

ISTANBUL TECHNICAL UNIVERSITY ★ GRADUATE SCHOOL OF
SCIENCE ENGINEERING AND TECHNOLOGY

**CALIBRATION OF MOBILE DEVICE MICROPHONES AND THEIR USE
FOR SOUND SOURCE IDENTIFICATION**



M.Sc. THESIS

Güner Berker RAHTUVAN

Department of Mechanical Engineering

Automotive Programme

JULY 2020

ISTANBUL TECHNICAL UNIVERSITY ★ GRADUATE SCHOOL OF
SCIENCE ENGINEERING AND TECHNOLOGY

**CALIBRATION OF MOBILE DEVICE MICROPHONES AND THEIR USE
FOR SOUND SOURCE IDENTIFICATION**



M.Sc. THESIS

**Güner Berker RAHTUVAN
(503161720)**

Department of Mechanical Engineering

Automotive Programme

Thesis Advisor: Dr. Osman Taha ŞEN

JULY 2020

İSTANBUL TEKNİK ÜNİVERSİTESİ ★ FEN BİLİMLERİ ENSTİTÜSÜ

**MOBİL CİHAZ MİKROFONLARININ KALİBRASYONU VE SES KAYNAĞI
TESPİTİ İÇİN KULLANILMASI**

YÜKSEK LİSANS TEZİ

**Güner Berker RAHTUVAN
(503161720)**

Makina Mühendisliği Anabilim Dalı

Otomotiv Programı

Tez Danışmanı: Dr. Osman Taha ŞEN

TEMMUZ 2020

Güner Berker Rahtuvan, a M.Sc. student of İTÜ Graduate School of Science Engineering and Technology student ID 503161720, successfully defended the thesis/dissertation entitled “CALIBRATION OF MOBILE DEVICE MICROPHONES AND THEIR USE FOR SOUND SOURCE IDENTIFICATION”, which he prepared after fulfilling the requirements specified in the associated legislations, before the jury whose signatures are below.

Thesis Advisor : **Dr. Osman Taha ŞEN**
İstanbul Technical University

Jury Members : **Prof. Dr. Özgen AKALIN**
İstanbul Technical University

Prof. Dr. Muammer ÖZKAN
Yıldız Technical University

Date of Submission : 15 June 2020
Date of Defense : 07 July 2020





To my family,



FOREWORD

First and foremost, I would like to devote this study to my loving family who was always supportive and has encouraged me during the hardest times. Their endless patience, unconditional love and support through this journey has always lightened the road ahead.

I am grateful to my thesis advisor, Dr. Osman Taha ŞEN, for sharing all his experience and knowledge with me throughout the thesis. His contribution and guidance has brought invaluable aspects to the study.

Furthermore, I would like to thank my colleagues at FORD OTOSAN for their exceptional support. Above all, I express my deep gratitude for the contributions, ideas and teachings of my supervisor Mehmet Volkan KAZANCI who provided all the resource I required throughout the thesis. Moreover, I am very thankful to Ercan BOZTEPE for lending me his valuable time and helping me through the experimental studies in Product Development NVH Laboratory.

Without the support of my family and friends, none of this work would be achieved. Thanks eternally to all of you for every sacrifice you made, for every memory we shared, for being the most important aspect of my life.

June 2020

Güner Berker RAHTUVAN
(Mechanical Engineer)



TABLE OF CONTENTS

	<u>Page</u>
FOREWORD	ix
TABLE OF CONTENTS	xi
ABBREVIATIONS	xiii
SYMBOLS	xv
LIST OF TABLES	xvii
LIST OF FIGURES	xix
ÖZET	xxi
SUMMARY	xxv
1. INTRODUCTION	1
1.1 Literature Review	2
1.1.1 The recording chain of mobile devices	3
1.1.2 Calibration of mobile devices for SLM purposes	5
1.1.3 Performance and accuracy of mobile devices for sound measurements	6
1.1.4 Sound source identification	8
1.2 Procedure of the Study	11
1.3 Objectives of the Thesis	13
1.4 Outline of the Thesis	13
2. BACKGROUND THEORY	15
2.1 Fundamentals of Acoustics	15
2.1.1 Frequency and period	16
2.1.2 Root mean square	17
2.1.3 Sound intensity	17
2.1.4 Sound power	18
2.1.5 Sound pressure and levels	19
2.2 Microphones	20
2.2.1 Microphone sensitivity	21
2.2.2 Frequency response	22
2.2.3 Dynamic range	24
2.2.4 Signal to noise ratio	25
2.2.5 Weighting	26
2.2.6 Microphone types	27
2.2.6.1 Carbon microphones	27
2.2.6.2 Crystal microphones	28
2.2.6.3 Dynamic microphones	28
2.2.6.4 Ribbon microphones	29
2.2.6.5 Condenser microphones	29
2.2.6.6 Laser microphones	30
2.2.6.7 Electret microphones	30
2.2.6.8 MEMS microphones	31
2.3 Sound Level Meters	33

2.3.1 Main sources of measurement errors	34
2.4 Calibration of Microphones.....	37
2.4.1 Pressure reciprocity calibration.....	37
2.4.2 Free field reciprocity calibration.....	42
2.4.3 Interchanging the microphones	43
2.4.4 Using a calibrator for comparison.....	44
2.4.5 Comparison pressure and free field calibration.....	46
2.4.6 Precision attenuator comparison method	46
2.4.7 Determination of frequency response with electrostatic actuators.....	47
2.5 Data Analysis and Processing	49
2.5.1 Types of signals.....	49
2.5.1.1 Deterministic signals	49
2.5.1.2 Random signals	50
2.5.2 Filters.....	51
2.5.3 Aliasing	52
2.5.4 Fourier transform.....	53
2.5.5 Leakage and windows	55
2.6 Sound Source Identification	56
3. EXPERIMENTAL STUDIES	59
3.1 Hemi Anechoic Room Measurements.....	59
3.1.1 Test setup within the hemi anechoic room.....	60
3.1.2 Results of the hemi anechoic room measurements	63
3.2 Test Vehicle Cabin Measurements.....	71
3.2.1 Test setup within the vehicle cabin	71
3.2.2 Results of the vehicle cabin measurements.....	72
3.3 Issue Resolution of a Test Vehicle with a Mobile Device	79
3.4 Microphone Comparison of Various Mobile Devices in a Test Vehicle	82
4. CONCLUSIONS AND SUGGESTIONS FOR FUTURE WORK	85
REFERENCES	89
CURRICULUM VITAE	95

ABBREVIATIONS

AC	: Alternating Current
ADC	: Analog to Digital Converter
ASIC	: Application Specific Integrated Circuit
dBFs	: Decibel Full Scale
DC	: Direct Current
EEMD	: Ensemble Empirical Mode Decomposition
FFT	: Fast Fourier Transform
FIR	: Finite Impulse Response
GPS	: Global Positioning System
HELS	: Helmholtz Equation Least Squares
IFA	: Instantaneous Frequency Attractor
IIR	: Infinite Impulse Response
MEMS	: Micro Electro Mechanical Systems
RD	: Run Down Maneuver
RMS	: Root Mean Square
RPM	: Rotations per Minute
SLM	: Sound Level Meter
SNR	: Signal to Noise Ratio



SYMBOLS

β	: Voltage Ratio
C	: Capacitance
d_m	: Distance Between the Acoustic Centers of the Microphones
dS	: Finite Surface Area within the Control Volume
e	: Open Circuit Voltage Level
f	: Frequency
f_s	: Sampling Rate of the Signal
g_i	: Preamplifier Gain
G_{xx}	: Auto Spectral Density
G_{xy}	: Cross Spectral Density
ρ	: Density of Air
I	: Sound Intensity
L_{eq}	: Equivalent Sound Pressure Level
L_t	: Time Weighted Sound Pressure Level
L_p	: Sound Pressure Level
L_W	: Sound Power Level
L_I	: Sound Intensity Level
l	: Length
L_i	: Sensitivity of i^{th} Microphone in Interchange Method
L_{corr}	: Correction Factor for Microphone Sensitivity in Interchange Method
L_M	: Gain Sensitivity of the Measurement System
r	: Field Vector
M_i	: Open Circuit Sensitivity of i^{th} Microphone
n	: Normal Vector to Surface Area
p	: Acoustic Pressure
p_{atm}	: Atmospheric Pressure
p_{ref}	: Reference Sound Pressure

R^2	: Coefficient of Determination
T	: Period of the Signal
u	: Particle Velocity
V	: Equivalent Volume of Microphone Cavity
w	: Angular Velocity
W	: Power
Z	: Acoustic Impedance of a Cavity
λ	: Wave Length
γ	: Specific Heat of Gas Inside Microphone Coupler
γ^2_{xy}	: Coherence Between x and y



LIST OF TABLES

	<u>Page</u>
Table 3.1 : B&K Type 4189 SPL (dB(A)) in hemi anechoic room.	63
Table 3.2 : Samsung SM-P587 SPL (dB(A)) in hemi anechoic room.....	64
Table 3.3 : iPhone X SPL (dB(A)) in hemi anechoic room.....	64
Table 3.4 : Background noise SPL (dB(A)) in hemi anechoic room.....	64
Table 3.5 : Octave band SPL (dB(A)) in hemi anechoic room for each device.....	67
Table 3.6 : The effect of calibration on each device in hemi anechoic room.....	69
Table 3.7 : B&K Type 4189 SPL (dB(A)) for each medium.....	72
Table 3.8 : Samsung SM-P587 (dB(A)) for each medium.....	74
Table 3.9 : iPhone X SPL (dB(A)) for each medium.....	74
Table 3.10 : Octave band SPL (dB(A)) in the test vehicle cabin for each device.....	75
Table 3.11 : The effect of calibration on each device in the test vehicle cabin.....	78
Table 3.12 : SPL (dB(A)) increments of each device for each amplitude increase....	84



LIST OF FIGURES

	<u>Page</u>
Figure 2.1 : Pressure variations above and below atmospheric pressure.....	15
Figure 2.2 : Wavelength and frequency illustrations under normal conditions.....	16
Figure 2.3 : Sine wave average and RMS values representations.....	17
Figure 2.4 : Generation of a 1-D spherical wave.....	19
Figure 2.5 : Overall sound pressure levels for common sound sources.....	20
Figure 2.6 : A basic structure of an electret and a MEMS microphone.....	21
Figure 2.7 : Frequency range and sensitivity of microphones.....	22
Figure 2.8 : Frequency responses of various microphone types.....	23
Figure 2.9 : Three types of microphone by their working fields.....	24
Figure 2.10 : Relationship between dB SPL and dBFs – dynamic range.....	25
Figure 2.11 : The SNR effect on a signal.....	26
Figure 2.12 : Acoustic weighting curves for humans.....	27
Figure 2.13 : Schematic and structure of a carbon microphone.....	27
Figure 2.14 : The working principle of a crystal microphone.....	28
Figure 2.15 : The working principle of a dynamic microphone.....	28
Figure 2.16 : An internal structure of a ribbon microphone.....	29
Figure 2.17 : Working principle of a capacitive mic.....	29
Figure 2.18 : The application of laser microphones.....	30
Figure 2.19 : Cross sectional view of an electret microphone.....	31
Figure 2.20 : MEMS microphone structure.....	32
Figure 2.21 : Typical structure of a MEMS microphone.....	32
Figure 2.22 : Performance criteria for Class 1 & 2 SLM and a typical SLM.....	34
Figure 2.23 : Amplitude – frequency plot of a trumpet and background.....	36
Figure 2.24 : Equivalent SPL with different windscreen diameters.....	36
Figure 2.25 : Pressure reciprocity calibration of condenser microphones.....	38
Figure 2.26 : A setup for interchanging microphones comparison method.....	44
Figure 2.27 : A comparison method setup for microphone calibration	45
Figure 2.28 : Adjustable multi turn potentiometer.....	47
Figure 2.29 : Electrostatic actuator frequency response measurement setup.....	48
Figure 2.30 : A typical electrostatic actuator with metal grid design.....	48
Figure 2.31 : Time frequency representation of a periodic steady state signal.....	50
Figure 2.32 : Time frequency representation of a transient signal.....	50
Figure 2.33 : Stationary and non-stationary random signals.....	51
Figure 2.34 : Frequency domain filter types.....	52
Figure 2.35 : The harmonics of a saxophone note with anti aliasing filter.....	53
Figure 2.36 : The harmonics of a saxophone note without anti aliasing filter.....	53
Figure 2.37 : The fourier transform and inverse fourier transform of a signal.....	54
Figure 2.38 : Spectral leakage effect of an FFT.....	55
Figure 2.39 : Minimized leakage with windowing.....	56
Figure 2.40 : Fourier spectrum of a vibration data in a facility.....	56
Figure 2.41 : Coherent output power spectrum to determine panel radiation.....	58

Figure 3.1 :	Hemi Anechoic room in Ford Otosan Gölcük facility.....	59
Figure 3.2 :	B&K Type 4231 acoustic calibrator.....	60
Figure 3.3 :	B&K Type 4189 frequency response curve.....	61
Figure 3.4 :	Pre Sonus Eris E5 & Avid Fast Track Solo.....	62
Figure 3.5 :	The initial test setup for hemi anechoic room measurements.....	63
Figure 3.6 :	1500 Hz (Left) - 4000 Hz (Right) harmonics in hemi anechoic room...	65
Figure 3.7 :	FFT spectrum of tested devices in hemi anechoic room.....	66
Figure 3.8 :	1/3 octave band analysis of tested devices in hemi anechoic room.....	67
Figure 3.9 :	Calibration line of Samsung SM-P587 in hemi anechoic room.....	68
Figure 3.10 :	Calibration line of iPhone X in hemi anechoic room.....	69
Figure 3.11 :	Calibration values of Samsung SM-P587 in hemi anechoic room.....	70
Figure 3.12 :	Calibration values of iPhone X in hemi anechoic room.....	70
Figure 3.13 :	Test vehicle 14+1 bus variant and its interior.....	71
Figure 3.14 :	Experimental setup within the test vehicle cabin.....	72
Figure 3.15 :	Visible harmonics in hemi anechoic room (Left) and vehicle (Right).	73
Figure 3.16 :	White noise frequency responses of microphones inside the cabin.....	75
Figure 3.17 :	1/3 octave band analysis of tested devices in the test vehicle cabin....	76
Figure 3.18 :	Calibration line of Samsung SM-P587 in the test vehicle cabin.....	76
Figure 3.19 :	Calibration line of iPhone X in the test vehicle cabin.....	77
Figure 3.20 :	Calibration values of Samsung SM-P587 in the test vehicle cabin.....	78
Figure 3.21 :	Calibration values of iPhone X in the test vehicle cabin.....	79
Figure 3.22 :	Order Spectrum of 4th gear run down maneuvers.....	80
Figure 3.23 :	FFT vs. Time spectrum for 4th gear run down maneuvers.....	81
Figure 3.24 :	FFT vs. Time spectrum for 3rd gear run down maneuver.....	82
Figure 3.25 :	White noise level readings of various mobile devices in test vehicle..	83
Figure 3.26 :	Amplitude distribution of various mobile devices in test vehicle.....	83
Figure 3.27 :	SPL increment distribution of each device for amplitude increases....	84

MOBİL CİHAZ MİKROFONLARININ KALİBRASYONU VE SES KAYNAĞI TESPİTİ İÇİN KULLANILMASI

ÖZET

Mobil cihaz teknolojilerinin hızlı ilerleyişinin de etkisiyle son yıllarda pek çok çalışmanın odak noktası, bu cihazlarda bulunan sensörlerin performansını arttırmak ve profesyonel amaçla kullanımını araştırmak üzerine olmuştur. Sensör teknolojisindeki kritik gelişmeler, tüm insanların en yakınında bulunan mobil cihazların artık yalnızca bir iletişim aracı değil, profesyonel bir ölçüm aleti olarak da kullanılabilmesine olanak sağlamıştır.

Bu tez çalışmasında, mobil cihazlardaki gömülü mikrofonların performansı irdelenmiş ve harici referans bir mikrofon ile kıyaslama metodu kullanılarak mobil cihaz mikrofonlarının kalibrasyonu incelenmiştir. Mobil cihaz mikrofonlarının performansı yarı yankısız oda ve farklı araç kabinleri olmak üzere çeşitli ölçüm ortamlarında ve cihazlarla test edilmiş, ölçüm yapılan ortamın mobil cihaz mikrofonlarının performansı üzerindeki etkileri araştırılmıştır. Kalibre edilen mobil cihaz mikrofonlarının profesyonel akustik uygulamalarda kullanımına dair öngörü edinilmesi amacıyla, ses kaynağının tespit edilmesine yönelik mobil bir uygulama geliştirilmiş ve Ford Transit üzerinde test edilmiştir.

İlk bölümde, mikrofon kalibrasyonu, mobil cihazların ses ölçüm aletleri olarak kullanımı ve ses kaynaklarının tespitini kapsayan geniş bir literatür araştırması yapılmış, mevcut metotlar incelenmiştir. Süreç içerisinde mobil cihazların mikrofon kalibrasyonu amacıyla geliştirilmiş endüstriyel bir cihazın henüz tasarlanmadığı gözlemlenmiş ve mobil cihaz kalibrasyonlarının Android veya IOS sistemlerinde tasarlanmış ses ölçüm uygulamaları üzerinde yapıldığı görülmüştür. Mobil uygulamaların ardında çalışan kodlarla birlikte cihazdan cihaza değişen mikrofon yapısı, işlemci ve filtrelerin etkisiyle alınan ses ölçümlerinde büyük farklılıklar olduğu bulunmuştur. Teknolojinin gelişimi ile birlikte mobil cihaz mikrofonlarında da iyileşme gözlemlenmesine rağmen bazı modellerde hala MEMS mikrofonların yetersiz kaldığı ve mobil cihazlar ile araç içinde ses kaynağı tespitine yönelik herhangi bir çalışma olmadığı belirlenmiştir.

Bölüm 2’de, mobil cihazlar üzerine bu çalışma kapsamında yapılan araştırmaların ardındaki teorik altyapı sunulmuştur. Temel düzeyde ses fiziğinden bahsedilmiş, günümüzde kullanılan mikrofon çeşitleri ve çalışma mekanizmaları üzerinde durulmuştur. Özellikle MEMS ve electret mikrofonlar günümüz mobil cihazlarında sıkça kullanıldığından, bu mikrofonların çalışma prensipleri yüksek önem arz etmektedir. Bu temellerin yanında, mikrofonların ve ses ölçüm cihazlarının kalibrasyonu üzerinde durulmuş ve mevcut metotlar üzerinden mobil cihaz mikrofonlarının kalibrasyonu bağlamında geliştirilebilecek yöntemler incelenmiştir. Tez süresince geliştirilen ve ses kaynağı tespiti amacıyla kullanılabilen mobil uygulamanın temellerini oluşturan veri toplama ve işlemenin özündeki noktalar kısaca belirtilmiş ve mevcut ses kaynağı tespit metotları listelenmiştir.

Bölüm 3'te, mobil mikrofon kalibrasyonu için öncelikli olarak yarı yankısız odada ve Transit kabinlerinde kurulmuş test düzeneklerinin detayları verilmiştir. Mikrofon kalibrasyonu, hassas bir işlem olduğundan her ekipmanın ve düzenek kritik önem teşkil etmektedir. Bu sebeple test düzeneğinin yarı yankısız odada ve Transit kabininde kurulumu tüm ayrıntılarıyla anlatılmıştır.

Deneyisel çalışmalarda Brüel & Kjær markasının tip 4189 kondenser mikrofon modeli referans mikrofon olarak kullanılmıştır. Test edilen tüm ortamlarda, ses ölçümü öncesi ve sonrasında Brüel & Kjær tip 4231 mikrofon kalibratörü yardımıyla mikrofonun kalibrasyonu LMS uygulaması üzerinden her seferinde dikkatli bir şekilde yapılmıştır. Mikrofon kablolarının ve diyaframının güvenilirliği, ana ölçümlerden önce yapılan referans sinyal ölçüm sonuçları incelenerek kontrol edilmiştir.

Mobil cihazların mikrofonlarının yönselliği yüksek olduğu için, test aracı kabininde ve yarı yankısız odada yapılan ölçümlerde sonuçlar referans mikrofon değerleri ile kıyaslanırken mikrofonların hepsi yerden aynı yüksekliğe ayarlanmış, ses kaynağına yönelimleri tripodlar yardımıyla birebir düzenlenmiş ve ses kaynağına uzaklıkları uzak alan şartını sağlayacak şekilde ölçülerek tüm mikrofonlar kaynaktan aynı uzaklıkta olacak şekilde sistem hazırlanmıştır.

Mobil cihaz mikrofonları ve referans kondenser mikrofon ile ölçülen değerlerde kıyaslamamanın doğru yapılabilmesi ve olası aliasing problemlerinin önüne geçmek amacıyla referans mikrofon ve mobil cihaz mikrofonlarının örnekleme frekansı aynı değere ayarlanmıştır.

Sinyallerde herhangi bir kirlilik olmaması için ses ölçümlerinde kullanılan sinyaller MATLAB ortamında yaratılmış ve doğrulukları Artemis programı üzerinde kontrol edildikten sonra sesler kaydedilmiştir. Referans sinyallerin yayılması esnasında bozunmaların önüne geçebilmek için, kalibre edilmiş harici bir hoparlör kullanılmış ve ses basınç düzeylerindeki artışlar bilgisayarın ses kartı üzerinde yapılmıştır.

Deneyisel çalışmaların ilki yarı yankısız odada referans ses basınç düzeyi değerlerinin belirlenmesi amacıyla belirtilen deney düzeneğinde yapılmıştır. Ölçümler sonucunda, mobil cihazlarda ölçülen değerlerinde cihaz bazlı kritik değişimleri gözlemlenmiştir. Bununla birlikte, mobil cihazların frekans yanıt karakteristikleri arasında büyük farklılıklar olduğu belirlenmiş ve bu durumun alınan ölçüm sonuçları üzerindeki etkileri irdelenmiştir. Yarı yankısız oda sonucunda alınan ölçümlerde kullanılan mobil cihaz mikrofon performansının referans kondenser mikrofon performansına yakınsayabildiği görülmüş ve mobil mikrofonlar için kalibrasyon değerleri hesaplanmıştır. Bununla birlikte, kullanılan mobil cihazlardan bir tanesinin akustik aşırı yüklenme noktasının, yarı yankısız odada test edilen değerlere yakın olduğu ve bu sebeple yüksek ses basınç düzeylerinde hatalı ölçüm yaptığı tespit edilmiştir.

Yarı yankısız odada kontrollü ortamda yapılan deneyisel çalışmaların ikinci parçası Ford Transit kabininde yapılmıştır. Buradaki ölçümlerde, gerçek dünyaya tekabül eden ve yansımaların mevcut olduğu bir ortamda mikrofon performansları kıyaslanmış ve test edilen ortamın ölçüm sonuçları üzerindeki etkisi incelenerek kalibrasyon değerleri belirlenmiştir. Bu aşamada kullanılacak araç kabini, kompleks yapı oluşturulması açısından Bus modeli olarak seçilmiştir. Kabin içinde alınan ölçümler sonucunda, mikrofon performanslarının genel itibarıyla yarı yankısız oda ile benzerlik gösterdiği, ancak kabin içindeki yansımalar sebebiyle ölçülen değerlerde yükselmeler olduğu gözlemlenmiştir. Bununla beraber, kabin içinde alınan ölçümlerde yalnızca belli bir frekansta çalınan sinyallerdeki tutarsızlığın yarı yankısız odaya kıyasla, araç içindeki kompleks yapıdan ötürü biraz daha yüksek olduğu belirlenmiştir.

Yarı yankısız odada ve test aracı kabininde alınan ölçümler sonucunda ulaşılan kalibrasyon değerleri, mobil cihazda ses kaynağı tespiti için kullanılan uygulama içerisine işlendikten sonra araç hareket halindeyken güç aktarma organı kaynaklı bir sorunun tespiti üzerindeki performansı irdelenmiştir. Aynı durumlar altında araç hızı kademeli olarak azaltılırken üç farklı kez ölçüm alınmış ve sorun kaynağı mobil uygulama yardımıyla tespit edilmeye çalışılmıştır. Alınan testler sonucunda, üç manevranın iki tanesinde ölçülen değerler, aracın sorunlu parçasına işaret eden tolerans bandı içinde kalırken, bir ölçümde mobil cihazın herhangi bir sorun tespit edemediği gözlemlenmiştir. Farklı bir Transit aracında aynı ölçümler tekrarlanmış ve elde edilen değerler incelenmiştir. İkinci araçtaki sorun daha belirgin olduğundan, mobil uygulamanın hata tespiti konusunda daha başarılı olduğu görülmüştür.

Alınan ölçümler sonucunda, özellikle 1000 Hz ve altındaki frekanslardaki belirgin tepe noktalarının mobil cihazlar tarafından başarılı bir şekilde tespit edilebildiği sonucuna ulaşılmıştır. Ancak, kullanılan mobil cihazın frekans yanıt karakteristiğine istinaden daha yüksek frekanslarda bozunmalarla karşılaşılabilceği de belirtilmiştir.

DeneySEL çalışmalar kapsamında son testler Ford Kuga kabini içerisinde farklı mobil cihazlar ile araç ses sistemi kullanılarak yapılmıştır. Bu testlerin yapılmasındaki amaç, farklı mobil cihaz modellerinin ölçtüğü ses basınç düzeyi değerleri arasındaki farklılıkların gözlemlenebilmesidir. Testler sonucunda, mobil cihaz değişimi kaynaklı ölçülen değerlerde büyük farklılıklar olduğu görülmüştür. Yarı yankısız odada ve Transit kabininde belirlenen referans kalibrasyon değerlerinin, bu sebeplerden ötürü tüm cihazlar için ayrı ayrı belirlenmesi gerektiği ve ortak bir değer kullanılamayacağı sonucuna ulaşılmıştır.

Sonuç olarak, mobil cihazlarda MEMS mikrofonlar kullanılmasına rağmen bu cihazların bazılarının, cihaz bazlı mikrofon kalibrasyonu sonrasında profesyonel ses ölçüm uygulamalarında yüksek kesinlik ve doğruluk ile kullanılabileceği belirlenmiş ve araç içinde mobil cihazlarla ses kaynağı tespitine yönelik örnek çalışmalarda başarıyla test edilerek önerinin doğruluğu ispatlanmıştır.



CALIBRATION OF MOBILE DEVICE MICROPHONES AND THEIR USE FOR SOUND SOURCE IDENTIFICATION

SUMMARY

With the advancement of mobile device technologies, many studies are focused on the improvement of the sensors within these devices thus searching a way to use mobile devices for professional applications. The critical improvements in these sensors allowed these devices to be used as professional measurement tools as well as a communication tool.

The main purpose of this thesis is to investigate the performance of built in microphones within the mobile devices and to calibrate the mobile microphone with comparison to a reference condenser microphone. The performance of mobile device microphones are investigated in different mediums to define the effects of these mediums on the measured sound pressure values. Moreover, the sound source identification of a vehicle is also conducted in this thesis with a developed android application on a mobile device after the calibration process is completed.

Firstly, a literature review on microphone calibration, use of mobile devices as sound level meters and sound source identification is conducted. It is found that a commercially available calibrator is not designed yet for the calibration of mobile device microphones. The current methods of calibration is also investigated and it is found that the mobile device hardware, software and filters greatly affect the measurement results.

The theoretical background on sound, microphones and their calibration, data acquisition and signal processing along with the sound source identification methods is also presented in the thesis briefly since these subjects become the foundation of this thesis.

Various experimental setups are prepared in order to test the capabilities of mobile microphones and to investigate the shift in their performance with regards to the measurement medium. The experimental setup is carefully constructed each time before the sound measurements to make sure every equipment is ready and working. A reference condenser microphone is compared to mobile device microphones during the first two experiments for signals that are generated on computer environment and transmitted via calibrated loudspeakers.

The experiments are conducted in a hemi anechoic room, a Ford Transit Cabin and a Ford Kuga cabin. The reference values are determined in hemi anechoic room and are the basis of other experiments. Later, the same setup is constructed in a Ford Transit cabin and the same experiment is conducted. The calibration coefficients are defined according to these two experiments. Moreover, the difference in behaviour of various mobile devices after a change in medium and their frequency response is also investigated. Acoustic overload point is found to be existent at lower sound pressure

levels for one of the tested mobile devices and the measurement results are found to be wrong in this device for values above this point.

After the calibration values are defined, the values are manually entered to the mobile application that is developed for the sound source identification of powertrain based issues in a vehicle. The application and the performance of mobile device is tested for various maneuvers in two different vehicles and issues. It is seen that the mobile application is able to define the root cause of the noise accurately for two of the three measurements conducted. Higher mobile device microphone performance in identifying the more significant noise sources is also identified as a results of these measurements especially for frequencies where the response of the microphone is flat.

The final experiment aimed to define the differences in behavior and measured sound pressure values for various mobile devices in a Ford Kuga cabin. As a result, it is found that there are significant differences in measured values for each mobile device. It is also deduced that the calibration values should be identified for each device model specifically as there is no general calibration value that can be applied to all devices.

Briefly as a result of this thesis, it is determined that some mobile devices can be used for professional sound measurement applications with high accuracy and precision after the device based calibration even though they are using MEMS microphones. The applicability of the argument and the performance of mobile devices on sound source identification within the vehicle is also proved with exemplary studies on vehicles under operating conditions.

1. INTRODUCTION

Traditionally, microphones with the required sensitivity, frequency range and polar directivities are used for professional purposes especially on the area of acoustics. The calibration of these microphones should always be conducted before the measurement takes place in order for the results to be accurate. There are many calibrators and systems available for the calibration of industry standard half inch and quarter inch microphones along with their respective adapters.

The advancement in the technology made many of the measurement devices more compact and mobile yet their accuracy is still one of the main issues that the engineers face today. Most of the people carry their mobile devices throughout the day and use them for sound recording, capturing images, measuring basic quantities, using the gyroscope and many other sensors for basic purposes. The use of mobile devices for professional purposes other than voice command or basic recordings is an area of high importance for many of the researchers.

Sound measurements and the assessment of noise with the mobile devices for professional purposes is the area of interest for this study. Since calibration is an aspect of utmost importance, a scheme should be found to calibrate the in built microphones of the mobile devices. The complexity of this task mainly comes from the unknown filters and various processors along with the diverse hardware options available for each of the mobile devices from tablets to mobile phones.

The capabilities of mobile devices as sound level meters are investigated in the past by a limited number of scientists and an equipment to calibrate the mobile device microphones is still not commercially available in the world. As newer technologies are used on mobile devices each year, the capabilities of in built microphones are increasingly being improved day by day.

The use of mobile devices for professional purposes brings a significant amount of cost reduction for measurement systems. In case of high microphone capability, mobile devices may eliminate the need for high price condenser or ribbon microphones

for a limited number of noise measurement applications. Also for calibration purposes, since the dependency to an external microphone calibrator is eliminated for mobile devices, this may also prove a great advantage. The root cause investigation of machinery, system or component failures are also currently done with professional high quality microphones. If in any way the mobile device microphone capabilities catch up with these high quality microphones or a scheme to use mobile devices for professional sound measurements are found, mobile acoustic root cause investigation will also be an advancement in the industry. Therefore, the investigation of the capabilities of mobile device microphones is an important area of research for acoustic engineers all around the globe.

1.1 Literature Review

The literature review is conducted in order to research the prior studies on the aspect of mobile device microphones and their usage areas. Possible in built microphone calibration methods are also investigated to get an overview of the current status in science with regards to this aspect.

The advancement in the technology brings the fact that the in built microphone capabilities of mobile devices are significantly increasing. In the literature, it is seen that many studies with various extents are conducted on the use of mobile devices as sound level meters. Among all in built microphones of mobile devices, the studies show that it is not possible to calibrate them with a calibrator. Some of the studies also include external microphones that can be used in conjunction with the mobile devices and their calibration can be conducted with a quarter inch B&K calibrator naturally but this doesn't resolve the fact that the calibration of mobile microphones are still not commercially possible.

Considering the professional applications in which mobile devices are used, there are not many practices of mobile device sound measurements for purposes other than scientific research. The nature and complexity of mobile devices make the issue a fog to be cleared before using these devices for professional sound measurement applications.

In the following section, the usage of mobile devices as sound level meters and the current studies on their microphone calibration is investigated. Moreover, the current

studies on the identification methods of sound sources are also described to get an overview of the process of sound source identification. Since these topics are the basis of the study, high importance is given to research the current studies and define the further steps of advancements in this area.

1.1.1 The recording chain of mobile devices

The smart devices are becoming widely popular with the advancements in technology. With the development of mobile devices, their area of usage is extended along with their technical capabilities. Thus the use of mobile devices as sound level meters and the capability of their microphones and processors are being investigated within the last few years.

The initial microphones that were used on mobile devices were electret microphones. As stated in Akustica (2011), the electret microphones have two main limitations, their non homogeneous acoustic characteristics and the large size of the microphones. The sensitivity and background noise levels of these microphones are not the same all around the microphone itself. Due to this problematic structure of the electret microphones, MEMS microphones are developed and are being used since 2010s with better sensitivities and homogeneous structures. The good dynamic range, small size, great signal to noise ratio and flat frequency response make MEMS microphones ideal for mobile devices [1].

The microphone properties of mobile devices are not the only aspects to be considered when the purpose of being a sound level meter is investigated. The ADC within the mobile device, the placement of the microphone inside the device also affect the sensitivity of these devices on sound level measurements. In addition to these parameters, mobile devices also use some audio processing techniques such as noise suppression, equalization and automatic gain control. According to Android (2013), automatic gain control is a process used to normalize the output of the received sound signal by increasing or decreasing the input from the internal microphone to match a preset level. This process allows the output signal level to be virtually constant [54].

Since the built in microphone properties of the mobile devices differ significantly, some external microphones can also be used with the audio input of these devices to bypass the internal microphones. These external microphones generally exhibit better directivity characteristics, better sensitivities and signal to noise ratios than the internal

microphones. The use of such microphones are the generally followed procedure to use the mobile devices as sound level meters. For instance, the mobile MicW i436 is an example of these external microphones to be connected to the audio port of the mobile device. As highlighted by Faberacoustical (2012b), this microphone has a flat frequency response and omni directional characteristic and is specially designed to meet Class 2 international sound level meters standards. The production size of this specific microphone also allows the calibration of the microphone with a sound calibrator [2].

Along with the MicW i436, some other external microphones are also being used with mobile devices for better recording capabilities. The iTestMic microphone that is produced by StudioSixDigital has an integrated microphone preamplifier, USB interface and analog to digital converters. The price of these microphones are higher when compared with MicW i436 due to better sound characteristics. These microphones also meet the international standards for Class 2 sound level meter specifications but are limited with iOS devices only. Another external microphone that is used with mobile devices is Dayton Audio iMM-6 with similar properties to MicW i436 with the exception of unavailability of Class 2 sound level meter certification. Kardous et al. (2016) stated that the comparison of MicW i436 with Dayton Audio iMM-6 gave +/- 1 dB results within most frequency ranges but the difference between these microphones can be minimized with calibration [3].

While using mobile devices for sound level measurement purposes, raw audio can't be obtained by default due to internal processing of the sound. Typically, audio decoder, a predetermined filter and a noise level calculator along with an integrated calibrator is included in sound level meter applications. By including these components in the sound level meter applications, digital audio metrics are aimed to mimic the same metrics in standard noise exposure instruments. The filter component is also another factor that needs to be used for precise measurements. Generally, A or C weighting filters are designed and used to meet the international IEC standards [4].

In order to improve the quality of the sound level measurement application, the noise levels can be manually adjusted by the user. According to a study on mobile noise measurements by Rana et al. (2010), as implemented in Ear-Phone application the sensors that are included in the mobile device (inertia sensor, GPS receiver, proximity sensor) can be used to detect the location of the mobile device. The location of the

mobile devices poses a great importance since it shapes the characteristics and accuracy of the measurements with regards to the real values. When the mobile device is located in a bag or pocket during the measurements, the application shouldn't take these measurements into account while calculating the results due to the low quality of these measurements [4].

The current studies prove that the measurement process and the credibility of the obtained results depend on the internal data processing steps that are built-in within the mobile device. The properties of mobile devices define the procedures to be followed to obtain the best results while using the mobile device as a sound level meter. Calibration of these mobile devices becomes much more important in this case to obtain accurate results during the measurements.

1.1.2 Calibration of mobile devices for SLM purposes

The calibration of professional sound measurement devices are normally conducted with internationally standard calibrators. The calibrators used in such cases are suitable for each microphone types with the use of appropriate adapters. In traditional calibration process, a standard sound pressure level of 94 dB is applied under a specified frequency of 1000 Hz and response of the microphone is calibrated accordingly [5].

For mobile devices such as phones however, this approach is not feasible since there are no calibrators applicable for these devices. The main purpose of the calibration in this case is to eliminate the errors that may arise due to the internal structure and processors of the mobile device. Stevens (2012) has the most extensive research on the calibration of such devices and follows a phone model specific approach to calibrate the devices. In the study, the calibration values for a few models are stored on the database and calibrated accordingly when the application launches [5].

The calibration process of mobile devices can be followed in a level dependent, frequency dependent or both level and frequency dependent procedures. Also as stated by Stevens (2012), a digital filtering technique with FIR or IIR filter design can be applied and the coefficients of these filters can be deduced from calibration measurements yet this study hasn't been conducted. In an investigation by Brown and Evans (2011), the application of Real Time Analyzer by Studiosixdigital is told to be implementing a frequency dependent algorithm yet no details are shared or described about the calibration process. Most of the sound level meter applications on the market

today are using factory default calibration values that are stored on a database or level dependent trimming algorithms to compensate for the effects in measurement chain of mobile devices [6].

The algorithms used today consist of a trimming and scaling operations. The trimming algorithm works by adding or subtracting the measured value with regards to a reference while the option of scaling operation just multiplies the response of the mobile device with a coefficient to match the responses with the reference signal. However, according to Stevens (2012), a combination of both methods can also be used to calibrate the mobile device. For this method, at least two points for calibration are measured. In his study, various calibration points are taken between 30-105 dB(A) and a white noise calibration is conducted with trimming and scaling algorithms. Factor is defined by measuring the slope between these values while y axis interception is used for the trimming [5].

Ventura et al. (2017) proposed a different calibration method by using the narrowband signals and pink noise reference signals given in a controlled lab. The anechoic chamber measurements are done comparing both specific models of android devices and a reference condenser microphone. As a result of estimating the bias and differences between mobile device and reference microphone measurements a calibration is done and results within ± 1.3 dB are obtained within the range of 45 dB and 75 dB [7].

Frigerio (2019) also discussed the use of mobile devices as sound level meters with working place findings. The gain is calculated for each mobile device within a room with 75 dB(A) steady noise level through the measurements when the closest difference between the reference sound level meter is obtained. The obtained results show that only 10 % of the measured devices fall within ± 0.5 dB(A) while 70 % of them are within ± 3 dB(A) of the SLM readings. The best results in this case are said to be obtained within 75-85 dB(A) range [8].

1.1.3 Performance and accuracy of mobile devices for sound measurements

The accuracy of standard sound level meters are defined with IEC and ANSI standards with regards to their capabilities. Class I microphones pose a better and flatter frequency response with less deviations from the real sound pressure values while Class II microphones are slightly worse than Class I microphones in term of their

frequency response. In order to use the mobile devices for professional purposes, their frequency response and sound measurement capabilities should converge to Class I and Class II microphones to be precise. There are no specified requirements or targets for mobile devices to be used for sound measurement purposes so far [9].

SafetyAwakenings (2013) investigated the accuracy of sound level measurement applications on IOS system using the iPad and the built in microphone. The measurements are stated to be taken outside of the acoustical laboratory with 4 types of noise sources including pure tone signal of 1 kHz and 8 kHz, 30 second sound recording and a prerecorded explosion. The default calibration settings are used for the sound level meter applications and the measurements are compared with a Class II reference microphone. Out of the 30 applications, 13 applications are proved to be inadequate to clarify the need for calibration of the mobile devices [9].

The noise tube application by Stevens (2012) is thought to be the most extensive study on the subject of mobile device sound measurements. The calibration of the system consists the testing of 16 noise levels of white noise signals in an anechoic room. Even though the calibration of the system was model specific for each mobile device, Stevens (2012) reports that an average difference of 1 dB up to 0.6 dB is found for noise levels below 100 dB(A). These results show that the capability of the calibrated mobile devices were not enough to match the strict Class II sound level meter specifications [5].

Serpanos et al. (2018) investigated the use case of mobile devices with sound level measurement apps and compared the results with a reference calibrated B&K Type 4189 microphone for noise level higher than 40 dB. The sound level meter applications are calibrated according to the reference signal that is generated using GSI Audiostar pro audiometer with 1000 Hz narrowband noise and white noise from 20 dB up to 100 dB for 8 different environments. The sound level meter and mobile device are put at 1.5 inch off the table able, 1.2 m from the speaker and 1.2 m from the floor. The outcomes of the study shows that the measurements are very inaccurate below 50 dB when using calibrated or uncalibted mobile devices for sound measurements. This is most probably due to the high internal noise, low sensitivity and accuracy of the mobile devices. The study shows that mobile devices are not capable of correctly measuring the noise in quiet environments [10].

Murphy (2016) also investigates the performance of sound level measurement applications with 100 smartphones both IOS and Android within a reverberation room to eliminate the directivity effect of the microphones. White noise with 3 different levels of 50, 70 and 90 dB(A) respectively is transmitted from a loudspeaker and is used to compare the signal with the reference microphone. As a result of this study, 3 out of 7 applications proved worrying results with being significantly below the reference noise levels. Moreover, the age of the phone, the deterioration of the built in microphones and the smartphone manufacturers are said to be important aspects that affect the result of noise measurements [11].

The placement effect of the mobile devices on the sound measurements are studied by Romain (2014) in his thesis work. Along with the placement effects, external microphones and external earsets are also used to analyze the sound signals. The effects of frequency dependent and level dependent calibration algorithms are studied and a level dependent calibration is used for sound measurement corrections. The sound level measurement application is written on Java structure for this study and the effects of calibrated – uncalibrated mobile device measurements are compared. As a result of this extensive study, the difference between a laboratory controlled environment and real world environment calibration remains unresolved [12].

1.1.4 Sound source identification

The identification of sound sources is of utmost importance for the diagnostics of machinery especially while searching for the root cause of the error state. There are many studies that are conducted on sound source identification yet only a few will be reviewed here as those are the ones that pose importance for the area of interest of this thesis.

Helmholtz least squares method is a popular method to detect the noise sources when used in conjunction with nearfield acoustic holography to visualize the acoustic pressure, intensity and velocity. The helmholtz least squares method has a similar working procedure as modal analysis since it shows the acoustic pressure as a superposition of spherical wave equations or any other shape functions that are the solutions to Helmholtz equation. In this method, the amplitudes of shape functions are determined by comparing and then matching the measured sound pressure level around the source and the assumed solution to Helmholtz equation. Benival et al. (2005)

discussed in their study the usage of HELS method on identifying the sound source. A vacuum cleaner is investigated in the study and noise diagnosis is done using this method. Sound pressure level measurements are initially done 2 cm away from the source and then are projected to 0.5 cm. Then, 56 microphones are used to measure the sound pressure level at 0.5 cm away from the vacuum cleaner and the main source of sound is found to be the canister vibration caused by the fan at low frequencies. In the study, a microphone array is used to conduct the near field sound measurements and the sound source is successfully determined [13].

Data processing techniques can also be used to identify the sound sources. Watanabe et al. (2016) have developed a sound source identification system to distinguish the noise source on sub assembly level by combining the sound pressure level measurements and timing analysis conducted on these measurements. In the study, the time data taken from the sound source is analyzed with a high resolution instantaneous frequency analysis. The instantaneous frequency is calculated from the temporal changes in phase with a higher resolution than short term Fourier transform. By using this method, they were able to distinguish the noise sources at a component level and detect the main noise source in the system up to 30 mm components. The exemplary study on the usage of this method is the investigation of a fuel pump along with its components. By using the IFA method, the injector, pump, inlet and outlet valve noises are differentiated from each other and identified using both frequency and time analysis [14].

Wang et al. (2019) also used signal processing techniques to identify the sound source. Ensemble empirical mode decomposition method is used to decompose a time series signal with the assist of white noise into an intrinsic mode function with different central frequencies. In the study, an electric bus powertrain is analyzed on a dynamometer with a 24 channel data acquisition system. The motor is operated on 2nd and 3rd gear ranges and the operating zones have been divided into 3 sections namely high speed low torque, low speed high torque and general operating zone. By using the EEMD method, the signals are decomposed and robust independent component analysis is used to extract the noise sources of interest. Then by using FFT and continuous wavelet transform functions on the data, the independent components of the motor are identified successfully. Even though the method seems complex, the usage of continuous wavelet transform along with the Fourier transform is a general

method that can be put the use on noise source identification projects since continuous wavelet transform can preserve phase information and provide adaptive frequency resolution [15].

Mathew et al. (2014) developed an algorithm to identify the noise sources. In the algorithm, there are 2 separate main blocks, sound detection and sound recognition. When an abnormal situation is sensed by the microphones at the area the sound signal is collected and is given to the detected as a special sound. The special sound is then sent to the recognition block and the noise is recognised as a previously heard noise like a phone ring. This is done by training the algorithm for both threshold calculation using background noise and acoustic fingerprint extraction from collected sound signals. The recognition phase is only conducted when an abnormal sound collected by the microphones and then frequency analysis is conducted to match the sound with a previously heard one. The disadvantage of this study is seen when the signal of interest is not unique If the algorithm is not trained for the typical sound source of interest, then the detection phase may not be completed successfully [16].

Engine fault detection is a special area of study in automotive industry that many companies put an emphasis on. Aly (2013) has studied a multi level signal processing technique and combined it with a search in a predefined database for 26 various engine faults. For this study, 10 accelerometers are placed on the engine while operating to pick up the vibration signals. A band pass signal has also been used to limit the frequencies to a range of 20 to 20000 Hz. A defective right phaser on an engine is investigate in the case study of the research. By running a wavelet packet transform on the vibration signal, the corresponding crank angle is determined. When the algorithm detects a high energy level on the wavelet vectors, it looks up the predefined database to find the corresponding error. If the program is able to find and identify the sound source by looking at the predefined table, then the description appears on the screen. If the signal of interest is an unknown spike, a warning message is also provided to the end user [17].

Verma et al. (2014) have conducted a study on condition based monitoring for industrial machines and the fault detection on platforms Windows Mobile and Windows Tablet. The acoustic sound measurements and data mining techniques are mainly used to identify the healthy and faulty states of the machinery and the developed smartphone application is trained to identify a reciprocating type air

compressor. The acoustic pattern recognition is the foundation of this study. The recognition task uses a supervised learning method and the application is trained by pre-recorded sound measurements to identify the state of the machinery. During the training and testing phases, there are a total of 5 phases conducted, namely data acquisition, data processing, feature extraction, feature selection and classification of the sound signal. As a result of the research, an inlet valve leakage is identified on the machinery by seeding the application with both the healthy and erroneous states before hand. The features regarding the inlet valve is extracted and most relevant ones are chosen for the classification of the signal. 18 of the 20 faulty states were identified on the machinery on the study [18].

1.2 Procedure of the Study

The main purpose of this thesis is to devise an experimental setup to calibrate the built in microphone of a mobile device and compare the capabilities of mobile device microphones with regards to condenser microphones in terms of accuracy in professional applications. A mobile application is also developed to identify the sound source with the measurements taken from a mobile device microphone.

Before taking the vehicle sound measurements with the mobile device to identify the sound source, initial experiments are conducted to define the capabilities of the mobile microphone and calibration curves are determined to increase the accuracy of the microphone. During the process, four different mediums are used. One of these mediums is the hemi anechoic chamber in which every condition is controlled and the other three mediums are vehicle interior cabins in which reflections and outside factors are also existent. The reason to use these mediums is to define the change in performance of the mobile microphones under real life conditions along with the experimental conditions. The following procedures are used to define the capabilities of mobile microphones and their use in professional applications;

- The test setup in hemi anechoic chamber is prepared carefully to minimize the reflections and to fully control the environment.
- A calibrated loudspeaker and an external sound card are used to transmit the signals in order to eliminate the errors that may arise from the hardware.

- The frequencies of interest are generated via MATLAB and their frequency spectrum is checked on Artemis software to prepare a pure and clean signal.
- B&K condenser microphones are taken as reference microphones. Their calibration is conducted before and after the experiment with LMS tool and external calibrator under 1000 hz 94 dB to increase the accuracy of the measurements.
- Squadriga II mobile measurement system is used with B&K microphones. The sensitivity inputs are done according to the levels of the signals of interest. Cable connections and system robustness is checked before the experiment.
- All measurements are taken at the same time with sound measurement devices attached to tripods. The tripods are placed with 1 meter height from the surface and 1.5 meters distance from the sound source to satisfy the far field conditions. The microphones are directed to the sound source during the measurements.
- The calibrated loudspeaker is placed at the same height with the microphones to eliminate the effect of difference in height for sound source and receivers.
- Vehicle interior cabin measurements are taken with the same equipment and the same test setup. The effect of difference in sound field on mobile microphone performance is investigated.
- The calibration curves are defined from the hemi anechoic and vehicle interior cabin measurements. White noise signal is taken as reference signal since it contains a wide range frequencies.
- Vehicle interior cabin measurements are repeated inside another vehicle with various mobile devices to compare the capabilities of mobile devices and the difference in their performances for sound measurement purposes.
- Calibration is conducted on the developed mobile application and vehicle measurements are taken under various maneuvers to identify the sound source. The performance of mobile devices in identifying the sound source is determined and compared for various maneuvers.

1.3 Objectives of the Thesis

In this study, it is aimed to calibrate a mobile device's microphone and increase the capabilities of the mobile device for professional applications. The performance of various mobile devices under hemi anechoic room and vehicle interior cabin is also investigated during the thesis. Moreover, the capabilities of the calibrated mobile device's microphone on sound source identification within the vehicle is also analyzed. The objectives of this thesis can be briefly summarized as follows;

- The investigation and comparison of the capabilities of an uncalibrated mobile device's microphone with a reference condenser microphone both in a controlled and an uncontrolled environment under various conditions.
- The investigation of the effects of reflection from nearby surfaces, background noise, sound field and variation of the mobile device on sound measurements.
- The effectiveness of mobile microphone calibration method and the improvement in their accuracy after the calibration will be investigated. Comparison of the calibrated mobile device with regards to a reference microphone under the same conditions will be done.
- Devising a method for sound source identification within the vehicle using the mobile device microphones.
- Assessment and comparison of each result to conclude the capabilities of the various mobile devices and determination of the effectiveness of using mobile devices for sound source identification purposes.

1.4 Outline of the Thesis

An initial introduction is given in Chapter 1 in order to present the literature review of advancements and previous studies on mobile devices for sound measurements. A wide investigation is performed on the subject of mobile device microphone calibration and their capabilities for sound measurement applications. A review of sound source identification methods and previous studies are also presented since the thesis aims to implement a sound source identification method with the use of mobile device microphones.

Chapter 2 explains the background theory behind the basic concepts that are used or investigated within the thesis. A brief review of sound characteristics and sound metrics is conducted. The working principle and measurement errors of sound level meters are briefly given since the calibration of the microphone needs to be conducted on a sound level meter application within the mobile device. Since various microphones are of interest for the experimental studies during the thesis, the working principles and performance metrics of microphones are researched. The calibration methods of microphones in the literature and industrial standards are also reviewed in chapter 2 in order to get an overview of the existent calibration methods. The chapter is finalized with a general view of data analysis and processing as the thesis aims to apply a sound source identification method with a mobile device.

In chapter 3, the experimental studies conducted both in a controlled hemi anechoic room and vehicle interior cabins are included. The effect of sound field and sound receiver variation on measurement results is investigated. Calibration curves are defined with regards to the measurements that are conducted in these mediums. Finally, the performance of a calibrated mobile device on identifying the sound source within the vehicle under various maneuvers is tested and results are presented.

In chapter 4, general discussions regarding the tests conducted in various sound fields with different mobile device microphones are done and conclusions are drawn. Recommendations for future studies are approached.

2. BACKGROUND THEORY

This chapter covers the general theoretical principles and methods that are involved during the thesis study. The physics of sound is briefly explained since the propagation of sound waves are important for the sound measurements of interest. Along with this, the working principles of various microphones are presented since the comparison of different microphones will be done throughout the research. Moreover, sound level meters and their calibration, basic data acquisition and processing techniques are also investigated. The background theory overview is finalized with a brief look on sound source identification methods.

2.1 Fundamentals of Acoustics

Sound and noise lies in the basics of acoustics. Sound, in short is defined as the pressure variation or the vibration of the particles in a specified medium. A graphical representation is given in Figure 2.1 to show the pressure variations around the atmospheric pressure. The vibrations vary according to the densities of the mediums, therefore the sound produced in each of these mediums are different from each other.

Noise, on the other hand is the perceiving of sound and is undesirable by the customer. The noisy disturbance should be avoided whenever possible to prevent environmental pollution since it is unwanted. Noise is a factor that needs to be eliminated for comfort. In order to achieve this, source, transfer path and receiver need to be identified and investigated using the sound and vibration measurements along with signal analysis. Therefore by following these steps, the root causes of noise can be identified and resolved.

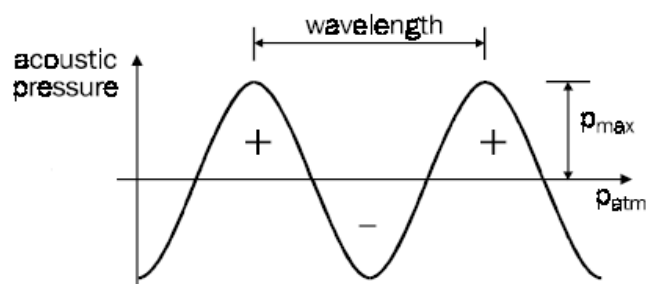


Figure 2.1 : Pressure variations above and below atmospheric pressure [19].

Sound and noise are distinguished from each other. Both results from the vibrations and pressure variations in the medium but from a point of view, what is noise to a person may not always be heard this way to another person, any sound can also be perceived as noise by somebody else, as an undesired sound. Just to put another example, an amplified classical music can be perceived as a beautiful sound by many, but when the amplification goes beyond comfortable limits, then it may become noise to some people.

2.1.1 Frequency and period

Frequency is generally defined as the number of vibrations or cycles in the relevant medium within a specified unit of time. In acoustics, simply this is the number of cycles per second and used with the unit Hertz (Hz). A sound measurement that is taken for a chosen duration is usually a combination of many different frequencies. Human ear is sensitive to frequencies starting from 20 Hz up to 20000 Hz.

Wavelength (λ) is affected by the frequency component of the sound. The distance that a sound wave travels within the duration of a single cycle is called wavelength. Higher frequencies have lower wavelengths.

Period is the time that the system takes to complete one cycle and has a strong relation with frequency. The inverse of frequency is period and is usually denoted as T (seconds). The relation between frequency, period and wavelength can be seen on Figure 2.2 for a better understanding. As it can also be seen from the figure, with increasing frequencies the wavelength decreases and the period of the signal linearly decreases.

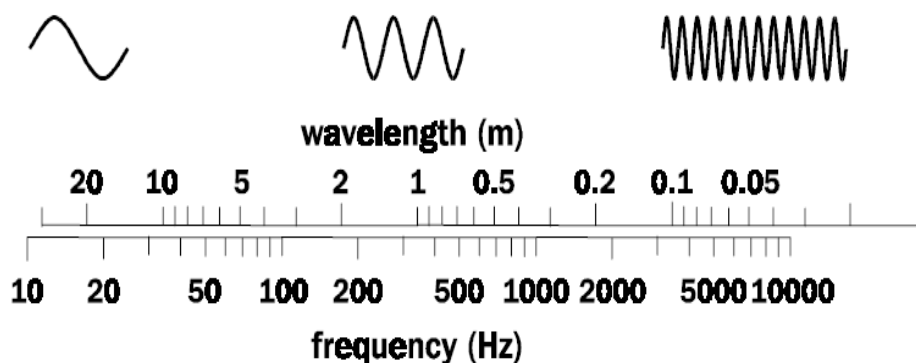


Figure 2.2 : Wavelength and frequency illustrations under normal conditions [20].

2.1.2 Root mean square

While analysing the signals, investigating the average amplitude of the signal may not give the best results. If we take a sinusoidal for reference on Figure 2.3, the average of the signal corresponds to zero considering it has a negative peak as well as a positive peak. If an average amplitude of 0 is considered for a sine wave, then it can be said that this is not a correct representation of the signal since it is not always 0. The RMS value, however is defined as the effective value since the square root of the time average is taken in order to define the signal. p_0 being the peak value for pressure, RMS can be calculated as below :

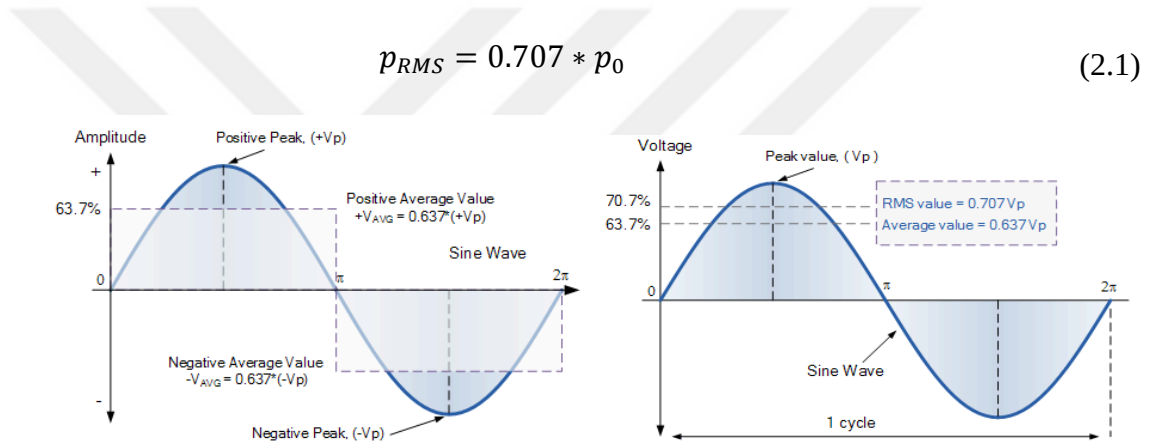


Figure 2.3 : Sine wave average and RMS values representations [55].

2.1.3 Sound intensity

As sound waves propagate through a medium, they also transmit the energy they carry within. Sound intensity is the time average rate of the energy that the sound waves transmit and is a vectorial property since the transmitted energy also carries direction information. The unit for sound intensity is W/m^2 .

The determination of sound intensity requires the knowledge of both the pressure and the particle velocity along with the directional information. The instantaneous sound intensity $I_i(r, t)$, at a specified location is an expression of the field vector r and the vectorial particle velocity $u(r, t)$.

$$I_i(r, t) = p(r, t) * u(r, t) \quad (2.2)$$

The overall sound intensity however, requires the time average of instantaneous sound intensity to be described. General expression for sound intensity is given below :

$$I(r) = p(r, t) * u(r, t) = \lim_{T \rightarrow \infty} \frac{1}{T} \int_0^T p(r, t) * u(r, t) dt \quad (2.3)$$

Sound waves that transmit the energy away from a source becomes planar as the travel path increases in the far field. When a planar wave is considered, the acoustic impedance $\rho c = \frac{p}{u}$ is a characteristic quantity that defines the wave and thus both the particle velocity and sound pressure are in phase during sound power transmission. Since scalar magnitude of the acoustic intensity is of significant interest in this case, the substitution of u into the equation 2.3 gives an alternative equation for intensity at location r :

$$I = \frac{\langle p^2(r, t) \rangle}{\rho c} \quad (2.4)$$

2.1.4 Sound power

The sound that radiates from a source also transmits the power that is contained within itself. This characteristic is called sound power and is described in Watts. Considering that the equation 2.4 holds true for radiated sound waves in general at a specific distance away from the source, the expression for sound power W can be stated as below :

$$W = \int_{S_1}^{S_2} I * n * dS \quad (2.5)$$

On the general expression for sound power W at equation 2.5, n is the normal vector to the surface of area S . This expression implies that the direction of propagation is an important factor that affects the sound power. Accordingly, some sources radiate more power in some directions than the others.

If the sound source is considered to be nondirectional, it is a spherical sound source. When such sources are of interest, the sound intensities at all points on an imaginary spherical surface around the source are equal as seen in Figure 2.4. The acoustic center of the sound intensity is the center of the source itself.

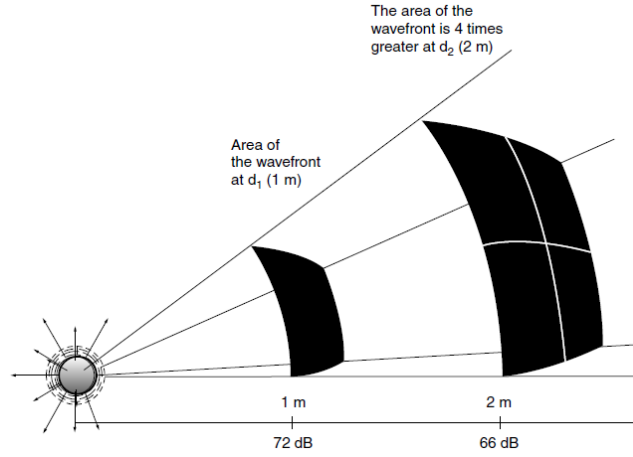


Figure 2.4 : Generation of a 1-D spherical wave [21].

The sound power of such source can be expressed with equation 2.6 :

$$W_s = (4\pi r^2) * I_s * (r) \quad (2.6)$$

Where I_s is the maximum sound intensity at radius r , W_s is the total sound power radiated by the source and is in Watts and r is the distance from acoustical center of source to the imaginary sphere surface. Considering equation 2.6 for line sources, the expression for the equation changes since for line sources, a cylindrical surface around a cylindrical sound source is investigated.

$$W_c = (2\pi r l) * I_c * (r) \text{ Watts} \quad (2.7)$$

Where W_c is the total power radiated by the cylindrical source of length l and is calculated in the unit of watts.

2.1.5 Sound pressure and levels

Pressure is another metric that needs to be considered for a propagating sound. Sound pressure can easily be detected but because of the wide range of sound pressures that a human ear can respond to, this characteristic quantity is not suitable to be used in graphs and tables. Due to this reason, a logarithmic scale is used to define the sound pressure in units rather than using a linear scale. Thus, the magnitude or the level of acoustical quantities such as power, pressure or intensity is given with a logarithmic scale called dB. While comparing these quantities in decibels (dB), a zero reference level is taken for a reference quantity and two levels are compared according to this reference level.

The level of sound pressure p is defined as L_p decibels above or below a reference sound pressure p_{ref} according to the equation 2.8 is given below :

$$L_p = 10 \log_{10} \left[\frac{\langle p^2 \rangle}{p_{Ref}^2} \right] = 10 \log_{10} \langle p^2 \rangle - 10 \log_{10} (p_{Ref}^2) \quad (2.8)$$

In order to define the reference sound pressure, the lowest sound pressure that a human ear can detect is considered. The sound pressure p_{ref} in this case is $2 \times 10^{-5} \mu Pa$. The result of L_p is dB in this case and this is the main quantity that is generally expressed in sound level meters that can be found in the industry. Some overall sound pressure levels can be seen in Figure 2.5 below for reference.

Sound pressure level (dB re 20 μPa)	Description of sound source	Typical subjective description
140	Moon launch at 100 m; artillery fire, gunner's position	Intolerable
120	Ship's engine room; rock concert, in front and close to speakers	
100	Textile mill; press room with presses running; punch press and wood planers, at operator's position	Very noisy
80	Next to busy highway, shouting	Noisy
60	Department store, restaurant, speech levels	
40	Quiet residential neighbourhood, ambient level	Quiet
20	Recording studio, ambient level	Very quiet
0	Threshold of hearing for normal young people	

Figure 2.5 : Overall sound pressure levels for common sound sources [19].

Sound power level and sound intensity level follow the same principle as the sound pressure level. The standard reference power level in this case is $W_0 = 10^{-12}$ watts while it is $I_0 = 10^{-12} W/m^2$ for sound intensity level. The corresponding equations 2.9 and 2.10 for sound power level and sound intensity level can be seen below :

$$\text{Sound Power Level} = L_W = 10 \log_{10} \left(\frac{W}{W_0} \right) dB \quad (2.9)$$

$$\text{Sound Intensity Level} = L_I = 10 \log_{10} \left(\frac{I}{I_0} \right) dB \quad (2.10)$$

2.2 Microphones

Sound measurements are mainly conducted in order to describe the related sound characteristics and for further investigations and analysis. These measurements are generally seen in terms of sound levels on graphics and are taken with electro

mechanical conversion systems called microphones. Microphones convert the acoustical vibrations and the energy to electric energy. Every microphone has a small diaphragm that is excited with acoustic waves and as a result of this excitation and the movement of the diaphragm, an electrical signal is generated. This electrical signal is passed through the internal converters of the microphone thus resulting in an output voltage that we can be heard as sound signals. A basic MEMS and an electric condenser microphone structure can be seen below on Figure 2.6.

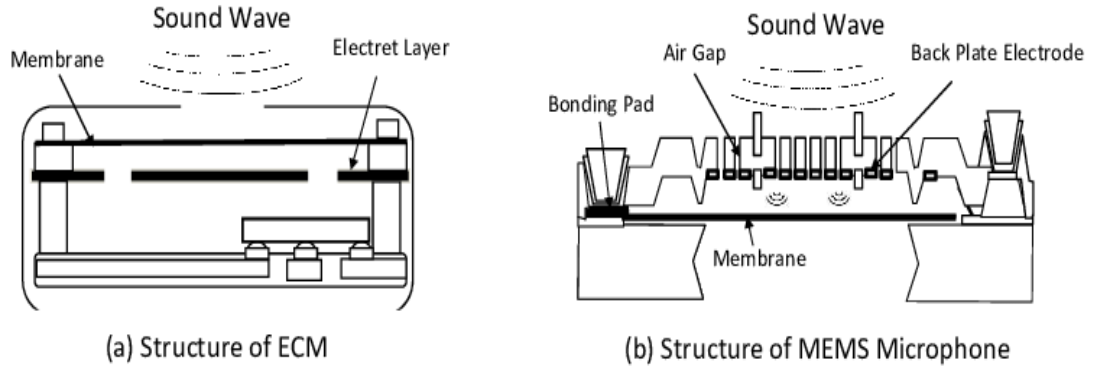


Figure 2.6 : A basic structure of an electret and a MEMS microphone [22].

2.2.1 Microphone sensitivity

Microphone sensitivity can be defined as an indication of the electrical output that is resulted from the sound pressure which is applied on the membrane. While considering the microphone sensitivity, 1 Pascal at 1000 Hz and the corresponding sound pressure level of 94 dB is used. In order to calculate the microphone sensitivity, the equation 2.11 below can be used. In this equation, e corresponds to the output voltage level in terms of RMS and p corresponds to the acoustic pressure on the diaphragm in terms of pascal.

$$Sensitivity (dB) = 20 \log_{10} \left(\frac{e}{p} \right) \quad (2.11)$$

Microphones are also rated with regards to their sensitivities. The microphones with higher sensitivities can gather quieter sounds since their diaphragms are more sensitive to pressure fluctuations on the membrane. Considering the area of usage, a higher sensitivity microphone is not always the best choice. If we take an example, a higher sensitivity microphone would be convenient in order to record various bird songs and to differentiate the birds later on easily but when this microphone is used in a conference room it may cause significant issues. Because contrary to the bird songs,

the microphone needed for this area should have low internal noise and medium sensitivity to distinguish the speeches only. As it can be investigated on Figure 2.7 the highest sensitivity can be achieved with largest microphones since they give the highest voltage output while the smallest microphones can measure higher frequencies.

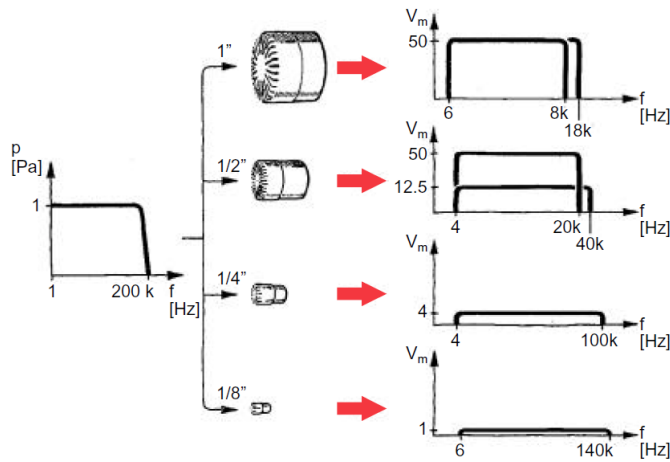


Figure 2.7 : Frequency range and sensitivity of microphones [51].

Most of the voltage levels created by the microphones are less than the reference voltage which takes 1 voltage = 0 dBV under 94 dB SPL as reference. Sensitivity is mostly negative because of this reason. As negative sensitivity reaches 0, it means that the microphone has higher sensitivity. For example, a microphone with -40 dB sensitivity has higher sensitivity than the one with -55 dB sensitivity.

2.2.2 Frequency response

Frequency response is an indication of the output that a microphone or a sensor gives between the minimum and maximum frequencies it can measure. In other words, the change in sensitivity of the microphone with regards to frequency is called the frequency response of the microphone. As shown on Figure 2.8, the frequency response of microphones become frequency dependent as the frequency of interest increases. Especially at high frequencies, the response of the microphone fluctuates. The limit of 2 dB below the reference 0 dB line is the cut off frequency.

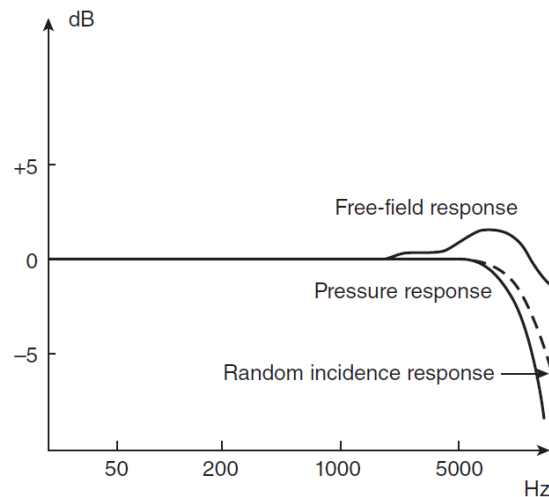


Figure 2.8 : Frequency responses of various microphone types [23].

Microphone types are characterised considering their frequency responses.

- a) Free Field Response Microphones
- b) Pressure Field Response Microphone
- c) Random Field Response Microphone

These three types of microphones are used for various situations under the effect of a sound field. Free field response microphones are generally used in anechoic chambers where sound comes from one direction to the microphone. These microphones compensate for the pressure build up on the membrane for the sound diffraction and other sounds which may interfere with the main signal [24]. The wavelengths of the affecting sounds are significantly large when compared to the physical size of these microphones. Thus, the presence of these microphones doesn't have an effect on the measured sound field, it just measures the real changes in pressure. The response of these microphones are called free field response.

A microphone with linear frequency response to the actual diaphragm pressure is called pressure response microphones. Contrary to the free field microphones, these microphones don't compensate for the pressure build ups on the membrane of the microphone. Thus, these microphones just measure the real pressure level on the membrane. A pressure field microphone can also be used instead of a free field microphone with the right orientation but this reduces the frequency range of pressure field microphones. When making these measurements on a diffuse field where the sound is reflected from every surface around the source, random field response

microphones are used. Some examples of these sound fields and the usage of these microphones can be seen on Figure 2.9.

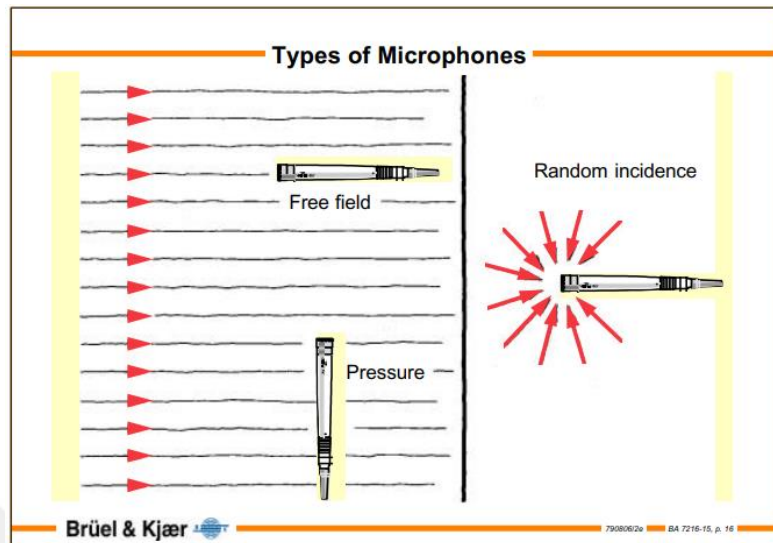


Figure 2.9 : Three types of microphone by their working fields [51].

2.2.3 Dynamic range

Dynamic frequency range of a microphone is defined as the difference between the largest and the smallest SPL which a microphone linearly responds to. The dynamic range of a microphone is not the same thing with SNR. SNR of a microphone measures the difference between a reference level of 94 dB SPL and the existing noise floor of the current medium but the microphone still responds linearly to these signals up to its acoustic overload point above 94 dB SPL.

Figure 2.10 shows the dynamic range of a microphone along with the difference between dB SPL and dBFs outputs of a microphone. The dynamic range here is the difference between the equivalent input noise level and the acoustic overload point of the microphone – as shown on the figure the difference between 29 dB SPL and 120 dB SPL. On the same figure, the acoustic overload point corresponds to 0 dBFs output signal. Briefly, dBFs describes how quiet a sound is with regards to a reference level. Since the full scale of the acoustic input is put to this value, the digital microphone has a sensitivity of -26 dB. This is due to the reference sound pressure level being measured at 94 dB SPL.

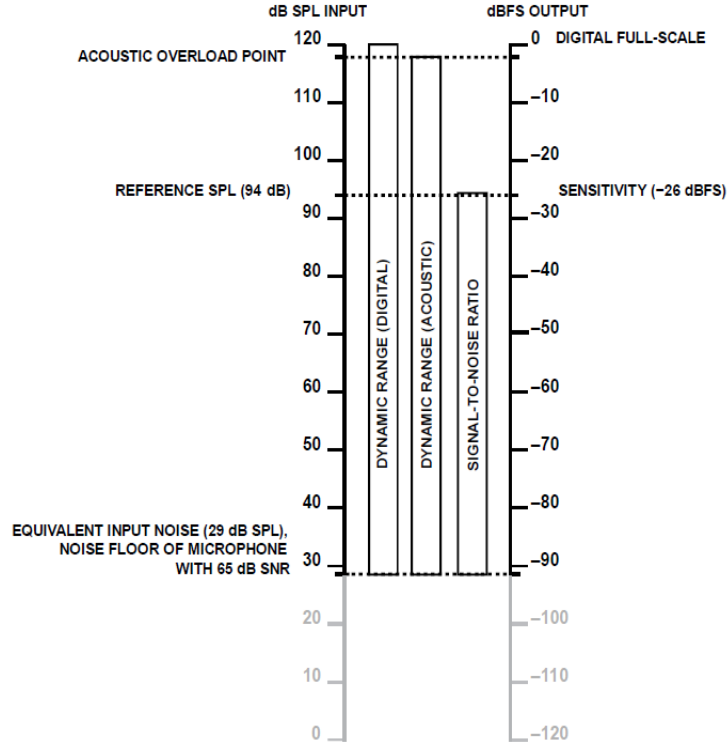


Figure 2.10 : Relationship between dB SPL and dBFs – dynamic range [25].

2.2.4 Signal to noise ratio

The signal to noise ratio (SNR) indicates the ratio of the reference signal to the overall unintended noise level in the microphone. The unwanted noise in the microphone is generally a result of the noisy and uncontrolled environment along with the self noise of the microphone. As this noise level increases, it also causes a reduction in the signal quality thus deteriorates the signal accuracy. The general expression for signal to noise ratio is given below on equation 2.12 :

$$SNR = 10\log_{10}\left(\frac{Energy_{Signal}}{Energy_{Noise}}\right) \quad (2.12)$$

SNR is generally calculated in a controlled environment such as an anechoic chamber under 1 kHz and 94 dB reference signal. In all cases, a higher signal to noise ratio is desired. The higher the SNR, the better the signal quality. To exemplify, an SNR of 70 dB is much better than an SNR of 30 dB. Figure 2.11 illustrates the effect of SNR on a signal. As the SNR is improved, signal can easily be differentiated from the noise within the environment.

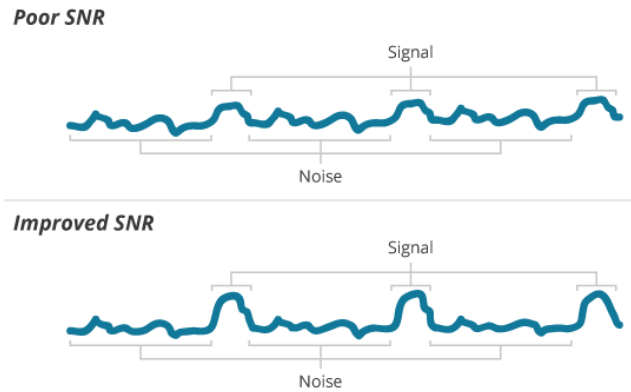


Figure 2.11 : The SNR effect on a signal [26].

2.2.5 Weighting

The apparent loudness of sound is a subjective metric of the sound which changes according to human ear's response. The loudness also changes significantly with shifts in frequency and sound pressure. Most of the sound level meters today have an option to apply weighting networks to mimic the response of human ear to various sounds. In general, the mostly used weighting networks are A, B and C weightings.

A weighting is the corresponding response of human ear at low sound levels. Most of the noise level measurements favor the use of A weighting since this is a good approximation of human ear's response to low level noises such as environmental noise. Even at higher noise levels such as the industrial noise which most workers are subjected to is also a function of A weighting. Similar to this system, B and C weighting networks are also used for various sound pressure levels of 55-85 dB and 85 dB respectively. Figure 2.12 is a representation of the differences between these weighting networks. As it can be seen, the weighting networks have different sensitive frequencies and levels. These are also widely used for diverse applications. For instance, while measuring the higher level noises such as an aircraft noise, C weighting is used. Many other weighting systems are introduced to the industry yet these 3 weighting networks are the most widely used systems.

Whenever a weighting network is used, a correction factor must be added to find the corresponding sound pressure levels. These correction factors can be found by looking at the acoustic weighting curves.

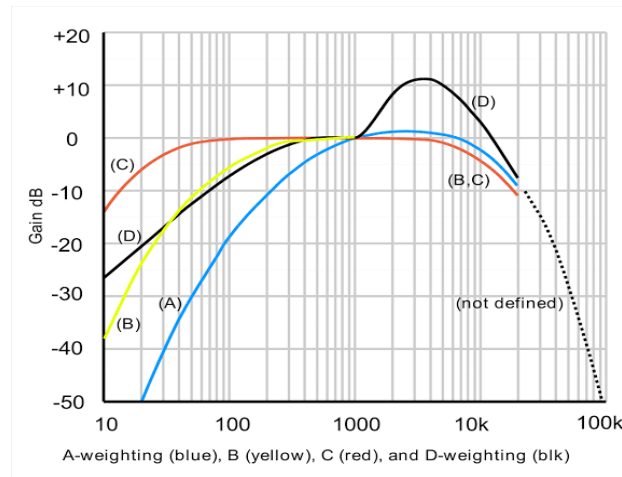


Figure 2.12 : Acoustic weighting curves for humans [27].

2.2.6 Microphone types

The microphones that are used throughout the history have varied a lot with the advancement in technology and the areas of usage. As newer microphones are developed with higher capabilities, there appeared to be many different types of microphones each for their special usage areas. This section will briefly go over the working principles of various microphone types to better understand the concept of sound measurement.

2.2.6.1 Carbon microphones

Carbon microphones are one of the oldest microphones. The first telephones that were used had carbon microphones inside them. The working principle of a carbon microphone and its physical example can be seen in Figure 2.13. The frequency response of these microphones are very bad considering today's microphones. These microphones also need a DC voltage to work and the current that goes through the carbon granules within them causes the microphone sensitivity to deteriorate in time.

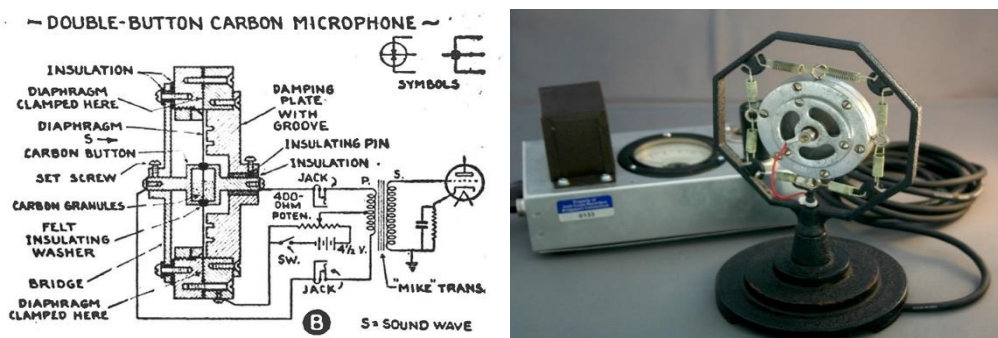


Figure 2.13 : Schematic and structure of a carbon microphone [56].

2.2.6.2 Crystal microphones

Crystal microphones were initially designed to get rid of the negative properties of carbon microphones back in 1940's. These were mostly used on radio and television broadcasting. Crystal microphones work with the principle of piezoelectricity.

On Figure 2.14 below, the working principle of a crystal microphone can be seen. A piezoelectric material, like rochelle salt is used in between two plates. Sound waves that hit the diaphragm vibrate the crystal and as a result of this vibration an output voltage is generated. These microphones are very sensitive to temperature and humidity effects.

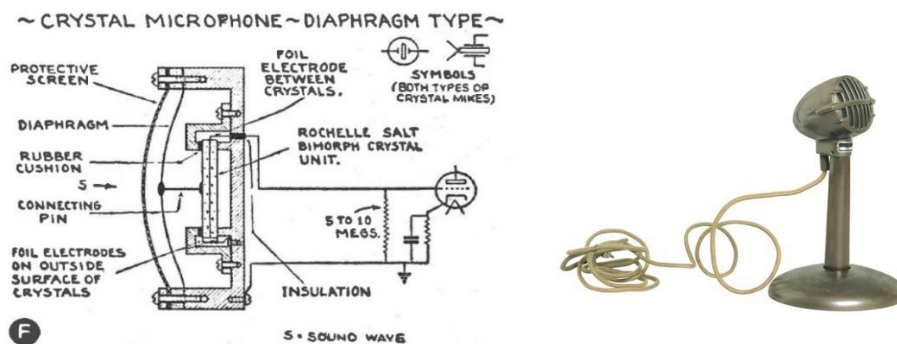


Figure 2.14 : The working principle of a crystal microphone [28].

2.2.6.3 Dynamic microphones

Dynamic microphones are also named as the coil or pressure microphones. As it can also be seen on Figure 2.15, dynamic microphones are consisted of a diaphragm and coil. When the sound waves hit the diaphragm, they cause the coil to move in the magnetic field and produce an output voltage. Since these microphones are affected with magnetic field, they need to be used outside these fields. Otherwise, some discrepancy can be seen on their frequency response characteristics.

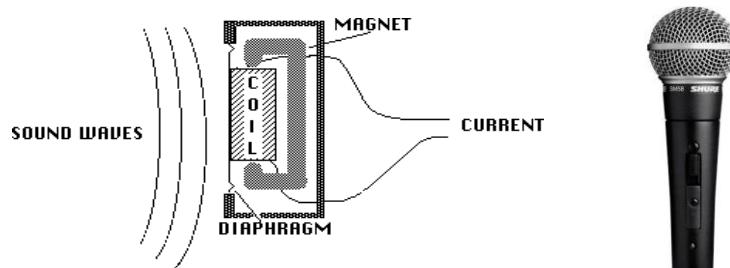


Figure 2.15 : The working principle of a dynamic microphone [57].

These are lower sensitivity microphones but they can be used in closer proximity with the source since they are more resistant to sound intensity. Dynamic microphones are widely preferred since they don't need an external power source and their cost is low.

2.2.6.4 Ribbon microphones

Ribbon microphones operate by moving an aluminum ribbon within a magnet as illustrated in Figure 2.16. The ribbon is placed inside the magnetic field that is created by the magnet inside the microphone. As sound waves vibrate the ribbon back and forth, this movement creates an output voltage thus the microphone works.

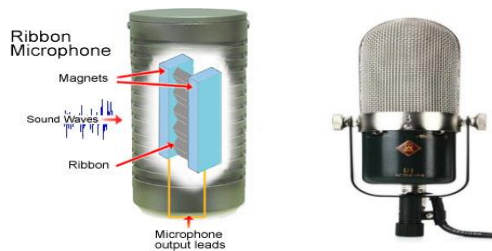


Figure 2.16 : An internal structure of a ribbon microphone [58].

Since ribbon microphones are sensitive to noise, wind and minor movements can cause noise in the measurement so these microphones are not suitable for outdoor. Moreover, since ribbon microphones have good frequency response characteristics in a wide range, they are used in studio recordings and radio broadcasting.

2.2.6.5 Condenser microphones

Condenser microphones have the capacitors that are located inside them. In condenser microphones, one of the capacitor plates is made up from a very light material as illustrated in Figure 2.17 in order for it to act as a diaphragm and vibrate as the sound waves hit it. As the sound waves hit the diaphragm, it moves closer to the second plate.

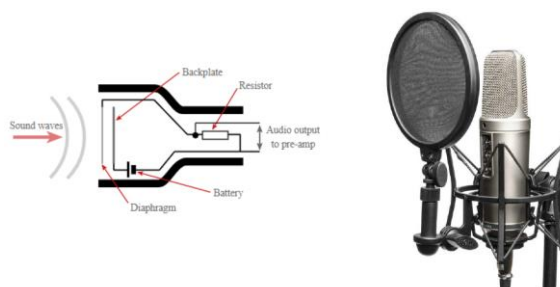


Figure 2.17 : Working principle of a capacitive mic [59].

Capacitive microphones are able to operate under very high sound pressure levels and they have very good frequency response. When compared with the durability of

dynamic microphones, capacitive microphones are more vulnerable to outside sources. Due to their low distortion and high sensitivities, they are widely used in many applications including TV broadcasting and scientific sound measurements. Condenser microphones also need an external power supply to be able to operate and they are very expensive in comparison to dynamic microphones.

2.2.6.6 Laser microphones

Every sound wave that hits a surface within a medium creates minor vibrations on these surfaces. For example, when a person within a room talks, the sound waves created by the speech vibrates the walls of the room. Laser microphones use this principle of vibration.

When a laser beam is directed to the vibrating surface, these minor vibrations cause small shifts in the reflected laser beams. Laser microphones operate by looking at these shifts and converting the changes to electricity. The reflections from vibrated surfaces are taken by the receiver and then filtered & amplified to output that as audio. Figure 2.18 shows an illustration of such laser microphones.

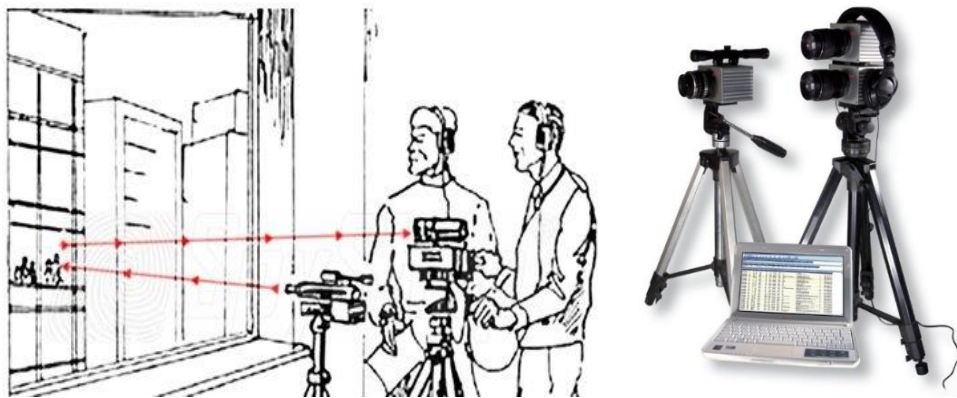


Figure 2.18 : The application of laser microphones [60].

Laser microphones are able to reproduce the audio by checking the minor vibration on the surface with a very high accuracy. Due to this reason, laser microphones are generally used in military applications and espionage.

2.2.6.7 Electret microphones

The working principle of an electret microphone is very similar with a condenser microphone. Electrets are semi conductor materials which constantly protect the positive and negative charge distribution in themselves. In electret microphones, the

electrets are coupled with a diaphragm. With the incoming sound waves, the diaphragm vibrates and these vibrations change the characteristic of electrets thus producing a voltage between the electrodes. The output from electret microphones is generally not high enough which is why an amplifier is used to observe the signal properly. As seen in Figure 2.19, a dust cover and an aluminum housing is used on electret microphones to protect them from external noise effects.

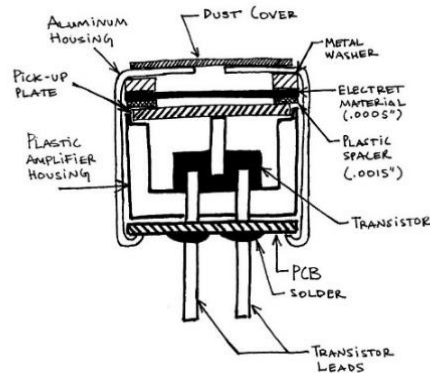


Figure 2.19 : Cross sectional view of an electret microphone [61].

Electret microphones are among the cheapest and simplest microphones available in the market today. Their good durability and small sizes make these microphones suitable candidates to be used for cell phones, tablets, hearing aid and other electronic devices. Even though their costs are very low, electret microphones aren't used during professional applications. Electret microphones don't always give accurate results due to erroneous measurements and non-existent frequencies that are seen if precautions are not taken. Moreover, additional amplifiers, cables and circuits to be used with electrets are not preferred setups for such professional applications.

2.2.6.8 MEMS microphones

MEMS microphones are developed as a result of the advancement in micro electro mechanical systems. These are very small microphones with high performances even under electrically noisy environments due to their high SNR. The use of MEMS microphones also reduce the unwanted effects of mechanical vibrations caused on the medium due to their assembly and mounting system which is seen on Figure 2.20.

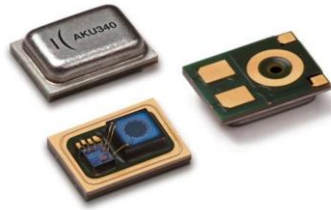


Figure 2.20 : MEMS microphone structure [62].

MEMS microphones are based on the same principle as condenser microphones since they both operate with the movement of capacitors. As seen on Figure 2.21, a MEMS microphone contains a diaphragm and a back plate along with the main microphone element MEMS chip and an internal analog to digital converter. Sound waves come from the specified sound holes drilled on the structure of MEMS and vibrate the diaphragm. As the distance between the diaphragm and back plate changes, this creates a voltage difference which is transmitted to MEMS chip. ASIC is used to convert the analog signals to digital signals thus working as an ADC within the microphone. The designated location of sound hole is subject to change for each MEMS microphone since design of these microphones differ significantly from each other.

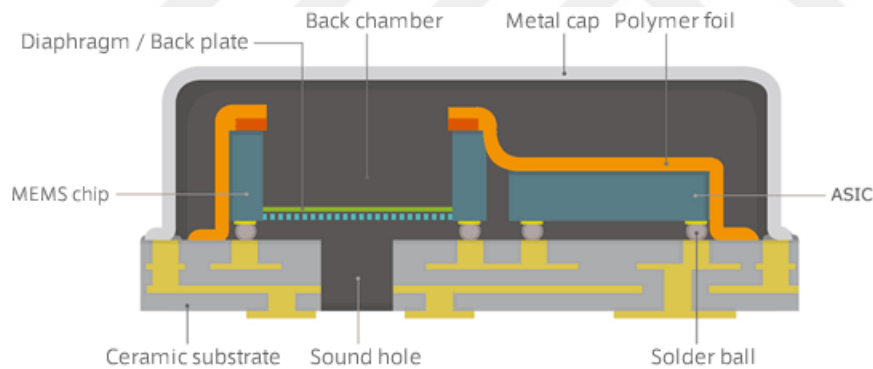


Figure 2.21 : Typical structure of a MEMS microphone [63].

MEMS microphones are mostly compared with electret microphones today since their market share is rapidly growing due to various advantages [64]. To exemplify, the reduced size of the microphones when compared with electrets provide an attraction for applications with space constraints. Secondly, the structure of MEMS microphones are more ideal for noisy environments since the effect of mechanical vibrations are less effective when compared with electrets. Overall, when the frequency response of electret condenser microphones and MEMS microphones are taken into consideration, a smoother and more straight frequency response over each operating frequency is seen on MEMS microphones.

Good sensitivity, high temperature stability, low power consumption, fine frequency response over a wide range and ease of use make the microphones ideal for many applications. Typical applications of MEMS microphones include audio recording, detection sensors, speech recognition, laptops and most important of all, these microphones are used nearly on every mobile phone as of today.

2.3 Sound Level Meters

Sound level meters are the primary tools to measure the sound pressure levels in a given medium. The sound level meters used today have dynamic ranges up to 100 dB which covers most of the sounds that exist in the environment. Apart from the microphones, the sound level meters have an ADC with a signal processor circuit board to convert the analog sound signal to a scale desired by the customer.

A complex sound level meter has sensitivity adjustment details which allows the tool to be calibrated. These measurement devices are also able to apply the A-Weighted or C-Weighted filters and without weighting options to display the sound pressure levels within the measurement period by averaging the sound levels. In order to get the most details out of a sound level meter, 1/3 octave band display and equivalent noise level quantities can also be included. Equivalent noise level becomes significantly important if the source of interest has fluctuations in time. equation 2.13 is used to calculate the equivalent continuous sound pressure level :

$$L_{eq} = 10 \log_{10} \left[\frac{1}{T} * \int_0^T 10^{\frac{L(t)}{10}} dt \right] \quad (2.13)$$

The continuous equivalent sound pressure level can also be calculated for A weighted noise levels using the same equation for a time period of t. Sound level meters are also distinguished by their ability to measure by looking at their dynamic range as well as their specified noise floors. These characteristics come from the chosen microphones that are integrated within the sound level meters. If a sound pressure level of 140 dB or above is chosen to be measured than a low sensitivity microphones should be used to get the best results while for lower sound pressure levels a higher sensitivity microphone will get better measurement results.

Sound level meters also have meter response times to be specified during the measurements. One of these options is called “fast” response whose primary aim is to

approximate the response of human ear to a noise but this option is used for very short periods up to a hundred milliseconds. Secondly, “slow” measurement with a time constant of 1 second can be used to determine the fluctuations in the measured sound. A third option comes with advanced sound level meters which calculate “impulse” response for impulsive type very short bursts of noises. The application in which a sound level meter is going to be used determines the response type of the microphones for the measurements.

Various classes of sound level meters are currently available in the industry as discussed in industrial standards IEC 61672/1-2013. This standard differentiates the sound level meters into two sections with regards to their accuracy. The performance limits and the accuracy differences of the various classes of sound level meters can be seen in Figure 2.22.

Performance criteria: tolerance limits at reference frequencies (for IEC61672-1:2013)	Class 1	Class 2
Frequency of 16Hz	+2.5dB, -4.5dB	+5.5dB, -∞dB
Frequency of 20Hz	+/- 2.5dB	+/- 3.5dB
Frequency of 1kHz	+/- 1.1dB	+/- 1.4dB
Frequency of 10kHz	+2.6dB, -3.6dB	+5.6dB, -∞dB
Frequency of 16kHz	+3.5dB, -17dB	+6.0dB, -∞dB



Figure 2.22 : Performance criteria for Class 1 & 2 SLM and a typical SLM [65].

The general differentiation is done as class 1 and class 2 for sound level meters. Class 1 sound level meters have precise microphones. Class 1 sound level meters are especially used for sensitive laboratory applications or accurate field measurements. Class 2 sound level meters have lower accuracy in comparison to Class 1 sound level meters. These are general purpose sound level meters and are used for general purposes. Overall noise level data or general field noise measurements are the primary applications for these sound level meters.

2.3.1 Main sources of measurement errors

When using sound level meters, in order to get good data from the measurements, main sources of measurement errors should be eliminated as much as possible. These measurement errors will be briefly reviewed on the following paragraphs.

The handling of the microphone affects the measurement results greatly. Especially if the sound level meter is placed on a vibrating surface, the signals that are read on the measurement device also includes the noise data from surface vibrations as well. In order to get healthy results from a measurement, a sound level meter should be kept away from any sources of vibration. Along with this precaution, microphones should be cared for in order to prevent any shocks or drops that may cause a significant change in microphone calibration.

Since sound level meters have built-in amplifier systems, these amplifiers should be kept away from any sources of magnetism or vibration in order not to see erroneous measurements. Moreover, the microphones should not be overloaded to avoid wrong readings. In order to prevent any overloading, the microphones should be selected according to their appropriate dynamic ranges as well.

The sound of interest should be within the manufactured microphone dynamic range, frequency and pressure amplitude limits. Otherwise, the sound level meter wouldn't give correct measurement outputs.

Background noise also affects the read sound pressure levels in the medium. A very basic method of measurement should be use to make sure if any background noise is present in the environment. The source to be measured should be put on off status and a measurement should be taken. If the background noise is less then 10 dB below the total level when the source is turned on, this proved that an interfering background noise is present in the environment. The effect of the presence of background noise in a sound measurement can be seen in Figure 2.23. The black line shows the trumpet and background noise combination while the blue line corresponds to the background noise only. The background noise contributes to the peaks that are measured for the related trumpet noise. Along with this first rule of source and background sound pressure levels, the sound pressure level difference between the background noise and total noise with a source should never go below 3 dB in order to get healthy and meaningful measurements [19].

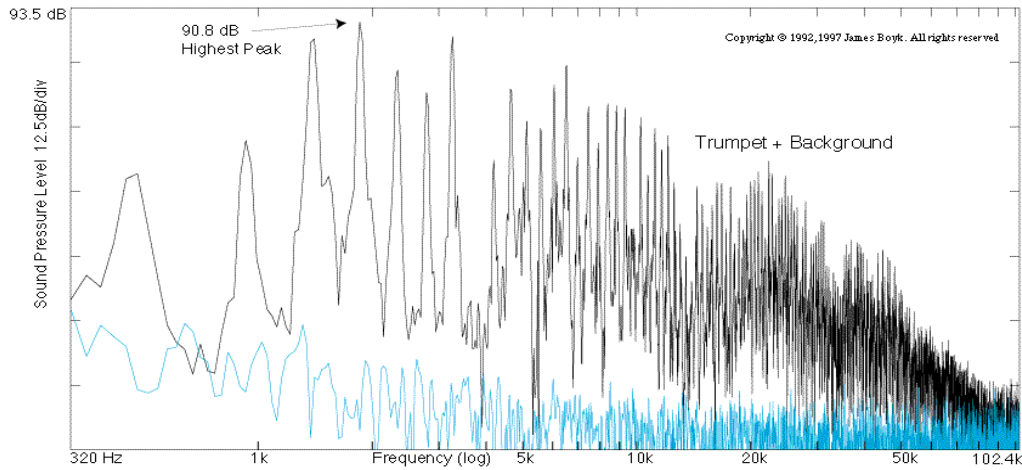


Figure 2.23 : Amplitude – frequency plot of a trumpet and background [29].

Wind noise is another aspect to be considered while taking measurements with a microphone. Even the light winds that below past a sound level meter's microphone may affect the measurement outputs especially below 200 Hz. The effect of wind noise can be reduced by using a windscreen of porous acoustic foam over the microphone. The drawback for this procedure is, if the wind is stronger than 20 km/h the method is not applicable and an alternative windscreen of a closed cubic or spherical enclosure should be used to prevent the effect of wind noise. The results of a study on the effect of windscreen on microphones can be investigated on Figure 2.24. As it can be seen, when windscreen is not present, measured sound levels are very high. With increasing windscreen diameters, the noise level decreases thus the effect of windscreen can be understood clearly.

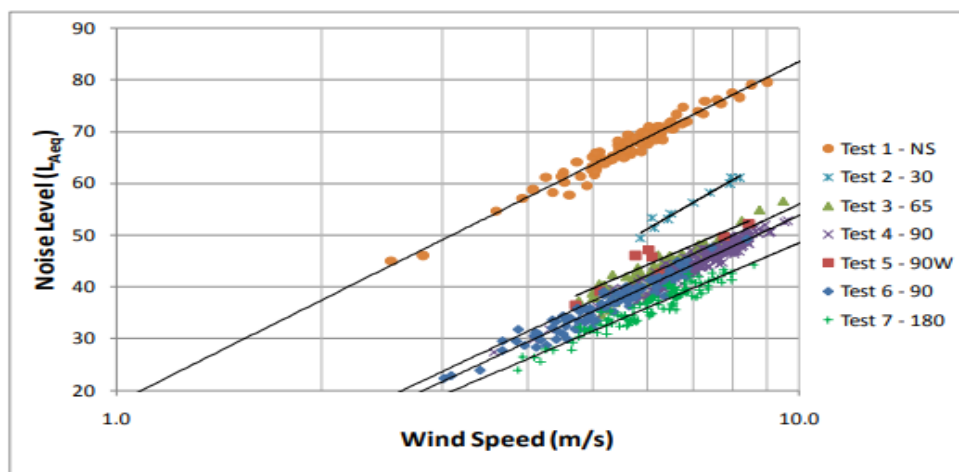


Figure 2.24 : Equivalent SPL with different windscreen diameters [30].

Temperature also affects the sensitive elements of the microphone along with the included batteries. As the temperature of the measurement environment increases, the battery life decreases. Moreover, every microphone has an operating temperature limit in which they obtain meaningful results. This operating range varies from $-40\text{ }^{\circ}\text{C}$ up to $150\text{ }^{\circ}\text{C}$ in some advanced sound level meters. Outside of these limits, the calibration of the microphones shift and unhealthy results can be seen as a result.

Some microphones are more vulnerable to humidity and dust, particularly the condenser microphones. Both of these effects cause self internal noise within the microphones, thus the microphones should be stored in a dry place or an enclosure. If the microphones are subjected to water, their frequency response characteristics will be changed.

While taking measurements, nearby objects may affect the radiation of sound due to possible reflections. If the object's dimensions are comparable with the wavelength of the sound that is being measured, the existence of such object may cause a distortion in the sound field and may cause erroneous measurements. In order to remove the effect of reflections, such measurements should be taken in an anechoic or semi-anechoic room with sound absorptive materials.

2.4 Calibration of Microphones

Microphones deteriorate over time due to their usage conditions, physical conditions and many other aspects as every other hardware does. In order to obtain the most accurate results with the microphones, calibration of the microphones before and after the measurements must be conducted. There are various methods of microphone calibration for industry standard microphones. The calibration methods incorporate different properties of sound and microphones and will be discussed on the following pages.

2.4.1 Pressure reciprocity calibration

Using sound level meters for scientific purposes requires high precision in measurements to be obtained even if human ear response to 1 dB change in sound pressure level is insignificant. In most of the applications, a professional microphone uncertainty should be better than or equal to 0.1 dB to be reliable [31]. For research purposes, the uncertainty of the microphones becomes more important and should be

better. One of the best methods to calibrate a condenser microphone is called reciprocity pressure calibration which is also called the coupler method. This method has an uncertainty which is less than 0.01 dB at 250 Hz under a controlled environment such as an anechoic room for reference condition [32]. If the calibration of such microphones is done without controlling the environmental parameters such as pressure, temperature and humidity, the uncertainty of reciprocity pressure calibration method falls to 0.05 dB at low to middle frequencies.

The reciprocity method is primarily based on measuring the product of the sensitivities of each pair of a set of three microphones. The related electrical and mechanical quantities are measured for these microphones and the absolute sensitivity of each microphone is then determined. The acoustic impedance Z_{AB} of a cavity while considering a pair of microphones, namely A & B is shown on Figure 2.25 and can be calculated as :

$$Z_{AB} = \gamma P_s [i\omega(V + V_{eA} + V_{eB})]^{-1} \quad (2.14)$$

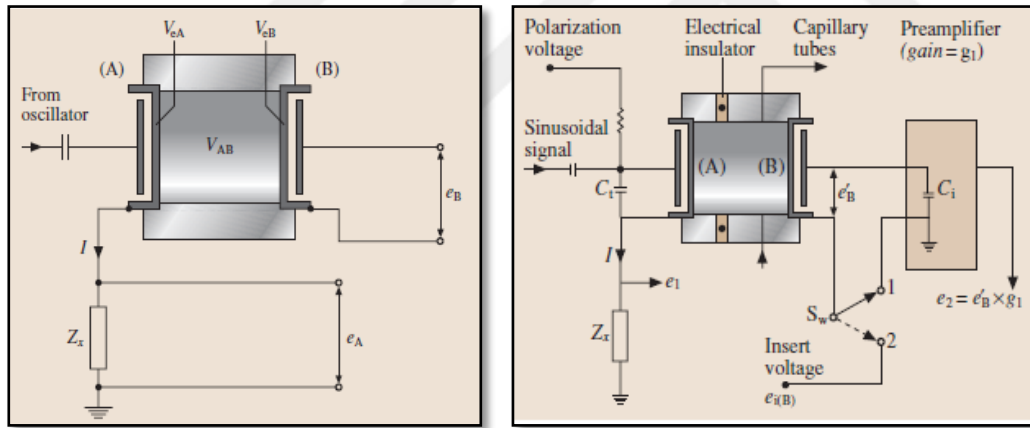


Figure 2.25 : Pressure reciprocity calibration of condenser microphones [33].

At equation 2.14, γ denotes the ratio of specific heats of gas inside the coupler, P_s is the barometric pressure, w is the velocity $2\pi f$, f is the frequency of the main driving signal, V , V_{eA} and V_{eB} are the equivalent volumes of the cavity, microphone A and microphone B respectively [34]. Equivalent volume can be described as the physical volumes of the cavity and the coupled microphones. The connection of alternating pressure p inside the cavity and the velocity of the volume u can be seen as described below :

$$p = uZ_{AB} = u\gamma P_s[i\omega(V + V_{eA} + V_{eB})]^{-1} \quad (2.15)$$

The equivalent volume of the cavity of microphones A and B can be calculated to define the alternating pressure p as stated in equation 2.16.

$$V_{AB} = V + V_{eA} + V_{eB} \quad (2.16)$$

It is seen in the reciprocity theorem that in a reciprocal transducer, the ratio of the volume velocity which was denoted as u in the relevant cavity to the input current I when the transducer is used as a sound source should be equal to the ratio of the open circuit voltage e_A across the electrical terminals to the sound pressure p which is acting on the diaphragm when the transducer is used as a receiver. When all the statements that this theorem defines are taken into consideration, the following equations can be written :

$$M_A = \frac{u}{I} = \frac{e_A}{p} \quad (2.17)$$

$$M_B = \frac{e_B}{p} \quad (2.18)$$

$$M_C = \frac{e_C}{p} \quad (2.19)$$

M_A , M_B and M_C on the equations above correspond to the open circuit sensitivities of each of the microphones A, B and C respectively. Multiplying equation 2.17 and 2.18 and substituting p from equation 2.15 gives :

$$M_A M_B = \frac{e_A i\omega V_{AB}}{I\gamma P_s} \quad (2.20)$$

Since $I = e_A/Z_X$; substituting this equation to equation 2.20 results in :

$$M_A M_B = V_{AB} k \beta_{AB} \quad (2.21)$$

Where :

$$k = \frac{i\omega Z_X}{\gamma P_s} \quad (2.22)$$

And :

$$\beta_{AB} = \frac{e_A}{e_B} \quad (2.23)$$

By using all 3 microphones of A, B and C in conjunction with each other, the equivalent volumes of V_{AB} , V_{BC} and V_{CA} can be found. After these equivalent volumes are found, these volumes are multiplied with the measured voltage ratios of β_{AB} , β_{BC} and β_{CA} , the sensitivity of these microphones can be defined according to the equations below :

$$M_A M_B = V_{AB} k \beta_{AB} \quad (2.24)$$

$$M_B M_C = V_{BC} k \beta_{BC} \quad (2.25)$$

$$M_C M_A = V_{CA} k \beta_{CA} \quad (2.26)$$

As a result of these equations, the open circuit sensitivity of these microphones can be calculated accordingly :

$$M_A = \left[\left(\frac{V_{CA} V_{AB}}{V_{BC}} \right) \left(\frac{\beta_{CA} \beta_{AB}}{\beta_{BC}} \right) k \right]^{\frac{1}{2}} \quad (2.27)$$

$$M_B = \left[\left(\frac{V_{AB} V_{BC}}{V_{CA}} \right) \left(\frac{\beta_{AB} \beta_{BC}}{\beta_{CA}} \right) k \right]^{\frac{1}{2}} \quad (2.28)$$

$$M_C = \left[\left(\frac{V_{BC} V_{CA}}{V_{AB}} \right) \left(\frac{\beta_{BC} \beta_{CA}}{\beta_{AB}} \right) k \right]^{\frac{1}{2}} \quad (2.29)$$

It is seen from the equations above that the sensitivity of microphones A, B and C can be determined if the voltage ratios are measured and if the equivalent volumes of all three pairs of microphones and the constant k are known. Without these variables, it is not possible to calculate the sensitivity of the microphones. The same equations for microphone sensitivities can also be given using the acoustic impedances of microphone cavities :

$$M_A = \left\{ \left[\frac{Z_{BC}}{Z_{CA} Z_{AB}} \right] \left(\frac{\beta_{CA} \beta_{AB}}{\beta_{BC}} \right) Z_X \right\}^{\frac{1}{2}} \quad (2.30)$$

$$M_B = \left\{ \left[\frac{Z_{CA}}{Z_{AB}Z_{BC}} \right] \left(\frac{\beta_{AB}\beta_{BC}}{\beta_{CA}} \right) Z_X \right\}^{\frac{1}{2}} \quad (2.31)$$

$$M_C = \left\{ \left[\frac{Z_{AB}}{Z_{BC}Z_{CA}} \right] \left(\frac{\beta_{BC}\beta_{CA}}{\beta_{AB}} \right) Z_X \right\}^{\frac{1}{2}} \quad (2.32)$$

In order to implement the pressure reciprocity calibration of the condenser microphones, a setup as described in Figure 2.25 can be used. The setup in this case consists of a driving microphone A and a receiver microphone B. Driving microphone A should be isolated electrically from the receiver microphone which is denoted as B to simplify the electrical measurements that need to be taken.

The preamplifier in the circuit has a gain of g_1 and a capacitance of C_i as seen on the Figure 2.25. The receiver microphone B is shunted by C_i and in this case the output voltage is modified by e'_B . When this exemplary setup is used, an insert voltage can also be applied to find the open circuit voltage when the switch in the circuit S_W is at position 2.

In addition to these components, a shunt capacitor C_t modifies the driving voltage of e_A to e_1 . The cavity has two capillary tubes as seen on the circuit. These tubes are mainly used for atmospheric pressure equalization. The sensitivity of each microphone in this case can be found as :

$$M'_A = \left[\left(\frac{V'_{CA}V'_{AB}}{V'_{BC}} \right) \left(\frac{\beta_{CA}\beta_{AB}}{\beta_{BC}} \right) k \right]^{\frac{1}{2}} \quad (2.33)$$

$$M'_B = \left[\left(\frac{V'_{AB}V'_{BC}}{V'_{CA}} \right) \left(\frac{\beta_{AB}\beta_{BC}}{\beta_{CA}} \right) k \right]^{\frac{1}{2}} \quad (2.34)$$

$$M'_C = \left[\left(\frac{V'_{BC}V'_{CA}}{V'_{AB}} \right) \left(\frac{\beta_{BC}\beta_{CA}}{\beta_{AB}} \right) k \right]^{\frac{1}{2}} \quad (2.35)$$

On these equations, the equivalent volume of the cavity is marked with a prime in order to indicate that the equivalent volume of capillary tubes included in the system are taken into consideration. The voltage ratio $\frac{e_B}{e_A}$ becomes $\frac{e_2}{e_1}$ on the circuit for practical implementation.

2.4.2 Free field reciprocity calibration

In free field reciprocity method for microphone calibration, a similar roadmap to pressure reciprocity is used. One microphone is driven as a sound source and the microphone pair is placed facing each other in a free field at a specified distance. While using this method, the acoustic centers of placed microphones should be accurately decided. For free field reciprocity, one of the microphones should be used as a sound source while the other one should be used as a receiver. If a third microphone is included for the calibration and is paired with the other two, the electrical impedances will carry enough information to find the related microphone sensitivities individually [35].

The free field sensitivity of each microphone is determined with the measurement of the electrical impedance between the microphones of interest. The electrical impedance is defined as the ratio of the open circuit output voltage that is observed between the terminals of the receiver microphone to the input current that flows through the terminals of transmitter microphone. The electrical impedance $Z_{e,12}$ can be calculated as below :

$$Z_{e,12} = \frac{u_1}{i_2} = j \left(\frac{\rho f}{2d_{12}} \right) M_{f,1} M_{f,2} e^{jk d_{12}} e^{\alpha d_{m12}} \quad (2.36)$$

On the equation 2.36, u_1 is defined as the open circuit voltage between the receiver microphone terminals, i_2 is the current that flows through the transmitting microphone's terminals, f is the frequency of the sound, d_{12} is the distance between the acoustic centers of the microphones, ρ is the density of air, $M_{f,1}$ and $M_{f,2}$ are the free field sensitivities of the microphones, d_{m12} is the distance between each of the microphones, k is the wave number ($k = 2\pi/\lambda$) and α is the coefficient that is used for the attenuation of air.

When a third microphone is included in the calibration, there are three sensitivity equations with three unknowns thus the individual sensitivity of each microphone can be calculated. The sensitivity for microphone 1 in this case, can be calculated using the equation 2.37 :

$$M_{f,1}^2 = \frac{2}{\rho f} \frac{d_{12} d_{13}}{d_{23}} \frac{Z_{e,12} Z_{e,13}}{Z_{e,23}} e^{jk(d_{12}+d_{13}-d_{23})} e^{\alpha(d_{m12}+d_{m13}-d_{m23})} \quad (2.37)$$

Using the same formula for the other 2 microphones, the sensitivities of the microphones are calculated. While calculating the sensitivity using this method, it should be noted that the acoustical centers of each one should be determined carefully according to the International Standards EN 61094-3 Annex A. Moreover, as it is also the case with the pressure reciprocity method, the environmental coefficients must also be found and included for corrections. This method is not very useful due to its time consuming nature and the difficulty of the setup.

2.4.3 Interchanging the microphones

First of these comparison methods involves comparing the microphones in a cavity or in a free field such as a small anechoic room. This method is also among the international standard IEC 61094-5 for microphone calibrations.

The difference in sound pressure levels that are seen on two microphones with close proximity diaphragms in a situation where microphones 1 and 2 are used in conjunction with preamplifiers 1 and 2 respectively can be calculated as :

$$L_{C12} = (L_1 + L_{m1} + L_d + L_{WA}) - (L_2 + L_{m2} + L_d + L_{WB}) \quad (2.38)$$

Where the sensitivities of each microphone are marked as L_1 and L_2 . L_{m1} and L_{m2} corresponds to the gain sensitivities of the measurement systems that are used for microphones 1 and 2 while L_d is the sound excitation level, L_{WA} and L_{WB} are denoted as the wave pattern effects of respective microphone locations. An exemplary setup for this comparison method can be seen on Figure 2.26 where an additional coupler is also used to fix the diaphragms of the microphones in close proximity.

When the microphones 1 and 2 are interchanged, the microphones are cross connected to each preamplifiers. Microphone 1 is connected to preamplifier 2, microphone 2 is connected to preamplifier 1. If this interchanging is done between the microphones, the sound pressure level difference also changes as stated below :

$$L_{C21} = (L_2 + L_{m1} + L_d + L_{WA}) - (L_1 + L_{m2} + L_d + L_{WB}) \quad (2.39)$$

The corresponding equations 2.38 and 2.39 are then subtracted from each other to find the sensitivity level of one of the microphones. If one of the reference microphone sensitivities are known, the other microphone's sensitivity can be deduced.

$$L_1 - L_2 = \frac{1}{2} (L_{C12} - L_{C21}) \quad (2.40)$$

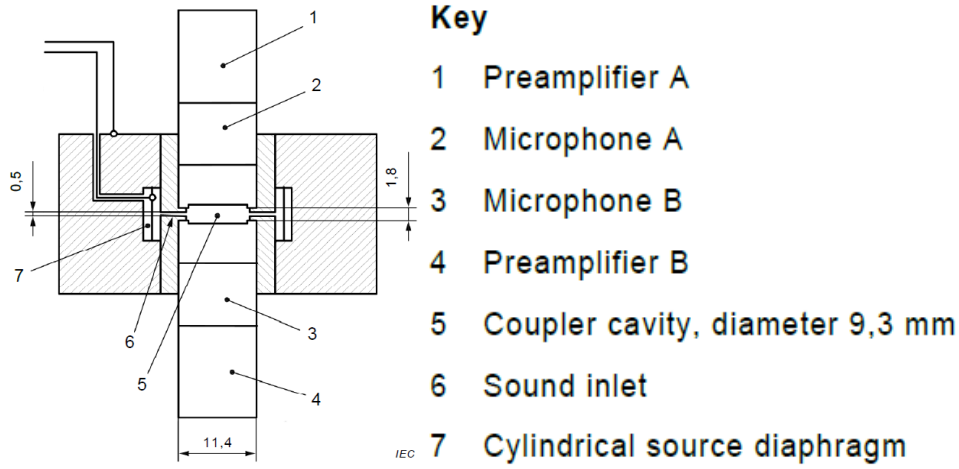


Figure 2.26 : A setup for interchanging microphones comparison method [37].

The previous calculations on this section considers that both the microphones and their input capacitances are identical along with the model of preamplifiers used during the measurement. If the input capacitances of the preamplifiers C_m are the same but the model of microphones used for the calibration are different, then a correction factor of L_{corr} needs to be added to the right hand side of the equation 2.40.

$$L_{corr} = 20 \log \left[\frac{C_2}{C_m + C_2} \right] - 20 \log \left[\frac{C_1}{C_m + C_1} \right] \quad (2.41)$$

The measurements should be conducted a few times for repeatability and uncertainty analysis. This is a low cost and effective method for microphone calibration when compared to reciprocity methods.

2.4.4 Using a calibrator for comparison

Calibrating the microphones with a calibrator system requires a reference microphone, a test microphone, a sound source with stable sound excitation characteristic or a calibrator that generates constant sound pressure, a microphone preamplifier and a sound level meter or an acoustical level indicator like a voltmeter. A similar setup for the comparison method with a calibrator can be seen below on Figure 2.27. The following steps are followed for the calibration of the microphones in this method :

- The reference microphone with a preamplifier is inserted to the calibrator and a specific sound pressure level reading of L_r is taken.

- The same measurement is taken after the reference microphone is replaced with the test microphone with unknown sensitivity and the level reading is written as L_t .
- The test microphone is then removed from the system and first measurement is taken once again with the reference microphone.



Figure 2.27 : A comparison method setup for microphone calibration [66].

The third measurement is conducted to find the validity of the measurement. If the first and the third measurements give the same result, this corresponds to an unchanged system between the measurements as requested. The sound pressure level difference $L_t - L_r$ is the open circuit sensitivity difference between each microphone if the conditions below are satisfied :

- The equivalent volumes of both the reference and the test microphone should be identical.
- The capacitances of both the microphones are the same in order not to cause any changes in measurements.
- Environmental conditions must remain the same during the course of the calibration process.

If these conditions are fully satisfied, the comparison method with a calibrator gives very precise results up to 0.05 dB. Otherwise, correction factors for these conditions should be substituted during the measurements. Moreover, this comparison method requires the preamplifier to have a very high input impedance up to gigaohms [38].

2.4.5 Comparison pressure and free field calibration

The method of pressure reciprocity for finding the unknown sensitivities of a microphone trio is the basis of this comparison method. An internationally accepted Class 1 or Class 2 microphone with a known sensitivity value and pressure response is used in this case to derive and calibrate the pressure response of an unknown microphone. The microphone to be calibrated for this comparison method should be compatible with the coupler to be used for calibration. For this procedure, a similar setup shown in Figure 2.27 is used. Microphone A is used as a sound source and the other two microphones of known and unknown sensitivities are used in succession with each other. After the setup is ready, a voltage of e_A is applied to the input of microphone A and the output voltage of e_B at the other end of the coupler is measured. Since one of the sensitivities is already known, the test microphone's sensitivity can be determined using the following equation :

$$M_{test} = M_{reference} - 20 \log \left(\frac{e_{Bref}}{e_{Aref}} \right) + 20 \log \left(\frac{e_{Bx}}{e_{Ax}} \right) \quad (2.42)$$

On equation 2.42, the sensitivities of test and reference microphones are denoted as M_{test} and $M_{reference}$ while e_{Bref} and e_{Aref} are the output open circuit voltage and input source voltages when the reference microphone is used at the other end of the coupler. Similarly, e_{Bx} and e_{Ax} are the output open circuit voltage and input source voltages when the microphone with unknown sensitivity is placed on the other side of the coupler. A volume correction may be required if the coupler size is significantly larger than the microphone diaphragm sizes but equation 2.42 can still be used to find the related microphone sensitivities for calibration.

2.4.6 Precision attenuator comparison method

In order to calibrate the microphones precisely, the method of calibrator comparison is developed and a calibrated attenuator or a potentiometer as shown on Figure 2.28 is used instead of an acoustical level indicator. The attenuator or the potentiometer needs to be calibrated to give precise results and the following procedure needs to be used for calibration :

- The sensitivity of the reference microphone is measured after inserting it to the calibrator. The sensitivity value is entered to the calibrated potentiometer and the indicator's reading is noted.

- Secondly, the test microphone with unknown sensitivity is used instead of the reference microphone. In order to find the sensitivity, initially the reading that is seen on the potentiometer screen is adjusted until it gives the same value as the reference microphone. When the indicator gives the same value, potentiometer shows the sensitivity of the unknown microphone.



Figure 2.28 : Adjustable multi turn potentiometer [67].

Even if this method is cost friendly, it still requires minor corrections due to the microphones that are used. If the microphones are of the same brand, this method gives very precise results when the uncertainties are taken into consideration.

2.4.7 Determination of frequency response with electrostatic actuators

This method has been standardised for frequency response measurement purposes for professional working standard microphones. A schematic for frequency response determination can be seen on Figure 2.29. The setup includes a D-C voltage supply, an AC sweep signal to produce the pressure on microphone diaphragm, and the standard circuit required for microphone operation.

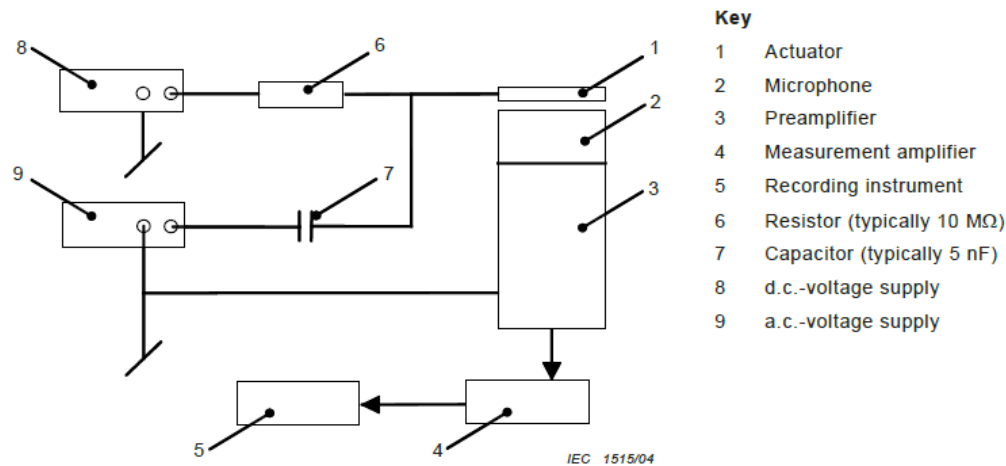


Figure 2.29 : Electrostatic actuator frequency response measurement setup [36].

A difference between the electrostatic actuator determined frequency response and the pressure reciprocity determined response can be seen due to the mechanical radiation impedance variations on the microphone diaphragms. Especially at higher frequencies, this difference on L Type microphones can get up to 1.5 dB [39]. As seen on Figure 2.30, using an electrostatic actuator with appropriate metal grid design for the selected microphone yields very precise results up to 0.1 dB uncertainty especially for higher frequencies around until 10 kHz for ½ inch and 1 inch working standard microphones.

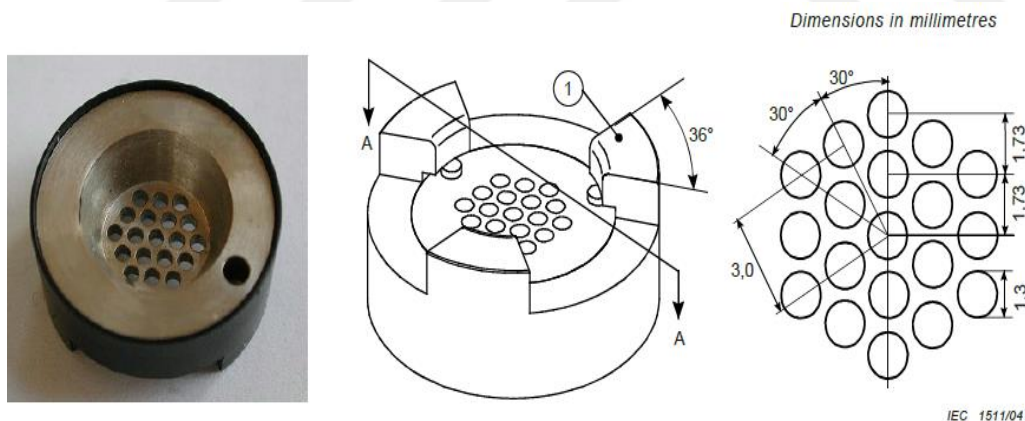


Figure 2.30 : A typical electrostatic actuator with metal grid design [36].

Along with the technical features presented and the economical attraction, there are a few more advantages of electrostatic actuators for calibration :

- In order to understand the shift from a nominal specified value, electrostatic actuators can be used to compare the frequency responses of microphones in a serial production line.
- Since the calibration is done in a closed area within the actuator, the electrostatic actuators are invulnerable to environmental changes such as

temperature and humidity. Due to this reason, actuators are a good way to understand the frequency response changes of the microphones with regards to environmental aspects.

2.5 Data Analysis and Processing

The measured signals in microphones, sound level meters or any other transducers need to be analysed to get meaningful results. Especially for every noise and vibration control purpose, further classifications and analysis need to be conducted. The reasons for analysis and further processing may be the assessment of environmental aspects, identification of a source, decomposition of sound signals and many others to give meaning to measured signals.

2.5.1 Types of signals

Analog time history signals are obtained through accelerometers and microphones considering the data of interest is an acoustical or vibration data. After the data is taken by these transducers, a signal conditioning equipment and a data acquisition system need to be used for conversion of these signal to digital domain. Time domain signals can be differentiated to two categories, namely deterministic data signals and random data signals.

2.5.1.1 Deterministic signals

Deterministic signals are the signals from which a mathematical equation may be derived. The future values of the signal may be determined according to the mathematical rule of the signals. Periodic signals are the most commonly used deterministic signals. These signals are periodic and repeat themselves after a time interval of T_p . Equation 2.43 defines the general rule for periodic deterministic signals.

$$x(t) = x(t \pm T_p) \quad (2.43)$$

Constant speed propellers, fans and any other rotating machinery fall into the periodic signal category. These signals are represented by means of a Fourier series and a series of harmonically similar sine waves. The representation of a periodic signal can be investigated on Figure 2.31. Since the statistical properties of mean, mean square and

spectrum of periodic signals don't change with time, these signals are also called steady – state.

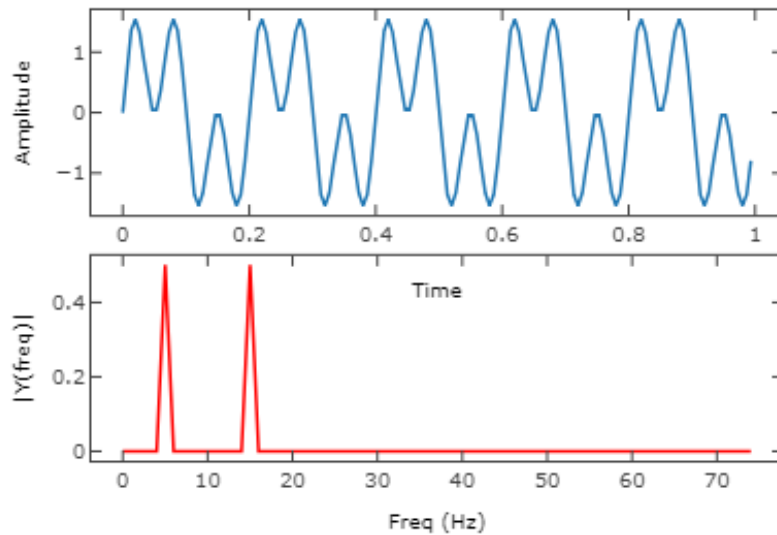


Figure 2.31 : Time frequency representation of a periodic steady state signal [68].

A second subcategory of deterministic signals is non periodic signals. These signals begin and end after a specified time interval and are called transient signals. Contrary to periodic signals, controlled impacts, sonic booms and other short time signals are transient. Figure 2.32 shows a representation of a transient signal that decays within a specified time.

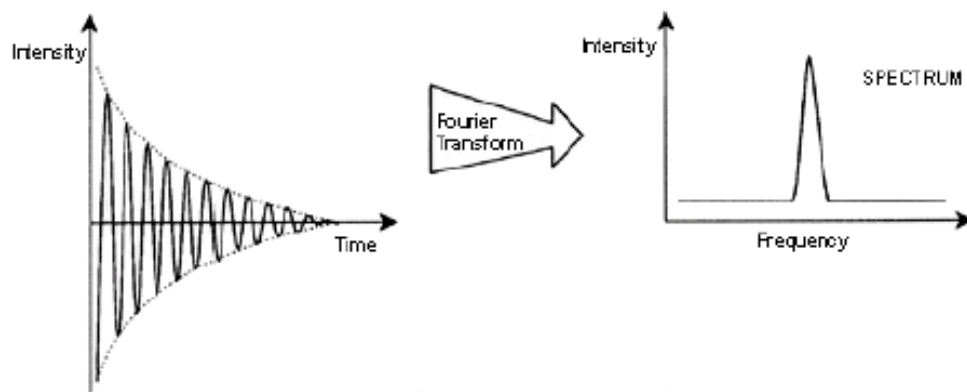


Figure 2.32 : Time frequency representation of a transient signal [40].

2.5.1.2 Random signals

Random signals are unable to be predicted using mathematical equations since they are not deterministic. When the machineries or the parts of interest are producing random signals whose averages don't change with time, these signals are named

stationary. Random stationary signals need to be defined with a continuous spectrum to be differentiated from steady state deterministic signals. While working on these signals, statistical sampling error are introduced to the measurement data due to the probabilistic properties of deterministic random signals. If the properties of a random signal change with time, the signal is said to be non stationary. More data is required to analyse non stationary signals since they are time independent. The typical characteristics of stationary and non stationary random signals can be seen on Figure 2.33 below. As it is clearly seen, non stationary random signal average properties change with time though stationary random signal properties don't.

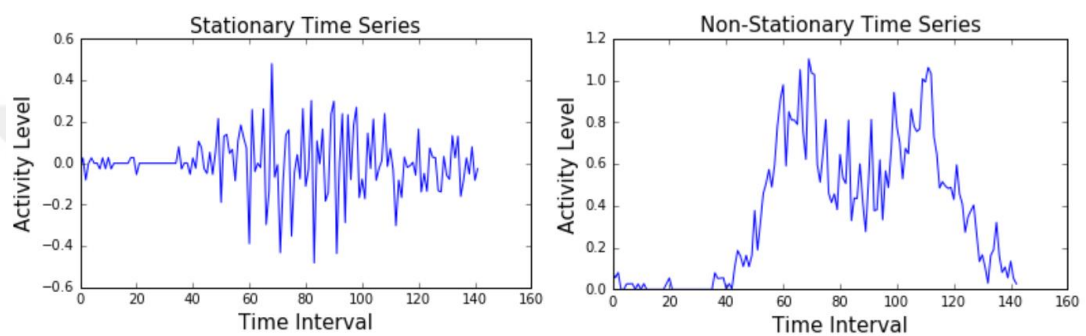


Figure 2.33 : Stationary and non-stationary random signals [41].

2.5.2 Filters

Digital filters are commonly used while processing the measured signals that are received with transducers. Digital filters are used for two purposes for analysing :

- Signal separation
- Signal restoration

The signal separation purpose is used for decomposing the measured noisy signals. The measured noise generally consists of a combination of many signals. In order to decompose the signal and obtain the raw signal of a specific occurrence, digital filters are used. These filters allow us to individually analyze each signal that makes up the final noisy signal. Signal restoration is another use of digital filters. Especially if the sound measurements are done with a poor equipment or the measured signal is distorted, digital filters are used to represent the original measured signal and remove the unwanted distortions.

On Figure 2.34 the frequency based digital filter responses can be investigated. These filters are used to block some frequencies completely while allowing other frequencies to pass without any changes. The low pass filter allows the signals below a specified

frequency to pass while blocking the other frequencies above this limit. The high pass filter on the other hand, blocks the frequencies below a specific limit while allowing the other frequencies above this limit to pass unaltered. The band pass filter allows the frequencies between specified limits pass without any alterations while band reject filter completely blocks the signals between specified limits and allows the other signals to pass without any changes. Since the filters are not constructed with ideal properties, there exists a transition band at cut off frequencies. As the order of the filter increases, the quality of the filter increases and the transition band is shortened.

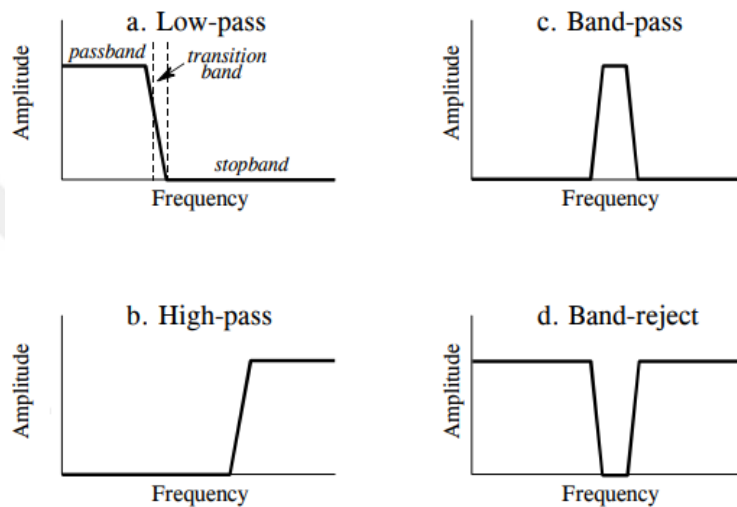


Figure 2.34 : Frequency domain filter types [42].

2.5.3 Aliasing

On digital signal processing, the importance of sampling rate can be seen on the phenomena of aliasing. In the case of sound signals, any input frequency which is higher than half of the specified sampling rate of f_s will cause ambiguities during representation. The half of the sampling rate is known as the Nyquist frequency. The sinusoidal inputs at frequencies Δf above the Nyquist frequency are the same as the ones Δf below the Nyquist frequency for a sampling rate of f_s . This ambiguity is called aliasing and is an unwanted phenomena in signal processing since it causes the wrong representation of a signal.

The aliasing effect of a sampled saxophone note can be investigated on Figure 2.35 and Figure 2.36. The saxophone sound is made up of harmonics since it's a very tonal instrument. The first portion of the figure corresponds to the filtered note with an anti aliasing filter. The sound signal is sampled at 2756 Hz and corresponding harmonics can be seen on the figure. However, if the anti aliasing filter is turned off, the fourth,

fifth and sixth harmonics are also seen on the figure. These harmonics are of higher frequencies and normally should show up at higher frequencies but due to the aliasing effect, the harmonics are seen at lower frequencies thus the signal is aliased.

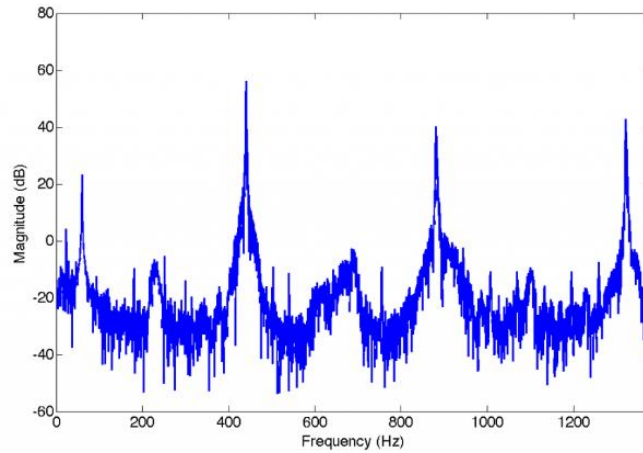


Figure 2.35 : The harmonics of a saxophone note with anti aliasing filter [69].

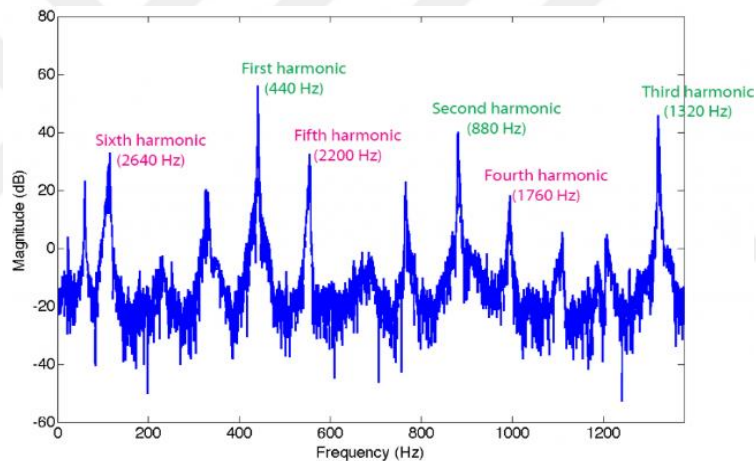


Figure 2.36 : The harmonics of a saxophone note without anti aliasing filter [69].

In order to prevent the aliasing of signals, a proper sampling rate needs to be used. Moreover, a low pass filter can also be used with a cut off frequency slightly lower than the Nyquist frequency [43].

2.5.4 Fourier transform

The Fourier transform is an extension of Fourier series and is widely used in signal processing. The Fourier transform is used to convert a time domain signal to a frequency domain. The frequency domain representation of signals is used to identify the frequency characteristics of a specific signal. The Fourier transform consists of real and imaginary parts and includes the negative frequency representations as well as positive frequencies. The positive and negative side of a Fourier transform is

symmetrical with respect to zero starting point. On Figure 2.37, a basic representation of the Fourier transform can be seen. A time domain signal in this case is converted to a frequency domain signal to analyse its characteristics. Inverse Fourier transform can also be used in this case to find the original signal as well. Both of these functions don't cause any information loss within the signals thus they are widely used. The Fourier transform of a signal of $x(t)$ can be calculated according to equation 2.44 :

$$X(w) = F[x(t)] = \int dt e^{-iwt} x(t) \quad (2.44)$$

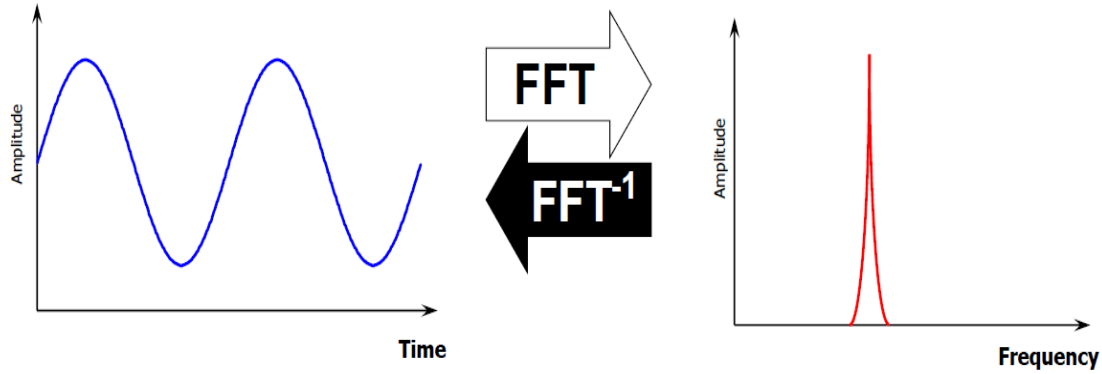


Figure 2.37 : The fourier transform and inverse fourier transform of a signal [44].

The Fourier integral includes all the corresponding negative and positive values. The most important few basic characteristics of FFT algorithms are defined below as referred to Cooley – Tukey Algorithm [45]:

- Number of data values for each FFT can be restricted to a power of 2. Namely, N being the number of data points $N = 2^p$ with p being between 8 and 12.
- The frequency resolution of each Fourier component can be defined as $\Delta f = \frac{1}{N\Delta t}$.
- The Nyquist frequency f_N can be seen after $k = \frac{1}{2}N$ th Fourier component.
- The first $\frac{1}{2}N + 1$ Fourier components until the Nyquist aliasing frequency are related with the last $\frac{1}{2}N - 1$ Fourier components after the Nyquist frequency. The relation can be defined as $X(k) = X^*(N - k)$ with $k = 0, 1, 2, \dots, N - 1$ and $*$ being the complex conjugate value.

- If only the positive one sided spectrum of Fourier components is required to be seen, this can be defined with $X(0)$, $X\left(\frac{N}{2}\right)$, $2X(k)$ with $k = 1, 2, \dots, \frac{N}{2} - 1$.

2.5.5 Leakage and windows

The FFT process provides very significant knowledge about the signal of interest but there are also limitations about the capabilities of FFT. When an FFT of a signal is taken, a finite set of data is analysed which means that the FFT assumes one period of a periodic signal and connects the ending points of these signals together. If the signal of interest has an integer number of periods, the FFT calculation provides correct information.

In most of the cases, the number of periods in a measured signal is not an integer. This results in end points being discontinuous for the signal. The discontinuities in this signal are seen as high frequency components on the FFT of the original signal and can be aliased between 0 and the Nyquist frequency. When the FFT of a discontinuous signal is taken, the energy leaks from one frequency to side frequencies. This effect on the signal is called spectral leakage. It causes the corresponding spectral lines to divide and spread into wider signals as seen on Figure 2.38.

The effect of spectral leakage can be minimized with a signal processing technique named windowing. Windowing functions allow to provide better knowledge about the signal by reducing the amplitude of each discontinuity especially at the boundaries of each signal. This is achieved by multiplying the time period of a measured signal with a varying amplitude to make the end points of each wave meet each other.

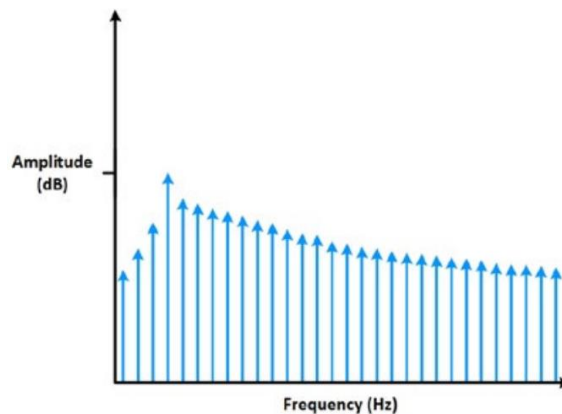


Figure 2.38 : Spectral leakage effect of an FFT [46].

However, when Figure 2.39 is investigated, if the windowing is applied correctly, it is seen that spectral leakages are significantly reduced. The appropriate windowing function is chosen according to the frequency characteristics of the measured signal.

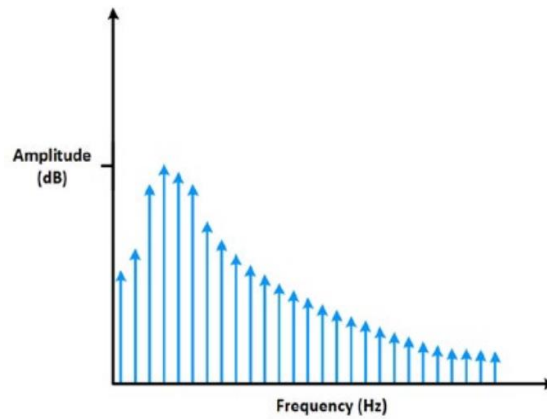


Figure 2.39 : Minimized leakage with windowing [46].

2.6 Sound Source Identification

The identification of sound sources is carried out by an extensive spectral analysis with the knowledge of rpm of the rotating machinery components that produce the noise in the case for periodic excitation. As it can also be investigated on Figure 2.40, each rotating machinery has various rotational frequency when the overall noise is investigated. The components that contribute to the overall noise level can be distinguished from each other in this case considering that they don't operate at the same frequency. The downside of this method is, it can't differentiate two components if they are operating at the same rotational frequency.

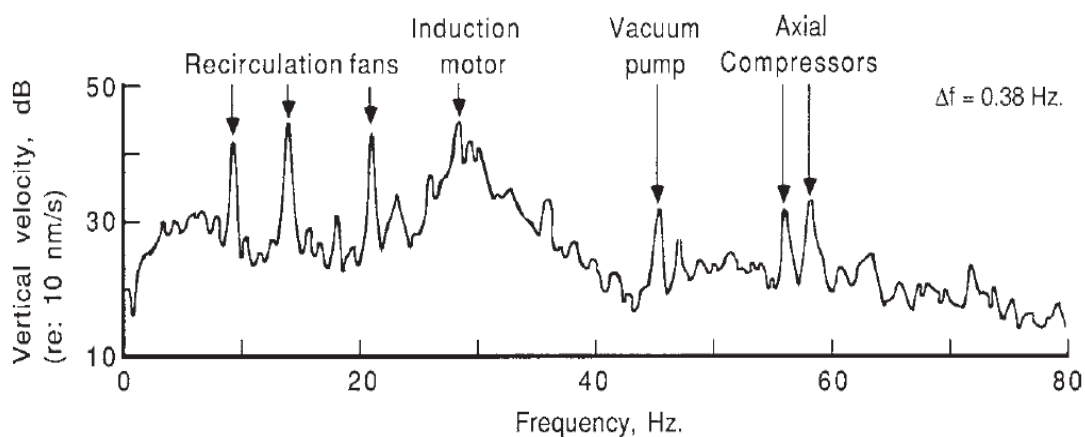


Figure 2.40 : Fourier spectrum of a vibration data in a facility [47].

In case of random excitation sources, the sound source identification method differs. Since the noise data is totally random, the identification process becomes significantly more difficult. The most mainstream way of finding the random excitation sources one by one is to turn off all the components, and turn on one by one to calculate their contribution. Since not every system allows us to follow this procedure, the coherent output power method can also be used. This method requires measuring the acoustical signal at the source locations for each of the noise sources. For each of these acoustical signals from the sources, the receiver signals $y(t)$ are also measured at the same time and coherence function between each source and receiver signals are calculated using equation 2.45. Following the determination of coherence function, the autospectrum of the receiver signal that is only resulting from a single i th source can also be found using equation 2.46 as given below :

$$\gamma_{xy}^2(f) = \frac{|G_{xy}(f)|^2}{G_{xx}(f)G_{yy}(f)} \quad (2.45)$$

$$G_{y:i}(f) = \gamma_{iy}^2(f) G_{yy}(f) \quad (2.46)$$

The equation 2.46 simply states the part of the autospectrum of the receiver that is resulted solely from a single source. This method is useful for separating non periodic signals in acoustic noise problems. In order to correctly apply this method, the procedure below needs to be followed [48] :

- The excitation sources must be measured accurately. The measurement noise in this case should be negligible. Any transducer that demonstrates a linear relationship with the sound excitation can be used for these measurements.
- The sources should be independent from each other from the aspect of statistics. In order for this method to be effective, there must be zero interference during the measurements and zero coherence among the measured source signals.
- Nonlinear effects and feedback should be negligible between the source and receiver signals.

The drawback of this method comes from the time delays between the source and receivers. The time delay may be significant when there is a measurement position difference between the source and receiver. In order to compensate for this time delay,

precomputation or a long measurement duration can be used [49]. The application of this method can be seen on Figure 2.41 in which a panel with an accelerometer attached is excited with random vibration and the resulting noise is measured with a microphone near the panel.

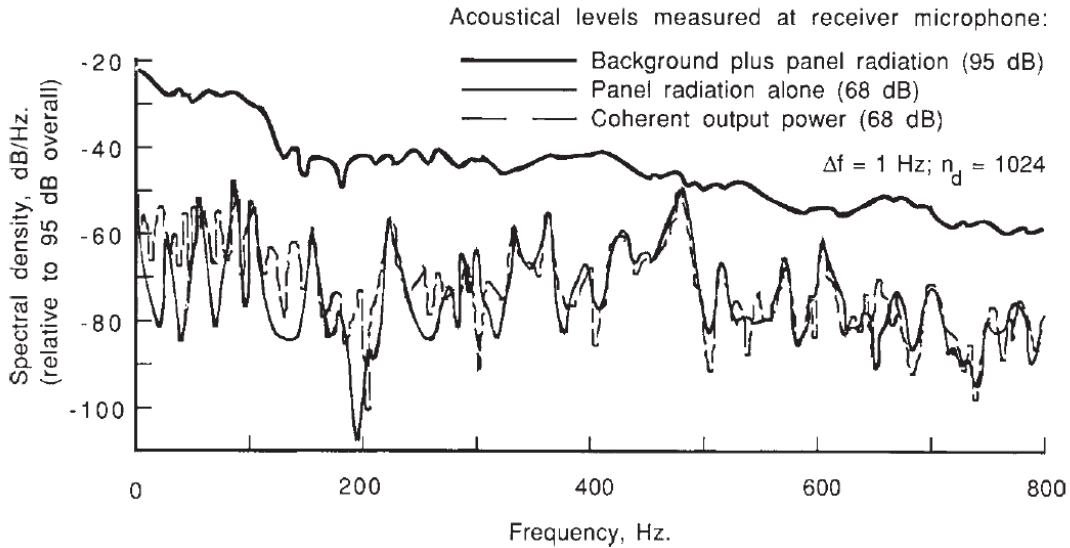


Figure 2.41 : Coherent output power spectrum to determine panel radiation [48].

The autospectrum of the radiated noise is shown on the Figure 2.41 as a thin line while the total noise that includes both the panel vibration and background noise is shown with a heavy solid line. The coherent output power is then calculated between the microphone and the attached accelerometer at the source and is drawn with a dashed line. As it can be concluded, the correct use of coherent output power method yields precise results by extracting the autospectrum of the panel noise from the overall noise.

The sound source identification methods described in this section are more likely to be used for systems which include few components and are mainly periodic. For more complex systems with higher numbers of variables involved, all factors affecting the propagation and the production of the noise should be identified and considered to prevent any erroneous calculations.

3. EXPERIMENTAL STUDIES

Experimental studies are handled in four different mediums. The first measurements took place inside the hemi anechoic room to define the reference values and the response of various microphones inside a controlled environment. The secondary measurements took place inside a Ford Transit cabin to simulate the response of these microphones under real world conditions. During the third measurement, the effectiveness of mobile device microphone under operating vehicle conditions is tested. The fourth experiment consisted of the investigation of change in mobile device with regards to its effects on SPL readings in Ford Kuga.

3.1 Hemi Anechoic Room Measurements

The professional measurements require purpose specific high quality condenser microphones in order to assess the sound metrics precisely. Before conducting any tests, the microphones are calibrated to correctly define the sensitivity of the microphone for the accuracy of the measurement. The calibrators are available for half and quarter inch microphones and they transmit the signal at 1 kHz and 94 dB to find the correct microphone sensitivity. The study aims to define the capabilities of mobile microphones and their calibration but a standard procedure for microphone calibration is not available. There are no calibrators that can be used for the calibration of mobile device microphones. In order to define the capabilities of mobile device microphones, the hemi-anechoic room at Ford Otosan Gölcük Facility is used as seen on Figure 3.1.



Figure 3.1 : Hemi Anechoic room in Ford Otosan Gölcük facility.

3.1.1 Test setup within the hemi anechoic room

The experiment consisted of a few white noise signals transmitted at various amplitudes. The signals are generated at MATLAB environment to be able to get the precise frequency spectrums correctly. 1 condenser B&K microphone and 2 built-in mobile device microphones are used for the initial measurements at the hemi-anechoic room :

- B&K Type 4189
- Samsung SM-P587 Tablet
- Iphone X

The B&K Type 4189 ½ inch free field condenser microphone has been calibrated using the B&K Type 4231 acoustic calibrator as seen on Figure 3.2 at 1000 Hz 94 dB SPL with LMS Test Lab software and the resulting microphone sensitivity is used for calibration. This calibration procedure needs to be followed before each measurement session in order to be sure about the precision of the measurement results.



Figure 3.2 : B&K Type 4231 acoustic calibrator [52].

The B&K Type 4189 ½ inch free field condenser microphone has been used in conjunction with HEAD Acoustics Squadriga II to collect the sound measurements. The frequency response curve below on Figure 3.3 is the original curve for the B&K condenser microphone. As seen from the graph, the frequency response is very linear across all frequency ranges. This is a very desired condition for the measurement microphones as the amplitudes are linear with increasing frequencies. Due to the flat frequency response, ease of calibration and precise measurement results, the B&K microphone is used as the reference microphone for the hemi – anechoic room measurements.

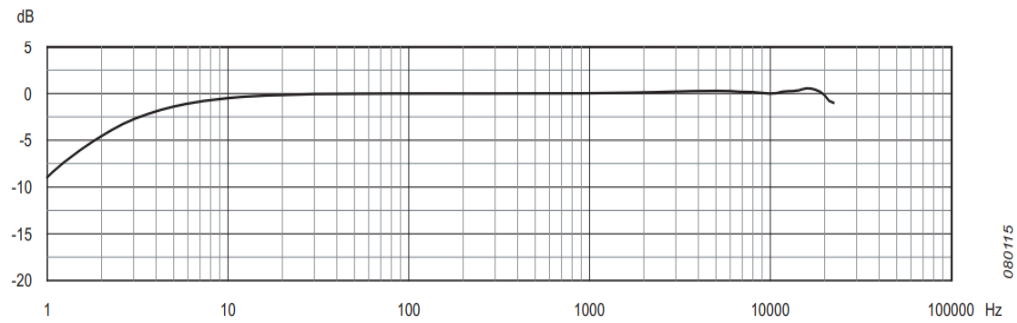


Figure 3.3 : B&K Type 4189 frequency response curve [50].

The other two microphones are the standard built – in microphones that are situated inside the iPhone X and Samsung SM-P587 tablet. The frequency response of these microphones are not commercially available to public yet since these devices are not designed for professional purposes, the frequency response is not expected to be flat as a professional condenser microphone. Contrary to the condenser microphones, the microphones situated inside these mobile devices are MEMS microphones – small chips inside the structure of the device. These microphones are not directional but due to their structure, their directionality is limited with the case of the device. Due to this reason, for the measurements the microphones of these devices are directed to the sound source inside the hemi-anechoic room.

The sound source for for the transmission of the signals is also important in order to define the capabilities of the microphones. The sound source used in this case is the Pre – Sonus Eris E5 studio standard high quality calibrated loudspeaker. These loudspeakers are able to transmit frequencies between 53 Hz – 22 kHz effectively [53]. Avid Fast Track Solo external sound card is also used in conjunction with the Eris E5 loudspeakers to control and generate the signals precisely. The capability of this setup has been tested with various dynamometer applications including engine tests before the measurements to make sure about the confidence of the system. The signals are transmitted at various sound pressure levels to define the variability in precision of the mobile microphones at different amplitudes. The amplitudes are controlled via the external sound card in order to adjust the changes as desired within the range of the loudspeakers. The amplification levels and all the settings of used hardware are kept constant throughout the measurement in order to clearly compare the values with regards to different amplitudes. The initial setup for the loudspeaker and external sound card is given below on Figure 3.4.



Figure 3.4 : Pre Sonus Eris E5 & Avid Fast Track Solo.

Two tripods are used to adjust the height of the Samsung SM-P587 tablet microphone and B&K Type 4189 condenser microphone. In the meantime, iPhone X is also held at the same height and distance from the loudspeaker during the measurements. The microphones are directed to the loudspeaker in order to eliminate any errors that may arise from the directionality of the sound. Moreover, the distance and height of the hardware has also been checked with a measurement tape.

To be able to satisfy the free field conditions, the microphones are placed at 1 meter height and 1.5 meters away from the sound source. The tablet has been attached to the tripod to prevent the microphone from moving during the measurements. All the other possible reflective equipments within room are covered with foam and absorber porous materials to eliminate the reflections from nearby surfaces as much as possible. All the signals are transmitted without changing the positions or the placement of these microphones. The test setup for the initial measurements can be seen on Figure 3.5 below.



Figure 3.5 : The initial test setup for hemi anechoic room measurements.

3.1.2 Results of the hemi anechoic room measurements

The measurements are conducted with 4 different signals, namely 1500 Hz, 4000 Hz, a sweep sine signal between 300-8000 Hz and white noise. White noise is also repeated once more with different amplification to increase the data points for calibration purposes. Background noise is also measured before each test to see the difference between the microphones in case of low sound pressure levels. The sound pressure levels stated below at Table 3.1 are the reference values taken with B&K condenser microphone. These values are the basis for all other measurements with various mobile device microphones within the same hemi – anechoic room. A weighting is applied to all the measurements below to simulate the human hearing.

Table 3.1 : B&K Type 4189 SPL (dB(A)) in hemi anechoic room.

Sound Level	Signals				
	Pure Tone (1500 Hz)	Pure Tone (4000 Hz)	White Noise (Normal)	White Noise (Amplified)	Sweep (300-8000 Hz)
10 %	51.11	62.12	52.28	58.61	51.87
30 %	72.32	82.33	71.90	78.29	71.34
50 %	79.53	89.92	80.51	85.93	79.28
70 %	87.15	95.44	85.68	91.32	85.15

While taking the measurements with B&K microphone, the same signals are collected with Samsung SM-P587 and iPhone X. The measured sound pressure levels of Samsung SM-P587 can be seen below on Table 3.2. When compared to the reference condenser microphone B&K Type 4189, these values are not accurate enough for professional measurements.

Table 3.2 : Samsung SM-P587 SPL (dB(A)) in hemi anechoic room.

Sound Level	Signals				
	Pure Tone (1500 Hz)	Pure Tone (4000 Hz)	White Noise (Normal)	White Noise (Amplified)	Sweep (300-8000 Hz)
10 %	64.45	71.02	67.20	73.56	60.75
30 %	81.98	88.65	86.08	89.43	81.76
50 %	85.76	96.63	89.85	89.74	88.50
70 %	92.63	99.22	90.11	90.03	92.53

Table 3.3 shows the measured values of iPhone X under the same conditions inside the hemi anechoic room. It is seen that iPhone X values converge better to B&K Type 4189 values.

Table 3.3 : iPhone X SPL (dB(A)) in hemi anechoic room.

Sound Level	Signals				
	Pure Tone (1500 Hz)	Pure Tone (4000 Hz)	White Noise (Normal)	White Noise (Amplified)	Sweep (300-8000 Hz)
10 %	47.82	57.78	51.91	57.21	53.97
30 %	71.23	80.45	69.95	75.63	71.68
50 %	80.26	87.44	78.32	83.63	79.18
70 %	86.00	93.06	83.52	88.91	85.00

The amplitudes that were measured for various signal types inside the hemi anechoic room were far higher than the noise floor of the medium. In order to see the performances of these microphones for very low sound pressure levels, the background noise of the medium is also measured. As expected, the results showed that under very low sound pressure levels, the performance of the mobile microphones are very low. This is mostly due to the poor SNR characteristics of the silicon MEMS microphones inside these mobile devices. The measurement results can be seen on Table 3.4 below.

Table 3.4 : Background noise SPL (dB(A)) in hemi anechoic room.

Device	Background Noise Level
Samsung SM-P587	25.32
B&K Type 4189	39.50
iPhone X	35.42

Level vs Time analysis is conducted on Artemis Suite for each the measurements over a specific time period. As it can be seen from the charts above, the iPhone X microphones are far superior than Samsung SM-P587 tablet microphones across all sound pressure levels. The highest difference between the B&K microphone and mobile device microphones are seen at lower sound pressure levels around 50-60 dB(A). Due to the test setup and the hardware that is used, for pure tone signals of 1500 Hz and 4000 Hz, 3rd or 2nd harmonics of these signals are seen on FFT vs. Time plots that adds up to the total value of these signals. Since the contribution of these harmonics are low to the overall level, these harmonics can be accepted as negligible. The frequency spectrum of these signals can be seen below on Figure 3.6.

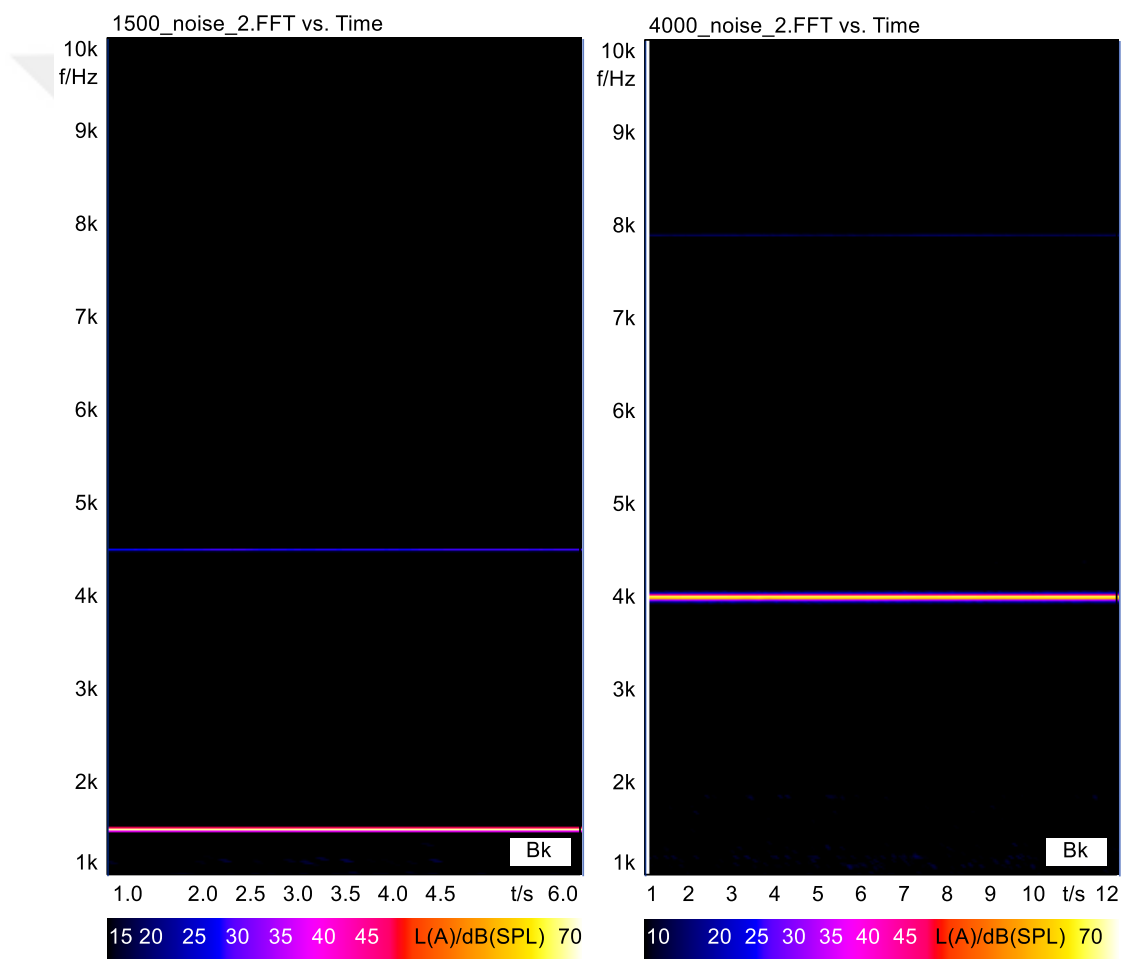


Figure 3.6 : 1500 Hz (Left) - 4000 Hz (Right) harmonics in hemi anechoic room.

Contrary to the traditional calibration method of 1000 Hz signal transmittal at 94 dB, a various signal of white noise is used for calibration to include the frequency response of these mobile microphones across all frequency ranges. Since these mobile microphones don't have a flat frequency response, the total frequency range needs to be included for calibration to define the difference in amplitude at various frequencies.

According to the test results, as seen on Figure 3.7, the Samsung tablet amplifies the results especially around 7-10 kHz which corresponds to a significant level difference between these frequencies. A more homogeneous distribution is seen on B&K Type 4189 which is the expected spectrum of the signal.

When the iPhone X data is investigated, it is seen that the in-built microphone of iPhone X is more capable than Samsung SM-P587 in every way including both the level and signal spectrum. As seen on Figure 3.7 below, the frequency response of iPhone X built-in microphones are similar to B&K reference microphone. With comparison to the Samsung SM-P587, the iPhone X microphone is more suitable to professional measurements even without the calibration.

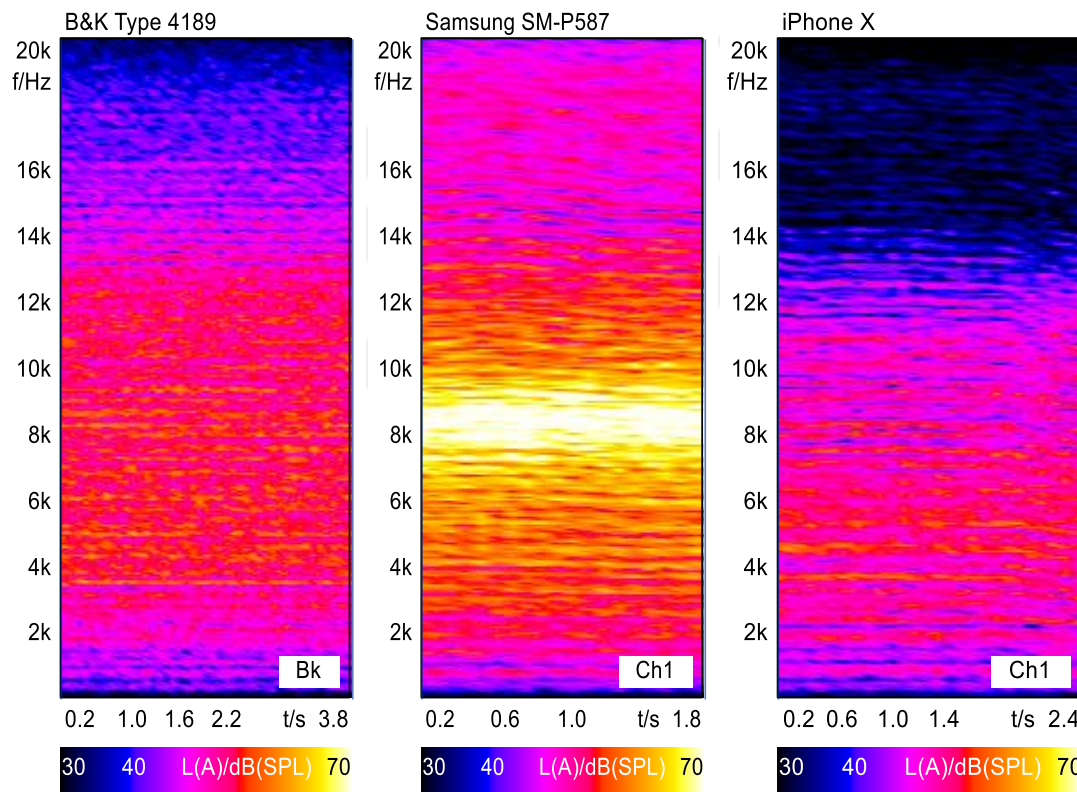


Figure 3.7 : FFT spectrum of tested devices in hemi anechoic room.

1/3 octave band frequencies are also investigated to precisely define the difference in various frequencies for these devices. These results strengthen the color maps for white noise signals as the higher frequencies around 7-10 kHz are dominated on Samsung tablet while for iPhone the amplitude distribution is similar to B&K reference microphone. The octave band analysis for white noise signal on each of these devices can be seen on Figure 3.8 below. The calibration of Samsung tablet is a must according to these results as the gap with the reference microphone is significant.

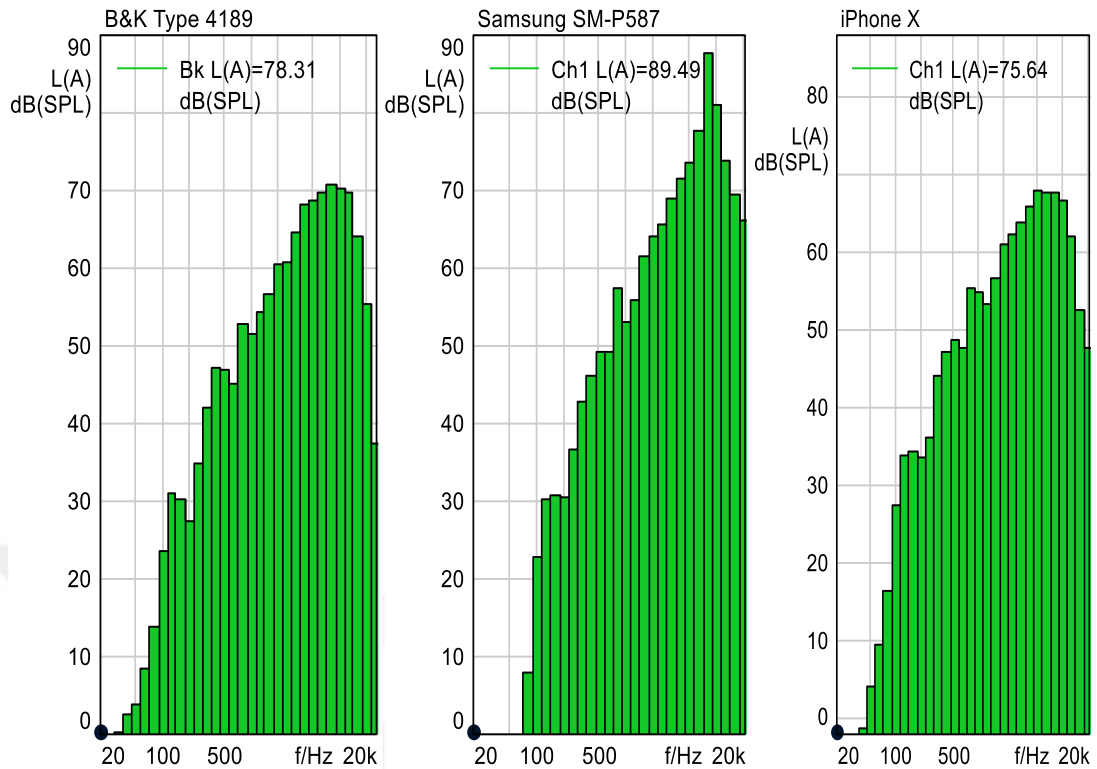


Figure 3.8 : 1/3 octave band analysis of tested devices in hemi anechoic room.

A brief overview of the amplitudes of octave band center frequencies for a specific white noise signal is given on Table 3.5. The table implies that the lower center frequencies are not obtained clearly with Samsung SM-P587 while iPhone X follows a similar distribution with B&K Type 4189.

Table 3.5 : Octave band SPL (dB(A)) in hemi anechoic room for each device.

Center Frequency	B&K Type 4189	Samsung SM-P587	iPhone X
31.5	1.87	-	-3.58
63	7.89	-	8.98
125	30.68	30.35	34.92
250	34.46	36.07	35.95
500	46.97	49.08	48.68
1000	51.99	53.15	55.41
2000	60.69	64.06	61.24
4000	68.24	71.49	66.17
8000	70.81	87.72	67.80
16000	64.24	69.56	52.61

A level based calibration algorithm is followed for the purpose of calibration of these mobile devices. Both for iPhone X and for Samsung SM-P587 tablet, the reference values are taken as the B&K microphone outputs. A linear interpolation is used to

calculate the calibration curves of these devices. The resulting line corresponds to the best possible performance of these mobile devices without the use of any external microphones under the controlled conditions of hemi-anechoic room. The initial calculations are conducted on the Samsung SM-P587 tablet. Since the acoustic overload point for the Samsung tablet is around 90 dB(A), the readings on Samsung tablet doesn't go over this value even if the signal amplitude is increased. This is a critical information since the tablet shouldn't be used over this sound pressure level. Figure 3.9 shows the calibration line of this Samsung SM-P587. The regression curve shows an R^2 value of 0.8899 for the correlation between B&K Type 4189 and Samsung SM-P587. The reason for R^2 value not to be higher is the conclusion that the overload point of the tablet is around 90 dB(A) limit.

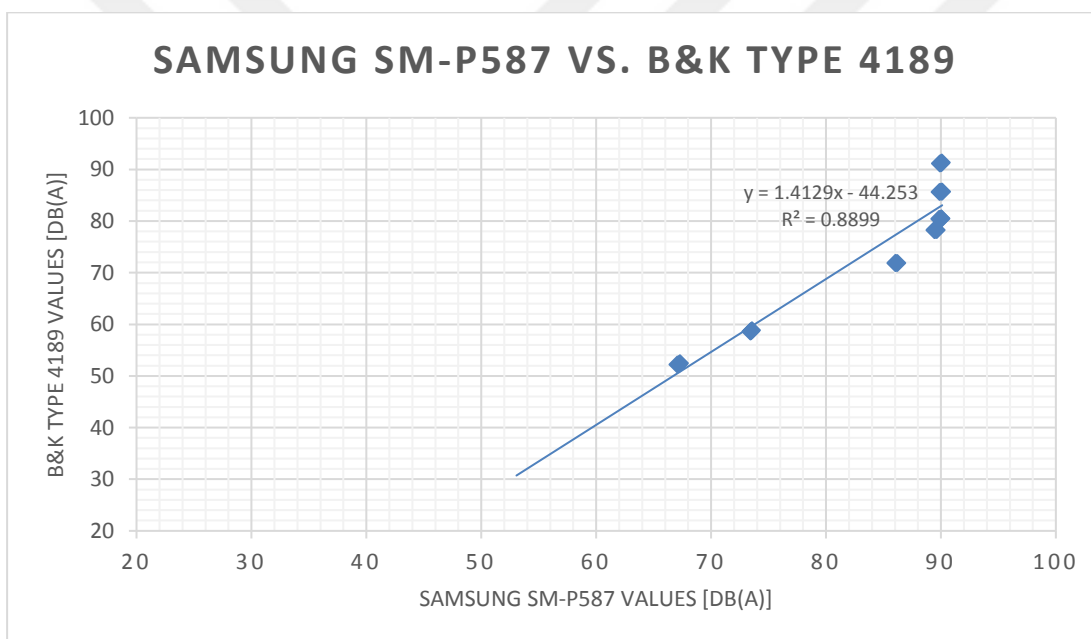


Figure 3.9 : Calibration line of Samsung SM-P587 in hemi anechoic room.

The same procedure is also followed for iPhone X as well. Since the readings on iPhone X were already similar to those on B&K microphones, a small adjustment on the readings give precise results. After the calibration, iPhone X can act as a professional sound level meter according to these results since the built-in microphones are nearly in line with the reference microphone. On Figure 3.10, the calibration line of iPhone X can be seen. The line shows very good linearity when compared with the reference results that were measured with B&K Type 4189 condenser microphone.

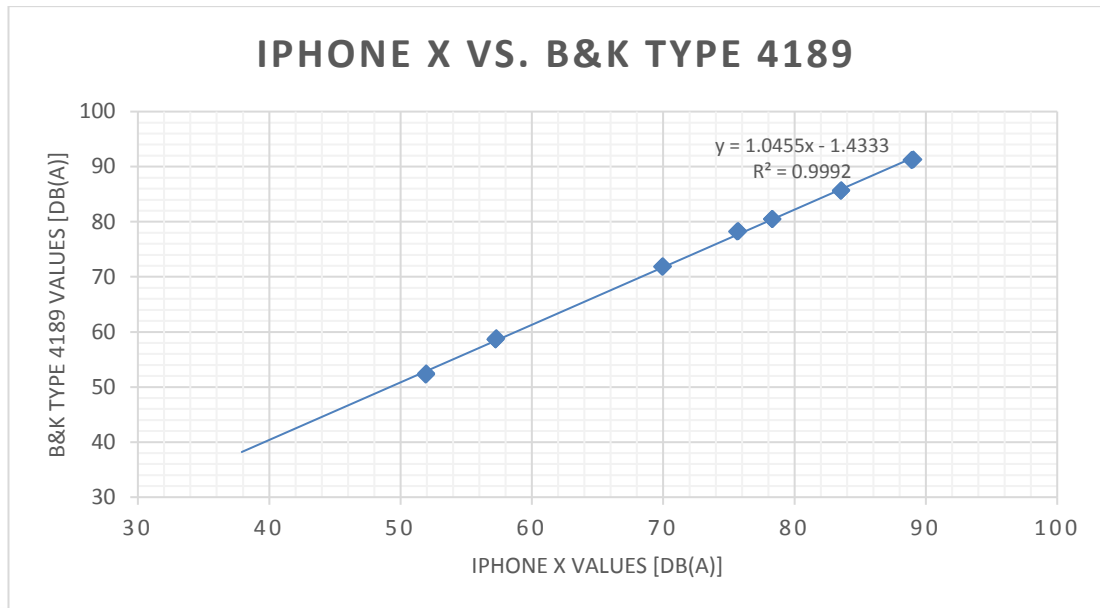


Figure 3.10 : Calibration line of iPhone X in hemi anechoic room.

The lower sound pressure levels are more different than B&K Type 4189 readings especially for the the Samsung tablet. The main reason behind this difference in sound pressure levels is the low signal to noise ratio of the built – in microphones. Since the android tablet has a low SNR, the low sound pressure levels are not identified accurately. An overview table of the effect of calibration on each device can be seen on Table 3.6 below.

Table 3.6 : The effect of calibration on each device in hemi anechoic room.

B&K Type 4189	Before Calibration		After Calibration	
	Samsung SM-P587	iPhone X	Samsung SM-P587	iPhone X
52.28	67.20	51.91	50.69	52.84
58.61	73.56	57.21	59.68	58.38
71.90	86.08	69.95	77.37	71.70
78.25	89.43	75.63	82.10	77.64
80.51	89.85	78.32	82.70	80.45
85.68	90.11	83.52	83.06	85.92
91.32	90.03	88.91	82.95	91.67

The calibration of Samsung SM-P587 is conducted on FON mobile application for Android devices. Since the application only supports Android devices for now, the calibration is only applied for this mobile device and iPhone X calibration values are left for later application releases. Since each of these mobile devices has various internal filters and processors, the sound pressure level readings at each of these devices are different. Therefore, different calibration values and curves are required for each of the mobile devices seperately. The calibration values of Samsung SM-P587

can be seen below on Figure 3.11. It is seen that at higher sound pressure levels, the microphone capabilities are very low for white noise signals.

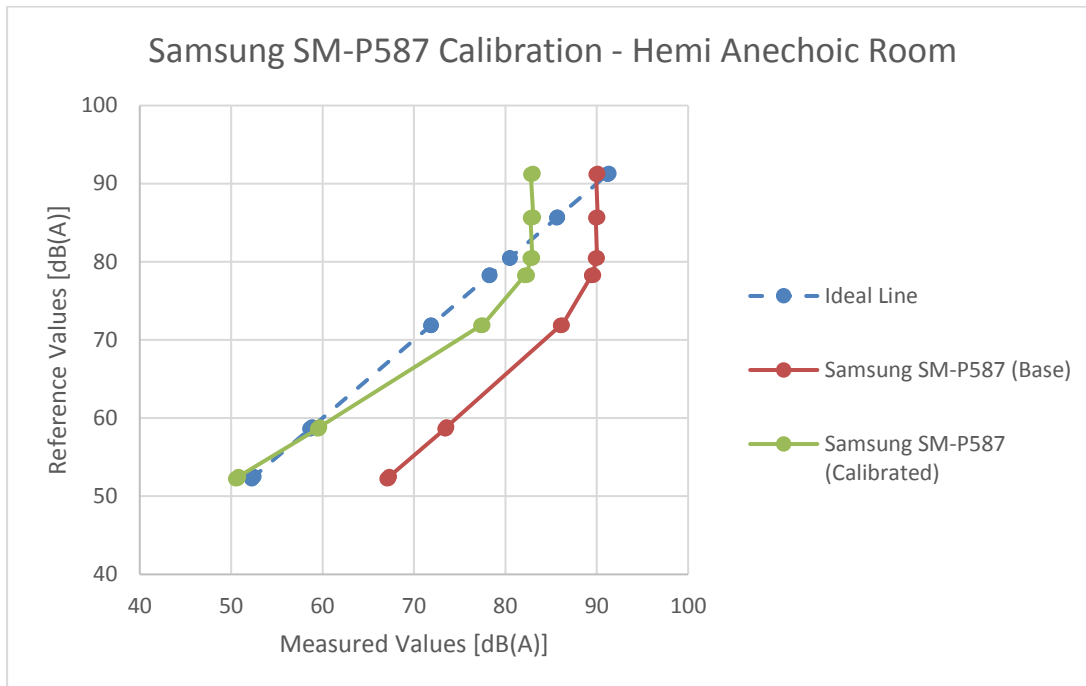


Figure 3.11 : Calibration values of Samsung SM-P587 in hemi anechoic room.

The same procedure is followed for iPhone X and the calibration lines are drawn in comparison to the ideal line for the hemi anechoic room measurements. Figure 3.12 clearly shows that especially with the calibration, iPhone X values are nearly identical to B&K Type 4189.

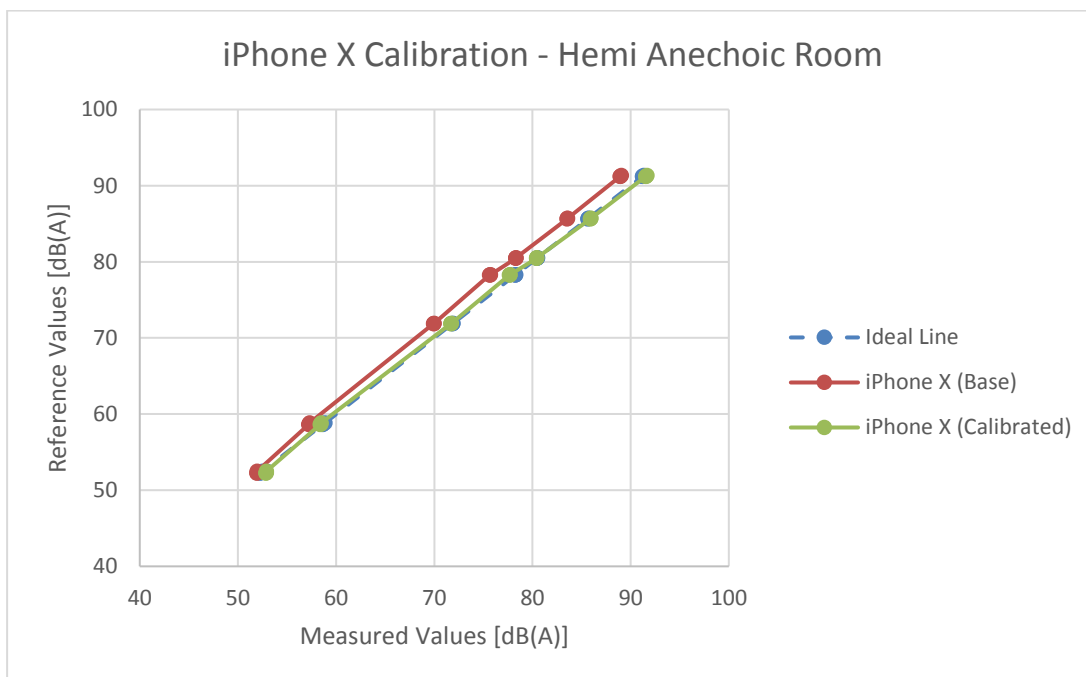


Figure 3.12 : Calibration values of iPhone X in hemi anechoic room.

The controlled testing is completed yet for the calibration of mobile devices, hemi-anechoic rooms are not always available. Therefore, an easier way for calibration of the mobile devices is needed to be found for practical purposes as well.

3.2 Test Vehicle Cabin Measurements

The controlled medium measurements are conducted within the hemi anechoic room and the results proved to be the reference values for the real world conditions. In order to simulate the behavior of these mobile microphones under real world conditions, the same test is conducted with the same equipment and settings in a different medium. A Ford Transit has been used as the vehicle cabin and the same procedure is conducted to find the results in this case. The difference between the controlled experimental medium results and a vehicle cabin with reflections is aimed to be found.

3.2.1 Test setup within the vehicle cabin

In order to find the response of the initial mobile devices in a cabin rather than a controlled environment, a test vehicle has been used. Since the most complicated variant in terms of interior trim of the test vehicle was a bus, the variant has been selected as 14+1 bus. The vehicle and test setup can be seen on Figure 3.13.



Figure 3.13 : Test vehicle 14+1 bus variant and its interior.

Tripods are used to adjust the height and distance of the mobile devices and microphones 1.5 meters away from the loudspeaker. The loudspeaker is also adjusted

next to the rearmost seat within the cabin at the same height and directly faced the microphones. Figure 3.14 shows the exact experimental setup inside the vehicle cabin.

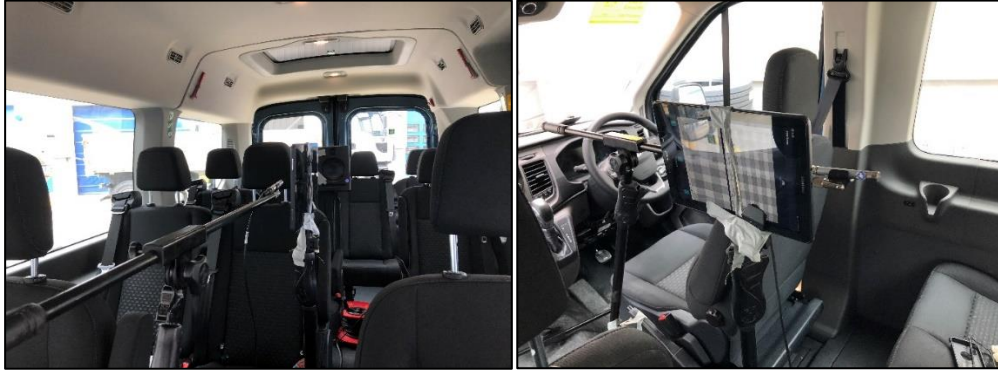


Figure 3.14 : Experimental setup within the test vehicle cabin.

The frequencies of interest were 1500 Hz, 4000 Hz, white noise and a sweep sine between 300 Hz – 8 kHz generated via MATLAB. Since the testing environment is not controlled, the signals are expected to generate reflections from surrounding parts and equipment inside thus producing a more complicated frequency spectrum.

3.2.2 Results of the vehicle cabin measurements

After the measurement setup has been prepared, the signals are played under seven different amplitudes from the loudspeaker. The amplitudes are adjusted from the internal sound card of the computer. The results and comparison of every signal measurement with B&K Type 4189 condenser microphone are given below on Table 3.7. The pure tone signals of 1500 Hz and 4000 Hz resulted in slightly higher amplitudes inside the hemi-anechoic room.

Table 3.7 : B&K Type 4189 SPL (dB(A)) for each medium.

Test Area	Sound Level	Signals			
		Pure Tone (1500 Hz)	Pure Tone (4000 Hz)	White Noise (Normal)	Sweep (300-8000 Hz)
Test Vehicle Cabin	10 %	55.34	54.76	53.83	52.19
	20 %	57.17	71.35	66.39	63.34
	30 %	67.69	79.57	73.09	71.66
	40 %	71.06	82.81	77.79	76.67
	50 %	77.40	86.77	81.65	80.27
	60 %	79.31	89.14	84.08	82.73
	70 %	82.59	93.85	86.72	85.52
Hemi Anechoic Room	10 %	51.11	62.12	52.28	51.87
	30 %	72.32	82.33	71.90	71.34
	50 %	79.53	89.92	80.51	79.28
	70 %	87.15	95.44	85.68	85.15

As seen from Figure 3.15, even though the third harmonic of 1500 Hz is and the reflections are more clearly seen inside the cabin on the frequency spectrum, the overall level of the cabin measurement for this frequency is lower than the ones measured inside the hemi anechoic room. The main reason for this is most probably due to the interior cabin and the hemi-anechoic room natural mode differences. For the reflections however, it is seen that the equipment inside the vehicle cabin affected the noise all around the frequency spectrum especially at lower frequencies.

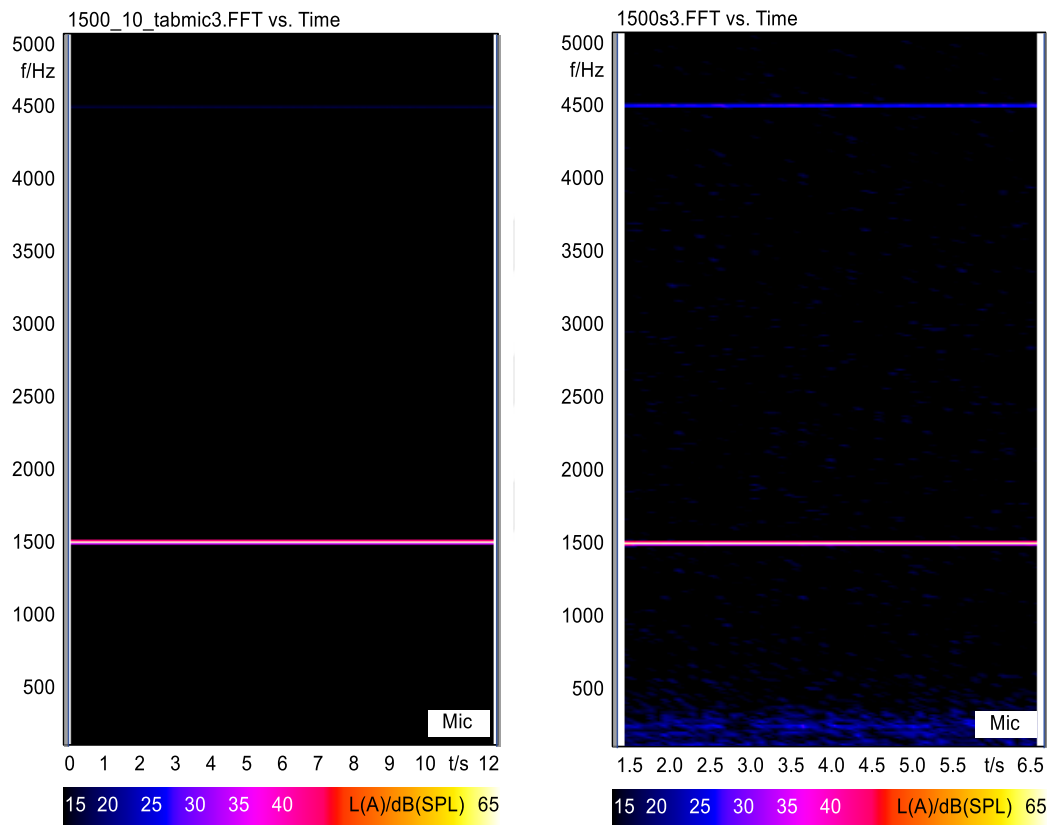


Figure 3.15 : Visible harmonics in hemi anechoic room (Left) and vehicle (Right).

With regards to the other signals and frequencies, it is seen that the overall levels measured inside the hemi anechoic room and vehicle cabin are similar but there is a slight increase in sound pressure level reading inside the cabin due to reflections and the cabin behaviour. When the performance of the mobile devices are taken into consideration, it is seen that the behaviour of mobile device microphones inside the cabin are unpredictable and results in worse results for pure tone signals when compared to the hemi anechoic room. Contrary to the pure tone signals, the swept sine between 300 Hz and 8 kHz gave similar results while the white noise response of the mobile device is better inside the cabin as it can be seen from Table 3.8 and Table 3.9 for Samsung SM-P587 and iPhone X respectively.

Table 3.8 : Samsung SM-P587 (dB(A)) for each medium.

Test Area	Sound Level	Signals			
		Pure Tone (1500 Hz)	Pure Tone (4000 Hz)	White Noise (Normal)	Sweep (300-8000 Hz)
Test Vehicle Cabin	10 %	65.83	68.52	70.89	61.80
	20 %	75.86	83.23	83.86	72.76
	30 %	81.71	92.03	88.19	82.29
	40 %	86.62	95.55	89.47	86.80
	50 %	88.53	96.86	89.65	89.30
	60 %	92.38	97.97	89.68	91.24
	70 %	93.77	99.25	89.76	92.73
Hemi Anechoic Room	10 %	64.45	71.02	67.20	60.75
	30 %	81.98	88.65	86.08	81.76
	50 %	85.76	96.63	89.85	88.50
	70 %	92.63	99.22	90.11	92.53

Table 3.9 : iPhone X SPL (dB(A)) for each medium.

Test Area	Sound Level	Signals			
		Pure Tone (1500 Hz)	Pure Tone (4000 Hz)	White Noise (Normal)	Sweep (300-8000 Hz)
Test Vehicle Cabin	10 %	64.27	61.02	54.02	54.93
	20 %	68.00	78.43	65.76	64.73
	30 %	78.04	80.65	72.22	72.34
	40 %	82.74	84.06	76.75	76.97
	50 %	83.77	92.10	80.43	81.06
	60 %	86.14	93.16	82.93	83.33
	70 %	90.05	96.26	85.42	85.96
Hemi Anechoic Room	10 %	47.82	57.78	51.91	53.97
	30 %	71.23	80.45	69.95	71.68
	50 %	80.26	87.44	78.32	79.18
	70 %	86.00	93.06	83.52	85.00

White noise signals are again used for calibration purposes since they naturally contain all the frequencies of interest for powertrain related error modes within the vehicle. The color spectrum is investigated initially to check the differences in frequency responses for each device inside the cabin. As seen from Figure 3.16, the amplification of Samsung SM-P587 tablet is again present and the iPhone X is still not good enough for frequencies above 14 kHz. The amplification of Samsung SM-P587 makes it nearly impossible to accurately measure the sound pressure levels that have a frequency between 4-12 kHz. This frequency range generally corresponds to high frequency turbo noise region. iPhone X however, shows a very similar characteristic to B&K Type 4189 around this region.

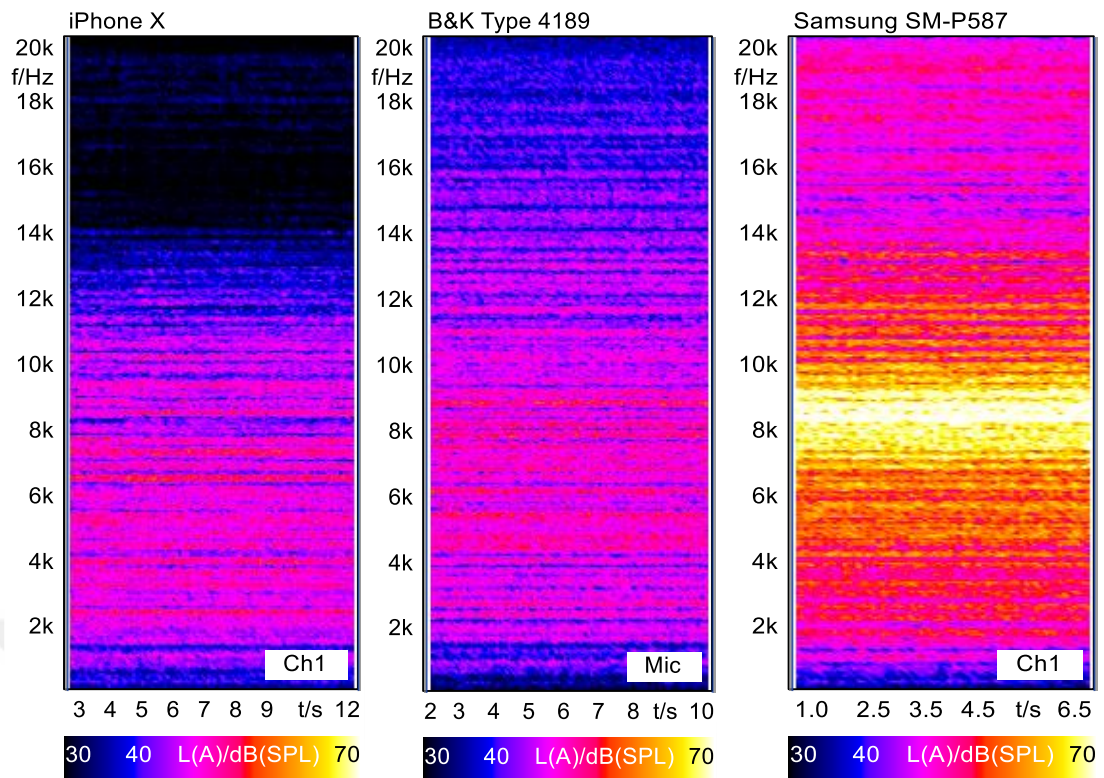


Figure 3.16 : White noise frequency responses of microphones inside the cabin.

Octave band frequencies have been investigated for the white noise with 30% amplitude to check the response of these microphones in detail. iPhone X and B&K Type 4189 respond similar under various frequencies yet as it can also be seen from the spectrum of the Samsung SM-P587, the higher frequencies are amplified significantly which deteriorates the overall results for this tablet.

Table 3.10 and Figure 3.17 below defines the differences of measured sound pressure levels. iPhone X dominates the Samsung SM-P587 tablet overall especially at frequencies above 1 kHz.

Table 3.10 : Octave band SPL (dB(A)) in the test vehicle cabin for each device.

Center Frequency	B&K Type 4189	Samsung SM-P587	iPhone X
31.5	25.17	-	19.04
63	30.50	22.81	31.59
125	33.34	34.13	37.47
250	39.11	41.97	42.17
500	45.57	48.63	48.18
1000	54.24	58.47	56.04
2000	61.80	67.97	63.05
4000	67.03	74.58	67.18
8000	69.83	87.80	69.15
16000	65.94	74.92	58.62

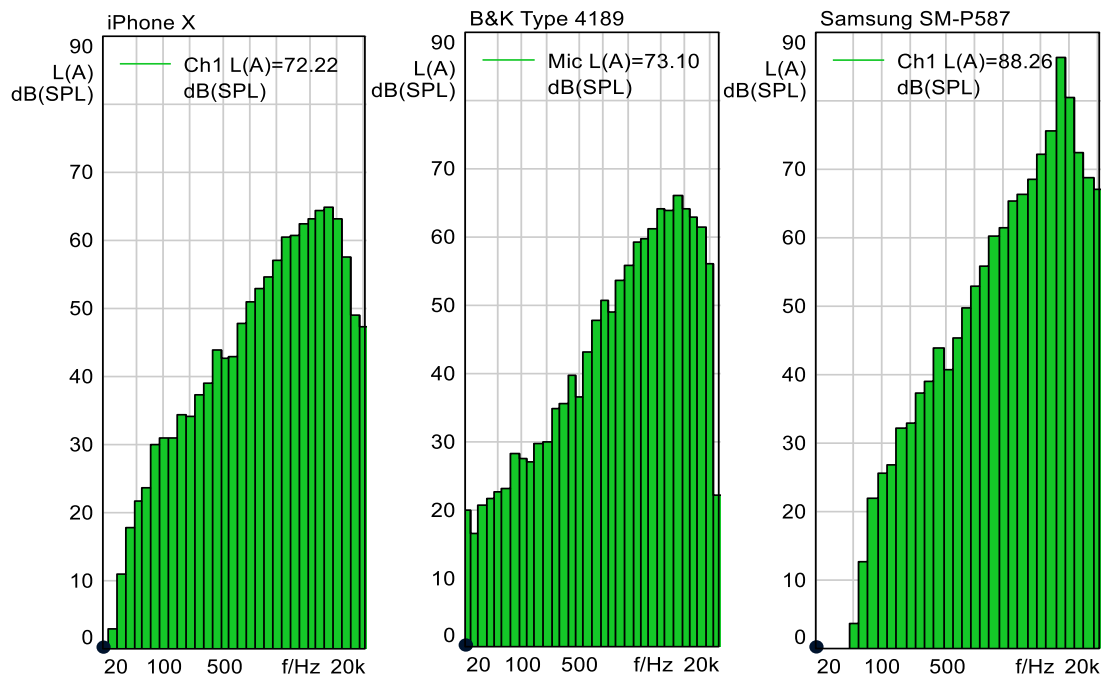


Figure 3.17 : 1/3 octave band analysis of tested devices in the test vehicle cabin.

It is seen from these results that the usage of this Samsung tablet or any other device with non-uniform frequency response will give faulty results. This may cause problematic measurements when the devices are used for professional purposes.

The level based calibration algorithm is again followed for the purpose of calibration of these mobile devices under an uncontrolled vehicle cabin environment. The reference values are taken as the B&K microphone outputs. The method of calibration is linear interpolation as stated at the previous section for hemi-anechoic room measurements. The calibration line for Samsung SM-P587 can be seen on Figure 3.18.

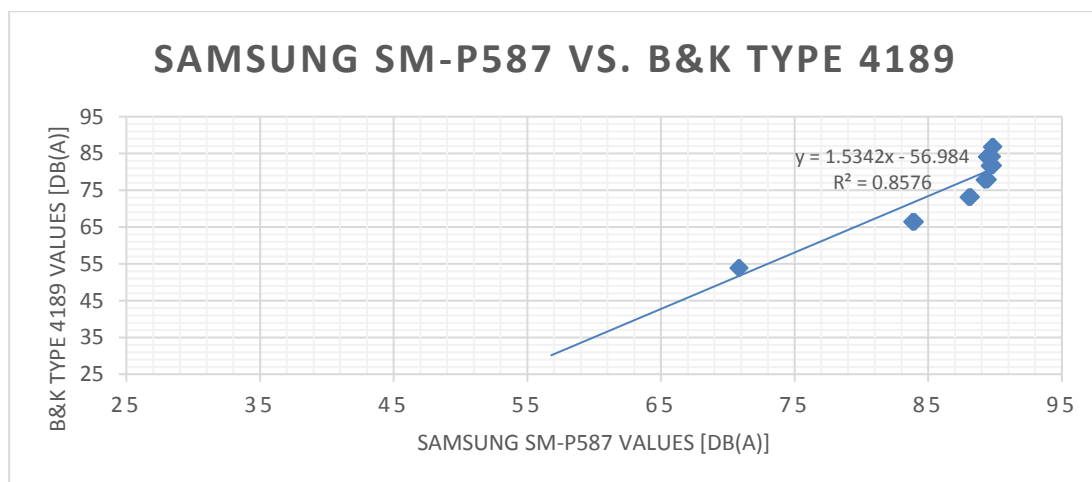


Figure 3.18 : Calibration line of Samsung SM-P587 in the test vehicle cabin.

The interpolation of measured values on Samsung SM-P587 to B&K microphone gives more accurate results yet the limitation of the android device under the influence of white noise remains constant. The android tablet gives erroneous results especially after the acoustic overload point of 90 dB(A). These results pose a minor advantage inside the cabin since the environment is uncontrolled and the reflections are present. The complex structure of the sound propagation inside the cabin converges the results slightly towards the B&K microphone. However, the use of the Samsung SM-P587 tablet under any testing environment will give inaccurate results to some extent due to the incapability of the device for sound measurement purposes.

The method of linear interpolation is again used to calculate the calibration line of iPhone X this time for the measurements inside the vehicle cabin. On Figure 3.19, the calibration line for iPhone X can be investigated.

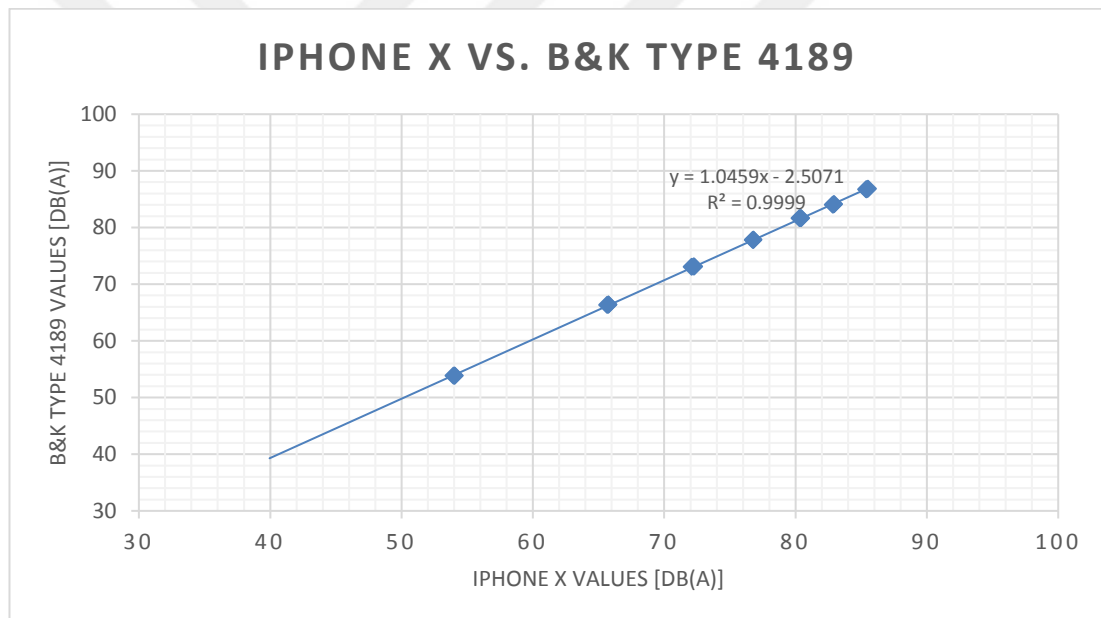


Figure 3.19 : Calibration line of iPhone X in the test vehicle cabin.

The measurement results of iPhone X gave very accurate results when compared to B&K Type 4189 microphone as expected from the previous hemi – anechoic room measurements. The case remains the same inside the interior cabin and the measurements converge towards the reference values inside the vehicle. After the calibration of iPhone X, the results show that the use of an iPhone X microphone is more than enough to be used for professional sound applications. As it can also be seen from the frequency spectrum of the microphone, especially if the frequency of interest is below 14 kHz, the frequency response of iPhone X is very accurate when compared

to similar mobile microphones. The overview of the effect of calibration on each device within the vehicle cabin can be seen on Table 3.11 below.

Table 3.11 : The effect of calibration on each device in the test vehicle cabin.

B&K Type 4189	Before Calibration		After Calibration	
	Samsung SM-P587	iPhone X	Samsung SM-P587	iPhone X
53.83	70.89	54.02	51.78	53.92
66.39	83.86	65.76	71.67	66.27
73.09	88.19	72.22	78.21	73.05
77.79	89.47	76.75	80.28	77.77
81.65	89.65	80.43	80.56	81.58
84.08	89.68	82.93	80.60	84.20
86.72	89.76	85.42	80.73	86.78

The initial measurements were conducted with the Samsung SM-P587 and iPhone X but as stated previously, calibration is conducted on FON application that was developed within Ford Powertrain NVH team. Since the application is only available for android devices for now, the calibration is done for the Samsung SM-P587 tablet only. The ideal values and the effect of calibration on Samsung SM-P587 tablet measured values can be seen on Figure 3.20. The results show that even if the calibration is conducted, the Samsung SM-P587 values are problematic especially for higher sound pressure levels. The hesitation of inaccuracy especially around 90 dB(A) remained existent during the measurements but proved an improvement for the results.

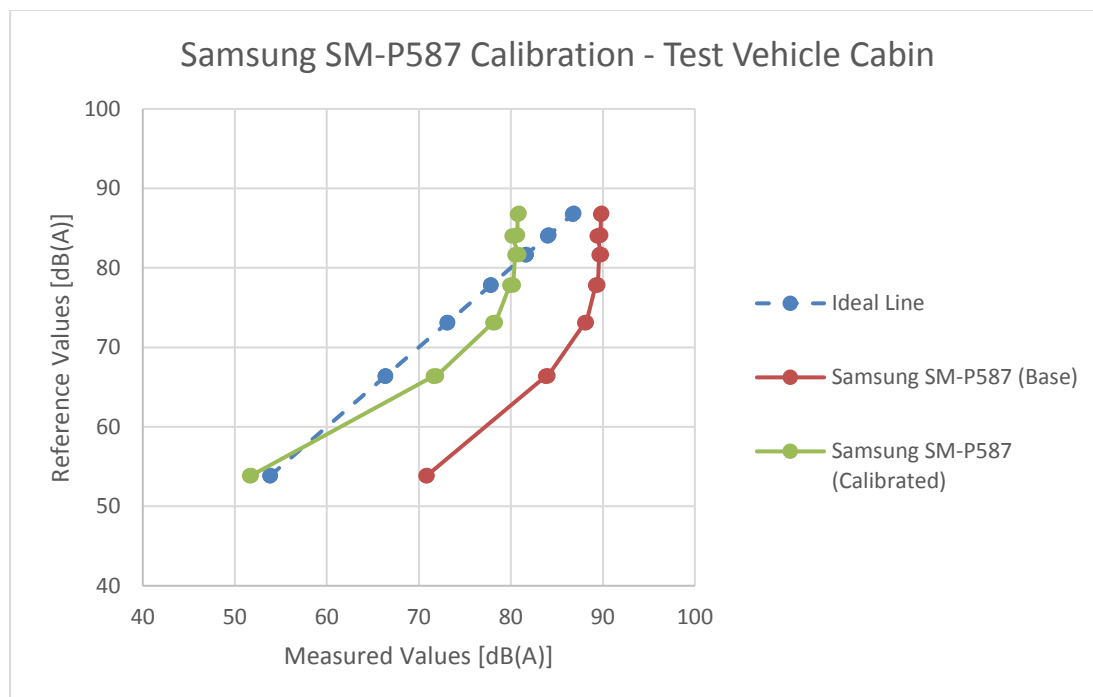


Figure 3.20 : Calibration values of Samsung SM-P587 in the test vehicle cabin.

Contrary to the Samsung tablet, the values showed clear similarity to B&K Type 4189 reference values on iPhone X. The performance of the in built microphones inside iPhone X and they proved to be suitable for such acoustic measurements. As it can also be seen from the Figure 3.21, after the calibration the measured values fit very good on the ideal line. Briefly, it can be said that the microphone of iPhone X is superior to Samsung SM-P587 tablet in terms of acoustic sound measurements.

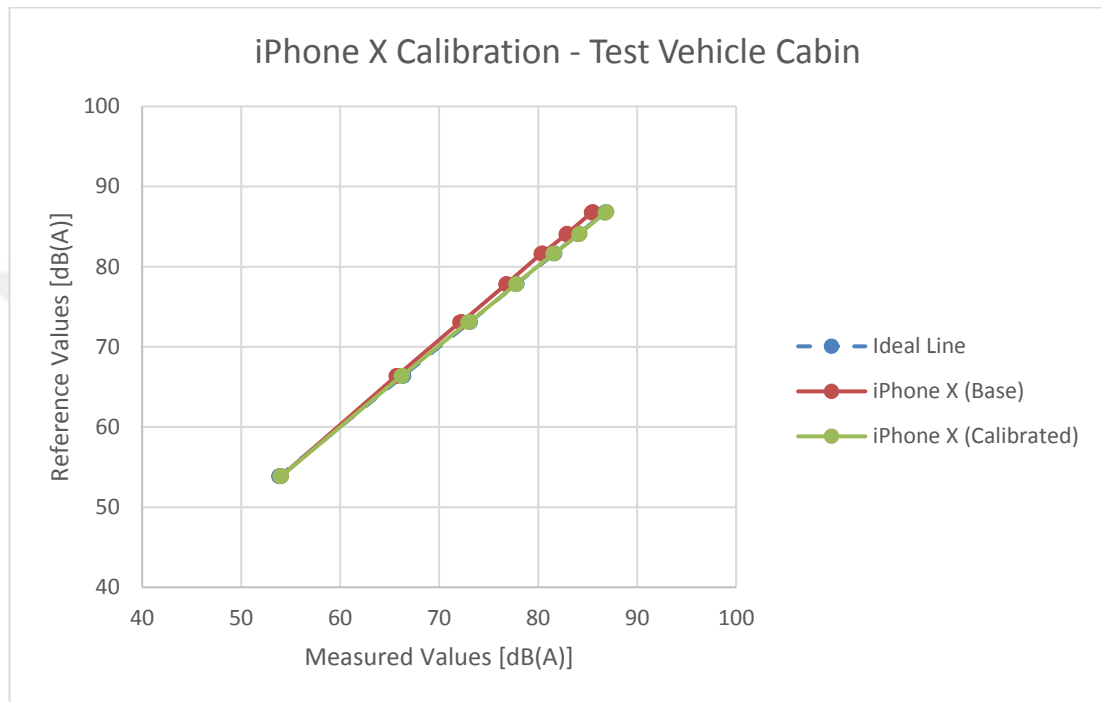


Figure 3.21 : Calibration values of iPhone X in the test vehicle cabin.

3.3 Issue Resolution of a Test Vehicle with a Mobile Device

FON – Ford Otosan Noise Diagnosis application has been developed in order to find the error states within the powertrain components of the vehicle thus producing a critical output for the end user about the current condition of the vehicle. This application is developed for android devices only as of today so mobile devices are used through the process to find the error state via audio signal processing.

In order to prove the effectiveness of using a mobile device for professional purposes, FON application setup has been done on the current android mobile device – Samsung SM-P587. The issue resolution with the mobile device requires the device to be stationed within the vehicle and measurements are conducted through the drive. When any powertrain related issue is seen inside the vehicle, FON picks up the noise and detects the problematic sound source.

A Ford Transit has been used for testing purposes of the effectiveness of FON application for powertrain related issue resolution inside the vehicle. The measurements are taken for each maneuver the driver makes including 2nd and 3rd gear run up and run downs along with a cruise at 5th gear.

After the conducted analysis, the issue is seen to be related with 6.4 th engine order which corresponds to the rear axle. Since the measurements are taken with the Samsung SM-P587, the rpm information at the time of maneuver was not available. Artemis Suite RPM Generator is used to generate the suitable rpm of the maneuver and order spectrum of the measured sound is analysed to check the orders at which the sound pressure shows peaks. Three run down maneuvers at 4th gear are analysed for the issue resolution as seen on Figure 3.22. Since the run down maneuvers were taken at a different time, there are some differences especially in sound pressure level readings but the order that shows the most distinct peak aside from the engine combustion is found to be around 6.4. The first RD maneuver is slightly below the tolerance limits for rear axle orders but 2nd and 3rd RD maneuvers find the correct orders. Considering the accuracy for finding the orders, Samsung SM-P587 proved to be capable of finding the correct problematic component for this rear axle whine problem.

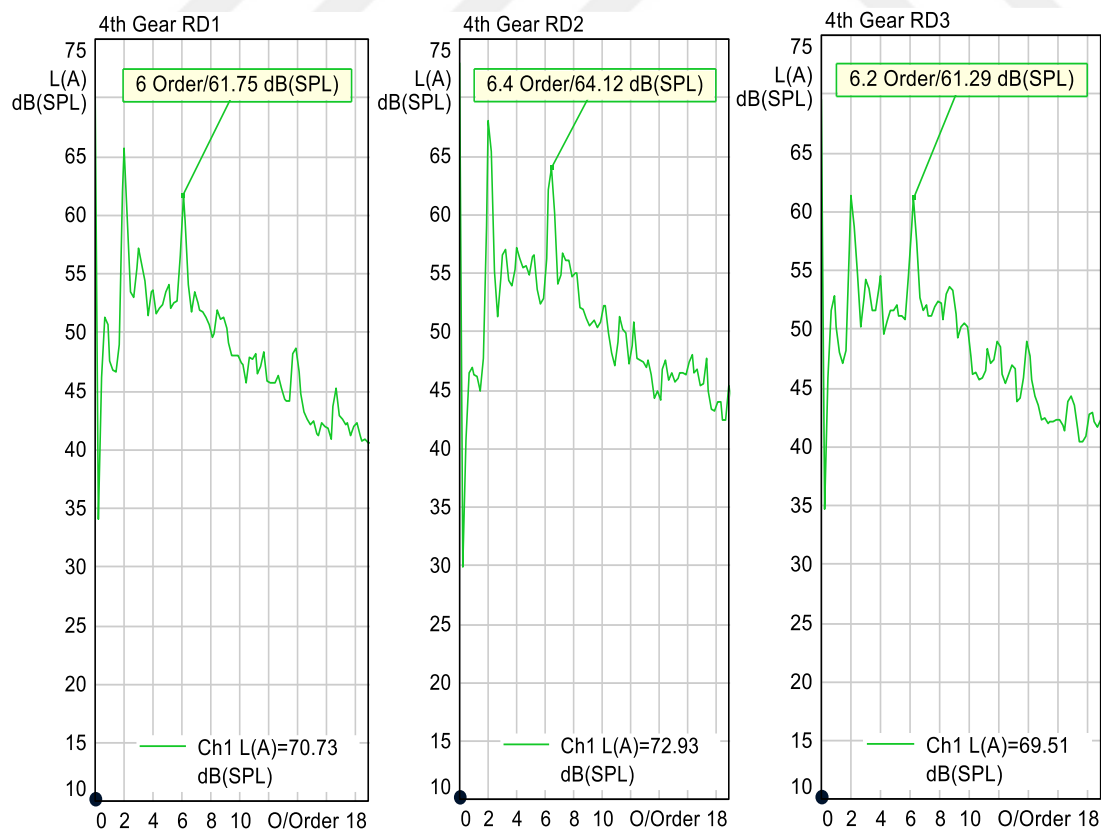


Figure 3.22 : Order Spectrum of 4th gear run down maneuvers.

With regards to the frequency spectrum, FFT vs Time analysis is conducted on Artemis Suite in order to define and verify the capability of Samsung SM-P587 for various frequencies for three run down maneuvers for fourth gear at a Ford Transit. As seen on Figure 3.23, the frequency spectrum of these run down maneuvers, a problematic order is seen clearly between 300-400 Hz area. Since this issue is very audible between these frequencies and the frequency range is around the area at which the tablet microphone is capable, the issue is easily seen.

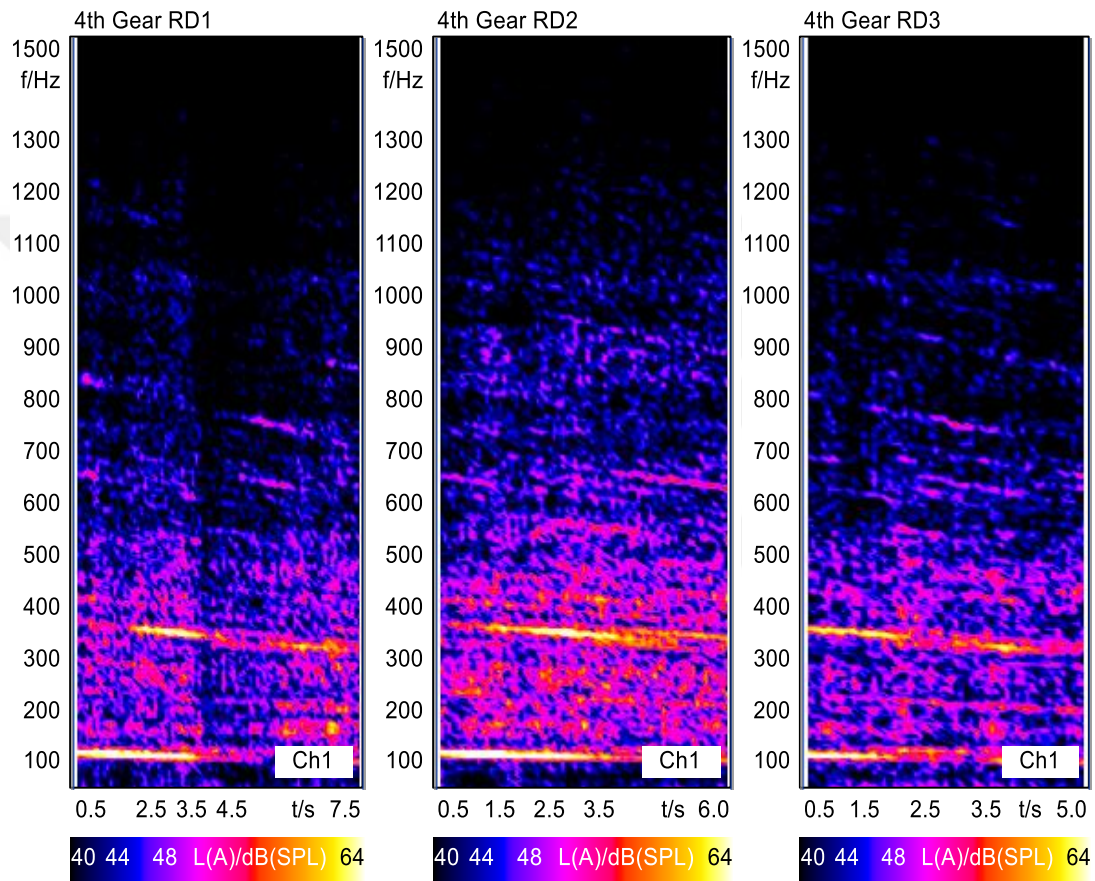


Figure 3.23 : FFT vs. Time spectrum for 4th gear run down maneuvers.

In order to verify the capability of the Samsung SM-P587 microphone, another problematic vehicle is tested with this device. The issue was again a whine which was observed during the third gear run down maneuver. The whine traces are clearly seen on the frequency spectrum on Figure 3.24 around 360 Hz and its harmonic of 720 Hz. Even though the microphone of Samsung SM-P587 proved to be amplifying the results at higher frequencies, the whine traces can easily be seen in this case. The reason for this performance can be explained by the frequency of interest for this error state. Since the problematic frequency is below 1000 Hz, we can clearly see the traces.

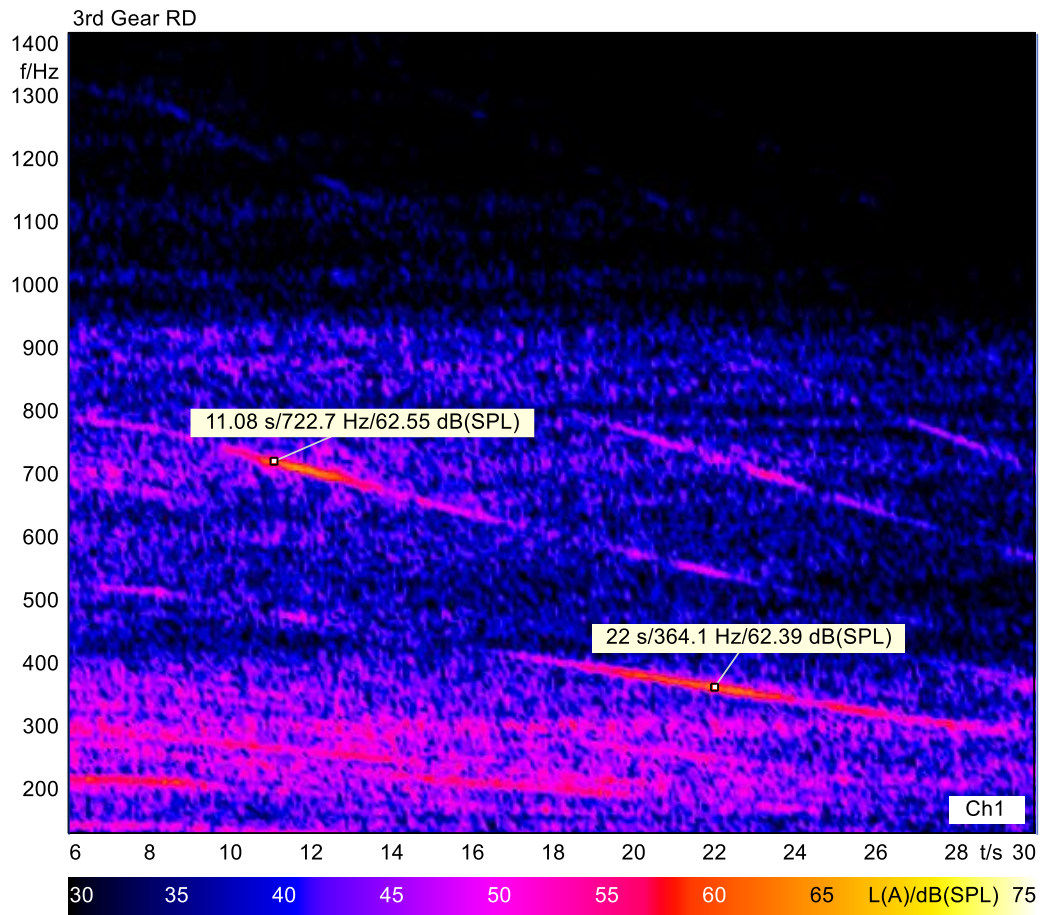


Figure 3.24 : FFT vs. Time spectrum for 3rd gear run down maneuver.

Since the whine issue was more audible and irritating on this second vehicle, the harmonics of this noise is also seen on the spectrogram. The use of a mobile device microphone is once more verified on a whiny vehicle with this measurement. Since the frequencies of interest were below 1 kHz for both of these issues, the problems were easier to defines. However, due to the non-uniform frequency response distribution of this mobile device, the issue resolutions may become harder to define with mobile devices even with the calibration.

3.4 Microphone Comparison of Various Mobile Devices in a Test Vehicle

After a significant variation in sound pressure level readings is seen between iPhone X and Samsung SM-P587 tablet, a further investigation is conducted in a Ford Kuga. The equipment and devices used for this experiment were different than the initial test setup. Ford Kuga 's interior loudspeakers are used to transmit the white noise signal generated in MATLAB with 5 different amplitudes. The amplitudes are controlled via the vehicle entertainment system within the vehicle. The devices used are the Samsung

SM-P587 tablet & iPhone X from previous test setup, General Mobile 5 Plus, LG K10 M250Y and Venus V3 5580 android devices. As it can be seen from Figure 3.25, there is a significant variation between the sound pressure level readings for the same amplitudes with various mobile devices.

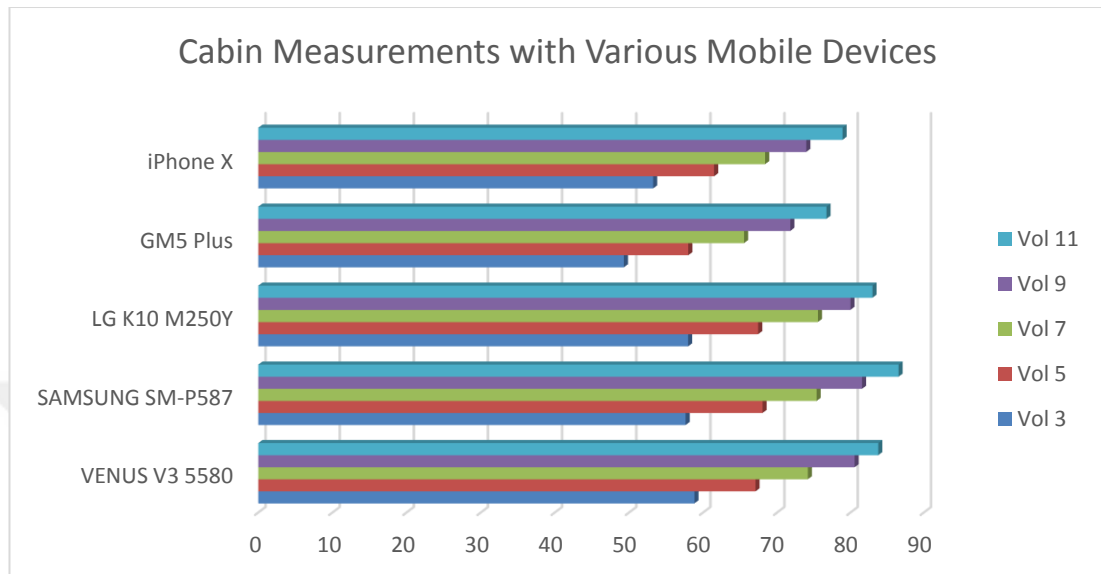


Figure 3.25 : White noise level readings of various mobile devices in test vehicle.

Figure 3.26 below shows the distribution of these amplitudes for each device. Since the difference between sound pressure level readings go up to 10 dB(A) for some mobile devices, it is not possible to measure the same value even if the signal amplitudes are the same.

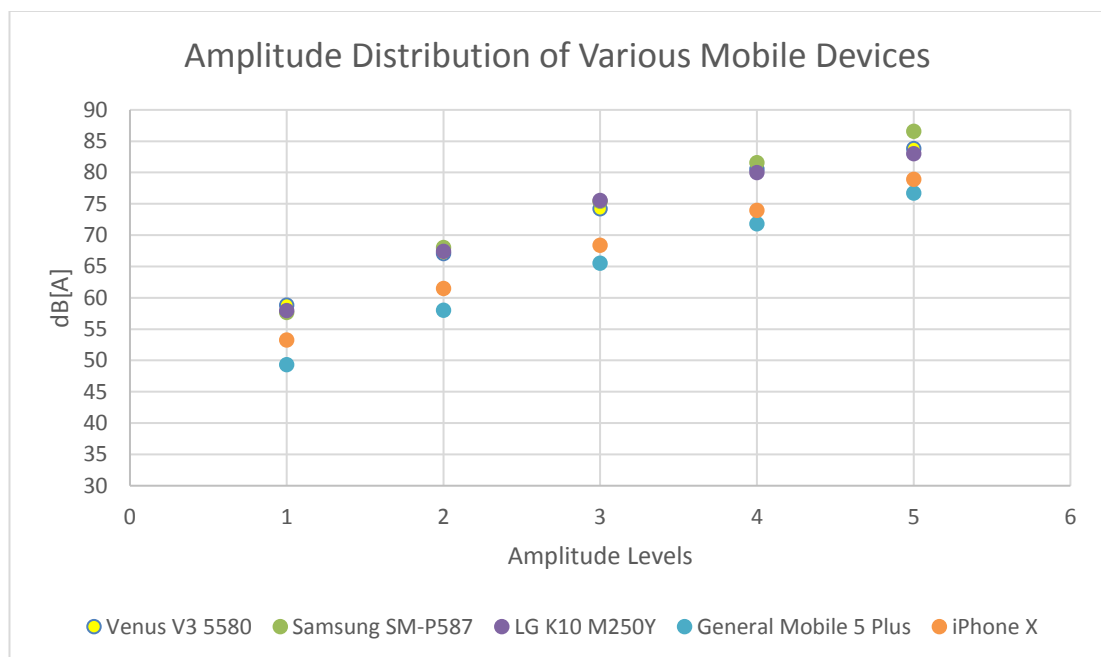


Figure 3.26 : Amplitude distribution of various mobile devices in test vehicle.

Since the sound pressure level readings were very different for each device, further research is conducted on the increase of SPL readings for each amplitude level change. The amplitude increase is done 4 times via vehicle entertainment system with equal increments and the increase in SPL readings is checked for each device as soon on Table 3.12 and Figure 3.27.

Table 3.12 : SPL (dB(A)) increments of each device for each amplitude increase.

Increase	Venus V3 5580	Samsung SM-P587	LG K10	GM 5 Plus	iPhone X	Mean	Standard Deviation
1	8.23	10.38	9.46	8.69	8.22	9.04	0.83
2	7.13	7.37	8.13	7.52	6.89	7.42	0.42
3	6.37	6.19	4.44	6.27	5.6	5.86	0.75
4	3.23	4.98	3.00	4.94	4.96	4.37	0.98

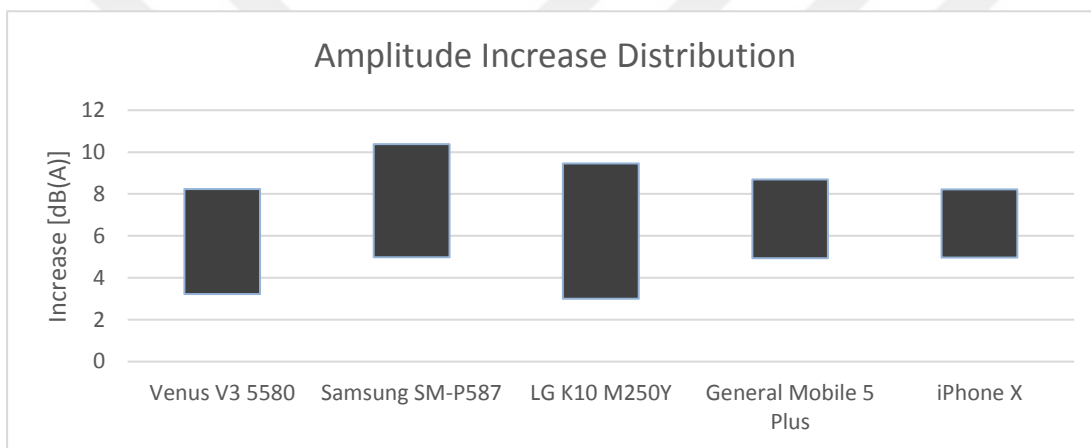


Figure 3.27 : SPL increment distribution of each device for amplitude increases.

As a result of these test conducted within the Ford Kuga, it is seen that the sound pressure level readings differ significantly for each mobile device since their microphones, internal structure and processors are different. The increase in sound pressure level readings are also not the same which means that the sensitivity of each device to a change in amplitude is also diverse. Due to this variation in measurement results, it is not possible to define a generally accepted calibration curve for every device. A device unique calibration is the best way to go for more precise measurements.

4. CONCLUSIONS AND SUGGESTIONS FOR FUTURE WORK

In this study, the calibration of mobile devices is studied in terms of a level based algorithm in which the effect of medium of measurement and the change in hardware is investigated. The measurements are conducted with a few mobile devices including a tablet. The reference sound pressure levels are structured using a pre calibrated B&K Type 4189 condenser microphone.

This thesis aimed to find the limitations, accuracy and performance of mobile devices in different mediums in comparison to a reference condenser microphone. Moreover, a level based calibration is done using the A weighted measurement to simulate the human hearing and implemented on an android mobile application. This study also investigated the performance of a mobile device for acoustic issue resolution and detection in a Ford Transit vehicle for whine noise.

It is clarified that the built in audio processing integrated within the mobile device and the limitations of internal MEMS microphones of these devices caused some erroneous measurements especially at high noise levels. The acoustic overload point of an android device is defined for white noise signals for these devices. Moreover, the frequency response of some mobile devices are not linear and distortions along with amplifications in their spectrum is existent.

From the noise level measurement results both in hemi anechoic room and Ford Transit cabin, it is seen that the behavior of mobile microphones are nearly the same in each of these mediums for swept sine signals. However, for pure tone signals it is seen that the response of mobile microphones are unpredictable and less accurate inside the vehicle cabin when compared to the hemi anechoic room. The reason for this difference in measured sound pressure levels is due to the uncontrolled environment inside the cabin and unpredicted reflections from the surrounding part.

The level based calibration that is implemented inside the mobile application proved to be improving the accuracy of the mobile microphones yet due to frequency distortions and acoustic overload points at higher amplitudes, calibration alone is not sufficient to converge the measurement accuracy of these mobile devices to

professional microphones. However, the more advanced mobile devices like iPhone X have better integrated microphones which in turn have better sound measurement capabilities. These more advanced mobile devices can be used for professional applications when calibrated up to some extent if there are no distortions in the frequency response of the device.

As an output of the measurements conducted inside a Ford Kuga cabin with various mobile devices, it is concluded that there is a significant variance in the performance of mobile microphones. Thus, the coefficients of a level based calibration for a single device can't be implemented to every mobile device. A device based calibration is the way to be followed in order to increase the measurement accuracy.

Analysis of performance of a calibrated mobile device under vehicle operation conditions showed that some mobile devices can be used for acoustic issue resolutions and detections if the problematic noise is significant. Even though the amplitude of the sound is slightly different, the significant noise peaks in frequency spectrum can be differentiated by these mobile devices. Thus, if the performance of the built in microphone is proved to be sufficient, the mobile device can be used for professional applications.

The uncertainties and limitations associated with this study for each condition are also identified. There are significant variations in measurements for each mobile device. In order to get a better look on the results, device quantity can be increased. It is seen that the presence of acoustic overload point is existent at higher levels in some mobile devices. The level of acoustic overload point is of high importance to define the capability of the built in microphones.

Since the built in microphones in mobile devices are silicon based MEMS microphones and are located inside the phone cases, they act like directional microphones. In order to get comparable results, the mobile device microphone location and direction of placement is also important.

Based on the results from this thesis, some recommendations can be made for further studies on the subject of mobile device calibration and their use for acoustic purposes. The first point is increasing the mobile device quantity that is tested. As the quantity of mobile devices increases, the overview on the variation of the measurements becomes more clear. Moreover, in order to define the effect of the measurement

medium on the results, the samples of real world environments can be increased along with the device quantity.

In this thesis, only a handful of noise detection applications are investigated within the vehicle. The behaviour of mobile device microphones for various professional applications and issue resolutions can be studied for an increased number of samples and issues for a more accurate view.

Finally, a frequency based calibration method can be devised to further increase the capability of the mobile device microphones. Moreover, the calibration process of the mobile device can be automatised for the convenience of the end user for each device. Thus, with a reference signal and a smart algorithm the calibration will automatically be conducted and the manual input of calibration values will be eliminated for each device. the calibration of mobile device microphones in this study has been conducted manually on each device with the help of a mobile application.

Consequently, alternative future researches would include the findings in this study and further improve the calibration process of mobile devices thus making way for further use of mobile devices in a wider range of industry applications.



REFERENCES

- [1] **Akustica, Inc.** (2011). *New Microphones Boost Acoustic Performance in Consumer Electronics*. Retrieved November 20, 2019, from <http://www.akustica.com>.
- [2] **Faberacoustical.** (2012). *Are you looking for a measurement microphone for your iPhone?*. Retrieved November 21, 2019, from <http://blog.faberacoustical.com/2012/uncategorized/are-you-looking-for-a-measurement-microphone-for-your-iphone>.
- [3] **Roberts, B., Kardous, C. And Neitzel, R.** (2016). Improving the accuracy of smart devices to measure noise exposure. *Journal of Occupational and Environmental Hygiene*, 13:11, 840-846. doi : 10.1080/15459624.2016.1183014.
- [4] **Rana, R. et. al.** (2010). Ear - Phone : An end-to-end participatory urban noise mapping system. *Proceedings of the 9th ACM/IEEE International Conference on Information Processing in Sensor Networks, IPSN '10*. 105-116. doi : 10.1145/1791212.1791226.
- [5] **Stevens, M.** (2012). *Community memories for sustainable societies The case of environmental noise*. Retrieved November 24, 2019, from https://www.researchgate.net/publication/316233429_Community_memories_for_sustainable_societies_The_case_of_environmental_noise.
- [6] **Brown, R. and Evans L.** (2011). Acoustics and smarphone. *Proceedings of Acoustics 2011, Paper Number 106*. Retrieved November 24, 2019, from http://www.acoustics.asn.au/conference_proceedings/AAS2011/papers/p106.pdf.
- [7] **Ventura, R. et. al.** (2017). Evaluation and calibration of mobile phones for noise monitoring application. *The Journal of the Acoustical Society of America*, 142, 3084.
- [8] **Frigerio, F. and Virga, V.** (2019). Noise metering via mobile phones limitations opportunities and findings in a workplace testing. *2019 IEEE International Conference on Environment and Electrical Engineering and 2019 IEEE Industrial and Commercial Power Systems Europe (EEEIC / I&CPS Europe)*, Genova, Italy, pp. 1-4.
- [9] **SafetyAwakenings.** (2013). *A Review Of 30 Sound (Noise) Measurement Apps*. Retrieved November 25, 2019, from <http://www.safetyawakenings.com/safety-app-of-the-week-42>.
- [10] **Serpanos, Y. et. al.** (2018). The Accuracy of Smartphone Sound Level Meter Applications With and Without Calibration. *American Journal of Speech-Language Pathology*. 27(4), 1319-1328. doi : 1. 10.1044/2018_AJSLP-17-0171.

- [11] **Murphy, E., King, E.A.** (2016). Testing the accuracy of smartphones and sound level meter applications for measuring environmental noise. *Applied Acoustics*, 106, pp. 16-22.
- [12] **Romain, D.** (2014). On the use of mobile phones and wearable microphones for noise exposure measurements : calibration and measurement accuracy. Retrieved November 25, 2019, from https://www.researchgate.net/publication/321038680_On_the_use_of_mobile_phones_and_wearable_microphones_for_noise_exposure_measurements_calibration_and_measurement_accuracy.
- [13] **Beniwal, R. et. al.** (2005). Seeing is Believing – A Better Way to Diagnose Noise Sources. *SAE Noise and Vibration Conference and Exhibition*. doi : 10.4271/2005-01-2507.
- [14] **Watanabe, M., Tanabe, Y., and Yoneya, N.** (2016). Development of High-Resolution Exciting Source Identification System. *2016 SAE World Congress and Exhibition*. doi : 10.4271/2016-01-1325.
- [15] **Wei, C. et. al.** (2019). Noise Sources Identification and Contribution Analysis of Electric Bus Powertrain over Complete Operating Region *SAE Technical Paper*. doi:10.4271/2019-01-0777.
- [16] **Alias, E., Naveen, N. and Mathew, D.** (2014). A Novel Acoustic Fingerprint Method for Audio Signal Pattern Detection. *2014 Fourth International Conference on Advances in Computing and Communications*, Cochin, 2014, pp. 64-68. doi: 10.1109/ICACC.2014.21
- [17] **Aly, F.** (2013). Engine Fault Detection Using Wavelet Packet Transforms. *2016 SAE World Congress and Exhibition*. doi : 10.4271/2013-01-1705.
- [18] **Verma, K. et. al.** (2014). Windows mobile and tablet app for acoustic signature based machine health monitoring. *2014 9th International Conference on Industrial and Information Systems (ICIIS)*, Gwalior, 2014, pp. 1-6. doi: 10.1109/ICIINFS.2014.7036655
- [19] **Bies, D., Hansen, C. and Howard, C.** (2017). *Engineering Noise Control*. Fifth Edition. CRC Press.
- [20] **Harris, M.** (1991). *Acoustical Measurements and Noise Control*. 3rd Edition. McGraw Hill, New York.
- [21] **Salter, M.** (1998). *Acoustics – Architecture, Engineering, the Environment*. William Stout, San Francisco.
- [22] **Yan, C. et. al.** (2019). The Feasibility of Injecting Inaudible Voice Commands to Voice Assistants. *IEEE Transactions on Dependable and Secure Computing*, March 2019, pp. (99):1-1. doi : 10.1109/TDSC.2019.2906165
- [23] **Brüel & Kjaer.** (1988). *Measuring Sound*. Nerum, Denmark.
- [24] **Wang, X.** (2010). *Vehicle Noise and Vibration Refinement*. Boca Raton, FL. CRC Press.
- [25] **InvenSense.** (2015). *Microphone Specifications Explained*. Retrieved November 28, 2019, from <https://www.invensense.com>.

- [26] **Rose, B.** (2019). *An introduction to beamforming with MEMS microphones*. Retrieved November 29, 2019, from <https://www.digikey.ee/en/articles/an-introduction-to-beamforming-with-mems-microphones>
- [27] **Merska, O., Mieczkowski, P. And Zymelka, D.** (2016). Low noise thin surface course – evaluation of the effectiveness of noise reduction. *Transportation Research Procedia*, 14, 2688-2697. doi : 10.1016/j.trpro.2016.05.445
- [28] **Shure.** (2019). *Primer on microphone types and operation*. Retrieved November 29, 2019, from https://service.shure.com/s/article/primer-on-microphone-types-and-operation?language=en_US
- [29] **Boyk, J.** (2000). *There's life above 20 kilohertz! A survey of musical instrument spectra to 102.4 kHz*. Retrieved December 9, 2019, from <https://www.cco.caltech.edu/~boyk/spectra/spectra.htm>
- [30] **Cooper, J.C., Leclercq, D. and Stead, M.A.** (2010). *Wind induced aerodynamic noise on microphones from atmospheric measurements*. Retrieved December 9, 2019, from <https://www.semanticscholar.org/paper/Wind-induced-aerodynamic-noise-on-microphones-from-Cooper-Leclercq/6a1fd10389fed3cee6199c0324a91dcacd4da331>
- [31] **Struck, C.** (2017). Measurement uncertainty and its application to acoustical standards. *Proceedings of Meetings on Acoustics*. 31. 032001. doi : 10.1121/2.0000775.
- [32] **Wong, G.S.K. and Embleton, T.F.W.** (1979). Arrangement for precision reciprocity calibration of condenser microphones. *The Journal of the Acoustical Society of America*, 66. 1275-1280. doi : 10.1121/1.383547.
- [33] **Wong, G.S.K. and Embleton, T.F.W.** (1995). *AIP Handbook of Condenser Microphones. Theory, Calibration and Measurements*. American Institute of Physics, New York.
- [34] **Schroeder, M. et. al.** (2007). *Springer Handbook of Acoustics*. 1st Edition. Springer Publishing Company, Incorporated.
- [35] **Barrera-Figueroa, S.** (2018). Free field reciprocity calibration of measurement microphones at frequencies up to 150 kHz. *The Journal of the Acoustical Society of America*, 144. 2575-2583. doi : 10.1121/1.5063815.
- [36] **International Electrotechnical Committee.** (2004). *Electroacoustics - Measurement Microphones Part 6 : Electrostatic Actuators for Determination of Frequency Response*. IEC, Geneva.
- [37] **International Electrotechnical Committee.** (2016). *Electroacoustics - Measurement Microphones Part 5 : Methods for pressure calibration of working standard microphones by comparison*. IEC, Geneva.
- [38] **G.R.A.S Sound & Vibration.** (2013). *Pistonphone Type 42AA / 42AA - S1 Instruction Manual*. Retrieved December 12, 2019, from https://www.gras.dk/files/m/a/man_42AA_42AA-S1.pdf
- [39] **Nedzelitsky, V.** (2007). Calibration of Pressure and Gradient Microphones. *Encyclopedia of Acoustics, Volume 4*. doi : 10.1002/9780470172544.ch157.

- [40] **Butler, E.** (2001). *Avance Beginners Guide*. Retrieved December 21, 2019, from <http://triton.iqfr.csic.es/guide/man/beginners/chap3-9.htm>.
- [41] **Sultan, K., Ali, H. and Zhang, Z.** (2018). Call Detail Records Driven Anomaly Detection and Traffic Prediction in Mobile Cellular Networks. *IEEE Access*, 6. 41728-41737.
- [42] **Smith, S.** (1997). *The scientist and engineer's guide to digital signal processing*. California Technical Publishing, USA.
- [43] **Hartmann, W. M.** (1998). *Signals, Sound and Sensation*. AIP Press, USA.
- [44] **Siemens.** (2017). *Digital Data Acquisition*. Retrieved January 3, 2020, from https://community.sw.siemens.com/servlet/fileField?entityId=ka54O00000PJtdQAG&field=Attachment__Body__s.
- [45] **Cooley, J. W. and Tukey, J. W.** (1965). An Algorithm for the Machine Calculation of Complex Fourier Series. *Math Computat.*, 19. 297-301.
- [46] **National Instruments.** (2015). *Understanding FFT and Windowing*. Retrieved January 5, 2020, from <https://download.ni.com/evaluation/pxi/Understanding%20FFTs%20and%20Windowing.pdf>.
- [47] **Beranek, L. L. and Ver, I. L.** (2005). *Noise and Vibration Control Engineering Principles and Applications*. John Wiley & Sons, Inc.
- [48] **Bendat, J. S. and Piersol, A.** (1993). *Engineering Applications of Correlation and Spectral Analysis*. Wiley, New York.
- [49] **Trethewey, M. W. and Evensen, H. A.** (1984). Time – Delay Bias Errors in Estimating Frequency Response and Coherence Functions from Windowed Samples of Continuous and Transient Signals. *Journal of Sound and Vibration*. 97(4), 531-540.
- [50] **Brüel & Kjær.** (2008). *½" Prepolarized Free - field Microphone Type 4189 Product Data*. Retrieved January 7, 2020, from <https://www.bksv.com/-/media/literature/Product-Data/bp2210.ashx>.
- [51] **Brüel & Kjaer.** (2012). *Pocket Handbook – Sound and Vibration*. Retrieved November 28, 2019, from <https://www.bksv.com/downloads/svpockethandbook/index.html>
- [52] **Brüel & Kjær.** (2014). *Sound Calibrator Type 4231 Product Data*. Retrieved January 7, 2020, from <https://www.bksv.com/-/media/literature/Product-Data/bp1311.ashx>.
- [53] **PreSonus.** (2019). *Eris E5 Studio Monitor Technical Sheet*. Retrieved January 10, 2020, from https://pae-web.presonusmusic.com/downloads/products/pdf/Eris_E5_Studio_Monitor_techsheel.pdf.
- [54] **Url-1** <<http://developer.android.com/reference/android/media/audiofx/AutomaticGainControl.html>>, Automatic Gain Control, date retrieved 21.11.2019.
- [55] **Url-2** <<http://www.electronics-tutorials.ws/accircuits/average-voltage.html>>, Average Voltage Tutorial, date retrieved 25.11.2019.
- [56] **Url-3** <<https://www.liberaldictionary.com/carbon-microphone/>>, Double Button Carbon Mic Interface, date retrieved 29.11.2019.

- [57] **Url-4** <http://artsites.ucsc.edu/EMS/Music/tech_background/TE-20/teces_20.html>, Microphones, date retrieved 29.11.2019.
- [58] **Url-5** <https://www.daviddarling.info/encyclopedia_of_music/R/ribbon_microphone.html>, Ribbon Microphone, date retrieved 30.11.2019.
- [59] **Url-6** <<https://www.mediacollege.com/audio/microphones/condenser.html>>, Condenser Microphones, date retrieved 01.12.2019.
- [60] **Url-7** <<https://www.signaldefense.com/>>, IR Protection, date retrieved 01.12.2019.
- [61] **Url-8** <<http://www.openmusiclabs.com/learning/sensors/electret-microphones/index.html>>, Electret Microphones, date retrieved 03.12.2019.
- [62] **Url-9** <<https://www.mouser.com/datasheet/2/720/DS26-1.04%20AKU340%20Datasheet-770074.pdf>>, AKU340 Bottom Port, Analog Silicon MEMS Microphone, date retrieved 03.12.2019.
- [63] **Url-10** <https://product.tdk.com/info/en/products/sw_piezo/mic/mems-mic/tech_note/tpo/index_mytdk.html>, MEMS Microphones, date retrieved 03.12.2019.
- [64] **Url-11** <<https://www.cuidevices.com/blog/comparing-mems-and-electret-condenser-microphones>>, Comparing MEMS and Electret Condenser Microphones, date retrieved 03.12.2019.
- [65] **Url-12** <<https://pulsarinstruments.com/en/post/difference-between-class-1-and-class-2-sound-level-meters>>, Difference Between Class 1 and Class 2 Sound Level Meters, date retrieved 05.12.2019.
- [66] **Url-13** <<https://dewesoft.com/products/services/acoustic-calibrations>>, Acoustic Calibration Services, date retrieved 12.12.2019.
- [67] **Url-14** <<https://www.amazon.com/WMYCONGCONG-3590S-2-502L-Adjustable-Multi-Turn-Potentiometer/dp/B01MRTXOGN?th=1>>, Adjustable Multi-Turn Potentiometer, date retrieved 14.12.2019.
- [68] **Url-15** <https://akshaysin.github.io/fourier_transform.html#.XgdPra4zbIU>, Edge detection in images using Fourier Transform, date retrieved 20.12.2019.
- [69] **Url-16** <<https://allsignalprocessing.com/2015/04/25/aliasing-of-signals-identity-theft-in-the-frequency-domain/>>, Aliasing of Signals – Identity Theft in the Frequency Domain, date retrieved 20.12.2019.



CURRICULUM VITAE



Name Surname : Güner Berker RAHTUVAN

Place and Date of Birth : İzmir 19.11.1993

E-Mail : berker.rahtuvan@hotmail.com

EDUCATION :

- **B.Sc.** : 2016, Istanbul Technical University, Faculty of Mechanical Engineering, Mechanical Engineering

PROFESSIONAL EXPERIENCE AND REWARDS:

- FORD OTOSAN Product Development / Body Interior Soft Trim Engineer (03.2017-11.2019)
Responsible for interior soft trim sound insulation parts from concept and design phase up to the final product release for Transit, Focus and Kuga projects. Involved in the design change, feasibility studies, cost reduction actions, launch management activities and the validation of part performance tests.
- FORD OTOSAN Product Development / Powertrain NVH Development Engineer (11.2019 - Present)
Responsible for performing NVH simulations, advising on issue resolutions and design options for powertrain components in order to achieve the program targets. The coordination and participation in NVH testing activities on site is also conducted on regular basis for development purposes.