EVALUATING THE PERFORMANCE OF AN ELECTRONIC HEARING PROTECTION DEVICE WITH SELECTIVE NOISE ATTENUATION.

BY

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ABSTRACT

Objective: This study evaluated the performance of the recently developed IIUM e-HPD prototype in noise attenuation and speech intelligibility scores. Method: This study used a quasi-experimental study design. Firstly, the applications of digital signal algorithms and their effect on speech intelligibility for in-ear type e-HPD were discovered through a systematic review. First, six databases were searched using the following key concept of terms: “hearing protection device” and “algorithm”, and “speech intelligibility”. Next, a Modified Acoustic Test Fixture (MATF) test was conducted to measure and calculate the insertion loss (dB) at different ear conditions (Open ear, Occluded ear, Record and playback algorithm and NLMS filter) at 500 and 2000 Hz at 80 and 90 dB SPL. Meanwhile, the Modified Hearing in Noise Test (MHINT) were conducted to measure the speech recognition threshold in noise (dB SNR) using The Malay Hearing in Noise Test speech material. The study compared the insertion loss (dB) between the ear conditions at at 500 and 2000 Hz at 80 and 90 dB SPL . The speech recognition threshold in noise was compared between the all four experimental conditions. Results: The systematic review revealed nine references testing the speech intelligibility of the in-ear type of e-HPD. These findings were identified under three themes: Population, algorithm and speech intelligibility tests. The insertion loss calculated showed the highest insertion loss achieved by the occluded ear at 500 and 2000 Hz. Negative insertion loss observed at 2000 Hz for the open ear, the occluded ear and the Normalised Least-Mean-Square (NLMS) filter with the NLMS filter were the lowest insertion loss calculated. RM-ANOVA was performed on the insertion loss (dB) from MATF test results it showed a significant difference ($p<.05$) between the main effect and pairwise comparison of the ear conditions, noise level, and frequencies of interest. Polynomial contrast showed significant linear, quadratic and cubic components ($P>0.10$) except at the interaction between condition, noise level and frequencies of interest. A one way RM-ANOVA of the MHINT on the ear conditions indicated a significant main ($p<.05$) while the pairwise comparison of the ear conditions showed a significant difference between the open ear and NLMS filter only. Conclusion: In summary, the developed e-HPD prototype was able to attenuate noise although not as comparable to an occluded ear while preserving the speech intelligility in noise compared with open ear.

خلاصة البحث

إن الهدف من هذه الدراسة هي تقييم أداء النموذج الأولي IIUM e-HPD المطور مؤخرا في توهين الضوضاء ودرجات وضوح الكلام. وقد استخدمت الدراسة تصميم شبه تجريبية منهجا للبحث. أولا، تم اكتشاف تطبيقات خوارزميات الإشارة الرقمية وتأثيرها على وضوح الكلام لنوع e-HPD داخل الأذن من خلال مراجعة منهجية . تم البحث في ست قواعد بيانات باستخدام المصطلحات الرئيسية التالية: "جهاز حماية السمع" و"الخوارزمية"، و"وضوح الكلام." ثم تم اجراء اختبار تركيبات الاختبار الصوتية المعدلة (MATF) لقياس وحساب فقدان الإدخال) (dB) في ظروف الأذن المختلفة (الأذن المفتوحة، والأذن المسدودة، وخوارزمية التسجيل والتشغيل، ومرشح NLMS) عند 500 و2000 هرتز عند 80 و90 dB SPL. وفي نفس الوقت، تم اختبار السمع المعدل في الضوضاء (MHINT) لقياس عتبة التعرف على الكلام في الضوضاء (dB SNR) باستخدام مادة الكلام الملايو للسمع في اختبار الضوضاء. قارنت الدراسة فقدان الإدخال (dB) بين ظروف الأذن عند 500 و2000 هرتز عند 80 و90 dB SPL. وتمت مقارنة عتبة التعرف على الكلام في الضوضاء بين جميع الظروف التجريبية الأربعة. ومن أهم نتائج أنها كشفت المراجعة المنهجية عن تسعة مراجع تختبر وضوح الكلام لنوع داخل الأذن من e-HPD. وحددت هذه النتائج في ثلاثة محاور: السكان، والخوارزمية، واختبارات وضوح الكلام. أظهر فقدان الإدخال المحسوب أعلى فقدان الإدخال حققه الأذن المسدودة عند 500 و2000 هرتز .كان فقدان الإدخال السلبي الذي لوحظ عند 2000 هرتز للأذن المفتوحة، والأذن المسدودة، ومرشح المربع الأدنى المعياري (NLMS) مع مرشح NLMS هو أقل فقدان الإدخال المحسوب. تم

أجراء RM-ANOVA على فقدان الإدخال (P<.05) من نتائج اختبار MATF حيث أظهر فرقا كبيرا بين التأثير الرئيسي والمقارنة الزوجية لظروف الأذن، ومستوى الضوضاء، والترددات ذات الأهمية. أظهر التباين متعدد الحدود مكونات خطية، وتربيعية، ومكعبية معنوية (P>0.10) باستثناء التفاعل بين الظرف، ومستوى الضوضاء، والترددات ذات الأهمية. أشارت طريقة واحدة RM-ANOVA من MHINT لظروف الأذن إلى وجود رئيسي مهم (P<.05) بينما أظهرت المقارنة الزوجية لظروف الأذن فرقا كبيرا بين الأذن المفتوحة ومرشح NLMS فقط. الخلاصة: باختصار، كان النموذج الأولي e-HPD المطور قادرا على تخفيف الضوضاء، وعلى الرغم من أنه لا يمكن مقارنته بالأذن المسدودة مع الحفاظ على وضوح الكلام في الضوضاء مقارنة بالأذن المفتوحة .

APPROVAL PAGE

I certify that I have supervised and read this study and that in my opinion, it conforms to acceptable standards of scholarly presentation and is fully adequate, in scope and quality, as a thesis for the degree of Master of of Medical Science (Audiology Sciences).

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DECLARATION

I hereby declare that this dissertation is the result of my own investigations, except where otherwise stated. I also declare that it has not been previously or concurrently submitted as a whole for any other degrees at IIUM or other institutions.

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LIST OF ABBREVIATION

AG	Array Gain
AGC	Adaptive Gain Control
ANC	Acoustic Noise Cancelling
ANOVA	Analysis of Variance
ANR	Acoustic Noise Reduction
ATF	Acoustic Test Fixture
ATS	Asymptomatic Threshold Shift
BS	Backstepping
CRD	Centre for Reviews and Dissemination
dB	Decibel
DNA	Deoxyribonucleic Acid
DOSH	Department of Occupational Safety and Health, Malaysia
DSP	Digital Signal Processor
DSP	Digital Signal Processor
EBM	Evidence-Based Medicine
EEP	Electronic Earplug
e-HPD	Electronic Hearing Protection Device
EPHPP	Effective Public Health Practice Project
FAAF	Four Alternative Auditory Feature Test
FBANC	Feedback Active Noise Control
FPGA	Field Programmable Gate Array
HINT	Hearing in Noise Test
HL	Hearing Loss
HPD	Hearing Protection Device
HTD	Hearing Test Device
IHC	Inner Hair Cell
IIUM	International Islamic University Malaysia
IREC	IIUM Research Ethics Committee
LMI	Linear Matrix Inequalities
LMS	Least-Mean-Square
MIRE	Microphone-in-Real- Ear
MJIIT	Malaysia-Japan International Institute of Technology
MRT	Modified Rhyme Test
NIHL	Noise-Induced Hearing Loss
NIPTS	Noise-Induced Permanent Threshold Shift
NITTS	Noise Induced Temporary Threshold Shift
NLMS	Normalised Least Mean Square
NRR	Noise Reduction Rating
OHC	Outer Hair Cell

ONIHL	Occupational Noise-Induced Hearing Loss
OSH	Occupational Safety and Health
OSHA	Occupational Safety and Health Act
PHP	Personal Hearing Protector
PICO	Population Intervention Comparison Outcome
PRGS	Prototype Research Grant Scheme
PRISMA	Preferred Reporting Items for Systematic Reviews and Meta-Analyses
PTA	Pure Tone Audiometry
PTS	Permanent Threshold Shift
QuickSIN	Quick Speech In Noise
RASTI	Rapid Speech Transmission Index
REAT	Real-Ear Attenuation at Threshold
ROS	Reactive Oxygen Species
S-HPD	Intelligent Hearing Protection Device
SINR	Signal-to-Noise-Plus-Interference Ratio
SNR	Signal-To-Noise Ratio
SOCSO	Social Security Organization
SPL	Sound Pressure Level
SPRINT	Speech Recognition in Noise Test
SPSS	Statistical Package for the Social Sciences
STI	Speech Transmission Index
TCAPS	Tactical Communications and Protective Systems
TTS	Temporary Threshold Shift
WRS	Word recognition scores

GLOSSARY

Electronic Hearing Protection Device The Electronic Hearing Protection Device (e-HPD) is a type of Personal Protective Equipment (PPE) that protects the user from noise. These devices are battery-powered and relying on analogue circuitry or digital processing, or a combination of both (J.G. Casali, 2010b).

Normalised Least-Mean-Square (NLMS) Filter Algorithm The NLMS filter is an adaptive algorithm developed and applied by researchers to obtain an adaptive filter for active noise cancellation. The NLMS algorithm uses adaptive filter weights according to the incoming audio data as it is being received and used to train the coefficients of the adaptive filter. This algorithm involves the computation of the output of a linear filter in response to the noise reference and the generation of the estimation error between this output and the desired response and accounts for the variation in the signal level at the filter output and selecting the normalized step-size parameter that results in a stable as well as fast converging algorithm

Acoustic Test Fixture Acoustic Test Fixtures is a device that approximates specific dimensions of an average adult human head and is used according to standards for measuring the insertion loss of hearing protectors. For this purpose, it includes a microphone arrangement for measuring sound pressure levels (Department of Standards Malaysia, 2004).

Acoustic Test Fixture Test The Acoustic Test Fixture (ATF) Test is a test method utilizing the ATF, a device with the same shape as a human head. The microphone of the ATF is inside the head, and a tube connects the microphone to the outside. The tube has the effect of an auditory canal in a real human ear. The test places the HPD in such a way as to cover the tube's opening fully to ensure that occlusion will occur. The signal analysis used the data from the microphone’s output during the analysis process.

Insertion Loss	Insertion loss is the difference in SPL measured using a microphone at a given location with and without the protector in place (Williams, Reeves, & Chen, 2016). The algebraic difference, in decibels, between the one-third octave band pressure level measured by the microphone of the acoustic test fixture in a specified sound field under specified conditions with the hearing protector absent and the sound pressure level with the hearing protector on with other conditions identical (Department of Standards Malaysia, 2004).
Hearing in Noise Test	The Hearing in Noise Test (HINT) is an adaptive measurement of the speech reception threshold (SRT) used to account for the distortion component of hearing loss and predict speech recognition performance at supra threshold level for a wide range of workplace noise characteristics and individuals.

CHAPTER ONE

INTRODUCTION

This chapter presents and discusses the background of the study. It explains that speech recognition is an essential component for workers in the industry. Additionally, as this study set to discover the performance of the developed prototype in attenuating the noise while allowing the speech to be heard, the problem statement was discussed. This chapter also presented the research questions, hypotheses and objectives. The significance of the study followed, highlighting how this study fills the gap in the research literature on the performance of Electronic Hearing Protection Device (e-HPD) with face to face communication in a noisy environment. Finally, the study's limitations were mentioned, followed by brief definitions of the key terms in this study.

1.1 BACKGROUND OF THE STUDY

Hearing Protection Device (HPD) or Personal Hearing Protector (PHP) was legislatively defined as a device worn by a person to prevent unwanted auditory effects from acoustic stimuli (DOSH, 2019). Workers exposed to industrial noise above the 90-dBA are legally obligated to use HPD to reduce the noise exposure (Katz, Chasin, English, Hood, & Tillery, 2015). Recently, Malaysia has gazetted the Noise exposure regulation (2019) under the Occupational Safety and Health Act (OSHA) which specified that worker needs to wear HPD when exposed to 85 dBA of noise.

This requirement was to protect the user from the unwanted effects of noise. One of the unwanted auditory effects of noise is the Occupational Noise-Induced Hearing Loss (ONIHL) or Noise-Induced Hearing Loss (NIHL). Therefore, the immediate mode of protection is by the reduction of the noise going into the user's ears. It can be through the usage of earplugs or earmuffs or a combination of both types of HPDs. These types of HPD is referred to as conventional or passive HPD. However, there are challenges faced using this type of HPD. One of them is that the HPD frequently resulted in blocking out desired speech from reaching the ear (Wagoner et al., 2007). Therefore, the user would face communication difficulties due to passive attenuation of the HPD, reducing all the sound going to the ears, including the possible important speech information such as warnings and critical.

To address this issue, several off-the-shelf HPD, or so-called electronic earplugs (EEP) or Electronic Hearing Protection Device (e-HPD), were developed to allow more effective communication among workers (Lee & Casali, 2017). These typee of HPD increased the usage of e-HPD among workers, musicians, and the military for their high flexibility and multiple functionalities such as active noise control (Brimhall et al., 2002) or adaptive gain control (Hotvet, 1996). e-HPDs commonly have a built-in Digital Signal Processor (DSP) to process the incoming signals in real-time (J.G. Casali, 2010). In addition, the DSPs utilize algorithms to probe with the noise from the environment, such as using Multiband Adaptive Gain control (Voix, Lezzoum, & Gagnon, 2015). Commercially available e-HPD in the Malaysian market primarily uses the muff type of e-HPD, incorporating technologies such as electronic amplification. However, electronic amplification can only be activated under a pre-determined threshold, thus affecting the communication capability in a noisy environment.

Considering that, International Islamic University Malaysia (IIUM) has developed a conceptual design of an e-HPD as an exploratory attempt to further improve functionality of the e-HPD by incorporating a dual-microphone beamformer with an optimal Wiener post-filtering algorithm into the e-HPD design. However, in a previous study (Selamat, Howe, Tang, & Razali, 2014), the conceptual design implemented applied all of the speech processing algorithms offline. Therefore, the direct implication of the findings to real-time speech processing scenarios may lead to unexpected outcomes. These unexpected outcomes are mainly due to the complexity and extensive computation time required by this algorithm. Besides, several HPD design variables, including but not restricted to the microphone separation distance, number of microphones, types of microphones, and crudely optimized filter parameters used in this study have an affect to the overall performance of the e-HPD.

The conceptual design was then developed into a proof-of-concept e-HPD prototype by incorporating a dual-microphone spatial filtering technique into the e-HPD prototype on a reconfigurable Field Programmable Gate Arrays (FPGA) platform (Oinonen, 2006) to improve workers hearing in noise. Compared to the above mention active HPDs, the main novelty of the prototyped e-HPD is that the spatial filtering technique concurrently processes the noisy speech signals recorded from both left and right microphones to achieve better spatial selectively of the desired speech. Furthermore, noise suppression was established on the concept that the travelling delay time of noise is larger than the travelling distance and time for the speech signal to reach the microphone array directly facing the speech (Valente, 1999). Therefore, this algorithm could be a substantial potential for the e-HPD to outperform the other noise suppression HPD products. As a result, workers wearing e-HPD could have better speech perception in noise than the other commercial active HPDsThis proof of concept prototype has been developed further by incorporating an adaptive filter as the algorithm. The development is due to the large size of the proof-of-concept prototype's FPGA platform.

1.2 STATEMENT OF THE PROBLEM

The intended use of the e-HPD prototype was to help industrial workers listening to speech in a noisy environment while protecting them from the harmful effect of loud sound. The performance in cancellation of noise and interference of the Wiener Filter algorithm for the conceptual design has been assessed by computing the array gain (AG). AG represents the improvement of signal-to-noise-plus-interference ratio (SINR) after the speech filtering. This AG was expressed as the ratio of the output SINR to the input SINR. The results show that the proposed dual-microphone beamformer with Wiener post-filter algorithm has been proven to effectively improve the SINR ratio compared to the enhanced Wiener filter. (Selamat et al., 2014).

The e-HPD algorithm successfully improved the spatial selectivity of the recording sensors on the desired speech along the microphone array steered direction. However, the direct implication of the result findings to real-time speech processing scenarios may lead to unexpected outcomes due to the complexity and extensive computation time required by this algorithm. This previous study also did not address the speech understanding of listeners in a noisy environment.

1.3 PURPOSE OF THE STUDY

This study evaluates the performance of the recently developed IIUM e-HPD prototype in noise attenuation and speech intelligibility scores.