The equal energy principle states that the permissible amount of time in loud environments is halved for every 3 dB increase in level. This *exchange rate* is intended to quickly allow for calculation of how much time one can safely spend in a given environment and stayed in debate for many years following the CHABA regulation. The 5 dB exchange rate attempts to account for the interruptions in noise exposure, presuming that some recovery occurs during these interruptions and was originally argued to be simpler to use in the field. In 1982 many leading investigators reviewed the available literature regarding the use of the equal energy rule and endorsed the 3 dB exchange rate as the most practical and reasonable method, this served as the basis for the international standard, ISO 1999:1990.

Today, the 3 dB exchange rate is used by NIOSH, the American Conference of Governmental Industrial Hygienists (ACGIH), is adopted by most Canadian provinces and the Canadian federal government (CSA Z107.56 2013), yet some legislation still uses the controversial 5 dB exchange rate.

In 1974, the U.S. Environmental Protection Agency (EPA) published a document entitled "levels" in which it was clearly stated:

> Decisions about how much noise is too much noise for whom, for how long, and under what conditions demand consideration of economic, political and technological matters far beyond the intent of the Levels Document (EPA 1974).

The EPA established an 8-hour level of 75 dB(A) to protect "public health and welfare with an adequate margin of safety". Much of this result was based on the work of Johnson, with the use of 4 kHz as the most sensitive indicator of hearing loss. Even though it was acknowledged that activities outside of the workplace, such as mowing the lawn or recreational activities such as listening to records could contribute to ones daily exposure the recommended exposure limit (REL) is designed for an 8-hour workday, five days per week over a 40-year working lifetime, while the other 16-hours in the day as well as weekends are assumed to be quiet.

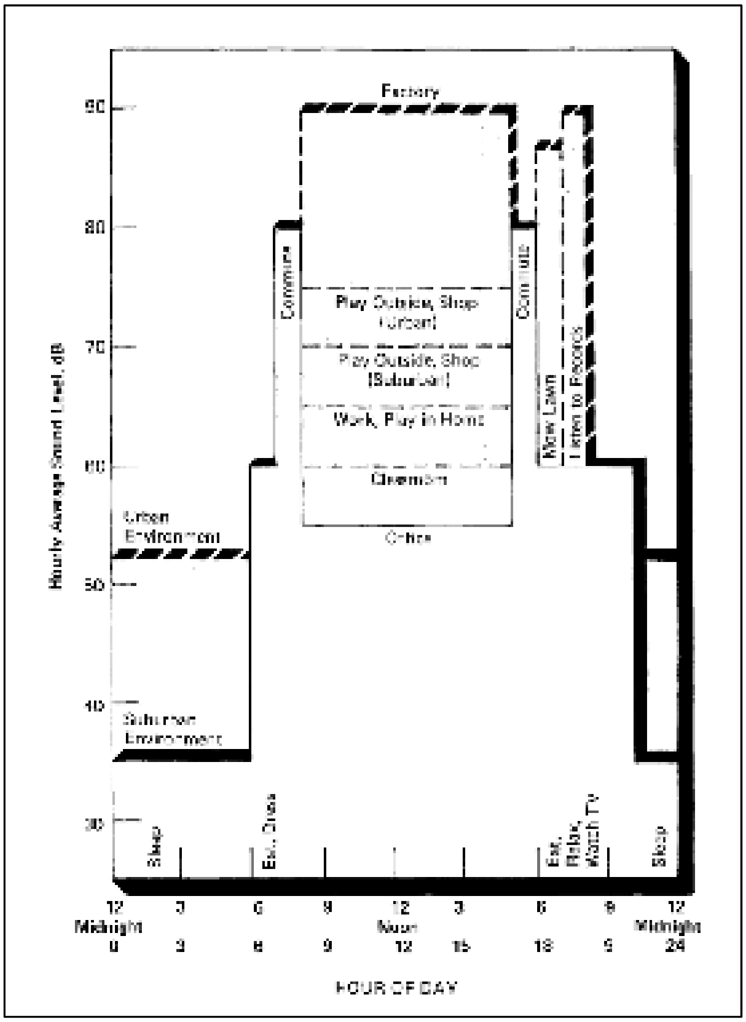

Figure 1.2 A 24-hour noise-exposure estimate graph from the "Condensed Version of EPA Levels Document"
Taken from EPA (1974, p.16)

Traditionally, noise exposure measurements are to be performed using a sound level meter at the 'center of head' location with the subject absent, although this is not universally agreed upon and the 'hearing zone', a hypothetical sphere of 30-cm radius enclosing the head, is also commonly used. It is expected that the measurement location used be noted.

Although the Occupational Safety and Health Administration (OSHA) and the National Institute for Occupational Safety and Health (NIOSH) were both put into place when President Nixon and Congress signed the Occupational Safety and Health Act of 1970, the two entities have greatly different roles. The OSHA is part of the U.S. Department of Labor and is responsible for developing and enforcing workplace safety and health regulations. It carries behind it the force of law and employers in the industrial sector are bound to comply with it. NIOSH conducts research and provides information, education, training, and recommendations regarding occupational safety and health, but is not in a position to regulate or enforce them. As such, the guidelines set forth by OSHA and NIOSH differ and are in constant debate. OSHA has a REL, which is the *Criterion Level* for the *Criterion Time,* of 90 dB(A) for 8-hours with an exchange rate of 5 dB, while NIOSH has a REL of 85 dB(A) for 8-hours with a 3 dB exchange rate. These values are based on each organization's perspective on 'excess risk'. In 1998, NIOSH presented its revised thoughts on occupational hearing loss where the excess risk of developing occupational NIHL for a 40-year lifetime exposure at the 85 dB(A) REL was 8%, while at 90 dB(A), the permissible exposure limit currently enforced by OSHA and the Mine Safety and Health Administration (MSHA) carried a 25% excess risk (Niquette 2009).

In Canada, provincial and federal legislation set limits. Most Canadian provinces use a criterion level of 85 dB(A) with a 3 dB exchange rate (as the NIOSH recommendation) with a few exceptions, the Canadian federal noise regulation uses 87 dB(A) with a 3 dB exchange rate, while Quebec uses 90 dB(A) with an exchange rate of 5 dB, the same as OSHA (CCOSH 2015).

The effects of noise on the human auditory system have been observed dating back as far as the 1700's. Previous instrumentation and data collection limitations, combined with high individual susceptibility, and the progressive emergence of NIHL, posed challenges in establishing direct cause and effect relationships. However, even with scientific data highly supporting that overexposure to noise leads to hearing loss, the many political aspects involved in the development of noise regulations remain.

1.5 Hearing science perspective

The study of noise induced hearing loss (NIHL) is challenging not only due to the difficulty in tracking of HPD attenuation and individual noise exposure but also because of the extreme complexity of the human auditory system. Individual susceptibility is highly variable and driven by various intrinsic biological factors of which the governing mechanisms are not entirely understood. It is beyond the scope of this thesis to describe what is known about these processes in great detail and as such the discussion that follows is designed to acquaint the reader with the basics, while detailed information can be found in Feuerstein & Chasin 2009; Kryter 1950; Dancer et al. 1992; Johnson et al. 2010. Although several methods for evaluating auditory health are in use today, pure tone audiometry remains the primary tool for NIHL assessment.

Unfortunately audiometry generally documents NIHL after damage has already occurred, at best allowing for prevention of further damage rather than prevention of damage which may not even be present on the audiogram yet. The three types of hearing changes that may occur following excessive noise exposure are noise-induced temporary threshold shift (NITTS), noise-induced permanent threshold shift (NIPTS) and acoustic trauma. NIHL is generally insidious in nature, occurring gradually and painlessly over time, making it difficult to notice before it is too late. Although, cases of acoustic trauma involving extremely intense noise, such as an explosion, can result in immediate permanent cochlear damage, rupturing of the eardrum and/or fracture of the ossicular chain.

As the name indicates, NITTS is the temporary reduction in hearing ability and is evident by elevated audiometric hearing thresholds. Studies have shown that the damage from broadband signals such as industrial noise and music, typically occurs in the octave band above the noise. Given the external ear's natural resonance in the range of 2-3 kHz, this is consistent with the classic audiometric "noise-induced notch" at 4 kHz, often exhibited by patients with NIHL. The amount and duration of NITTS is a function of the duration and intensity of the noise exposure. The symptoms can be relatively brief, or may extend up to several hours or days and range from a slight reduction in hearing sensitivity to as much as 40 dB. During this time there are a variety of underlying physiologic changes. The outer hair cells (OHC) of the cochlea are the first affected, they may swell and rotate, become distorted or fused, resulting in the their inability to maintain proper cell function and possible permanent loss.

> Noise-induced permanent threshold shift (NIPTS) occurs when there is less than a full recovery from NITTS. This may be a fairly common occurrence, with small amounts of permanent damage taking place following each of many NITTS experiences (Feuerstein & Chasin 2009).

The effects of these very small changes may be imperceptible relative to the subjective improvement during NITTS recovery, yet this damage appears to be cumulative. Progressive damage may lead to degeneration of auditory nerve fibers and to changes within the central auditory system.

The cumulative effects of noise exposure have been revisited in recent years. Gates et al. found that audiometric hearing thresholds of older men (ages 58 to 80), who were exposed to noise earlier in their lives, increased more than men without history of previous noise exposure (Gates et al. 2000). This finding was subsequently confirmed in a population study of 70-year-olds and most recently extensively examined in controlled laboratory studies on mice. Kujawa and Liberman suggest that noise-induced damage to the ear has progressive consequences that are considerably more widespread than revealed by conventional threshold testing and that noise exposure is more dangerous than previously assumed (Kujawa & Liberman 2009).

Estimates for the risk of developing NIHL are statistically computed from population studies and are referred to as *damage risk criteria*. They serve as the basis for recommending noise exposure limits based on noise levels and duration of exposure, versus the percentage of people expected to develop significant hearing loss as a result of those exposures. The definition of significant hearing loss comes into question. For instance, the environmental protection agency (EPA) considers any hearing threshold shift of more than 5 dB at 4,000 Hz to significantly impact general quality-of-life. On the other hand, occupational regulations are typically aimed at preventing the development of ‘material hearing impairment’. This occupational definition varies but is typically considered hearing loss greater than 25 dB averaged across the frequencies of 500 through 2,000 or 3,000 Hz, and sometimes only meeting this criteria if present in both ears (Niquette 2009; Johnson et al. 2010).

Further investigation of the effects of noise exposure on the human auditory system remains necessary, as understanding the underlying mechanisms is important in the further prevention of NIHL and updating standards and legislations to current best practices. Although, keeping in mind that “the setting of specific limits on exposure to noise is a political decision, with results that vary between jurisdiction depending on economic and sociological factors” (EPA 1974).

1.6 Revisiting temporary-threshold-shift studies

A set of recent studies confirmed that the magnitude of temporary threshold shift (TTS) depends on the frequency distribution and impulse content of the otherwise nearly energetically equivalent noise exposure. It was also shown that recovery from TTS in an environment of 70 dB(A) occurred slower than in silence. The results can be seen in Figure 1.3, where the first test series ‘TS I’ represents exposure to white noise 94 dB(A) L_{eq} for 1 hour and recovery in quiet, ‘TS II’ additionally includes 900 impulses of 5 ms duration at 113 dB(A), and ‘TS III’ is the same as ‘TS I’ followed by 3 hours of white noise at 70 dB(A). In each case there was a significant change in the recovery times, but in traditional L_{ex} calculations the impulses and 70 dB(A) do not significantly contribute to the noise exposure. Since the impulse content has a very short duration (5 ms), it has a negligible L_{eq} contribution when ‘averaged’ over the 4 hours of the study. Similarly, the exposure to 94 dB(A) L_{eq} for 1 hour remains dominant when the additional 3 hour 70 dB(A) L_{eq} exposure is added (Irle et al. 1998).

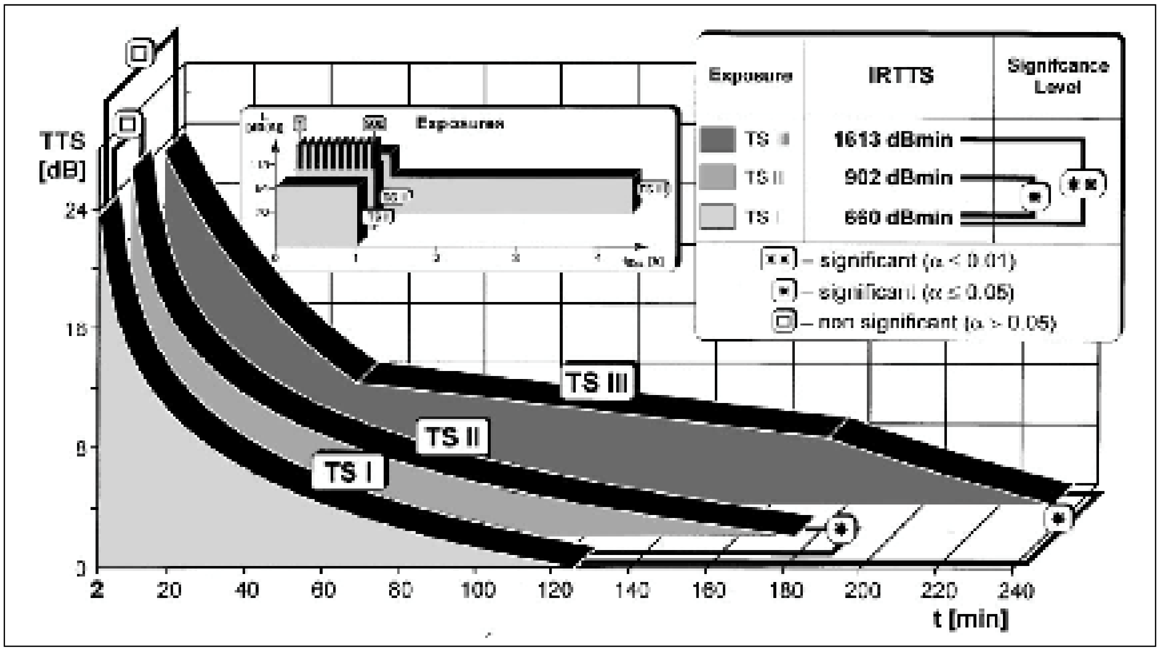

Figure 1.3 TTS recovery times for three different types of noise exposure. 'TS I' is the recovery pattern for exposure to white noise 94 dB(A) for 1 hour ~125min. 'TS II' including 900 impulses at 113 dB(A), and 'TS III' white noise at 94 dB(A) for 1 hour followed by 70 dB(A) for 3 hours
Taken with permission from Irle et al. (1998, p.460)

The results of the study also confirmed large inter-subject variations. It was observed that (TTS_2) values (TTS measured 2 min. post-noise exposure, at 4 kHz) ranged from 15-28 dB with a recovery time of 40-165 min (given the exposure to white noise stimulus of 94 dB(A) L_{eq} for 1 hour). Even larger variations were seen with the introduction of impulse noise (TTS_2 = 16-36 dB, recovery 55-240 min.) (Irle et al., 1998).

A similar study with exposure to various types of music, varying in frequency distribution and transient dynamics: Classical, Chinese and House music revealed different magnitudes of TTS, as seen in Figure 1.4 below.

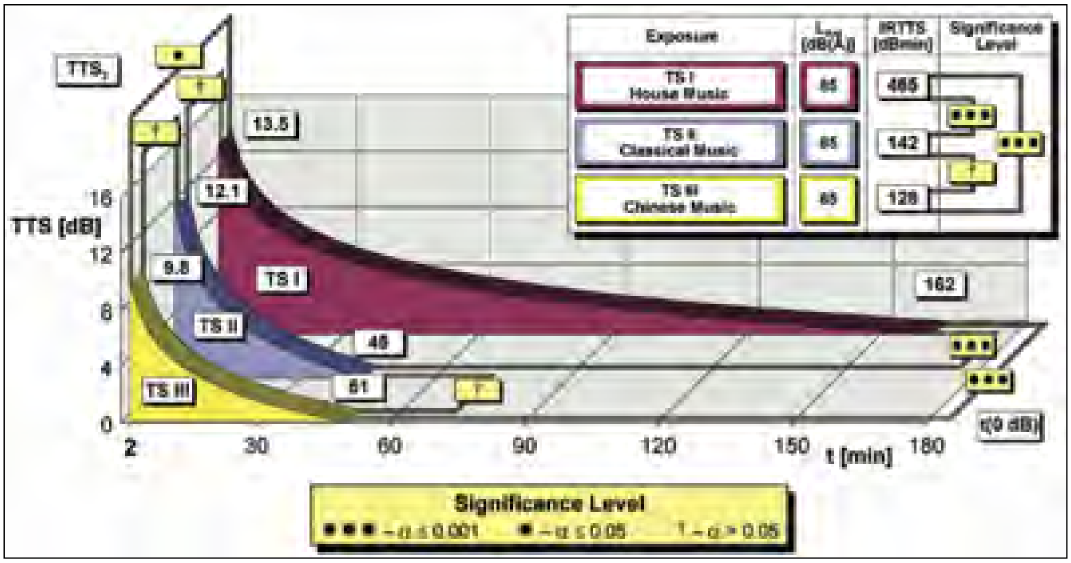

Figure 1.4 Recovery times from exposure to various genres of music. European classical music, Chinese Music, and modern House music. Significantly longer recovery times were observed after exposure to House music
Taken with permission from Strasser et al. (2008, p.413)

The authors reasoned that the 'consistently high levels' and lack of 'breaks' in the House music could be a reason for the prolonged recovery times as well as the significantly more amount of energy in the low frequency bands (Strasser et al. 2008).

The results of these TTS studies support that the frequency distribution and impulse content of the noise exposure should be considered in assessing the damage risk. The study also confirmed the importance of ensuring proper background noise levels (< 70 dB(A)) for recovery from noise exposure (Irle et al. 1998).

1.7 Psychoacoustic noise dosimeter model

A recent series of papers describe the development of a 'psychoacoustic noise dosimeter' (PND) model. The model takes into account the excitation of the basilar membrane occurring in the inner ear, has provisions for impulse noise using time constants for the acoustic reflex, and includes the metabolic and structural components of asymptotic threshold shift (ATS) to predict TTS as well as recovery time given a particular noise exposure (Kostek et al., 2012;

Kotus et al., 2008). The growth and decay variable for the model was selected using exponential regression on data collected from small groups (~30 subjects). The results of the model versus measured data, including standard deviation, can be seen in Figure 1.5.

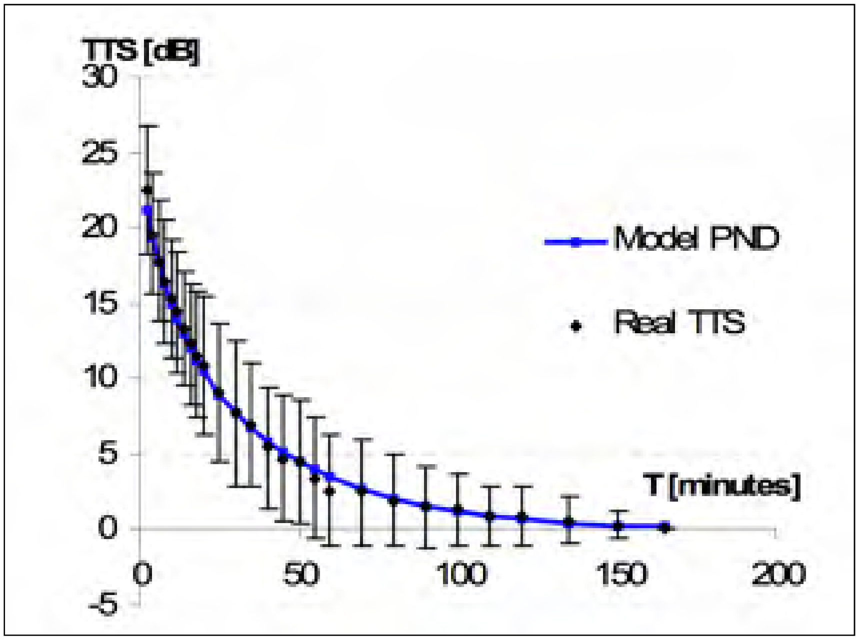

Figure 1.5 Psychoacoustic noise dosimeter model prediction vs. measured TTS after exposure to 94 dB(A) of white noise for an hour
Taken with permission from Kostek et al. (2012, p.6)

The model is intended for real-time use and showed promising results in initial field studies at several dance clubs. While the model fits the average values considerably well, the inter-individual variability remains quite large.

The model predictions for 85 dB(A) for 8-hours of various noise (Pink, Brown, White) produces different degrees of TTS, as shown in Figure 1.6.

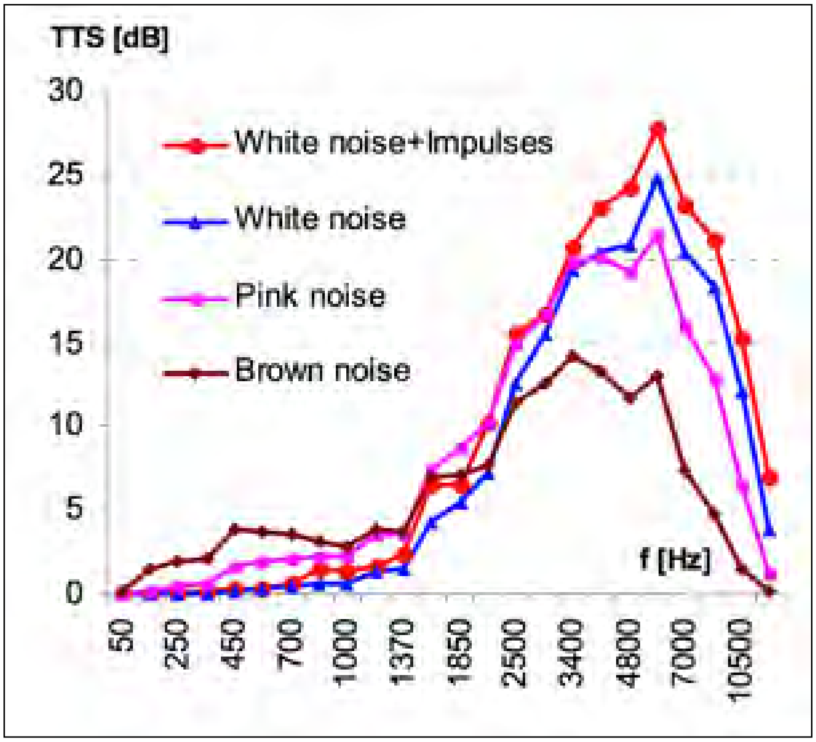

Figure 1.6 The TTS estimations of the PND model for 85 dB(A) 8-hour exposure to : White Noise + Impulses, White Noise, Pink Noise and Brown Noise
Taken with permission from Kostek et al. (2012, p.7)

This coincides with the other recent TTS studies examining the frequency distribution of noise and inclusion of impulse noises (Irle 2008).

1.8 Summary

The single-number Noise Reduction Rating (NRR) label on HPDs is notoriously misrepresentative of actual field performance. The tracking of individual noise exposure is challenging as it requires the knowledge of the HPD attenuation and the monitoring of environmental sound pressure levels (SPLs), or the direct measurement of the effective (protected) levels, i.e. behind the HPD. Traditional dosimetry equipment cannot account for proper fit or use of HPDs. Current legislations do not address 24-hour metrics, they are designed around an 8-hour workday, five days per week over a 40-year working lifetime, while the other 16-hours in the day, as well as weekends are assumed to be quiet.

Recent studies examining the cumulative effects of noise exposure suggest that noise-induced damage to the ear has progressive consequences that are considerably more widespread than revealed by conventional threshold testing (Kujawa & Liberman 2009). Studies measuring TTS post exposure to various signals reveal sensitivity to frequency distribution of the noise, impulse content and background noise level during recovery (Irle et al. 1998). A psychoacoustic noise dosimeter model taking into account frequency distribution and impulse noise showed promising results at predicting TTS post exposure in several night clubs (Kostek et al. 2012). The author presumes one of the reasons such a promising algorithm has received poor adoption is due to the lack of tools to prototype and test it in the field.

This thesis contributes to the existing literature by addressing the following research questions:

- Is it feasible to design a device capable of monitoring comprehensive 24-hour in-ear noise dosimetry, while interfacing with personal music players (PMPs) or communication devices?
- Is it feasible to design an open framework of hardware and software to collect comprehensive field data and prototype noise dose algorithms?
- What would be a possible adaptation of a noise dosimetry algorithm for 24-hour use?

CHAPTER 2

METHODOLOGY

The proposed approach relies on the tracking of individual noise exposure by combining the measurement of the effective protected levels (i.e. behind the HPD) with the measurement of the unprotected exposure levels and HPD attenuation. A digital HPD was designed for this purpose. The system consists of two main components, the earpiece instrumentation hardware and the accompanying signal processing electronics, currently contained within a belt-pack. This chapter details the methods, algorithms and tools used in the development of this system. Starting at the human interface, the in-ear electroacoustics platform (Section 2.1) consists of the ear tip and transducers embedded inside. The transducers require the measurement and application of correction factors (Section 2.2) this is followed by a description of the signal processing algorithms for continuous in-line verification of HPD attenuation (Section 2.3) and noise dose with recovery calculations (Section 2.4). The electronic hardware platform is detailed (Section 2.5) along with the development and manufacturing process (Section 2.6) and programming and debugging (Section 2.7). Finally, the Matlab toolbox is described along with the integration with mobile tools and recorded audio file analysis (Section 2.8).

2.1 In-ear electroacoustic platform

This section describes the interface with the human ear, the ear tip (Section 2.1.1) and the electroacoustic components integrated into the earpiece (Section 2.1.2).

2.1.1 Ear tip

The ear tip's purpose is to provide maximum attenuation with a comfortable acoustic seal in the ear canal. Several types of commercially available solutions exist, two different options were selected, a custom-fit silicone and a universal fit foam.

The custom-fit silicone HPD is developed by Sonomax Technologies Inc. (Montreal, Canada). The unique feature of the SonoFit^{TM} platform is a mass-deployable instant (5 minute) custom fit, capable of high attenuation. It is generally regarded as comfortable and very similar earpieces have been successfully used in brain plasticity research, which required them to be worn continuously for eight days (Schönwiesner et al. 2009). Unfortunately, it can suffer from fit verification problems just as traditional custom HPDs, where an impression of the ear canal is taken by an audiologist and sent to a lab for molding and assembly of the final earpiece. These traditional customs do not guarantee a perfect fit, as Brian Fligor recently concluded while verifying their performance (Fligor 2012).

Closed-cell foam provides universal fit, good comfort and is also capable of high attenuation levels but performance is greatly contingent on proper use. Berger refers to this as the 'inadequate roll-down' (Berger 2011). Although it is possible for an individual to reuse them several times, they are generally regarded as single-use.

2.1.2 Electroacoustic transducers

The purpose of the electroacoustics is to measure acoustic levels just outside the HPD on the faceplate of the device and in the user's ear canal, as well as provide a speaker as in a traditional earphone for PMP playback or communication. Thus, each earpiece is to have a pair of microphones and a speaker. The goal was to place the faceplate of the device as close to the entrance of the ear canal as possible, in an effort to avoid disturbing the acoustical effects of the pinna, centering an omnidirectional microphone flush with the faceplate. The inner-microphone was placed as close as possible to the ear tip's sound bore, facing the eardrum, to minimize the necessary tubing length.

The electroacoustic components were selected based on best performance available in the size and form-factor necessary to embed in such small earpieces. A matrix of the dimensions and acoustic specifications of all commercially available transducers was constructed. With the help of a Mechanical Engineering intern, several configurations were mocked up, the final of which is shown in Figure 2.1. The universal fit option is a similar variation of this design, featuring a sound-bore nozzle that the foam ear tip is press-fit over.

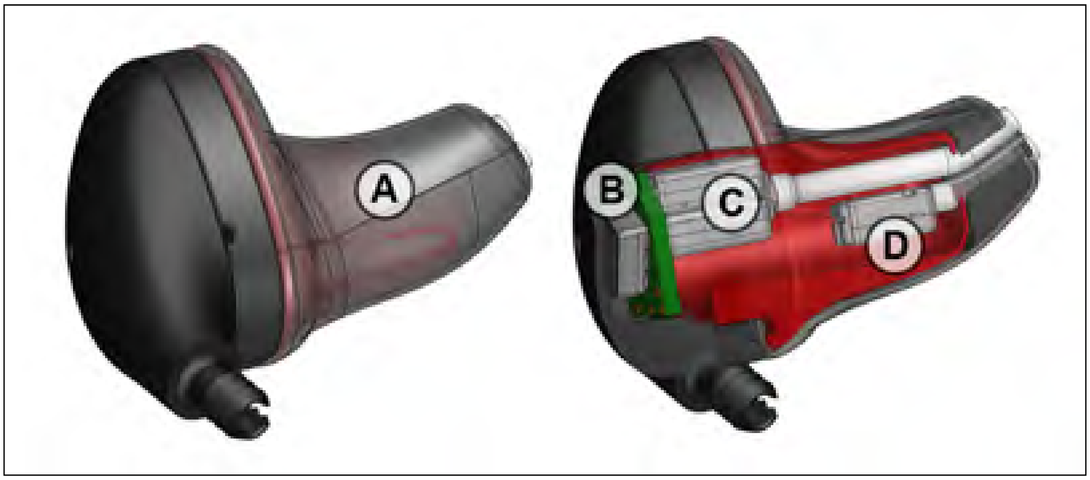

Figure 2.1 CAD prototype of ARP earpieces: A. Sonomax (Montreal, Canada) SonoFit™ earpiece. B. The outer-ear microphone (OEM) Sonion (Roskilde, Denmark) 66AF31 C. Dual balanced-armature speaker Knowles TWFK series. D. The inner-ear-microphone (IEM) Knowles (Itasca, Illinois) GA38 series

The final transducer selection consisted of miniature electret condenser microphones: Knowles GA38 for the inner-ear-microphone (IEM) and Sonion 66AF31 for the outer-ear-microphone (OEM), as well as the Knowles TWFK series balanced armature receiver. Several such prototype earpieces were hand-assembled by the author.

While much work has gone into the selection, sourcing and integration of appropriate electronics for this platform, the practical implications of such a portable system had to be taken into account. It is understood that with further technological advancements the earpieces could benefit from miniature microphones with higher dynamic range and as Sonomax refines its product lines, the earpieces themselves can also be updated.

2.2 Correction factors and signal processing algorithms

Traditional noise exposure assessment and dosimetry use free field sound pressure levels. In order for noise exposure calculations using the proposed earpiece to be compliant with existing data and legislations, correction factors are applied to estimate an equivalent free-field level.

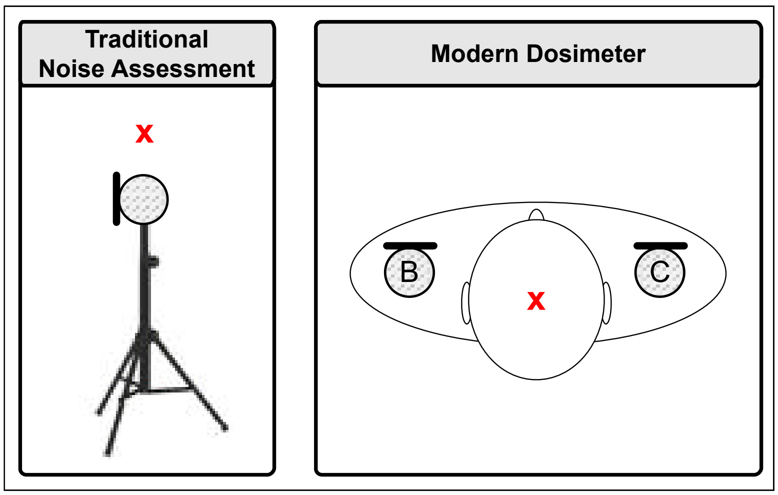

Figure 2.2 Measurement locations for traditional noise exposure assessment and modern dosimetry. A. Sound level meter at the location of center of head with subject absent. B and C, left and right shoulder worn dosimeter

Figure 2.3 illustrates the pressure variables at different locations in the open and occluded ear (the ‘prime’ symbol is used in the latter case) and will be used as a reference throughout the next few sections.

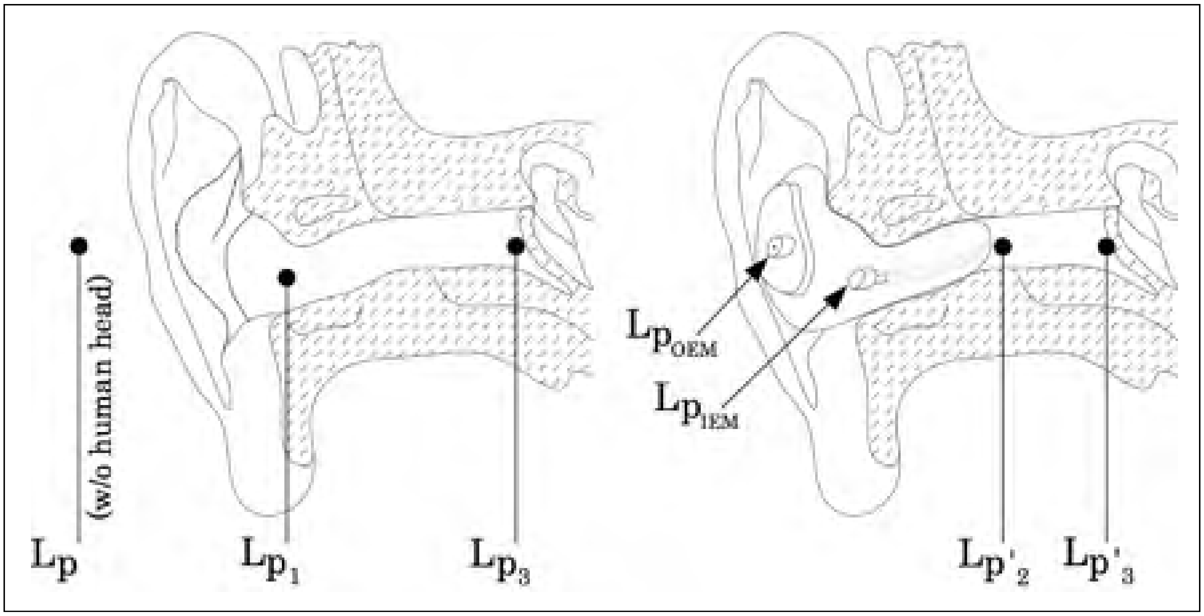

Figure 2.3 Pressure variables at different locations in the open (left) and occluded ear (right). Lp is at the center of head without subject present, Lp_1 is at the entrance of the open ear canal. Lp_3 is at the tympanic membrane. Lp_{OEM} is the outer-ear microphone (OEM), located on the outer faceplate of the prototype HPD, Lp_{IEM} is the inner-ear microphone (IEM) embedded inside the prototype HPD. Lp'_2 is at the tip of the mic probe tube opening and Lp'_3 is at the tympanic membrane, both in the occluded condition

2.2.1 Correction for the occluded ear canal

Two correction factors are applied to the occluded ear canal IEM measurements (Lp_{IEM}) to obtain a free-field equivalent level (Lp):

1. Transfer Function of the Outer Ear (TFOE);
2. Occluded Ear Canal Resonance & Probe Tube Effect (OER).

The TFOE represents the amplification caused by the resonance in the open ear canal, which is individual and varies with the geometry of the human head, torso, pinna, and length of ear canal. ISO 11904-1 states the approximated correction factors in one-third octave bands. This Transfer Function (TF) can be measured on a human subject or approximated by measurement on a head and torso simulator (HATS) and is represented by:

$$TFOE = Lp_3 - Lp \tag{2.1}$$

where Lp_3 is the open tympanic SPL and Lp is the free-field sound pressure level at the 'center of head' location without the subject or HATS present. The OER is dependent on the length of the probe tube and the length of the residual part of the ear canal between hearing protector and eardrum. This is dependent on HPD model and on the user's morphology and is represented by:

$$OER = Lp_{IEM} - Lp_3^{'} \tag{2.2}$$

where Lp_{IEM} is the SPL measured inside the occluded ear canal using the IEM and $Lp^{'}_3$ is the occluded-ear tympanic SPL. Incorporating the two correction factors above, the equation relating the SPL measurement in the occluded ear canal (Lp_{IEM}) to the predicted free-field equivalent is:

$$L_{pFF,IEM} = Lp_{IEM} - TFOE - OER \tag{2.3}$$

this computed $L_{pFF,IEM}$ value is A-weighted (L_{pA}) and used in calculating the estimated free-field equivalent level at discrete time increments:

$$L_{Aeq(IEM),T} = 10\log\left[\frac{1}{T}\sum_{i=1}^{N} t_i 10^{L_{pAi}/10}\right] \tag{2.4}$$

where i is the i^{th} increment, N is the total number of increments, t_i is the duration of the i^{th} increment (ex. 1 second), L_{pAi} is the A-weighted sound level for the i^{th} increment and T is the sum of the individual time increments (e.g. 8 hours).

2.2.2 Correction factor measurement

The correction factors involved vary according to the geometry and dimensions of the human body. In order to avoid some of these cumbersome and delicate measurements, the Bruel & Kjaer Type 4128C – Head and Torso Simulator (B&K HATS) was used as an approximation. Although this increases the uncertainty associated with the prediction, it is much more practical for field use. The validity of these measurements is discussed below.

Two measurements of the aforementioned correction factors were performed, first the transfer function of the outer ear (TFOE) and second the occluded ear canal resonance and probe tube effect (OER). The TFOE correction factor is the transfer function (TF) between the B&K HATS microphone and B&K Reference microphone. The TFOE data in Figure 2.4 is adapted from a study at the Danish Technical Institute (Snaidero et al. 2011). The OER correction is the TF between B&K HATS microphone and IEM of HPD, while fitted on the HATS.

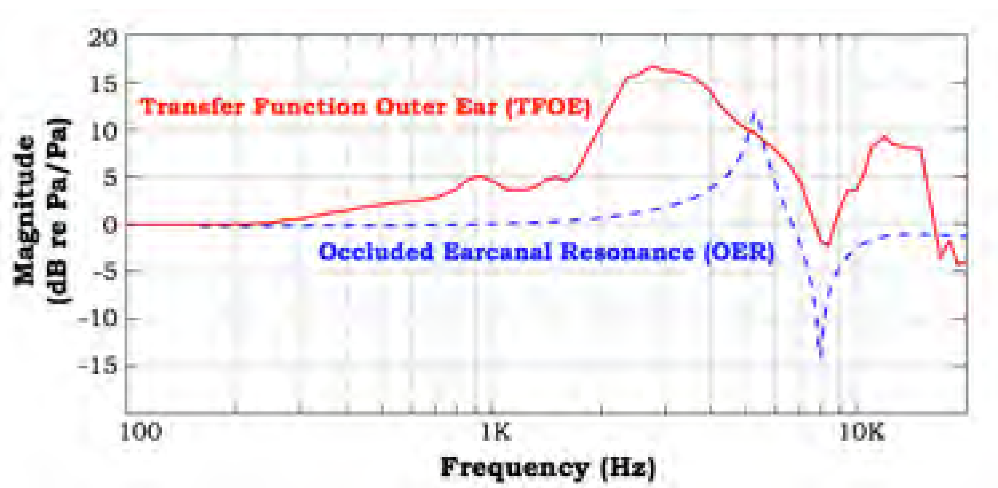

Figure 2.4 Correction factors applied to the measurements using the in-ear microphone (IEM) of the prototype HPD earpiece. The transfer function of outer ear (TFOE), solid red curve, representing the amplification caused by the resonance in the open ear canal is adapted from (Snaidero et al. 2011), a free-field measurement of B&K HATS Type 4128C. The Occluded Ear canal Resonance (OER) is the TF between B&K HATS microphone and IEM of HPD, as measured while fitted on the HATS

The head and torso diffraction (HTD) is the transfer function between the OEM of HPD (Lp_{OEM}) while fitted on the HATS and the free-field reference microphone at the 'center of head' position without the HATS present (Lp). The use of the HTD correction factor should be determined based on the particular application. In diffuse field conditions dB(A) errors beteen Lp_{OEM} and Lp are expected to be minor. In the direct field the HTD will dependent on the supporting structure, sound source location and spectrum of the noise (Byrne & Reeves 2008).

The measurement of the OER reveals a peak, presumably caused by a resonance between the tip of the mic probe tube opening Lp'_2 and the eardrum Lp'_3. Similarly, the most distinct minimum (anti-resonance) is most likely due to a resonance between the MIRE microphone Lp_{IEM} and the tip of the mic probe tube opening Lp'_2 . These resonances are a function of the length of the probe tube, and the length of the residual part of the ear canal between HPD and eardrum. Hence, they are dependent on user morphology and HPD model. Both the resonance and anti-resonance results are in agreement with Bockstael et al's results, who compared the HATS to 19 human test subjects using a different custom HPD. This implies that the OER measured (Figure 2.4) is a good approximation of human ears having average dimensional parameters (Bockstael et al. 2010).

2.3 Continuous in-line verification of HPD attenuation

The location of the reference microphone measuring the unprotected exposure level is on the outside of the user's HPD near the entrance of the ear canal (Lp_{OEM}). The presence of the subject in the acoustic field can be accounted for with the Head and Torso Diffraction and Pinna Effect (HTD):

$$HTD = Lp_{OEM} - Lp_{FF,OEM} \tag{2.5}$$

Thus the free field corrected value for the OEM measurements becomes:

$$Lp_{FF,OEM} = Lp_{OEM} - HTD \tag{2.6}$$

where $Lp_{FF,OEM}$ is the corrected free field level measured just outside the HPD (Lp_{OEM}). Using both free field corrected levels an $NR_{(FIELD)}$ is determined at the initial fitting of the HPD by the user. This value is a prediction of noise reduction (NR), based on a well-fitted HPD that is later used as an anchor to validate proper fit:

$$NR_{(FIELD)} = Lp_{FF,OEM} - Lp_{FF,IEM} \tag{2.7}$$

This $NR_{(FIELD)}$ value, along with the unprotected level ($Lp_{FF,OEM}$) is used to calculate the predicted free-field equivalent exposure level:

$$Lp_{FF} = Lp_{FF,OEM} - NR_{(FIELD)} \tag{2.8}$$

This Lp_{FF} value is then A-weighted and used to calculate the predicted equivalent exposure levels over time:

$$L_{Aeq(NR),T} = 10\log\left[\frac{1}{T}\sum_{i=1}^{N} t_i 10^{L_{pAi}/10}\right] \tag{2.9}$$

when the HPD is not inserted properly this predicted $L_{Aeq(NR)}$, based on the $NR_{(Field)}$ value, will largely deviate from $L_{Aeq(IEM)}$, the estimated free-field equivalent level based on actual measured in-ear levels, this is used to track proper fit over time and further study the relationship between predicted and actual measured levels. When the difference between these two values rises above 10 dB (experimentally selected) an ‘improper fit’ warning can be triggered, notifying the user:

$$L_{Aeq(IEM)} - L_{Aeq(NR)} > 10dB \tag{2.10}$$

2.4 Real-time dose calculation including cumulative and recovery effects

For the user's convenience dose is calculated as a percentage. Using the traditional dose calculation (Berger 2003):

$$D = \frac{100}{T_c} \sum_{i=1}^{N} t_i 10^{\left[L_{Aeqi} - L_c\right]/q} \tag{2.11}$$

where T_c is the *criterion time*, L_c is the *criterion level*, L_{Aeqi} is the A-weighted equivalent level for the i^{th} interval, N is the total number of intervals, and t_i is the duration of the i^{th} interval, and q is the normalized exchange rate (i.e. q=*Exchange Rate/log2*). Exposure at the *criterion level* (L_c) for the duration of *criterion time* (T_c) will yield a 100% noise dose.

The proposed dose including auditory recovery calculation is performed in parallel:

$$\begin{cases} D = \dfrac{100}{T_c} \displaystyle\sum_{i=1}^{N} t_i 10^{\frac{\left[L_{Aeqi} - L_c\right]}{q}}, & if\ L_{Aeqi} \geq L_t \\ D = \dfrac{100}{T_{cr}} \displaystyle\sum_{i=1}^{N} -t_i * K_r * 10^{\frac{\left[L_s - L_{Aeqi}\right]}{q_r}}, & if\ L_t > L_{Aeqi} > L_s \\ D = \dfrac{100}{T_{cr}} \displaystyle\sum_{i=1}^{N} -t_i * K_r, & if\ L_s \geq L_{Aeqi} \end{cases} \tag{2.12}$$

where L_t is the threshold above which dose is to be accumulated, typically around 75 dB SPL, L_s is the *effective silence level*, defined as the highest sound pressure level at which the ear can still recover at its fastest rate K_r (in %/h.). q_r is the *recovery exchange rate*. The *recovery criterion time* (T_{cr}) is defined as the amount of time (in hours) it takes to recover from a 100% exposure. Several options for this parameter are illustrated in Figure 2.5 below. Current legislation assumes the 16-hours outside of work are in relative quiet (solid blue line on graph).

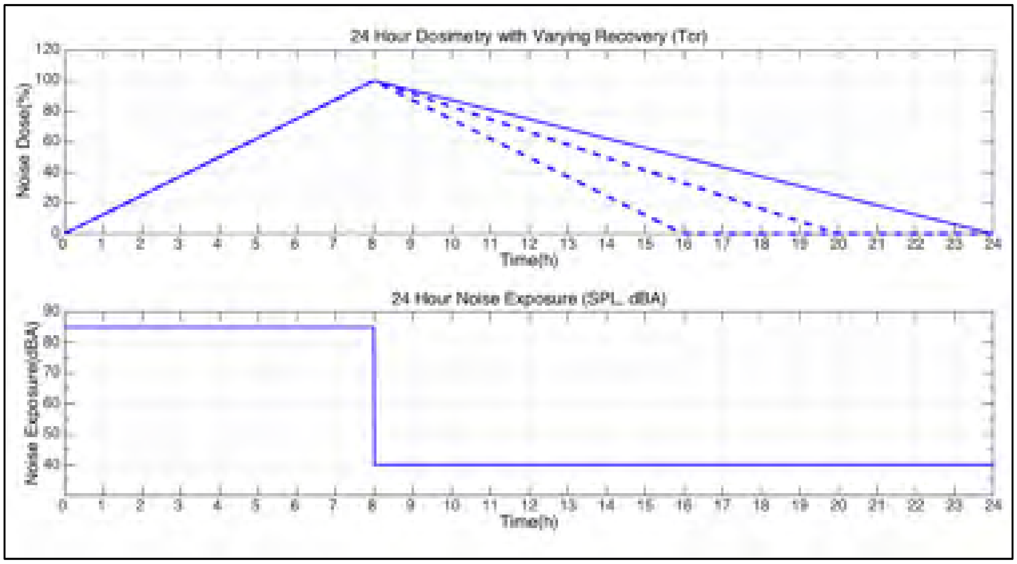

Figure 2.5 Effect on the noise dose of adjusting the recovery criterion time T_{cr} to 16, 12 and 8 hours after exposure to 85 dB(A) for 8 hours (100% noise dose using the NIOSH dose criteria) followed by exposure to 40 dB(A) for 16 hours

In equation 2.12 above three cases are defined, when L_{Aeqi} (the A-weighted equivalent level for the i^{th} interval) is above threshold L_t, dose is accumulated in the traditional fashion. When L_{Aeqi} is below L_t threshold but above the effective silence threshold (L_s), recovery takes place at a rate inversely proportional to the energy ratio of exposure level L_{Aeqi} and the effective silence level L_s affected by a recovery exchange rate, q_r. When L_{Aeqi} is below the effective silence level L_s, threshold, the maximum recovery rate Kr has been reached and linear recovery takes place as a function of the time spent in silence. This proposed dosimetry algorithm is an alternative to the traditional one that assumes a complete recovery has taken place on the beginning of the next exposed work shift and forces resetting the dose back to 0%, without accounting for the effective off-work exposure. A case-study demonstrating the algorithm is presented in the results chapter (Section 3.2.2).

2.5 Electronic hardware platform

2.5.1 Specifications

The specifications of the hardware required interfacing with the proposed earpieces as well as a communication device or smartphone. Each earpiece has 2 microphones and a speaker, thus the minimum input/output configuration was: 4 mic inputs, 2 headphone outputs, 1 line input and 1 line output.

The primary specifications were small-size and low-power as the envisioned final-form factor was a behind-the-ear (BTE) unit, similar to many hearing-aids. It was decided to source chips that would fit in such a design and prototype on them directly. A Bluetooth connection to interface with a smartphone was desirable to enable data-visualization and limit the number of hardware input devices on the belt-pack itself.

Finally, due to the multi-disciplinary nature of auditory research, the platform would not solely be used by programmers, thus it was attractive to have a DSP platform that could be developed without requiring Assembly or even C coding.

2.5.2 Design

The hardware design began with a review of relevant commercially available technologies including: audio codecs and DSP chips, microcontrollers and electroacoustic transducers. This work was published and presented as a poster at National Hearing Conservation Association (NHCA) (see APPENDIX I, p. 77). These commercially available components served as a basis for defining the specifications of the prototype. The prototype hardware is referred to as Auditory Research Platform (ARP), each main component is detailed in the following sections and illustrated in Figure 2.6 below, consisting of:

- Earpieces: Two microphones and speakers each, integrated into an HPD developed by Sonomax Technologies Inc. (Montreal, Canada);

- Audio-Codec: DSP and ADC/DAC with headphone amplifiers;
- Microcontroller: USB , Bluetooth, SD memory and Android^{TM} connectivity;
- Peripheral Devices: 4 Red/Green LED's, 3-way toggle switch;
- Power Management: Li-poly battery and Voltage regulators;
- Android: Touchscreen UI and data connectivity.

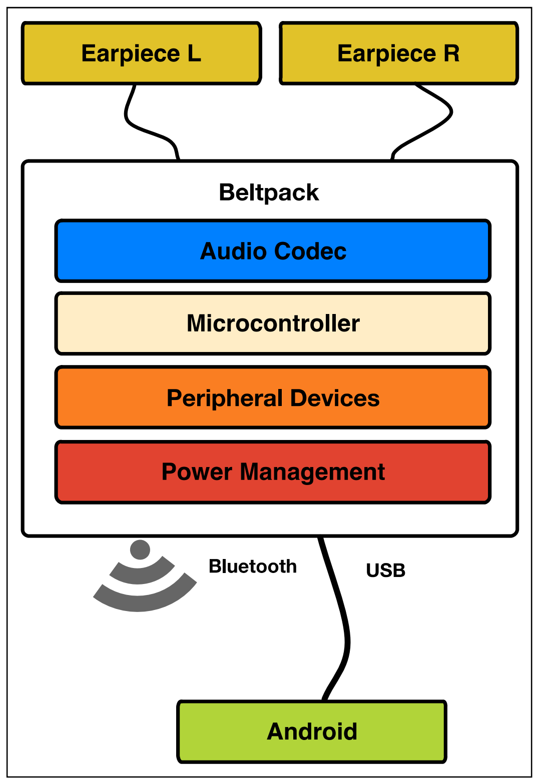

Figure 2.6 High-level overview of Auditory Research Platform (ARP) components, consisting of: Earpieces, Belt pack with Audio Codec, Microcontroller, Peripheral Devices, Power Management and Bluetooth/USB Android connectivity

2.5.2.1 Audio codecs and digital signal processors

The DSP audio codec, from here on referred to simply as the DSP, is responsible for all of the audio conditioning and processing in the system. Since the final platform was envisioned as a BTE device and this belt-pack was considered a stepping-stone, it was decided to choose a DSP that could be embedded inside the earpiece itself. There are few off-the-shelf products that fit this specification as most hearing aids use proprietary technologies. The final choice was an Analog Devices chip targeted at mobile audio, in a small package (5 by 5 mm). A block diagram showing the key functions and input/output stages is shown in Figure 2.7.

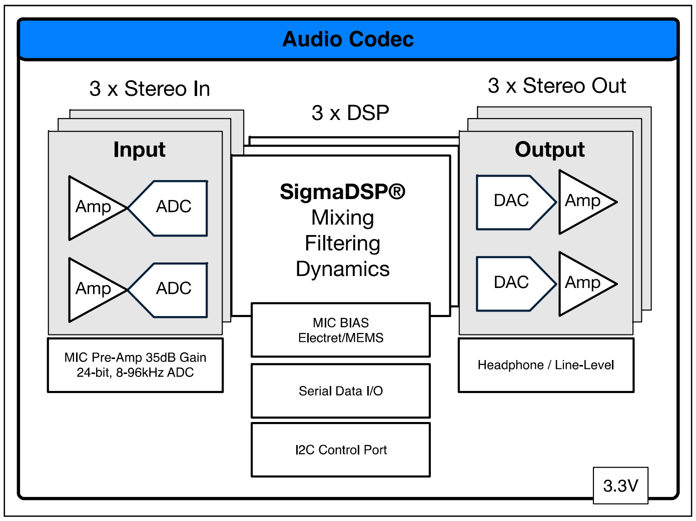

Figure 2.7 Block diagram of digital signal processor (DSP) used in ARP, the Analog Devices (Cambridge, MA) ADAU1761

There are three of these chips in the ARP, one for each earpiece and one for the smartphone downlink/uplink audio. To maximize flexibility the printed circuit board (PCB) is designed with headers such that audio from the earpiece miniDIN connector or smartphone can be routed to any DSP. Audio is also passed from one chip to another on the serial data stream, described in Section 2.5.2.3. The DSP is programmed using Analog Devices (Cambridge, MA) SigmaStudio Integrated Development Environment (IDE), a visual interface similar to schematic capture, a screenshot is shown in Figure 2.8. Once programmed, the DSP parameters can be set using the microcontroller, via hardware buttons or Android UI.

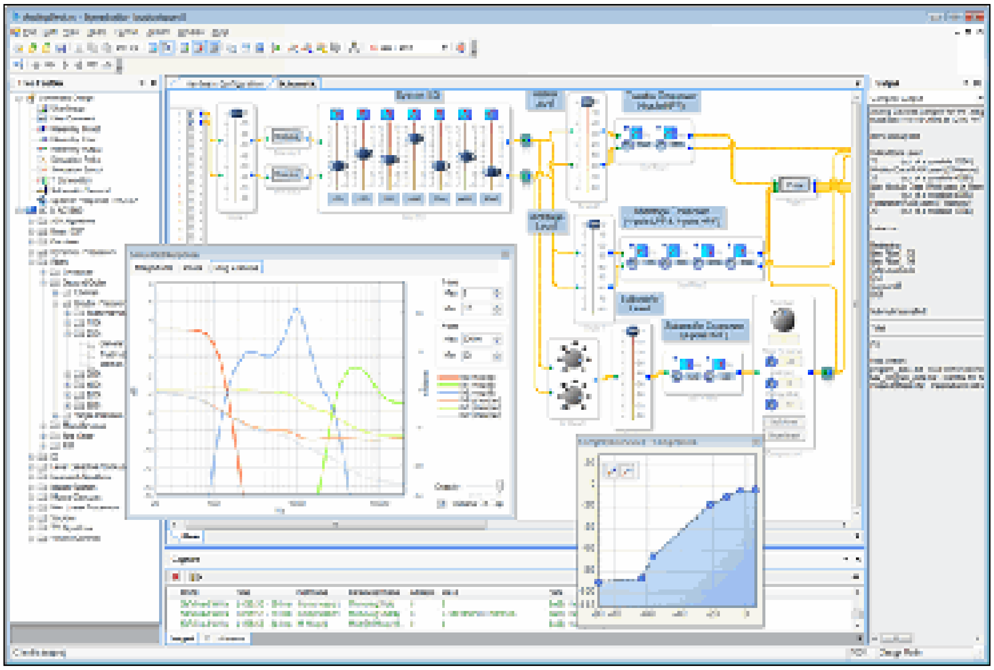

Figure 2.8 Screenshot of Analog Devices (Cambridge,MA) integrated development environment (IDE) SigmaStudio™
Taken from Analog Devices (2008)

2.5.2.2 Microcontroller

One of the primary functions of the microcontroller is to boot the DSP, as the volatile nature of this DSP requires it to be re-programmed upon every power-cycle. Other requirements for the microcontroller were interfacing with general-purpose-input-output (GPIO) devices, providing Flash memory, a USB and SD-card interface, as well as support for the Android Open Accessory protocol. An overview of the primary microcontroller functions is in shown in Figure 2.9.

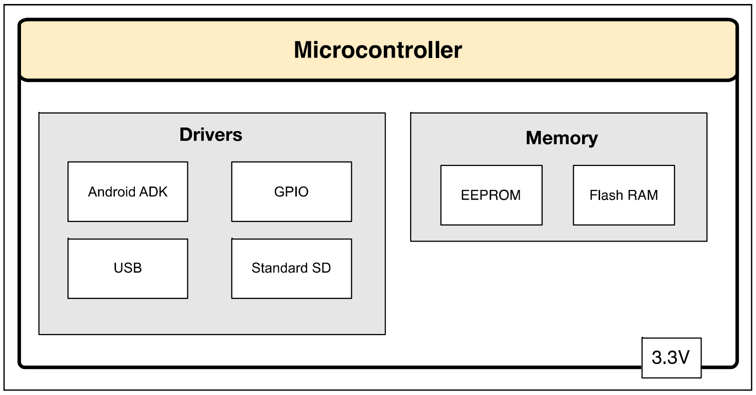

Figure 2.9 Overview of the microcontroller functions in the ARP consisting of: drivers for Android ADK, USB, SD Card and other GPIO as well as Flash and EEPROM memory

There are many commercially available chips that fit the requirements of the project, the PIC24 was selected primarily due to a thriving on-going open-source project IOIO (Ben-Tsvi 2015). The framework created by this community was leveraged to jump-start the ARP development.

2.5.2.3 Communication protocols

The primary communication protocol used is the 2-wire serial communication bus I^2C. The DSP is programmed on the I^2C bus, either using the Analog Device's USBi programmer or when booting from the PIC. Once flashed I^2C is further used to communicate between the PIC and ADAU data registers, for example when using the 3-way switch or the Android user interface.

The Serial-data-input-output (SDIO) port of ADAU1761 is used to transmit an 8-channel TDM stream from one DSP chip to another. Thus the third ADAU in the chain can manipulate data sampled by the first and second chip, providing greater programming flexibility without hardware changes.

Serial-peripheral-interface (SPI) is used for SD card access. This interface allows for reading and writing to the SD card. Unfortunately high-speed writing is not supported by SPI, thus writing multi-channel high sample rate audio is currently not feasible.

Debugging on a real-time system is inherently difficult, using the terminal to read messages while running the device can be helpful, for example confirming parts of the code have executed and verifying addresses when establishing I^2C connections. Serial messages are sent over the RS-232 protocol from the PIC UART port and received using an RS-232 to USB adapter connected to the PCB header.

2.5.2.4 Peripheral devices

To increase portability and robustness, peripheral devices were kept to a minimum without sacrificing the desired functionality; an overview is shown in Figure 2.10. All of the peripherals, aside from audio signals on the 3.5 mm jack, are connected to the PIC microcontroller, which serves as the input/output hub.

The hardware input/output consist simply of a 3-way switch and 4 bi-color red/green LEDs. The 3-Way toggle switch serves as a volume up/down and clicking in cycles between functional modes. The four bi-color LEDs are individually addressable, each LED can be green/yellow or both orange, example uses are displaying the currently selected mode or current noise dose.

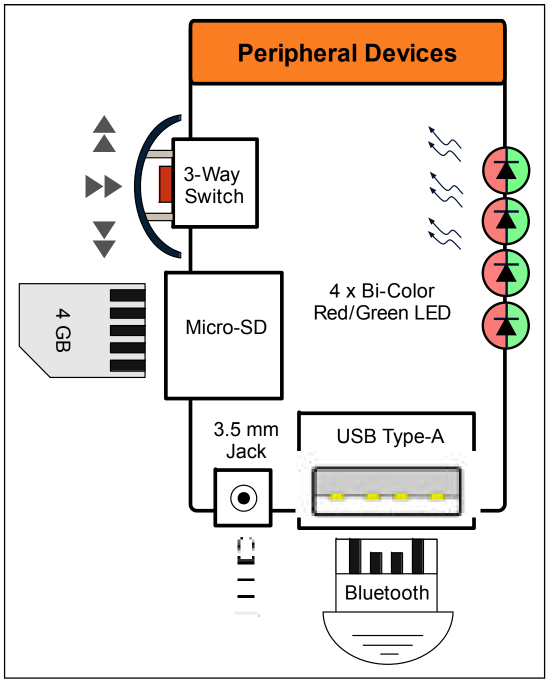

Figure 2.10 Overview of the peripheral devices on the ARP consisting of: 3-way toggle switch, Micro-SD card reader, female 3.5 mm jack, USB Type-A and 4 bi-color green/red LEDs

The Micro-SD memory slot is interfaced to the PIC using a Microcontroller library for the FAT file system. Cards up to 4GB (Class 2) have been tested by the author, although audio cannot be written directly to the card, an L_{eq} value calculated by the DSP could be written to the card.

The USB Type-A receptacle is currently only used for Bluetooth dongle or Android connection. Future host functionality can be implemented as described in the open-source IOIO project (Ben-Tsvi 2015).

The 3.5 mm jack has a header on the PCB for individual custom audio routing. The configuration used by the author is to interface with a standard smartphone, providing a line-level stereo input (for music or downlink communication signal) to the DSP to process or pass-through to speakers in earpieces and a mono microphone signal (uplink for communication), detailed in the ARP 101 document (CRITIAS 2015).

2.5.2.5 Power Management

The power management section is responsible for battery charging and providing two regulated power rails, 3.3V and 5V. An overview is shown in Figure 2.10, consisting of a battery, charge controller, voltage-regulator, DC/DC converter, LEDs, power switch and USB receptacle.

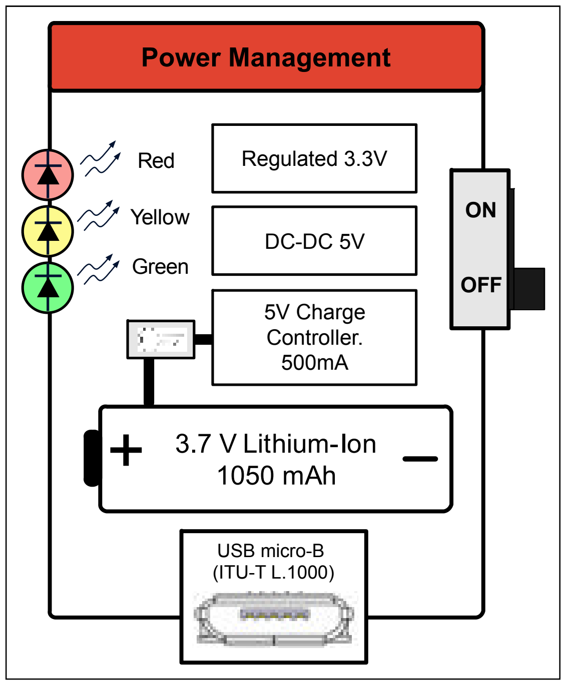

Figure 2.11 Overview of the power management section of the ARP consisting of 3 status LEDs, a USB micro-B plug, 5V charge controller, 3.3V power regulator, 5V DC-DC converter, and power switch. The key attached to the battery prevents reverse polarity connection, which has embarrassed many great engineers

A micro-USB receptacle was selected as the charging port per an ITU recommendation for standardization of cell phone chargers.

The largest consumer off-the-shelf battery available at the time was selected; it is a replacement battery for a remote control device. The capacity is rated at 1050 mAh and a simple battery test yielded 13 hours of playback time with both earpiece speakers playing back white noise and all MICs enabled. Ideally, a battery about twice the width can fit, a custom size could be ordered but minimum quantities were too large for these initial 10 prototypes. The charge controller is configured for 500 mA, typical for computer USB ports, which charges the current battery in approximately 2.5 hours. For faster charging, with the use of a power adapter, it can be increased to 1000 mA by swapping a resistor on the PCB.

The low dropout voltage regulator is configured for 3.3V, providing a continuous load current up to 500 mA, for the Audio Codec and Microcontroller. The DC/DC buck/boost converter is used to step up the battery 3.7V to 5V for devices plugged in as USB-on-the-go.

There are three LEDs signaling power status: powered on (red), charging (yellow) and charged (green). Finally, there is a large on-off switch with a heavy throw to avoid accidental trigger.

2.5.2.6 Android

Interfacing with a smartphone offers many UI advantages (seen in Figure 2.12) including: buttons, sliders, visualization, graphing, logging & sharing data over wireless networks. Currently only basic functionality is demonstrated in the ARP skeleton App for Android. Once the App is opened the connection to the hardware is automatically established over Bluetooth or USB. The USB connection provides lower latency and higher bandwidth.

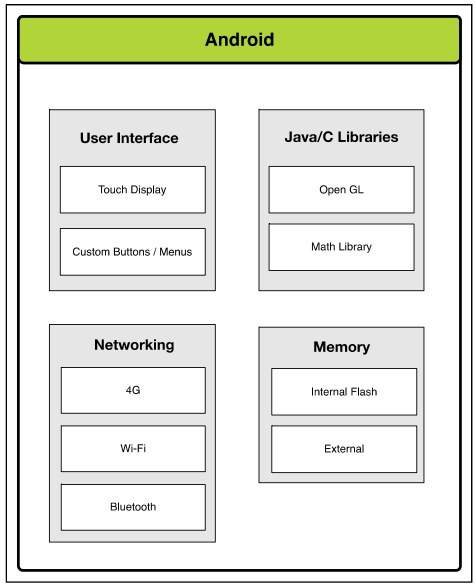

Figure 2.12 Overview of the main Android functions used: user interface, Java/C libraries, networking and memory

The skeleton app provides simple use of buttons, sliders and reading/writing data to DSP registers. Reading RMS values from the microphone signals is demonstrated and some attempts have been made to graph the results, but quite a bit of development remains to complete the interface envisioned by the author, shown in Figure 2.13.

Figure 2.13 Graphic design mock-up of envisioned ARP app interface. Screen 1 to provide direct control over ARP DSP settings, Screen 2 for graphing of L_{eq} and noise dose over time (day, week, month) and Screen 3 for mapping L_{eq} history

The app would provide three main uses. Screen 1 providing direct control over DSP audio settings for example, changing modes and adjusting levels. Screen 2 for visualizing the noise dose and L_{eq} over time (daily, weekly, monthly, etc.). Screen 3 leveraging the GPS of the smartphone and mapping L_{eq} history on a map. Hopefully the skeleton App and mock-up inspire further development of the open-source project.

2.6 Development and manufacturing

The definition of the specifications and functional block diagrams of the system allowed for the finalization of the parts list and schematic capture (Section 2.6.1). At this stage a local third party company was outsourced for the PCB Layout and manufacturing and assembly (Section 2.6.2). Finally leading into the validation, embedded software programming and debugging (Section 2.7). An overview of the flow of the development and manufacturing process is shown in Figure 2.14.

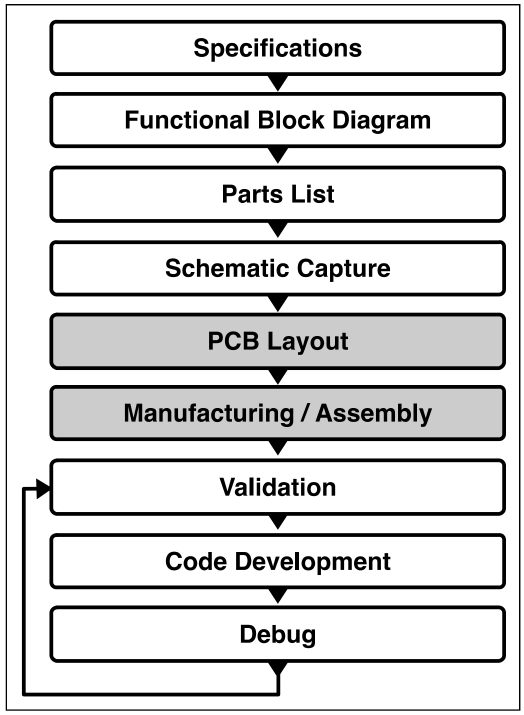

Figure 2.14 Overview of the flow of the ARP development and manufacturing process. The shaded steps (PCB layout and Manufacturing/Assembly) indicate steps outsourced to a local company

2.6.1 Parts list and schematic capture

The parts list essentially serves as a bill of materials. Generating the parts list and schematic capture was done in parallel as each component often had dependencies. As such the datasheets were closely inspected, and the components listed one by one.

Altium Designer, a commercial package licensed by the University was used for schematic capture. Using the block diagram as a guide while carefully referring to the datasheets the electrical connections were drawn one by one. It was quite important to become intimately familiar with each component during this stage as it would be the last time to make changes and a mistake at this stage would be costly. Several revisions resulted in the final schematic detailed in the ARP documentation (CRITIAS 2015).

2.6.2 PCB layout, manufacturing and assembly

The final design called for two PCB's, stacked one on top of the other, inter-connected with a data header and press-fit sliding inside the belt-pack enclosure. This provided for some tight tolerances on mechanical specifications.

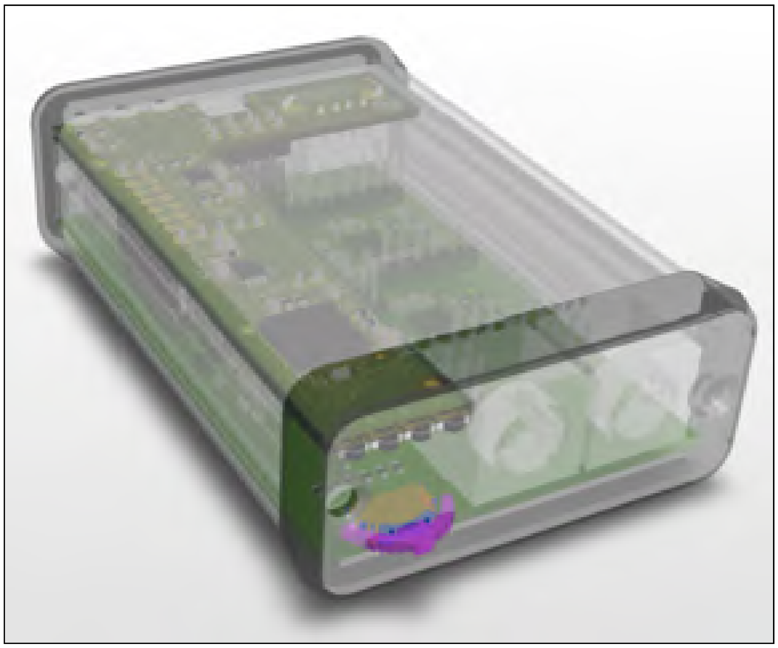

Figure 2.15 Render of both ARP PCBs stacked inside the enclosure, approximate dimensions: 3 x 2 x 1 inches. (CRITIAS 2015)

Due to the complicated mechanical design a local company was contracted to help layout the PCBs, sitting hand-in-hand with a PCB designer provided a good experience for the author without carrying all the risk. The DSP has a fairly complex package that requires proper heat dissipation. Each PCB has 4 layers, top and bottom signal layers and ground and power planes. Since the quantities were low (10) the prototypes were hand assembled using a reflow oven and precision soldering iron.

2.6.3 Hardware validation

The first priority following manufacturing and assembly was examining the power supply section, followed by the debugger communication with DSPs and Microcontroller. Finally verifying the audio paths and the rest of the GPIO and peripherals. Some errors were discovered, the most well disguised issue turned out to be swapping of the two data pins (D+ & D-) on the USB connector. Aside from a few other small oversights the boards are functioning as expected, the design and manufacture was a success.

2.7 Embedded software programming and debugging

There are three components that require software development. The DSPs, the Microcontroller and Android, each have their own integrated development environment (IDE) software and programming/debugging procedure.

This section provides a brief overview of the process, hopefully easing the potential intimidation of diving into the specific instructions and resources provided in the ARP 101 document (CRITIAS 2015).

For developing the audio chain, the DSPs are programmed using SigmaStudio and the Analog Devices USBi programmer. It is possible to leave the ARP in the enclosure with earpieces attached as long as the USBi is connected to the 3-pin I2C header (P25) on the ARP PCB and to a computer running SigmaStudio. Each DSP can then be independently programmed, by selecting its specific address or all three DSPs can be in one project. A skeleton project featuring the three DSPs is downloadable from the ARP website. When the 'Play' button is pushed in SigmaStudio the project is compiled and the DSPs are programmed over the I^2C connection. At this point if the device is not disconnected some parameters remain adjustable in the SigmaStudio environment such as filter gains or compressor settings, while monitoring the levels in the SigmaStudio GUI.

Any parameters that can be adjusted in SigmaStudio in real-time can be further assigned to the hardware toggle button or to an Android UI. When a control or button is used in SigmaStudio the corresponding addresses and values written to the DSP are displayed in the 'capture' window. These are the values that should be specified in the appropriate lines of code in the MPLAB® X skeleton project, for the toggle button and LEDs, or the Eclipse Android skeleton project.

The ARP can be disconnected from the programmer and used, but if the device is power-cycled the DSP memory is cleared and must be reprogrammed again. This is generally fine for development purposes.

The PIC microcontroller can program the DSP upon every power cycle. This requires exporting the project files (C-headers) from SigmaStudio and importing them into MPLAB X project, recompiling and re-flashing the PIC. Thus each time the PIC is powered on it will program the DSPs. The PIC can also be re-flashed to configure the functionality of the LED's or 3-way toggle switch to read or write to specific registers on the DSP.

For Android development it is recommended to follow the standard Google instructions to set up and test the development environment. The ARP skeleton Android project has comments describing the location of the buttons and sliders.

Programming basic DSP control on the ARP can be as straightforward as following the manufacturer's basic setup instructions for each development environment, opening the ARP skeleton demo applications, searching the comments and inserting DSP register values in the code and reprogramming the device. Details are provided in the ARP 101 document (CRITIAS 2015).

2.8 Matlab dosimetry toolbox

In order to reach a broader audience and inspire more research and discussion surrounding the topic of 24-hour noise dosimetry, an open-source Matlab toolbox implementing the proposed dosimetry algorithm was developed. The toolbox also computes other traditional noise exposure computations (L_{eq}, L_{ex}) (CSA Z107.56 2013). This section describes the use of the toolbox.

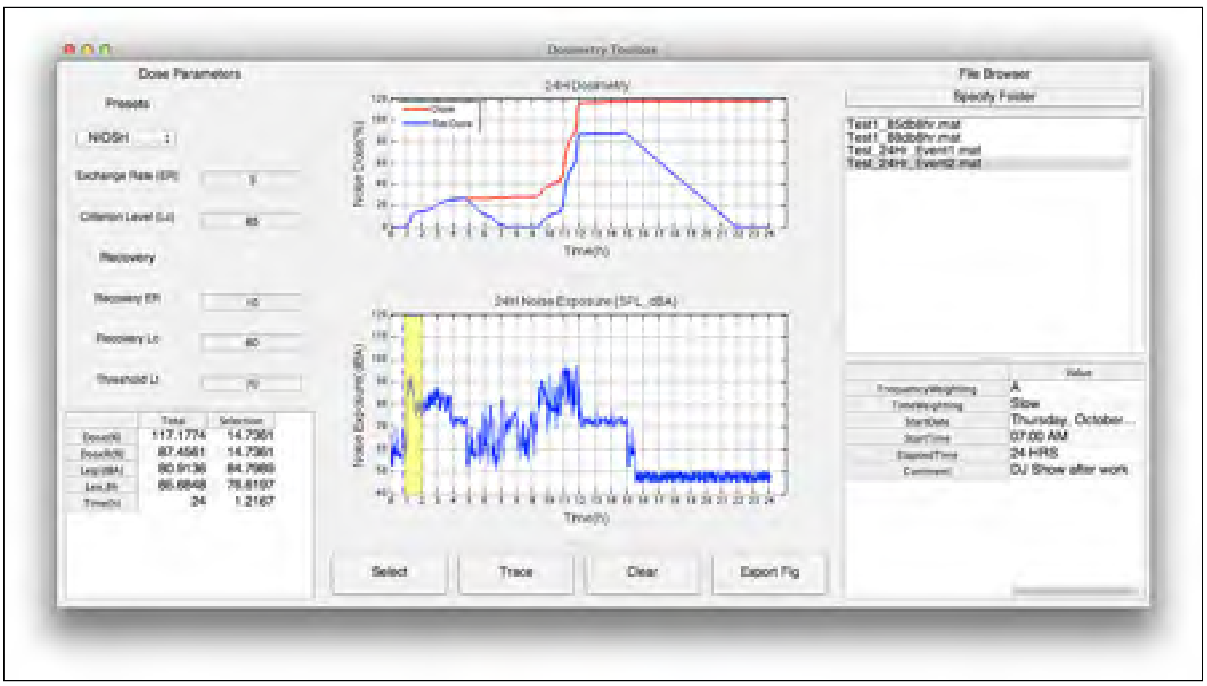

Figure 2.16 GUI screenshot of contributed ‘Dosimetry Toolbox’ for Matlab. The left side ‘Dose Parameters’ section allowing for user specification of parameters. The table below displays the results of the calculations. The center of the GUI features two graphs, on the top dose with and without recovery over time and below the noise exposure. The right side of the GUI is used for specifying the folder containing the MAT log files, selecting the file and displaying parameters and comments of the file

The GUI can be broken down into three sections: dose parameters, graphs and file browser. The dose parameters section, on the left side of the window, enables the selection between several presets: ‘NIOSH’, ‘OSHA’ and ‘EPA’, using a drop-down menu. The parameters immediately below: the *criterion level* (L_c) and *exchange rate* (*ER*) reflect the preset change and can be manually adjusted for a custom calculation. The recovery parameters of the proposed algorithm including the *threshold* (L_t), *recovery exchange rate* (q_r) and *effective silence* (L_s) are manually specified by inputing values into the text boxes.

Immediately below the parameters section a table displays the results of the noise dose and L_{eq} calculations, the first column is for the entire time period of the loaded file and the second is for a selection on the noise exposure graph, should there be one. The calculations featured in the table are:

- Dose (%) – traditional dose computation using preset or custom parameters;
- DoseR (%) – proposed dose with recovery algorithm;
- L_{eq} (dB(A)) – standard L_{eq} calculation (CSA Z107.56 2013);
- $L_{ex,8}$ – the L_{eq} normalized to 8 hours (CSA Z107.56 2013);
- Time (h) – time in hours.

On every parameter change the table and the graph section, in the middle of the GUI, are updated. The top graph displays the 'Noise Dose' while the bottom graph displays the 'Noise Exposure'. A selection can be made on the 'Noise Exposure' graph by clicking the 'Select' button, immediately below the graph, and drawing (click & drag) a rectangular box on the graph. A yellow box highlighting points on the graph appears and the results table is updated to reflect the selection. This selection is cleared using the 'Clear' button. Similarly, any single point is highlighted by the use of the 'Trace' button. The figure is exported to a new window, for further analysis or printing, by the use of the 'Export' button.

The final component of the toolbox is the browser section on the right. The 'Specify Folder' button is used to browse and select a folder containing MAT files for import into the toolbox. These MAT files are simply Struct variables with a specific format. The format was modeled to be compatible with a commercially available iOS app described in the next section and a template file is included with the toolbox, facilitating the creation of these MAT files from existing noise exposure data sets. Once the folder has been selected, the filenames of all MAT files in the folder are loaded into the list box. Single-clicking a file loads its' parameters into the table below the browser. Any parameters in the Struct of the mat file are displayed. This includes any user-defined fields such as: Location, Date, Time, Comments, etc. Double-clicking the file loads the exposure data, performs all the calculations and displays the results in the dose table and graphs.

2.8.1 Mobile noise exposure assessment

The Dosimetry Toolbox for Matlab supports import from a third party iOS application, SoundMeter Faber Acoustical[1] (Provo, Utah). This section describes the logging and export functions, suggests a few instrumentation configurations and provides reasons for selecting this particular app.

A recent NIOSH study using the iPhone 5S built-in microphone and FaberAcoustical SoundMeter App showed a max deviation of 2.5 dB(A) from a reference microphone when tested up to 90 dB(A) SPL (Kardous & Shaw 2014). This implies that the build in microphone can be used to at least determine if hearing protection should be worn and in some cases as a noise exposure assessment tool. For measurement of higher SPLs external microphones can be interfaced using the 1/8” audio jack or in some cases the Apple Lightning to USB Camera Adapter[2]. It is even possible to emulate a shoulder-worn dosimeter using an external microphone, cable and clip, as shown in Figure 2.17.

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[1] At the time of writing the logging and export features require the ‘Pro’ version of the app or can be added via separate in-app purchases.

[2] At the time of writing the MiniDSP UMIK-1 was the only microphone supported and documented by FaberAcoustical to work with Apple’s Camera Connection Kit.

Figure 2.17 Replicating a shoulder-worn dosimeter, using the iPhone 5S running Faber Acoustical SoundMeter with the MicW i436 including extension cable and clip accessories

The microphone MicW i436[1] manufacturer specifications:

- 7 mm electret;
- Omnidirectional polar response;
- Frequency response within Type 2 IEC specifications;
- Max SPL 130 dB (MicW 2013).

---

[1] MicW is a division of BSWA Technology Ltd., a measurement microphone company founded in 1998. Beijing, China.

Figure 2.18 iPhone 5S running SoundMeter with the MiniDSP (Hong Kong) UMIK-1 connected via the ‘Lighting to USB Camera Adapter’

The MiniDSP UMIK-1 (used the for the sleep portion of the case-study in Section 3.2.2) manufacturer specifications:

- 6 mm electret;
- Omnidirectional polar response;
- Factory sensitivity and frequency calibration file;
- Max SPL 130 dB (Umik 2013).

Measurement setups such as these, as opposed to traditional lab equipment, make reasonably accurate noise assessment tools more readily available to the general public and can be used to provide valuable insight into personal noise dosimetry.

The SoundMeter app (with appropriate in-app purchases) can be used to log SPL levels at a user-defined increment (minimum of 0.1 sec.) with selectable Frequency weighting (Flat, A, C) and Averaging time constants (Fast, Slow and Impulse). Once the measurement is complete it can be exported as a MAT file and imported directly into the dosimetry toolbox (Section 2.8). There are a few settings to note in the ‘Tool Options’ menu, shown in Figure 2.16, in order to properly export data for the Dosimetry Toolbox to interpret. Although the

‘Log Interval’ value will be read by the toolbox upon import, 1 second was the standardized value used by the author for all measurements, as it is the minimum value supported by the iOS app for 24-hour logging. Also, the ‘Log Levels’ option should remain selected, these are the levels the used by the dosimetry toolbox, as opposed to the running L_{eq} value.

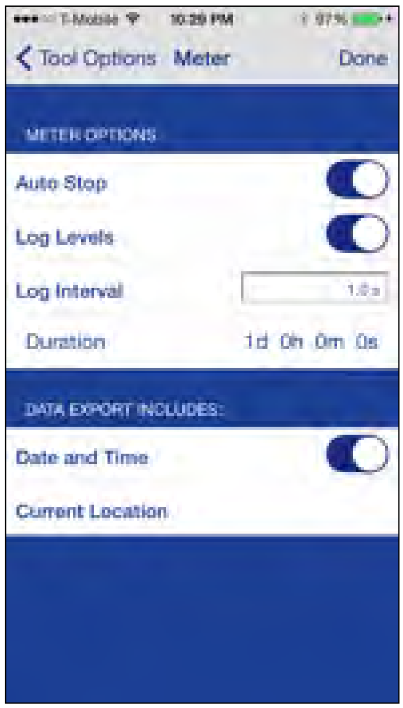


Figure 2.19 ‘Tool Options’ menu of the SoundMeter app. ‘Log Levels’ should remain selected and a ‘Log Interval’ of 1 second is recommended for export to the Dosimetry Toolbox

To export the log for import into the dosimetry toolbox the MAT file format should be selected and the filename must not begin with a number or contain any special characters. It is recommended to use underscores, distinct names and keep numbers at the end of the filename to retain good legibility in the toolbox. Figure 2.17 shows the MAT export screen.

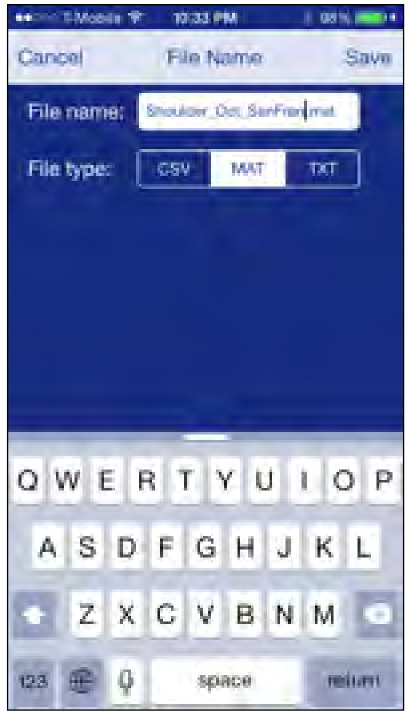


Figure 2.20 Specifying the Filename in the export menu of SoundMeter app.
For import into the Dosimetry Toolbox, filenames should not begin
with a number nor contain special characters

The primary reasons for selecting this particular application were:

- Built-in iPhone 5S default sensitivities were within a few dB(A) of a reference microphone in a recent NIOSH study (tested up to 90 dB(A));
- Support for external microphones via headphone jack or Apple camera connection kit[1];
- Selection amongst all built-in microphones[2] on iPhone 5/6 and iPad Air.
- An option for manual input of sensitivity (for built in or external microphones) including a built in calibration feature;
- Logging capability with MAT file export (requires ‘pro’ version or in-app purchases).

[1] At the time of publication the only documented supported USB microphone is the MiniDSP UMIK-1.
[2] iPhone 5 and 6 have three selectable built-in microphones (Bottom/Front/Back).

2.8.2 WAV file analysis and import

A convenient way of exploring in-ear noise dosimetry without the need for specialized hardware, outside of the earpieces themselves, is recording the raw signals using a digital field recorder. This is in fact how some of the levels used in the case-study (Section 3.2) were attained. In the spirit of encouraging others to explore the proposed dosimetry algorithm this section briefly describes the process of using VSLM, a third party open source toolbox for Matlab to pre-process a WAV file for import into the Dosimetry Toolbox (Muehleisen & McQuillan 2013).

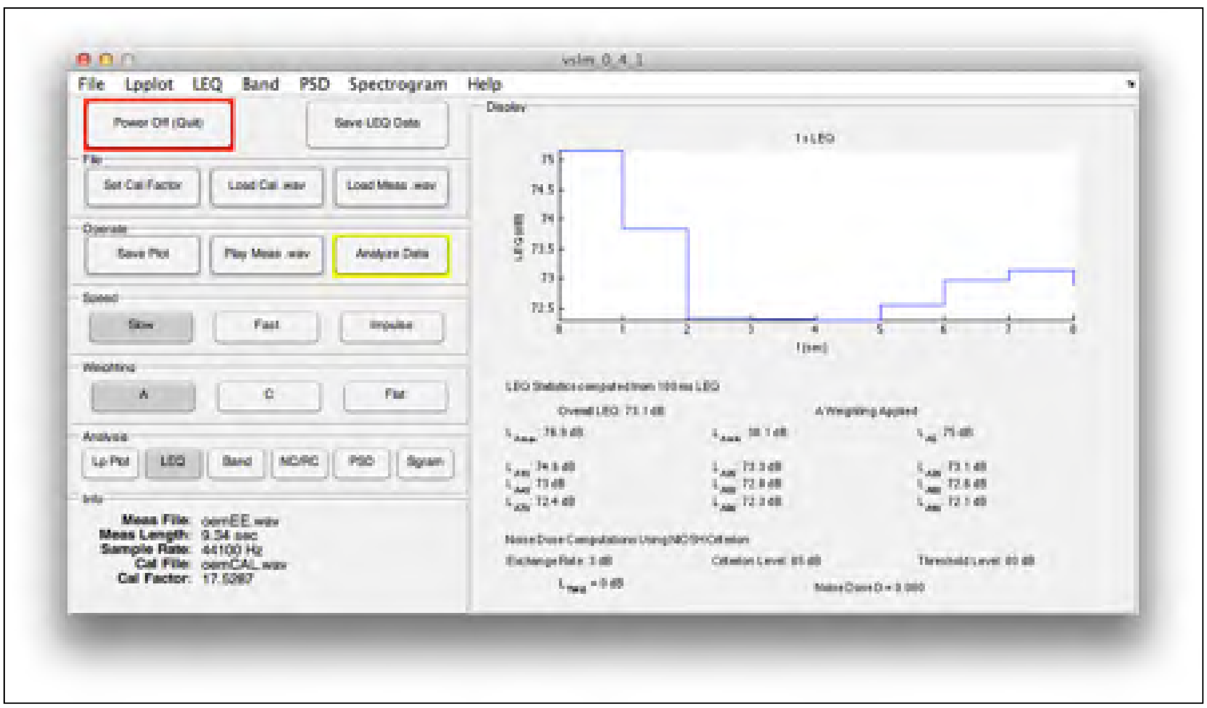


Figure 2.21 Screenshot of virtual sound level meter (VSLM), a third party toolbox for Matlab, which can be used to compute L_{eq} values from a WAV file, given a calibration factor or calibration recording, for further import into the Dosimetry Toolbox

The sensitivity of the microphone and signal chain gain or a calibration file recorded on the field recorder is necessary. Any necessary correction factors such as TFOE should be applied to the WAV file first. The following steps compute and export the L_{eq} from a WAV file:

- Press *‘Load Meas .wav’* button;
- Press *‘Load Cal .wav’* or *‘Set Cal Factor’* button;
- Select Analysis type as *‘Leq’* (default integration time is 1s);
- Ensure *‘A’* Weighting is selected;
- Press *‘Analyze Data’* button;
- Press *‘Save Leq data’* button to export as a txt file, ex. *‘leq_output.txt’*;
- Close/Exit VSLM.

Importing the data into the Dosimetry Toolbox is now simply a matter of formatting the data into the expected Struct. After importing the L_{eq} txt file into Matlab by one of several methods, such as executing the following lines of code:

```
leq = importdata('leq_output.txt');
Template.Data.LevelLog = leq.data( :,1);
Template.Data.TimeOffset = leq.data( :,2);
Template.Data.Parameters.Comment = ‘Good time for a comment’;
Save(‘Template.mat’,’Template’) % variable name and MAT filename must match.
```

This MAT file will now appear in the Dosimetry Toolbox, when it’s parent folder is selected with the ‘Specify Folder’ button as described in the previous Section 2.8.

CHAPTER 3

RESULTS

The results of this thesis are embodied in the Auditory Research Platform (ARP) hardware and the Matlab dosimetry toolbox, this chapter summarizes them with respect to the original objectives. The first sections detail the finalized states of the hardware development (Section 3.1) and proposed dosimetry algorithm (Section 3.2). The next section provides an overview of the Matlab Dosimetry Toolbox, including the Matlab GUI itself (Section 3.3) and finaly a brief discussion. The schematics and software code are downloadable on the CRITIAS website (CRITIAS 2015).

3.1 Open-Source Hardware Platform

3.1.1 Hardware Objectives

The original hardware objective was to assess the feasibility of designing and manufacturing a battery-powered belt-pack DSP system and accompanying custom earpieces, in order to prototype and conduct personal in-ear noise dosimetry research in the field. The device was to be worn throughout the entire day, integrating with the user's personal media player or communication device and featuring a pass-through communication and awareness mode, while continuously tracking noise exposure. The assumption was that if the device easily blended into the user's lifestyle, it would be adopted, enabling comprehensive 24-hour noise dosimetry. Four desired functional 'modes' of the system were defined:

- Serve as an HPD with passive attenuation provided by earpiece;
- Run the dosimetry algorithm and log data;
- Provide audio playback, from a personal media player device;
- Interface with a communication device (i.e. smartphone or handheld radio);
- Provide a pass-through mode, i.e. blending in sound using the external microphones for face-to-face communication and situational awareness of the environment.

Any combination of mixing these modes was also desirable, for example pass-through during audio playback. The device battery was to last all day and continue to monitor background environmental noise levels while charging at night, ideally in close proximity to the user, such as on a nightstand.

3.1.2 Hardware Status

The finalized ARP hardware closely resembles the original vision; ten units have been assembled and verified for all basic functionality, the finished prototype can be seen in Figure 3.1.

Included software applications (Assembly/C/Java) demonstrate all basic audio data-streams and UI control between the various components of the system (Microcontroller, DSP, Android) and serve as 'skeleton' programs, providing a starting point for further development.

Figure 3.1 Photo of the completed ARP hardware interfaced to Android tablet

Simply put, the ARP is a multi-channel (6-stereo input & 6-stereo output) portable DSP system, programmable through Analog Device's visual SigmaStudio™ integrated development environment (IDE). The DSPs can be directly controlled from the IDE, allowing 'live' audio manipulation when interfaced to a computer using the Analog Devices USBi programmer. The SigmaStudio IDE attempts to facilitate DSP programming by providing many 'building blocks' that are visually linked together, similar to schematic capture. This includes:

- Dynamics processing (compressor/expander);
- Level mixing and metering;
- EQ (parametric, shelf, biquad, etc.);
- Low-level math functions (add, subtract, divide, etc.).

Any of the interactive controls in SigmaStudio can be assigned to an Android UI, or the hardware 3-way toggle switch. This is accomplished by noting the address of the control in

SigmaStudio and specifying it into the appropriate field in the provided demo Android application. The volatile nature of this DSP requires it to be re-programmed upon every power-cycle and it does not contain built-in EEPROM memory. Thus, in order to boot into the desired functionality, an external controller is necessary, in this case the PIC24 Microcontroller. Once the desired functionality has been achieved in SigmaStudio the application is exported as C-headers. These C-header files are included into the demo PIC24 MPLABX project, re-compiled and re-flashed on the PIC. Upon powering-up the device, the PIC initializes itself and then boots the DSPs. An example of the development process:

- Use SigmaStudio graphical environment to program the DSPs ‘live’;
- Note the addresses of desired interactive controls;
- Export the project as C-headers;
- Import the C-headers into MPLAB project and re-flash PIC;
- Modify the Eclipse Android project with address of UI controls;
- Power-up and use in the field;
- Optionally, connect Android using USB or Bluetooth for read/write of DSP parameters using UI.

Most of the functional modes previously defined have been implemented and although the ARP hardware DSP supports the dosimetry algorithms, there remain some embedded coding challenges to implement them. Upon building the platform it became clear that a rigorous validation task would be necessary. In order to aid in prototyping and validating the proposed dosimetry algorithm an open-source Matlab toolbox was developed. The algorithm is described in the next section followed by the Matlab toolbox itself (Section 3.3).

3.2 Dosimetry Algorithm

3.2.1 Algorithm Objectives and Description

The objective of the algorithm was continuously computing noise dose while accounting for auditory recovery. The development of the algorithm is presented in the methodology chapter. The following description is an overview of the finalized dose with recovery expression, followed by an explanation of the configurable recovery parameters and the justification behind the default parameters. Although the proposed dosimetry algorithm was previously presented in the Methodology (Chapter 2) it is shown here again for reference:

$$\begin{cases} D=\frac{100}{T_c}\sum_{i=1}^{N} t_i 10^{\frac{\left[L_{Aeqi}-L_c\right]}{q}}, & if\ L_{Aeqi} \geq L_t \\ D=\frac{100}{T_{cr}}\sum_{i=1}^{N} -t_i * K_r * 10^{\frac{\left[L_s-L_{Aeqi}\right]}{q_r}}, & if\ L_t > L_{Aeqi} > L_s \\ D=\frac{100}{T_{cr}}\sum_{i=1}^{N} -t_i * K_r, & if\ L_s \geq L_{Aeqi} \end{cases} \tag{3.1}$$

where L_t is the threshold above which dose is to be accumulated, T_c is the *criterion time*, T_{cr} is the *recovery criterion time*, L_c is the *criterion level*, L_{Aeqi} is the A-weighted equivalent level for the i_{th} interval, N is the total number of intervals, and t_i is the duration of the i_{th} interval, and q is the normalized exchange rate i.e. (*q=Exchange Rate/log2*). Exposure at the *criterion level* (L_c) for the duration of *criterion time* (T_c) will yield a 100% noise dose.

There are three possible cases: when L_{Aeqi} (the A-weighted equivalent level for the i_{th} interval) is above threshold L_t dose is accumulated in the traditional fashion. When L_{Aeqi} is below L_t threshold but above the effective silence threshold (L_s), recovery takes place at a rate inversely proportional to the energy ratio of exposure level L_{Aeqi} and the effective silence level L_s affected by a recovery exchange rate, q_r. When L_{Aeqi} is below the effective silence level L_s, threshold, the maximum recovery rate Kr has been reached and linear recovery takes place as a function of the time spent in *effective silence*. This proposed dosimetry algorithm is an alternative to the traditional one, that assumes a complete recovery has taken place on the beginning of the next exposed work shift and forces resetting the dose back to 0%, without accounting for the effective off-work exposure.

The recovery parameters remain configurable. The author attempts to provide reasonably conservative default values that are based on the following observations. The threshold level (L_t), as defined in ANSI S1.25-1991, is the level above which dose is accumulated, NIOSH recommends 75 dB(A) while OSHA uses 80 dB(A) (Niquette 2009). On the other hand, 70 dB(A) is the maximum level the EPA recommended for 24-hour exposure (EPA 1974). This also happens to be approximately the level that causes no TTS for 24-hour. The TTS study presented in the literature review (Chapter 1) showed that recovery from 100% dose took longer when in background of 70 dB(A). Thus, recovery took place, but at a slower rate. As such, the default value of L_t = 70 dB(A) is proposed, it likely varies individually and could be expected to lie in the range of 65-75 dB(A) SPL.

The effective silence level (L_s) is defined as the highest sound pressure level at which the ear can still recover at its fastest rate Kr (in %/h). Assuming that this level can be thought of as a quiet bedroom, a placeholder level of 40 dB(A) was used in the case-studies above. The EPA levels document recommends an SPL of 50 dB in the 4000 Hz octave band or an A-weighted level of approximately 60 dB(A) (EPA 1974).

The most difficult values to recommend (due to the lack of literature) are the *recovery exchange rate* (q_r) and *recovery criterion time* (T_{cr}). Current legislation is based on a 16-hour recovery in 'relative quiet', this would make for a *recovery criterion time* (T_{cr}) of 16-hours. Extending this assumption to *recovery exchange rate* (q_r) would result in a value of 6 dB, which is likely quite conservative. Considering an exposure representative of 100% dose (94 dB(A) for 1 hour), TTS recovery can take place in the range of 45-200 minutes. This has been shown to depend on many factors including: frequency and impulse content of the stimulus signal as well as interpersonal suscebtibility (Irle et al., 2008). It is also important to remember that recovery from TTS alone does not imply that full auditory recovery has taken place (Kujawa 2013). Thus using TTS recovery time directly in place for T_{cr} is not advisable. These recovery parameters, T_{cr}, L_s and q_r certainly deserve further investigation and the proposed default values are simply provided as a starting point.

3.2.2 Demonstration of the Dosimetry Algorithm

To serve as a demonstration of the proposed dosimetry algorithm, a 24-hour case study is detailed in this section. As previously described in the hardware results (Section 3.1.2), the implementation of the dosimetry algorithm was not completed on the ARP hardware, hence the case study is assembled from several separate exposure levels the author gathered in the field and concatenated into one continuous 24-hour period.

The *commute* levels were recorded using a 4-channel field-recorder, with a setup similar to the one described in a recent IRSST study. The *work* noise-exposure was supplied by the author of the IRSST study (Nélisse et al. 2010). In both cases the audio wav files were post-processed in Matlab to apply the correction factors, in-ear recorded vs. free-field estimate, as described in the Methodology chapter. The L_{eq} levels were then computed using a Matlab toolbox as described in Section 2.8. The other exposure levels (*café, lunch/dinner, movie*) were captured using a Larson Davis Spark 706RC shoulder-worn dosimeter, and the *Sleep* levels using the iOS app, as described in Section 2.8.1. If the L_{eq} sampling interval used for all collection methods is the same, in this case 1 min., a 24-hour scenario can be created by simply concatenating the appropriate arrays and generating the appropriate time-vector. The result after post-processing in the Matlab Dosimetry Toolbox including extra annotations is shown below in Figure 3.2.

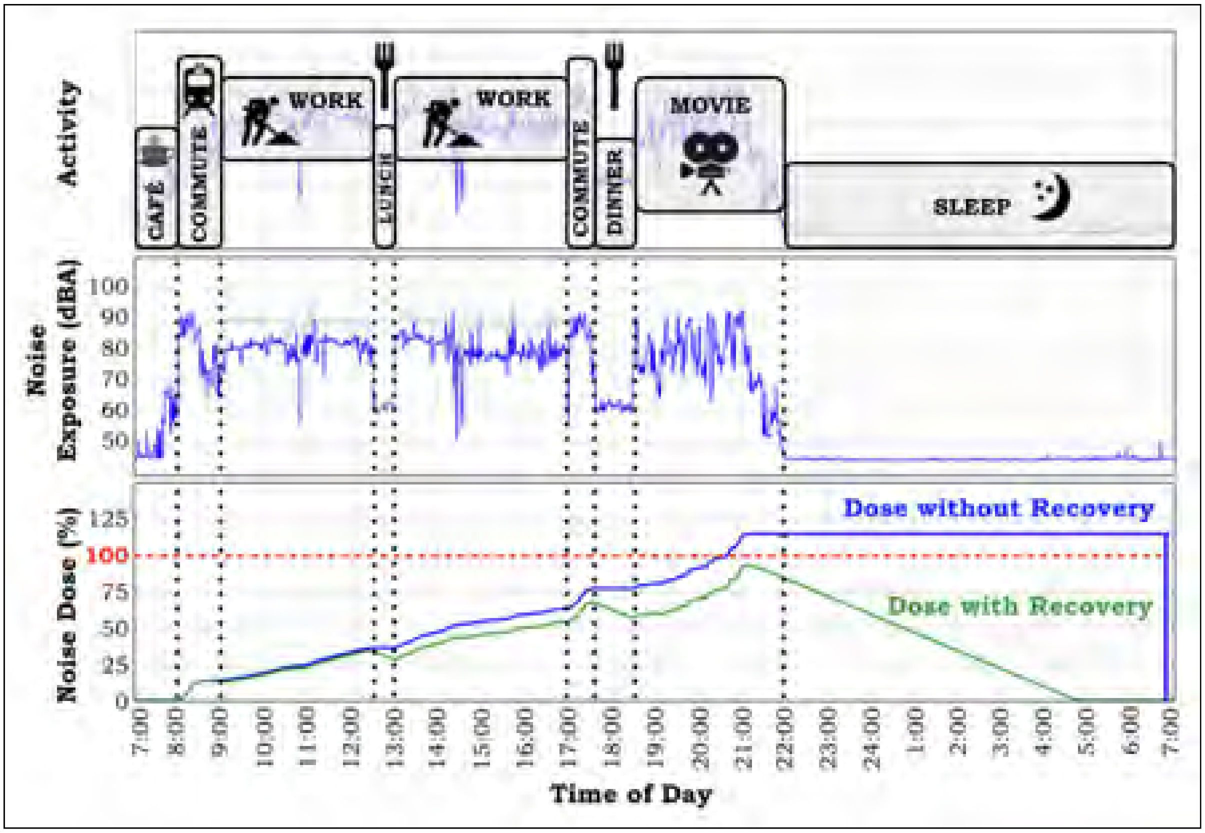


Figure 3.2 Case-study 'Scenario 1' demonstrating
24-hour dosimetry algorithm with recovery

Following along with the Activity and Noise Exposure subplots in Figure 3.2, it can be seen that the user starts the day at 7:00 AM with a *Café*, followed by a *Commute* on the subway while listening to their PMP and arriving at *Work* in an industrial occupation. The blue curve is the 'effective', i.e. protected level as measured under the HPD, while the shaded grey exterior levels are as measured just outside the HPD (data shown for one ear). Around 12:30 PM the user takes *Lunch* in a relatively quiet environment (60 dB(A)), then returns to *Work* for another shift before leaving home on the same *Commute* (90 dB(A)) as the morning, while again listening to their PMP. *Dinner* is in a relatively quiet environment and then an outing to the local movie theater to see *Lord of the Rings*, followed by a walk home and then finally *Sleep*.

Examining the Noise Dose subplot of Figure 3.2, it can be seen that the slope of the line changes at various points in the day, corresponding to the rate of accumulation or recovery for that given environment. In this case, recovery takes place during: *Lunch*, *Dinner* and *Sleep* activities. For example, the *Commute* is quite loud (90 dB(A)) thus the noise dose slope steeply rises. The *Lunch* is relatively quiet (60 dB(A)), thus the recovery is gradual, and finally during sleep the linear recovery rate is reached. With the default recovery values, as discussed in the previous section, the slope of the line during *Lunch* and *Dinner* indicate recovery but not as fast as during *Sleep* in the quiet bedroom. There are two important scenarios to consider.

In ‘Scenario 1’, illustrated above in Figure 3.2, an individual leaves work with a 55% noise dose level, yet still risk over-exposure by the end of the day if they went to see a movie. Traditional dosimetry stops here and is reset back to 0% when returning to work the next day. For illustrative purposes the traditional NIOSH criteria (meant for 8-hour use) is applied for the rest of the day and would yield a dose of 113% by the end of the day. Using the proposed recovery algorithm, the noise dose would not exceed 100%, as the recovery time between the noisy activities is considered.

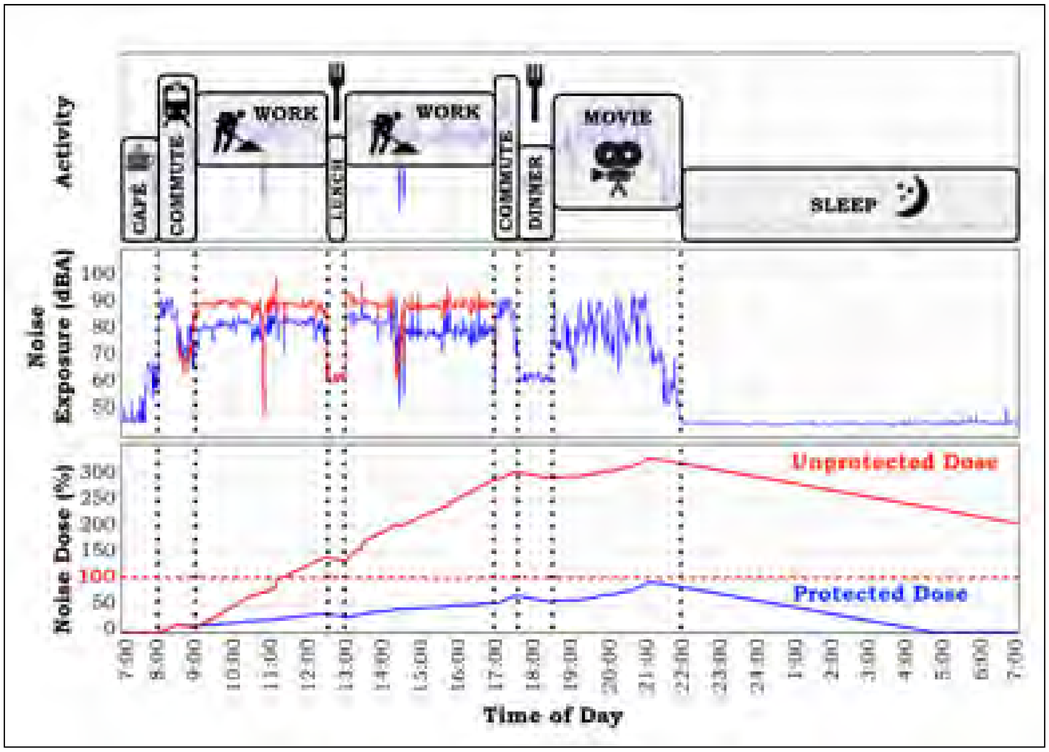


Figure 3.3 Case-study 'Scenario 2', demonstrating
24-hour dosimetry algorithm with recovery

In 'Scenario 2', illustrated above in Figure 3.3, the user had not worn their HPD at work, removed it throughout the shift, or improperly wore it, they could reach a noise dose over 300%. In this case it would require much more time for complete recovery, possibly not occurring before the following work shift. It is indeed the authors personal belief that some individuals continuously exposed to such high noise doses, for example live sound engineers not wearing hearing protection, could be potentially living in a permanent state of TTS.

3.3 Matlab Dosimetry Toolbox

The objective of the Matlab toolbox was analyzing field-data, calculating standard noise dosimetry and testing the proposed 24-hour dosimetry algorithm. The toolbox contains functions, as well as test vectors, for calculation of:

- Combination of L_{eq} values;

- L_{eq} to L_{ex} conversion;
- L_{eq} to Dose conversion;
- Dose to L_{eq} conversion.

These computations have been implemented as described in Canadian standard Z107.56 and each includes an example test calculation (CSA Z107.56 2013). A GUI for analysis of SPL/L_{eq} noise exposure datasets is also included. The GUI calculates standard L_{eq}, L_{ex} and Dose parameters (using the above command-line functions) as well as the proposed dose with recovery algorithm. All dose parameters are adjustable, as the aim of this thesis was to provide tools and a method for 24-hour noise dosimetry, without advocating the use of specific damage risk criteria. The toolbox is complete with four example noise exposure dataset MAT files:

- *'Test1_85dB8hr'*: Useful in validating the dose presets. 8 hours of 85 dB(A) followed by 16 hours of 60 dB(A);
- *'Test1_88dB8hr'*: Useful in validating the dose presets. 8 hours of 88 dB(A) followed by 16 hours of 60 dB(A);
- *'Test_24Hr_Event1'*: 24-hour example case study, as described above;
- *'Test2_24Hr_Event2'*: A similar 24-hour case study with a different occupation (lab and mp3 player use) and effective levels from a music festival event in the evening.

3.4 Discussion of Results

It is widely documented in literature that more insight into recreational noise exposure including PMPs is necessary (Fligor, Portnuff 2013). It is also well established that laboratory HPD noise reduction estimates are poorly correlated with what is experienced in the field and individual fit testing only provides a momentary 'snapshot'. Finally, no clear guidelines and methods exist for continuous 24-hour noise dosimetry. The ARP device and algorithm attempt to address each of these issues and show promising laboratory results, while also reaffirming previously established ideas from literature:

- The combination of activities such as listening to PMP during a commute and going to the local movie theater, can in-fact be hazardous when examined in a 24-hour context in combination with a noisy occupation;
- Wearing an HPD properly in a noisy environment can be effective, but wearing it continuously is important;
- Comprehensive 24-hour dosimetry is an instrumentation challenge and integration into commonly used devices could enable much greater insight into individual noise exposure.

CONCLUSION

Conventional ‘personal’ noise dosimeters do not easily interface with personal media players (PMPs), communication and hearing protection devices (HPDs), nor are they designed for 24-hour use. As such, tracking ‘effective’, i.e. behind the HPD or earphone, exposure levels is not straightforward. This thesis presented the development of a measurement method and hardware platform for performing comprehensive, occupational and recreational, 24-hour in-ear noise dosimetry including auditory recovery. The hardware interfaces with PMP or communication devices and features a pass-through situational awareness mode. Preliminary laboratory validation studies were presented and showed good potential for field-testing. More research and development is encouraged through the open-source contributions.

The objective of this thesis is assessing the feasibility of designing a device capable of monitoring comprehensive 24-hour in-ear noise dosimetry, while interfacing with personal media players (PMPs) or communication devices. The basic functionality of the hardware has been verified and the author believes the general objective has been met. The ARP DSP supports all defined features, easily integrates with a PMP or smartphone via 1/8” audio jack and the lightweight belt-pack is easy to carry all day.

The first specific objective included the ability to collect noise dosimetry field data and test dosimetry algorithms on the hardware platform. Although the framework for this objective has been developed, in hindsight completing all the embedded programming and associated comprehensive electro-acoustic validation tasks fell out of the scope of the thesis, thus this objective was not completely met.

The second specific objective is adapting a noise dosimetry algorithm for 24-hour use, including auditory recovery. Although the proposed algorithm has been developed and implemented in the Matlab toolbox, it is hard to assess the validity of such an algorithm without extensive field and lab testing. Default values for the all the tunable parameters have been proposed based on available literature but they should continue to be refined in future work.

The third objective is designing a Matlab toolbox to analyze field-data, perform standard noise dosimetry calculations and test the proposed 24-hour dosimetry algorithm. The Matlab toolbox can be used to assess the algorithm and refine the tunable parameters. Various methods of importing data have been described in the Methodology chapter (Section 2.8). The author's hope is that the contributed framework of tools inspires more research and discussion surrounding the topic of 24-hour noise dosimetry.

The current device is affected by two main instrumentation limitations. First, the validity of using approximated compensations on an individual basis and second the electroacoustic limitations of the ARP earpiece components. Due to the inherent complexity in measuring the resonance of an occluded ear canal and TFOE, the HATS approximations are used. These human morphological differences can be expressed as a normal distribution, but individual differences could be significant. As new methods emerge facilitating better compensation approximations, such as estimations based on physical dimensions, or less cumbersome measurement techniques, the parameters of the system can be updated. The second main instrumentation limitation is the acoustic validity of the ARP earpieces in the context of dosimetry. Although a comprehensive acoustic validation and calibration should be performed on the ARP, the most likely limiting factors are the transducer selection and placement in the earpieces. As new miniature transducers become commercially available the earpieces themselves can be updated.

Despite these limitations the proposed method and device show promising results. The development of such a platform should enable the revisiting of damage risk criteria and further research into the underlying mechanisms of NIHL such as recovery rate, effective silence, own voice contribution, and ultimately individual susceptibility.

RECOMMENDATIONS

Throughout the development of the ARP several interesting future research topics surfaced. This chapter briefly describes them to inspire future work and as a reminder to the author to return to these topics in the future, including: impulse noise analysis, own voice contribution to noise dose, defining the ‘effective quiet’ level and individualizing the psychoacoustic noise dosimeter.

Impulse noise analysis

In a study of dosimeters in impulse environments, Kardous et al. identified three problems with current noise dosimeters: the limitations of the microphone (dynamic range, peak pressure), the uncertainty of dose-impulse relationship, and the time weighted average (TWA) and dose calculation not properly accounting for impulses. He also suggested that new dosimeters should be capable of retaining the original impulse waveform, for further spectral analysis. The inclusion of these features and integration of new microphones is planned for future revisions of the system as the technology becomes available (Kardous et al. 2005).

Own voice contribution to noise dose

There are two different phenomena regarding the user’s own voice contribution. First, the user’s influence on the measurement of dose accumulated by an externally worn dosimeter, such as the traditional mid-shoulder position, or as in the proposed device the earpiece microphones. The first influence is largely a factor of the environment and percentage of time the user spends speaking. In high background noise environments, such as an industrial setting, the human voice has little contribution to the overall levels. However, in medium to low noise level environments, the measured dose could be influenced by sound originating from the user’s voice. Two methods for estimating this contribution are implementable on the current digital HPD, the binaural method and a most recently published statistical method. The binaural method uses the self to other ratio, meaning that all measurements ‘center’ of the two dosimeters are assumed to be due to the wearer's own voice. The statistical approach is based on the background noise level in the room and the percentage of time the user spends speaking to estimate the vocal effort based on the Lombard effect (Lindstrom et al. 2009; Ryherd et al. 2012).

The second contribution is to the IEM, the amplification of the user's own voice inside the ear canal picked up by the IEM due to the occlusion by the HPD is also experienced while a user is speaking or chewing. Figure 5.1 shows the octave band A-weighted SPLs measured using the earpiece in-ear microphone (IEM) while one of the authors vocalized the vowel 'EE'. The SPL in the ear canal reached over 100 dB at 250 Hz. Depending on how much time the user spends performing these actions, they could potentially account for a significant contribution to the noise exposure.

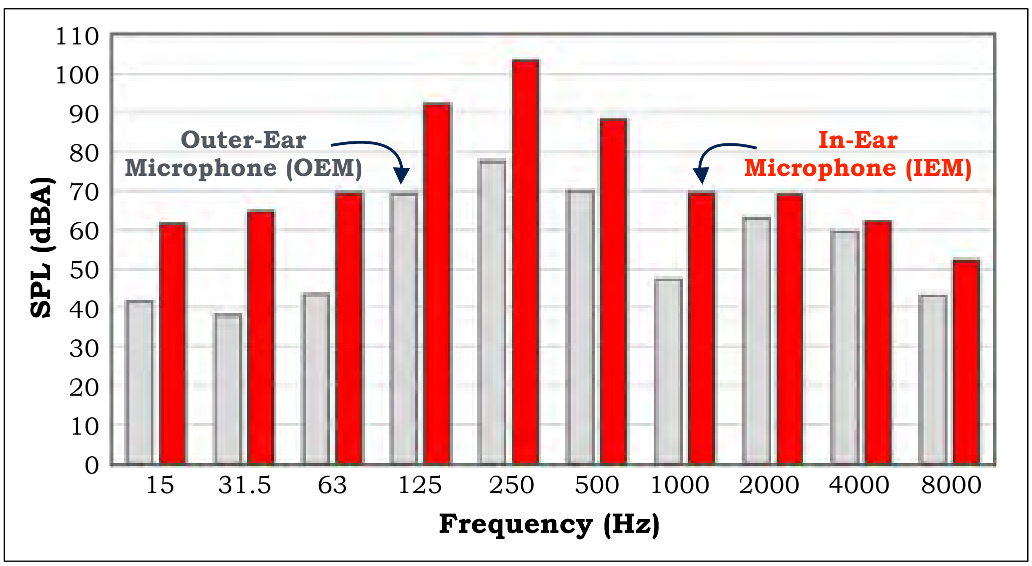


Figure 5.1 Octave band SPL as measured in the occluded ear canal of the author using IEM of proposed device while vocalizing the vowel "EE" at a mildly elevated level, demonstrating the magnitude of the occlusion effect in this particular case

Defining 'effective quiet' level

Although it is well known that time in a relatively quiet environment is necessary for the auditory system to recover from noise exposure, the exact level and duration is not known.

Current occupational legislation is based on a 16-hour recovery time under a noise level of 75 dB(A). The validity of this recovery rate and effective silence level is unknown and has not yet been studied in great detail. The EPA's document 'levels' cites 70 dB(A) as the maximum 24-hour permissible exposure, to protect the general public with a slight margin of safety, a possible 60 dB(A) 'effective quiet' level is mentioned (EPA 1974).

Individualizing the psychoacoustic noise dosimeter

Another recommendation for further development is the implementation of the 'psychoacoustic noise dosimeter' (Kostek et all 2013), in order to further study the effects of spectral and temporal characteristics of noise exposure. Ultimately these recovery parameters could be individualized, potentially from a TTS to dose feedback loop, similar to the envisioned sequential flow-chart shown in Figure 5.2. This would involve the field measurement of TTS at various stages of dose recovery, which is technically feasible to implemented on-board the ARP.

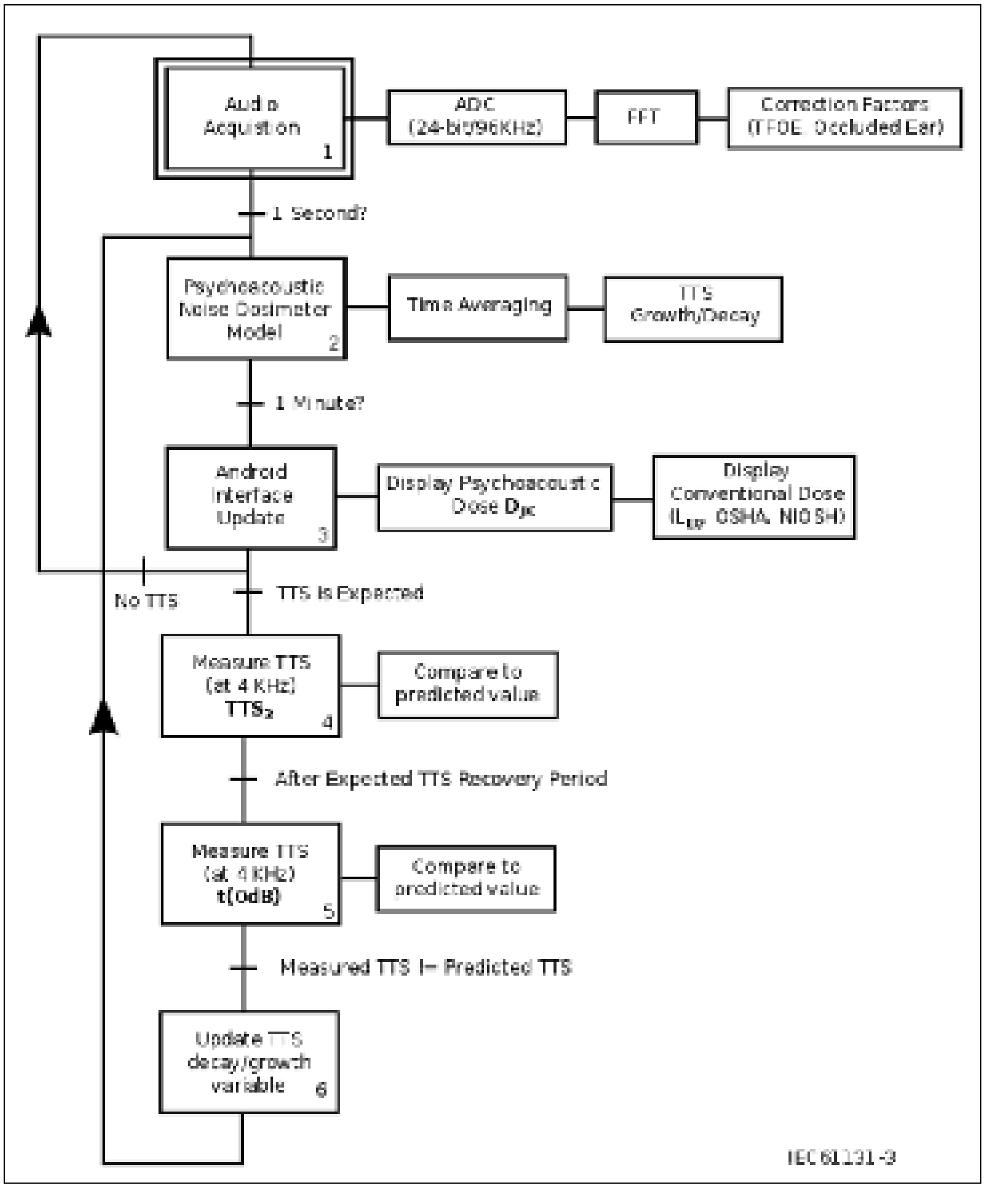
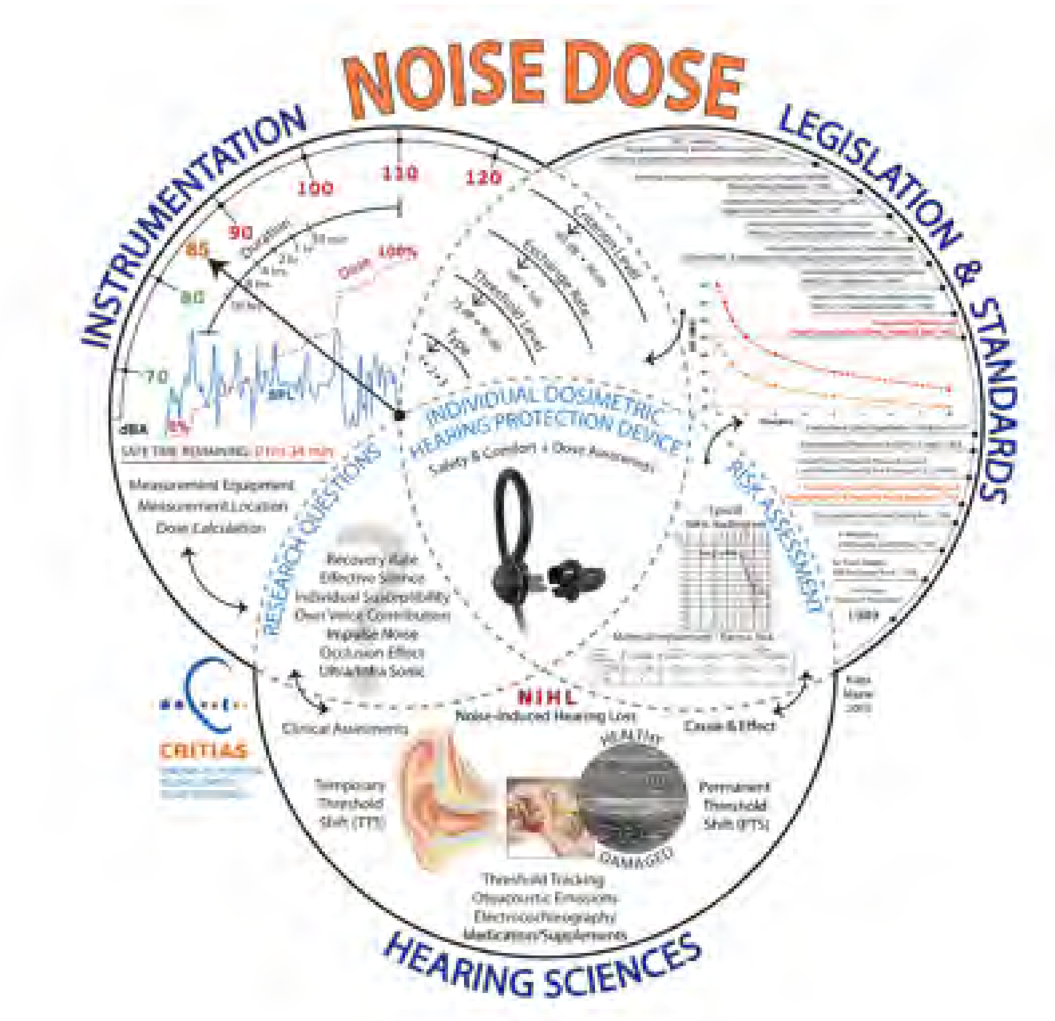


Figure 5.2 Sequential function chart of the proposed dose and recovery algorithms. Block 1 is the sampling of the noise signal using the earpiece microphones, Block 2 is the "Psychoacoustic Noise Dosimeter Model" (Kostek et al., 2012), Block 3 is the Android software interface, Block 4 and 5 represent the in-situ TTS measurement using the ARP hardware and Android interface to update the variables in the dosimeter model (2) on an individual basis

APPENDIX I

Poster I: Noise Dose



APPENDIX II

Publication I: Implementing 24-hour in-ear dosimetry with recovery

Proceedings of Meetings on Acoustics

ICA 2013 Montreal
Montreal, Canada
2 - 7 June 2013

Noise

Session 1pNSa: Advanced Hearing Protection and Methods of Measurement II

1pNSa8. Implementing 24-hour in-ear dosimetry with recovery

APPENDIX III

Publication II: Development of an Individual Dosimetric Hearing Protection Device

Development of an Individual Dosimetric Hearing Protection Device

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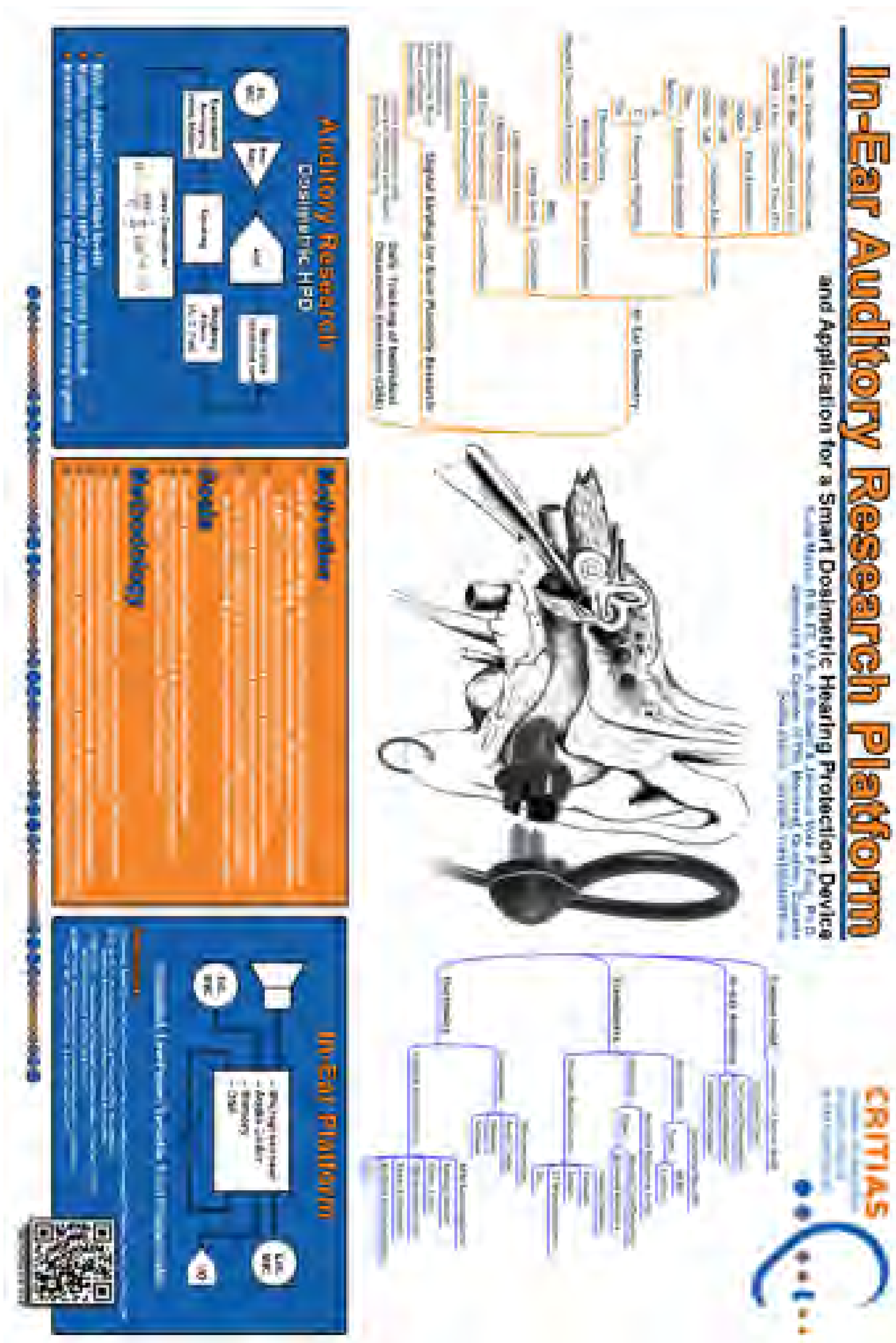
In an effort to increase hearing protection compliance in dangerously noisy environments, two common issues will be addressed: proper fit and tracking of individual exposure level. A custom electronic hearing protection device (HPD) has been implemented on a commercially available platform developed by Sonomax Technologies. Measurement of individual exposure requires either the knowledge of exposure level and HPD attenuation or the direct measurement of the effective (protected) levels, or preferably a combination of both, as in the current prototype. This paper will discuss the development of such a dosimetric HPD while identifying instrumentation challenges, revisiting some hearing science aspects, and highlighting several key research questions related to the risk assessment of Noise Induced Hearing Loss in current standards and legislation. Finally, some preliminary laboratory validation studies will be presented.

APPENDIX IV

Publication III: In-Ear Auditory Research Platform



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